ADVISORY GROUP ON

Computer Graphics

A Review of Multimedia Networking

Alan Taylor and Madjid Merabti

sima

Support Initiative
for Multimedia Applications

Part of the JISC New Technologies Initiative

Contents

1. INTRODUCTION

A demonstration of multimedia application sharing between California and Copenhagen seen recently [EIUF94] took several days to set up and cost about 100,000 dollars in time, equipment, and communication costs.

When was this? Not in 1984, but late in 1994. We are told that the Information Super Highway is driving towards us in the fast lane, but no one can quite say from which direction. Trials of video on demand are continuing but roll out of services have been delayed. We are told that home services on the highway will have to be kept simple for us to take in, and will be affordable. In education multimedia learning packages have been mounted on file-servers, use of the World Wide Web is widespread, and some video conferencing is being used.

So networking of computer based multimedia is possible. However network delivery of multimedia is not new, the broadcast television system has been available since the 1950's. The difference now is that the information which has previously been presented on radio, TV, or in books, is now being presented or can be accessed by computer networks. Or looking at this development in another way computer technology has almost caught up with the broadcasters. The drive to deliver multimedia information over a computer network is fuelled by protagonists of the Information Superhighway who need multimedia to justify the requirement for the highway. Multimedia producers are also looking for a wider market and lecturers are excited both by the new learning environments offered and possible efficiency gains.

1.1. Networked Multimedia Applications

Multimedia can roughly be defined as a technology that enables humans to use computers capable of processing textual data, audio and video, still pictures, and animation. Applications range over entertainment, education, information provision, design e.g. CAD/CAM, co-operative working such as video conferencing, application sharing, remote working and virtual reality experiences.

Multimedia applications for computers have been developed for single computing platforms such as the PC, Apple Mac and games machines. The importance of communications or networking for multimedia lies in the new applications that will be generated by adding networking capabilities to multimedia computers, and hopefully gains in efficiency and cost of ownership and use when multimedia resources are part of distributed computing systems. Widening of access to multimedia sources and potential markets in multimedia, video and information are commercial driving force for networking multimedia.

The reality of networking multimedia is that :-

> The characteristics of multimedia make heavy demands on storage and transmission systems.

> Data compression can be used to reduce the demands of multimedia, particularly of video and audio on these systems, but usually at the expense of some loss in the detail compared with the source and at extra

cost.

The ways in which users or participants in multimedia sessions access multimedia or connect with others have important consequences for the storage and transmission systems. For instance multimedia learning material can be accessed directly from a server during a class or downloaded to student machines prior to a session. The demands on a connecting network are very different in each access mode.

The cost of transmitting multimedia information will determine the pace of development of networked multimedia applications

The availability of standards for multimedia networking, particularly for inter-working between applications, the development of networked applications, and interworking between networks are essential to reduce the complexity and level of skill required in using multimedia.

1.2. An Example

Using your desktop conferencing from Fujitsu which works on ISDN at 64 Kbps, and the Olivetti PCC video conferencing, you want to do some work with a colleague at Imperial College in London who fortuitously has a Super JANET Asynchronous Transfer Mode (ATM) connection at 34 Mbps, and has a workstation with UNIX based desktop conferencing and a codec for H.320 compatible video conferencing. Hard luck, you can't. While the video conference systems are compatible, both use H.320, the ISDN network cannot connect right through the ATM network. Also the desktop conferencing systems use different standards for sharing applications, whiteboards etc. and would not inter-work even if directly connected.

1.3. Report Structure

The rest of this report is divided into twelve sections where the different subsystems that impact on multimedia networking are reviewed. Section 2 reviews the requirements of multimedia before introducing some of the issues to be considered by users in section 3. Section 4 reviews the development in compression standards before leading to a an investigation of networking - from hardware to software technology (section 5 to 10). In section 11 we review some of the networked multimedia systems. Finally section 12 points to the leading edge research and developments efforts and issues in networking.

2. USER REQUIREMENTS FOR MULTIMEDIA

2.1. Human - Computer Interface

The standards of reproduction for computers which are desirable have been set by the publishers of books, music, Walt Disney cartoons and television producers. With the development of High Definition TV and beyond, it is likely that there will be a continual increase in the demands placed on computer based multimedia systems.

The current PAL standard in the UK delivers video in 625 lines at 25 frames/sec. High Definition TV delivers video in 1250 lines with a higher horizontal resolution at 25 frames/sec and requires about five times the information rate as the current PAL system.

Multimedia applications like any other application, appliance or tool, benefit from being easy to use, with minimal training or self learning. The need for a well designed human - computer interface, which may be screen or audio based is well accepted.

2.2. Access, Delivery, Scheduling and Recording

Television channels can be changed at the touch of a button. On demand access times to computer information need to be below one second to be usable in real time. Alternatively the delivery of information at a later time is acceptable if it can be scheduled, as in a TV broadcast schedule, or a first class postal letter. Scheduling the delivery of multimedia information has not been widely implemented. Scheduling can have advantages for users over on demand delivery. In a learning situation times can be defined for class attendance by a lecturer. In open learning situations learners can control their programme by requesting a multimedia unit at a convenient time.

Just as we can record a TV film on a VHS recorder, some multimedia computer users will wish to record a film, session, or learning experience for future reference.

2.3. Interactivity

Interactivity, meaning the ability to participate in a video or audio process on a computer, by changing its behaviour or appending comments has become very important in multimedia. Some of this popularity stems from the perception of computer games as being enjoyable because they are interactive, and some from work done in education which shows that some types of learning becomes easier, and is retained more permanently if the learner participates in some way with the learning material. Computer based multimedia needs the same degree of interactivity that a school exercise book, or a laboratory experiment has in order to remain credible as a learning medium. The generation of computer based virtual reality is an extension of this process. The incorporation of interactivity is really the job of the application designer. The incorporation of interactivity is assisted if the network is capable of two way communication, and for some applications the sense of interactivity is aided by the ability to deliver a moving picture, or a sound very quickly, so that a sense of two way human participation can be generated. Real time video conferencing is an example.

2.4. Educational requirements

An Open Learner needs to be able to use any multimedia application at any time. However since open learning is often undertaken in centres, the use of audio and video require particular thought. Obviously several sets Sound Blaster driven speakers will disturb those learners working on a computer based self test! Similar considerations occur for users of multimedia in a class situation, such as a language teaching application. Not only will a number of students be performing similar activities at the same time on a network but the lecturer must decide whether to control the activities via the media of the computer. The use of multi-party desktop conferencing with the lecturer chairing the running of the conferencing session, showing selected parts of a video etc. is a case in point.

Distance learners or users of multimedia will also be capable of having same impact on a network as several students playing the computer game Doom at lunchtime. Additionally the co-ordination of a learning activity must also be done by a lecturer over the network. So the role of the chair in multi-party video conferencing is crucial.

2.5. Cost

In education the main costs visible to multimedia users to date have been the cost of the computer platform, the CD-ROM, and the software. Network costs are usually borne centrally within an institution, or by JANET nationally. The increased cost of providing sufficient network capacity and ability of new networks to charge on the basis of bandwidth used mean that individual users will increasing have to consider the costs of access when designing or using multimedia applications. Additionally information providers, electronic publishers, etc. will begin to incorporate charging mechanisms in their systems [Nelson94].

The educational user ideally needs the costs of use to be well defined, in advance so applications teaching journalism, which make repeated access to Reuters databases would not be viewed favourably by educational managers.

Cost benefit analysis of multimedia distance learning, and open learning proposals will increasingly become the norm.

3. ISSUES FOR USERS

3.1. Characteristics of multimedia

Multimedia can be as simple as a few images with some accompanying text to a multimedia presentation using video clips, sound, images animation and text. Multimedia files to use a lot of data when in a digital format. Video is the most demanding. A PAL signal when digitised can require a data rate of 170 Mbps. Audio is less demanding but still requires 1.3 MByte for a 1 minute clip using a Sound Blaster Pro system at 22 kHz sampling rate. Still images require use more data proportional to their size. Synchronisation of sound and video is important. Sound is likely to breake up if parts of it are lost or delayed in storage or transmission.

Video is less vulnerable to loss (depending on the application), but still requires all of the picture to be on the screen at the same time and is also vulnerable to jitter. Jitter could be controlled in some applications if the sender of the isochronous video data time stamps each piece of data when it is generated, using a universal time source, and then sends the data to the receiver. The receiver reads a piece of data in as soon as it is received and store it. The receiver processes each piece of data only at the time equal to the data's time stamp plus the maximum transit delay. Thus isochronicity of the video would be restored.

An example estimate of the requirements made by voice and video on an ATM network is given below.

Parameter	**Interactive Voice**	**Non-Interactive Video** @ 30Mbps
Delay	200 msec	1000 msec
Jitter	1 msec	5 msec
Throughput	8.8 Kbytes/sec	4.1 MBytes/sec
Average Throughput	3.9 kbytes/sec	4.1 Mbytes/sec
Packet sequencing required	Yes	Yes
Absence of packet duplication	Yes	Yes
Setup time	0.8 sec	15 sec

(Ferrari   RFC 1193   Requirements for Real-Time Services   November 1990)

To summarise, multimedia data is large, sensitive to delay and loss of data.

To accommodate these characteristics techniques used by the telecommunication networks to carry telephone and television traffic are required. These include compression of data, and methods of timing the transmission and replay of multimedia. Data networks and computers have been built in a different way (they are asynchronous) to telephone and TV networks (which are isochronous).

3.2. Compression

Compression algorithms and techniques are critical to the viability of multimedia networking. Uncompressed digital television requires about 140 Mbps. Since few users have this sort of network access compression is the only hope for the widespread deployment of digital video and multimedia. Compression techniques depend on algorithms implemented in software or hardware. The use of hardware is important still to enable rapid compression, and also speeds de-compression. At this time the cost of hardware is still high, from £200 to £350 for a MPEG video compression PC card. Sound cards can implement proprietary compression, and software only video compression is available in products like Microsoft Video for Windows, or for UNIX workstations.

While compression can ease the demands on networks and storage media there are several trade-offs. Since some compression techniques remove information considered to be less important a loss in resolution may result. Once material is compressed the algorithms may prevent access to single frames of video for viewing or editing. The cost of complex hardware and software and compression and decompression delay are other factors important to users.

Different uses require different compression methods. Video conferencing must be done in real time so fast encoding and decoding is needed. This is the aim of the H.261 standard. Video film distribution via cable networks, radio or CD is essentially a playback process, so encoding is not time critical, and decoding should be easy to implement to reduce consumer costs. The MPEG standards address these applications.

MIDI encoding of audio notes is not really a compression method, but almost another form of media.

Inevitably, successful compression techniques encourage the design of applications which require higher bandwidths still, such as Super Definition TV which will also require appropriate compression.

3.3. Storage

Multimedia requires high capacity storage systems. But over and above the continuous improvement in storage capacity of magnetic and optical devices multimedia raises issues relating to the format in which audio, video and data should be stored. A VHS tape stores video and audio in a structured analogue form. Digital storage techniques have hitherto used random access techniques, which do not suit the time structured format of audio and video. Since bandwidth is invariably limited in most networks users need to consider the option of local storage of multimedia data for subsequent playback. For instance the Internet is capable at present of delivering only low quality live video and audio. Local storage could enable higher quality playback for the end user. Storage systems are evolving to meet this requirement. For instance the ATML DiskBric is optimised for multimedia data streams and ATM interfaces with a 4 to 8 Gbytes capacity.

Storage devices such as CD ROM's need to be able to provide data at high speeds and in large chunks with low access times. Current CD ROM's can transfer data at around 300 kbyte/sec or higher, hold 600 Mbyte of data and have an access time of around 300 milli-secs. For some applications this is only just adequate. Even hard disc technology is strained by multimedia demands. The IEE working group P1394 is studying the use of 125 microsecond frame based storage and retrieval on disc drives.

3.4. Bandwidth

Multimedia applications, particularly those using video and images demand large bandwidths. However bandwidth for the foreseeable future will be limited. The limitations arise from the cost of installing optical fibre transmission, terminal equipment complexity and speed, tariffing regimes, switching speeds, and increasing numbers of users sharing equipment and networks. Photonic switching in trunk networks and the use of ATM on optical fibres will provide higher bandwidths [Midwinte94], but the growth of networks such as the Internet demonstrate that demand will always exceed provision.

Consequently users and applications need to formulate their demands so that bandwidth is used appropriately and efficiently. The current support for higher bandwidth offered by network technologies will be discussed in section 4.0

3.5. Quality of Service

The availability of multimedia resources places new demands on the service that a network must provide. The most important of these are the bit error rate, the packet or cell loss, delay and delay variation. Network resources need to be committed to mulitmedia data streams to accommodate the peak bit rate, mean bit rate and burstiness of the data stream. Until the advent of ATM networks users have had to live with the characteristics of the network to which they are connected. ATM provides the means to specify requirements in advance through an application. Users of ATM networks will be allocated different bandwidths and quality of service according to the application in use. For instance an audio application would request a circuit with low delay to ensure adequate voice quality. Other networks are also now capable of adapting to user requirements. An ISDN network [Ovum94 can provide additional ISDN channels on demand for higher data rates.

3.6. Platform Support

Multimedia also makes new demands on the workstations used to reproduce audio and video. Processor speeds operating systems, displays, storage medium and network interfaces and application must all be capable of handling multimedia. The Multimedia PC specification defines a defines a 33 MHz 386SX processor with 4 Mbyte of RAM, a VGA graphics card, a sound card, a large hard disc and standard peripherals as being the minimum level of machine needed. Apple Macintosh and UNIX workstations already come with many of these features. Increasingly multimedia features are being incorporated into computer motherboards to reduce the need for plug-in cards. Considerable investment is needed particularly in education to provide large numbers of multimedia ready machines.

There is no ideal platform for multimedia. All vendors are hoping that their products will benefit from the demand for multimedia. A limiting factor in the use of multimedia over networks will be not only the suitability of the networks but the availability of multimedia machines connected to the networks.

3.7. Inter-operability

The inter-operablity of platform hardware, networks and applications and multimedia formats are a major issue for multimedia users.

Providing networking standards are implemented on each platform, inter-operablity between different platforms can be achieved. It is more difficult to enable applications to use different networks, and to integrate multimedia applications in a modular fashion. Recently bodies like the Interactive Multimedia Association (IMA) and the Multimedia Communications Forum (MMCF) [MCF] have been formed to develop technical solutions to multimedia inter-operablity.

The objectives of the MMCF are to develop :-

End to end networked multimedia communication solutions independent of applications and transport technologies

Extensible Application Programming Interfaces and protocol infrastructure to support end to end multi-vendor inter-operability. This type of software has been termed 'middleware because it sits between user applications and the complexities of file formats, storage mechanisms, and networks.

The MMCF are developing a reference model for multimedia architecture to allow easy application development for independent software producers.
The IMA has undertaken some work in co-ordinating multimedia file format standardisation. This is a difficult task demonstrated by the large number of formats for audio, images and video.

Audio encoding schemes number about twenty. The most important are based on u-law, A-law and ADPCM coding using 4, 8 or 16 bits per sample:-

Sound Blaster .VOC
Sun/NeXT/DEC .AU
Windows .WAV
Sounder/Soundtools .SND
Amiga .8SVX .IFF
Apple/SGI AIFF files

Still images come in many formats. Additionally some formats support from 16 to several million colour shades. Common formats include:-

Windows Bitmap .BMP
Graphic Interchange Format .GIF
TARGA .TGA
Joint Picture Experts Group .JPEG or JPG
TIFF
PCX

PhotoCD .PCD
and for Binary images such as facsimile JBIG

Worldwide there are fifteeen video formats for analogue TV. High Definition TV is close to implementation. Digital video is governed by the CCIR-601 standard.at a bit rate of 165 Mbps. Since this is too high for most users a number of compression schemes have been developed, some proprietary, and others as international standards. The key ones are : -

Picturetel SG3
Compression Labs CD-V
Motion JPEG
Video conferencing H.261
Microsoft AVI Video for Windows
Apple Quicktime
Intel Indeo DVI
Phillips CDI
ISO MPEG-1, MPEG-2, MPEG-4

The Digital Audio Video Council is hoping to play the same role in relation to video on demand.

IBM while supporting these efforts have published a proposed LAKES multimedia kernel. IBM hope to license this software to companies developing multimedia applications which will work in a wide variety of environments and networks.

All these efforts are aimed at masking the network type and file format from the users application. If communication hardware suppliers can provide drivers to interface with this 'middleware' and application developers can write to a common interface, the current single platform, single network, single vendor characteristics of much multimedia will disappear.

4. COMPRESSION STANDARDS

Uncompressed PAL video as a digital signal needs 140 to 270 Mbps, while uncompressed digital HDTV needs 1.2 Gbps. A digitised colour picture at 35 mm film resolution needs about 80 Mbyte. One minute of 8 bit sound sampled at 22 kHz needs 1.3 Mbyte. These requirements are large for storage and transmission in computer networks. Fortunately many applications can live with video or sound that is not perfect. It is therefore possible to produce coding schemes that will drop certain information, or make a guess at likely values. Such coding can result in compression ratios of up to 200 times, so PAL video can be compressed to 1.5 Mbps. Compression must be done to a standard to enable decoding. However their are many compression methods. Microsoft Video for Windows, Quicktime are proprietary examples. A standard called JPEG is used for still colour images. For video conferencing a standard called H.261 which is part of the H.320 standards family is used. Further video standards are MPEG-1 and MPEG-2 which gives a higher quality picture to MPEG-1, but at a higher data rate. MPEG is likely to be important for video on demand. An MPEG decoder card is available for about £300. Compression of video to MPEG can now be done in real time, but the hardware to support this is expensive.

4.1. JPEG Compression

The JPEG (Joint Picture Experts Group) [ISO10] is the first international digital compression standard for multi-level continuous tone still, black and white or colour images. It typically compresses images to 1/10 or 1/50 of their original size. It is based on use of a discrete cosine transform and requires same level of processing to compress and decompress an image. JPEG aims are :-

To be applicable to any kind of continuous tone digital source image.
To be able to be implemented in hardware or software at reasonable cost.

To support the following modes of operation :

Sequential encoding, i.e. left to right and top to bottom scanning.
Progressive encoding, using multiple scans so that the image builds up gradually.
Lossless encoding, in which the compresses image can be decompressed to be identical to the original.
Hierarchical encoding in which the image is encoded at multiple resolutions, so that lower resolution displays can be accessed without having to decompress the full resolution image.

Most implementations have been only of sequential encoding. A 10 MHz JPEG chip can typically compress a full page 24 bit colour 300 dpi image from 25 Mbyte to 1 Mbyte in about one second. JPEG takes each lock of 8 x 8 source image samples and codes them into coefficients. The most important coefficients are then preserved. JPEG compression and decompression by different systems are not guaranteed to be the same, but a an accuracy test is available.

JPEG has been used for full motion video by compressing each frame of the video. For a 640 x 480 pixel 24 bit colour JPEG compresses each frame of about 1 Mbyte by about a factor of 50, which results in a data rate of about 5 Mbits/sec for 30 frames/sec. For this high data rate most implementations use a small window in the PC of 256 x 240 pixels which reduces the data rate by a factor of five. Intel DVI compression does the same. Motion JPEG does not support audio compression which must be done separately.

4.2. MPEG Compression

The MPEG (Motion Picture Experts Group) [ISO11] has so far defined two compression algorithms.: MPEG-1 and MPEG-2. A common misconception is that MPEG-2 is a replacement for MPEG-1. Each algorithms has been specifically targeted at different bit rates. MPEG-2 runs at higher bit rates than MPEG-1. There are no firm constraints in either algorithm and it is possible to run MPEG-1 video at very high rates. MPEG requires more processing power to compress video than decompress, so it is ideal for video film distribution. MPEG-1 chips on the market provide about a 200:1 compression to yield VHS quality video at 1.2 to 1.5 Mbps

The MPEG specifications allow manufacturers to implement different proprietary, but MPEG compliant algorithms. There is therefore no guarantee the the output quality of MPEG encoders will be the same. Basically the user pays for what they see. MPEG takes advantage of temporal redundancy in video pictures by specify three type of pictures:-

Intra Pictures or I-pictures are coded using only the information present in the picture itself using cosine transforms. I-pictures use about two bits per coded pixel and are used about every two seconds. Predicted Pictures or P-pictures are coded with respect to the nearest previous P or I-picture and use forward prediction of the video picture content. Bi-directional or B-pictures that use a past and future picture as a reference. B-pictures provide the most compression and average out noise. Typically two B-pictures will separate a P-picture.

Pictures may not be sent in the order in which they are displayed, if reference pictures are needed for reconstruction. MPEG also provides for synchronisation of audio and video streams.

Display Order

I	B	B	P	B	B	P

Video Stream Order

I	P	B	B	P	B	B

Stream Versus Display Ordering

Motion compensation is a technique used to enhance the compression of P and B-pictures by examining the spatial difference between pixel blocks within the picture.

MPEG-2 uses many similar techniques, and allows for many different resolutions and frame rates. It also takes advantage of motion prediction between video fields, enabling higher compression ratios. Both MPEG-1 and MPEG-2 can run at reduced resolutions by reducing the number of pixels to be encoded by a factor of two before compression.

There are optimal choices for the use of MPEG-1 and MPEG-2. Below link rates of 3.5 Mbps MPEG-1 provides better video quality (VHS standard) if the pixel input is reduced as described above. However for data rates above 5.0 Mbps MPEG-2 provides better quality, Super VHS or above video.

4.3. ITU-T H.261 Compression

The ITU-T recommendation H.261 specifies a video codec for any number of 64 kbps channels between 1 and 30 combined. There are other H. series standards linked to video conferencing, and they are all subsumed under the general recommendation H.320. Video conferencing systems built to these recommendations should be able to inter-operate. (and most do !) The H.221 and H.222 standards describe the frame and signalling structure in 64 kbps channels. The channel can be shared dynamically between video, voice and data.

There are two formats for video telephony, common intermediate format (CIF) and quarter common intermediate format (QCIF). All codecs must support QCIF. Compression is achieved by several methods. Inter-picture prediction eliminates picture information that has not changed between frames. A discrete cosine transform is also applied to individual frames. Audio is coded according to G.722 ITU-T recommendations and combined in the same channel. Chips are available to implement both H.261 and MPEG-1 in hardware.

The H.221 recommendation is a secure synchronous procedure which allows the control of several 64 kbps channels of audio and visual information and the setting up of multi-point calls. See the table below.

Audio Visual Services covered by ITU-T Recommendations

INITIAL SET	FUTURE SET
Narrowband videophone (1 and 2 x 64kbps)	Video Mail
Broadband video phone	Videotex with picures and sound
Narrowband video conferencing (m x 384kbpsand n x 64 kbps)	Video Retrieval
Broadband video conferencing	High resoution image retrieval
Audiographic Teleconferencing	Video distribution services
Telephony	
Telesurveilance	

A number of video conferencing products based on the H.320 standard are available, largely for use on ISDN. A selection is described in section 11.0

4.4. AVI, CDI and Quicktime

All these standards are proprietary. Digital Video Interactive (DVI) has been developed by Intel for the PC. A CD-ROM disc can store 20 minutes of motion video. It consists of two algorithms. The first plays back at 10 frames per second, the second at 30 frames per second. The faster algorithm requires specialist compression facilities. Hardware support is needed for playback of DVI.

Microsoft have a standard called Audio Visual Interleaved which supports small video windows at up to 15 frames per second. Apple also have a software based system called Quicktime providing 160 x 120 pixel playback at 15 frames per second.

Phillips have developed a player for full motion video discs which can manage a data rate of 1.2 Mbps of video. MPEG based techniques are used.

5. TRANSMISSION MEDIA

5.1. Copper Conductors

Copper conductors twisted together are the basis of the telephone network. As used in the analogue telephone network, with modems turning digital data into analogue tones, the data rate is limited to around 28 kbps. The introduction of the Integrated Services Digital Network (ISDN) led to the use of improved modulation and coding schemes. In standard ISDN the user can access up to 128 kbps of data at distances up to 100 metres from an ISDN socket [BT94].

Local Area networks also use copper cable twisted pairs. Ethernet running at 10 Mbps can also operate at up to 100 metres or more, depending on the grade of the cable. These physical networks will be available and economic to operate for many years. For this reason considerable effort has gone into upgrading the network protocols that can be used on them for high bit rate real time multimedia. See section 7.0 below.

A recent development is Asymmetric Digital Subscriber Lines (ADSL) [Bellcore92] technology which is aimed at using two wire copper loops at data rates of 1.544 Mbps in the network to user direction and about 600 kbps from the user to network. This is achieved by using better modulation and coding techniques. The driving force behind this technology is the delivery of on demand compressed VHS quality video to the home by Telecommunication operators wishing to compete with cable TV operators. If this technology becomes commercially available (it is undergoing trials) then the opportunity exists to deliver several ISDN B channels or even low speed ATM to the subscribers premises.

In the short term this technology is unlikely to be useful to the academic community.

5.2. Coaxial Cable

Coaxial cable is still an effective medium to deliver multimedia. Cable TV networks use coaxial cable. BT uses coaxial cable to deliver some of its Kilostream services, and Local Area Ethernet networks can operate over coaxial cable to the 10BASE5 and 10BASE2 specification. In general coaxial cable enables longer distance transmission at higher data rates than twisted pair cable, but is more expensive. One video conferencing provider has taken advantage of the potential additional bit rate available over coaxial cable and designed a proprietary video conferencing system for use over a LAN (C-Phone). Because of the higher cost of coaxial cable new LAN installations which are to be based on copper technology most often use twisted pairs.

5.3. Optical Fibre

Optical fibre transmission has been a strong enabling factor in the creation of a high capacity digital network on which networked multimedia applications will depend. On campus, the provision of optical fibre enable the creation of a high capacity backbones to Local Area Networks. However optical fibre provision to the staff desktop or student seat is still more of an exception than a rule due to the higher costs. In principal optical fibre can carry almost any type of traffic, at high data rates. It should be noted that there are different type of optical fibre, single mode, multi-mode, with different sizes, and attenuation characteristics. When upgrading from one technology to another, e.g. FDDI to ATM this may become important.

5.4. Radio Systems

Radio systems are likely to impact on multimedia in two ways. Firstly radio technology operating at microwave frequencies is available to provide wireless local area networks. Use of this technology for delivery of real time multimedia material should be treated carefully, because radio links are susceptible to fading, interference, random delays etc. For non real time use this technology is likely to perform as well as current Ethernet LANs. Secondly the introduction of digital mobile systems termed 'GSM' from a French acronym means that mobile users will have access to the same digital networks as fixed users. So in theory a mobile user will be able to connect to an ISDN users application, or an ATM users, with the only difference being one of speed. At present the bit rate of the mobile link is about 9.6 kbps. This may increase. With improvement in compression techniques the possibility of video conferencing on the move arises. Whether academics will need to adapt their video conferencing equipment to communicate with students stuck an the bus and late for a lecture is a matter for speculation!

6. SERVICE PROVIDERS

6.1. Public Telecommunication Operators (PTO)

British Telecom is still obviously the main provider of communications in the UK. Services such as Kilostream and Megastream from BT are already extensively use to connect networks. There are synchronous channels and can carry data, voice, and compressed video. There are two disadvantages to fixed links. Firstly that only two locations can be served, i.e. each end of the link. and secondly that they incur an installation and rental charge of several thousand pounds depending on distance. A rule of thumb is that if a Kilostream link of 64 kbps is to be used for less than 3 to 6 hours per day then ISDN may be more economical and flexible.

BT is also the only PTO able to provide ISDN on a national basis. New services such as Switched Multi-megabit Data Service (SMDS), Frame relay and ATM are becoming available, using the installed national optical fibre network as the common carrier. BT were chosen to provide the network for the SuperJANET pilot. Initially this was based on their 34 Mbps Plesiochronous Digital Hierarchy, but is moving to the 155 Mbps Synchronous Digital Hierarchy [Janet93 ; Clyne94]. BT are also providing Internet and video conferencing services. At present government policy prevents BT from offering video services to the home, but much of the technical work being undertaken by BT such as Asymmetric Digital Subscriber Lines (ADSL) technology anticipates the lifting of this regulation.

With the liberalisation of telecommunications in the UK a number of other alternatives to BT are available. Mercury, Energis, Water companies and others are hoping to tap regional and national market niches. The basic technologies available from these operators is very similar to those of BT. One such market niche may be the linking of educational sites, schools etc. for the distribution of multimedia learning material.

6.2. Cable TV Providers

Cable TV networks are being installed in many areas of the UK. many of these have an optical fibre backbone, and hence have the potential of carry other traffic then television distribution. Whether this is possible depends on the equipment installed, and the architecture of the system. For instance a top down tree and branch network will be difficult to adapt to two way broadband services.

However a number of providers are offering analogue telephony service, with ISDN as a possibility in some cases. Potential exists for the glass fibre infrastructure to support Metropolitan Area Networks (MANs) linking educational institutions and related organisations.

The cable industry's broadband integrated services network architecture is based on a hierarchical deployment of network elements interconnected by broadband fibre optics and coaxial cable links. Starting at the home, a coaxial cable tree-and-branch plant provides broadband two-way access to the network. The local access coaxial cable plant is connected at a fibre node, which marks the point in the network where fibre optics becomes the broadband transmission medium. The multiple links from the fibre nodes reach the head end, which is where existing cable systems have installed equipment for origination, reception and distribution of television programming. The head ends are in buildings that can accommodate weather protection and powering facilities, and hence represent the first natural place into the network where complex switching, routing and processing equipment can be conveniently located. Cable networks will continue to be asymmetric, and they will continue to deliver analogue video. But digital capabilities are being installed and a significant upstream bandwidth is rapidly being activated. The deployment of optical fibre deeper into the network is making the shared coaxial plant more effective in carrying broadband traffic in both directions. The recent announcement in the USA by Continental Cablevision and PSI to provide Internet access services is one example of the many uses that these two-way broadband capabilities can provide.

If compressed digital video is the way to deliver future video programs (including interactive video, video on demand, and a whole menu of other applications like computer supported collaborative work, multi-party remote games, home shopping, customised advertisement, multimedia information services, etc.) will be made available. In this sense the Cable TV providers will play a role in the provision of an infrastructure for multimedia capable networks.

7. LAN NETWORK TECHNOLOGIES

7.1. Ethernet at 10 Mbps

There is a vast installed base (about 40 million Ethernet nodes) of 10 Mbps Ethernet and 4 or 16 Mbps Token Ring Local Area Networks (LANs) using coaxial cable or twisted copper wire. Most new LANs used twisted copper wire cable. Ethernet uses a contention method to enable workstations attached to the same cable to share the data bandwidth on the cable. Nodes transmit to the network on demand, but continually monitor the network to see if another node is transmitting at the same time. If this occurs both nodes cease transmission, and try again later at random intervals. The throughput with such a system is limited to about half the available bandwidth or 5 to 6 Mbps. Multimedia file servers can be attached to such LANs. Both Novell and Microsoft have developed software to support video on these traditional LANs.

Video in an Intel DVI or MPEG form requires 1 to 2 Mbps per user, so it is clear that only a handful of users can run video applications simultaneously. Audio requires lower bandwidth but is sensitive to unpredictable delays. For instance an intensive file transfer or video stream could prevent an audio application from transmitting onto the LAN for several tens of milliseconds, which is sufficient to reduce the intelligibility of voice. Improved performance is possible if LAN segments are divided up into segments with only a few attached users. Switched Ethernet takes this approach to the limit by enabling only one user to access a single segment which is then connected to a higher capacity network at a central hub by a switch. The price per port of Ethernet switches ranges from £400 to £1000. The use of a switch permits the filtering of packets based on address. Switching times need to be fast enough to deliver packets at the basic 10 Mbps Ethernet line rate. An Ethernet switch would be equipped with a higher speed ATM or FDDI interface to other networks. However even with one user per Ethernet Switch port or segment contention between different applications on the same machine or incoming and outgoing traffic can occur.

7.2. Fast Ethernet

The IEEE has set out the aims of the 100 Mbps Ethernet LAN standard [Rame93] as follows:-

- Line Rate of 100 Mbps
- 100 metre distances permissible to the Ethernet Hub
- Category 3 to 5 twisted pair operation
- Equivalent error rates to 10 Mbps
- Compliance with electromagnetic compatibility standards
- Simultaneous support of 10 Mbps and 100 Mbps
- Use of RJ-45 connector

Most of the 10 Mbps parameters including the CSMA/CD media access protocol will remain unchanged and the standard is being named 100Base-T. Another 100 Mbps standard called 100VG-AnyLAN [LAN95] has also been proposed and changes the media access protocol to a demand priority system. Fast Ethernet products are only just starting to appear. Use of both standards will require a change out of workstation and hub cards. The cost of 100 Mbps Ethernet cards will be targeted to be comparable with high performance 10Base-T cards.

7.3. FDDI

The Fibre Distributed Data Interface (FDDI) [Minoli93] was the only standards based technology operating at 100 Mbps for some time. FDDI has experienced slow market penetration due to the high cost of cards, still around £600. The first version of FDDI was developed as a campus trunk network for data.
Logically FDDI consists of a dual ring, but it may be implemented as a physical star. Key feature of FDDI include :-

- Shared medium based on a token passing medium access control.
- Compatibility with IEEE 802 LANs.
- Ability to use a wide range of physical medium including multi-mode fibre, single mode fibre, shield and unshielded twisted pair.
- Operation at 100 Mbps
- Support for 500 workstations
- Maximum fibre length of 200 km.
- Ability to allocate bandwidth dynamically so that both 'synchronous and asynchronous' services can be provided.

An upgraded FDDI standard called FDDI II has been designed. In addition to the data, packet switched mode in FDDI an isochronous circuit switched service is made available by imposing a 125 micro second frame structure. The 100 Mbps bandwidth can be split between packet data and up to fifteen isochronous channel operating at 6.144 Mbps each.

7.4. Iso-Ethernet

Architecturally, Iso-Ethernet, also known as IEEE 802.9 [Minoli94] is a multiplexing of four separate channels of information: a 10 Mbps packet channel (P) with the IEE 802.3 CSMA/CD media access protocol, 6.144 Mbps of isochronous information organised in 96 B channels of 64 kbps each, one 64 kbps D-channel for signalling, and a 96 kbps M-channel for maintenance. Additionally framing is added to allow for synchronisation with a wide area network. The complete data stream is modulated with 4B/5B coding. The total bandwidth is similar to that used in 10Base-T Ethernet so require similar cable technology and hardware to existing Ethernet systems. A typical network will consist of Iso-Ethernet terminals connected to LAN hubs. The hubs may be connected by a backbone network.

Iso-Ethernet combines the best properties of current IEE 802 LAN and ISDN networks. Multimedia applications requiring isochronous channels can use any combination of ISDN B channels for audio and video according to the desired quality requested. Wide area interfaces from synchronous data channels such as Megastream or SMDS can be connected to the Iso-Ethernet Hub. Narrow band basic rate ISDN and Primary rate ISDN can provide Wide Area connections using Q.931 signalling. ATM protocols can be added. The cost of Iso-Ethernet connections will be targeted to be competitive with existing high speed Ethernet cards.

At least one video conferencing application developer has an interface in development for this type of LAN.

7.5. Proprietary LANs

To enable video conferencing over existing LAN cabling some vendors have adopted proprietary solutions which replace the existing network hardware, but retain the cable infrastructure. On example is the C-Phone system [Griffin94] which uses additional modulation above the 10 Mbps Ethernet spectrum to carry video and audio channels. Another technique is the emulation of ISDN over Ethernet CSMA/CD employed by TELES [Schindler94]. This is satisfactory for the video component of video conferencing but the audio component must be carried over the telephone.

Such solutions are of limited applicability and have been superseded by the Iso-Ethernet LAN.

7.6. Local ATM

ATM is really a network protocol because it operates at layer 2 of the OSI model. The physical transport of ATM has been standardised for fibre optics at 155 Mbps, 100 Mbps and synchronous digital networks. Standards are recently been approved for 51 Mbps and IBM with about 20 other vendors has re-submitted a proposal for 25 Mbps ATM transport.

The implementation of an local ATM network requires an ATM switch connected to a high speed backbone. Initially users can remain on 10 Mbps Ethernet segments connected to the ATM switch. As requirements rise, direct higher speed native ATM connections can be provided. Use of existing applications that run well over Ethernet should be possible in the connection-less ATM class of service. effectively emulating an Ethernet LAN.

While some organisations will be able to provide optical fibre to the desk top for delivery of 155 Mbps, many have unshielded twisted pair (UTP) Ethernet LANs. The 25 Mbps specification may be worthwhile implementing in such situations if the cost of adapter cards and ATM switches is comparable with quality Ethernet cards. It is likely that the cost of implementing 51 Mbps over twisted pair cables will be considerably higher [CT95].

One such ATM adapter card for an ISA bus PC is able to operate over Category 3 UTP wiring. It provides NDIS-3, ODI and native ATM socket type application programming interfaces. It is anticipated to cost around $400.

One pair of a cable is used for transmission, another for reception and token ring technology. ATM service classes are implemented in part by allowing different priority queues. Transmission is compatible with 16 Mbps Token ring physical components. A switch to support the adapters is available with 12 ports. Two port 155 Mbps modules are available for trunk connections. An Ethernet transparency module permits LAN emulation.

Obviously this type of low cost ATM implementation is very new and will require evaluation, but other vendors are certain to follow, so the option of low cost local ATM is a real possibility. At least one video conferencing application developer has an interface in development for this type of LAN.

8. WAN NETWORK SERVICES

8.1. Frame Relay

Frame relay is a connection oriented services operating at n x 64 kbps or 2.048 Mbps. It has evolved from X.25 packet switching and aims to reduce network delays, protocol overheads and equipment cost. Error correction is done on an end to end basis rather than a link to link basis as in X.25 switching. Frame relay can support multiple users over the same line and can establish a permanent virtual circuit or a switched virtual circuit.

Like ATM it is a protocol which must be carried over a physical link such as a Kilostream or Megastream link. While useful for connection of LANs, the combination of low throughput, delay variation and frame discard when the link is congested will limit its usefulness to multimedia.

8.2. SMDS

The Switched Multi-megabit Data Service (SMDS) [King93] is a new switched broadband data service. One of the first users of the service in the UK have been SuperJANET sites. SMDS provides a switched connectionless data service at speeds of 34 Mbps (at present) for connection of LANs. It uses variable length packets up to 9188 bytes in length, each of which carries an address in the E.164 format (ISDN uses this address format too) SMDS packets are transported in the public network using the Distributed Queue Dual Bus (DQDB) IEE 802.6 standard which uses packets fixed at 53 bytes. There is some overhead from this conversion process which reduces the bandwidth available to users to about 75% of the line speed. There are several access classes that limit the sustained data rate and burst data rate that can be injected into the network by a user. These access restrictions may result in discard of packets that exceed a certain limit. SMDS does not support timing. The higher speeds of SMDS will be of benefit to multimedia applications seeking to transfer large volumes of data quickly, but the lack of a time structure will reduce the video conferencing quality obtainable.

8.3. The Integrated Services Digital Network (ISDN)

In the real world the delivery of multimedia requires a widespread network capable of delivering at high data rates. The current implementation of ISDN in the narrow band form is the best access and delivery medium available. ISDN is seen by many in the industry as the ramp through which multimedia networking will gain acceptance. The installed base of ISDN is growing rapidly (30,000 line per month in Germany). ISDN is able to provide connections throughout the world. In Europe the Euro-ISDN agreements between operators is valuable.

ISDN offers point to point delivery, network access, and network interconnection for multimedia. Different data rates from 64 kbps up to 2 Mbps are commercially available which can meet many needs for transporting multimedia. Call set-up times are under one second.

ISDN will be the feeder network for broadband ISDN based on ATM standards. Initially the ISDN and ATM networks will be overlaid on top of each other, but users of ISDN will eventually be able to call an ATM user directly and be allocated an appropriate amount of bandwidth. The development of 'middleware' will enable applications to communicate over mixed networks.

Although ISDN could be cheaper, particularly in the UK (currently £300 to connect), it is likely to be cheaper than ATM connections and more widespread in availability for a long time. It is therefore an important tool in bringing multimedia applications to a wide range of users. The idea that multimedia can only be delivered on broadband networks is erroneous as the assertion that only a Macintosh can deliver multimedia.

The cost of ISDN hardware was high, but is now decreasing. Terminal adapters are available from £400 upwards, and PC cards for £300 upwards. Video conferencing cards cost around £3000, (BT's VC8000 card). Costs of ISDN equipment are much lower in Germany and some of these products are beginning to appear in the UK under the Euro-ISDN banner.

British Telecom are pursuing a strategy to make ISDN the preferred option for all multiple (2 or more) exchange line requests by the mid 1990s. ISDN is accessed through one of two services, named by the CCITT as Basic Rate Access (BRA) and Primary Rate Access (PRA).

Basic Rate Access (BRA) provides an ISDN user with simultaneous access to two 64 kbps data channels using the existing twisted pair copper telephone cable. The B.T. basic rate ISDN service is called ISDN2. The connection cost of ISDN2 is currently £300. Rental for the equivalent of two PSTN telephone lines is £384 per year.

Each data channel is referred to as a B-channel and can carry voice or data. Another channel, the D-channel, operates at 16 kbps and is used for signalling between user devices and the ISDN. The total data rate of BRA is therefore 144 kbps. The two B-channels and the single signalling channel give rise to the term '2B+D'. BRA is also referred to as I.420, after the CCITT recommendation. Basic rate ISDN is intended for low capacity usage, such as that required for small businesses.

British Telecom's primary rate ISDN service is known as ISDN30. This service is generally available throughout the UK and is based on the CCITT recommendations for primary rate ISDN. Mercury Communications Limited also offer a primary rate service known as 2100 Premier. Although this service is largely based on CCITT recommendations, it still utilises the some proprietary signalling.

Primary rate access can carry 30 independent voice or data channels, each at 64 kbps. The structure has a 64 kbps D-channel for signalling between devices and the network, and a 64 kbps channel for synchronisation and monitoring. The total data rate of PRA is 2.048 Mbps.

Primary rate access is often referred to as '30B+D' because of the number of B-channels and D-channels, or I.421 because of the CCITT recommendation from which it is taken. This form of access is primarily intended for use in situations which require a large transmission capacity, such as when organisations make voice and data calls through an Integrated Services PBX.

There are two standard ISDN connectors. For accessing basic rate ISDN, an RJ-45 type plug and socket (similar to a telephone plug) is used using unshielded twisted pair cable. Access to primary rate ISDN is through a coaxial cable.

The ISDN passive bus, which can be a maximum of 1 km in length, is a cable which in user premises. It enables up to eight user devices to be attached to the basic rate ISDN interface. Since there are only two B-channels, only two of the eight devices can communicate at any one time. For this reason, each device must contend for access to the passive bus.

ISDN signalling information, carried in the D-channel, is used to establish, monitor and control ISDN connections between users as well as instigating, the audible ringing or engaged tones.

The ISDN numbering system is similar to the contemporary telephone numbering system. Each B-channel has its own unique directory number which allows access to different terminal types (such as telex or facsimile devices). Each terminal type has an identity code which ensures that it only communicates with similar terminals.

The equipment available for ISDN includes Terminal Adapters, ISDN internal computer Terminal Adapter cards, Video Conferencing PC cards, and LAN access gateways or bridges, some of which are based on PC cards or stand alone boxes. Products are available from in this country from the USA, UK, France, and Germany. The market for ISDN is most developed in Germany.

Internal Terminal Adapters from Germany will all inter-work with each other, products developed in the UK are all totally and individually proprietary and will not inter-work in many cases. Many manufacturers are awaiting the dust to settle on the competing application programming interface standards from the European PTT body ETSI.

It is possible to avoid all the problems of API standards for internal computer adapters by using an external ISDN Terminal Adapter. Since the speed of most serial ports on a PC has been limited to about 19.2 kbps until recently, this approach has not been viable. However recently internal PC cards which will work asynchronously up to 115 kbps have appeared, which could have applications in multimedia work when used with an appropriate external Terminal Adapter.

8.4. ATM

The ATM technology referred to in section 7.6 is equally effective in Local and Wide Area Networks. However in the Wide Area context is one of many possible services offered by telecommunications operators which have been mentioned in this section. The costs of public ATM provision are not yet known. The costs of the first phases of SuperJANET which employs ATM between some twelve institutions over 34 or 155 synchronous digital links from BT have been funded by Research Councils. It is reasonable to assume that costs of subscribing to ATM services will be related to the required bandwidth and other user requirements such as quality of service. Competitors to ATM will include fixed links, Frame Relay and SMDS.

Wide area network interfaces will operate at 155.52 Mbps and 622.08 Mbps, both requiring optical fibre interfaces. The standards for ATM were first developed by the International Telecommunications Union (ITU) in the 1980's. Co-odination of the implementation of ATM is in the hands of the ATM Forum which consists of a wide cross section of companies. The ATM Forum has developed implementors specifications to try to insure that equipment manufactured by several companies can inter-operate. The latest is the UNI Specification (Version 3.1).

ATM uses small constant size packets to reduce and control delay. Control of the priorities of packets in ATM switches enables guaranteed delivery of information. ATM can emulate ISDN channels and Ethernet characteristics. ATM is seen as a universal technology which can be used over physical LANs and WANs and may be able to carry both asynchronous and isochronous data. ATM may also be delivered over ISO-Ethernet.

ATM network technology has strong industrial support and is already carrying traffic over the academic SuperJANET network. ATM can support different speeds, traffic types and quality of service matched to applications. ATM cells coming from a user are guaranteed delivery at the other end with a high probability and low delay. A cell is a short block of data 53 octets in length including 5 octets overhead. The performance aims are :-

A cell loss ratio of less than 1.7 x 10 -9

A cell transfer delay across the network of 150 micro second per switch. plus transmission and propagation time.

A cell delay variation for one cell in 10^8 of no more than 250 msec

ATM users have a dedicated connection to a high speed ATM switch. Switched virtual circuits are set up by the switch to a destination. Additionally ATM users can select a preferred network provider to service the connection.

ATM signalling establishes a "hard state" in the network for a call. "Hard state" implies that the state of a connection in intermediate switching equipment can be set and once established it will be maintained until a message is received by one of the ends of the call requesting a change in state for the connection. As a result, an ATM end system (this could be a workstation with an ATM adapter or a router with an ATM interface) receives guaranteed service from the ATM network. The ATM network is responsible for maintaining the connection state. ATM termination points must be responsible for changing the state of the connection, and specifically informing the ATM network to establish, alter, or close the connection.

Each ATM end point in a network has an ATM address associated with it to support dynamic connection establishment via signalling. These addresses are hierarchical in structure and globally unique. As a result, these addresses are routed. This allows ATM networks to eventually support a large number of ATM endpoints once a routing architecture and protocols to support it become available.

Several classes of ATM service have been defined:-

Constant bit rate, connection-orientated with timing
Variable bit rate, connection-orientated with timing
Variable bit rate, connection-orientated without timing
Variable bit rate, connectionless without timing
Un-restricted variable bit rate, connection-orientated or connectionless

Each of these categories are further specified through network provider objectives for various ATM performance parameters. These parameters may include cell transfer delay, cell delay variation, and cell loss ratio. The connection traffic descriptor specifies characteristics of the data generated by the user of the connection. This information allows the ATM network to commit the resources necessary to support the traffic flow with the quality of service the user expects. Characteristics defined in the ATM Forum UNI specification include peak cell rate, sustainable cell rate, and maximum and minimum burst sizes.

The variable and constant bit rate, connection-orientated with timing services are most appropriate to the transport of real time multimedia. However other services could be useful for transfer of multimedia material in less than 'real' time.

9. MULTIMEDIA AND INTERNET PROTOCOLS

9.1. Existing Internet Protocols

The Internet is a more of a phenomena than a network, but is important when discussing multimedia because a popular Internet Application, the World Wide Web is capable of accessing and displaying multimedia formats such as pictures, audio and video. The current Internet has thrived and grown due to the existence of TCP implementations for a wide variety of classes of host computers. These various TCP implementations achieve robust inter-operability by a "least common denominator" approach to features and options.

The system of connected networks which comprise the Internet has also been used to carry live audio and video. Extensions to the TCP/IP protocols currently used have been proposed as Real Time Protocols (RTP). Broadcast of audio and video has taken place on the Multicast Backbone (MBONE), by allocating higher priority to audio and video information from within routers.

The MBONE is being developed as a technology for low cost multimedia. Multicasting within MBONE enables multiple destinations to share the same information without replication. Internet routers and workstation software require some modifications to support multicasting. A virtual network has been implemented over the IP network to bypass routers which do not support multicasting, and to enable some bandwidth to be reserved for multicasting. However audio and video on the MBONE must still compete with other traffic on parts of the network. This limits the quality of both the voice and video obtainable.

However current transport protocols exhibit some severe problems for high performance, especially for using hardware support. Existing protocols require a processing overhead which takes longer than the transmission time on high speed networks. For example, TCP places the checksum in the packet header, forcing the packet to be formed and read fully before transmission begins. ISO TP4 is even worse, locating the checksum in a variable portion of the header at an indeterminate offset, making hardware implementation extremely difficult.

Special purpose transport protocols have been developed. Examples include special purpose transport protocols such as UDP (user datagram protocol), RDP (reliable datagram protocol), NVP (network voice protocol), PVP (packet video protocol) and XTP (Xpress Transfer Protocol), XTP fixes header and trailer sizes to simplify processing and places error correction in the trailer so that the code can be calculated while information bits are being transmitted. Flow, error and rate control are also modified in XTP. Examples of XTP applications include :-

A video-mail demo over XTP/FDDI that uses a proprietary Fluent multimedia interface and standard JPEG compression. This PC-based demo delivers full frame, full colour, 30 frames/s video from any network disk to a remote VGA screen.

Voice can be multicasted over XTP/FDDI. A simple multicast is distributed to a group with a latency of around 25 ms, where the latency represents delay from the voice signal from the microphone to the audio signal to the speaker.

Commercially, Starlight Networks Inc., migrated a subset of XTP into the transport layer of its video application server. By using XTP rate control, full-motion, full-screen compressed video is delivered at a constant 1.2 Mbps, over switched-hub Ethernet to work stations. This network delivers at least 10 simultaneous video streams.

The Internet physically depends on the capabilities of the underlying networks. If TCP/IP protocols are to be used in a world equipped with ATM capable of transporting audio and video efficiently then any adaptation of current TCP/IP protocols will need to be tailored to the needs of multimedia.

9.2. New Internet Protocols

A successor to the current version of TCP/IP Version 4 is being discussed. While some of motivation behind this is due to the need to increase the address space available, the opportunity is also being taken to review the need to increase the performance of Internet protocols, for multimedia applications. The Internet Engineering Task Force has resolved to move towards develop a replacement for the current TCP/IP Version 4 called IPng (IP next generation) [Brazdziunas94]. Effective support for high quality video and audio streams is one of the critical capabilities that is being called for to capture the attention of network operators and information providers of interactive broadband services (e.g., cable television industry and partners). Such additional features will also help overcome resistance to change. The intention is that IPng should last for the next 20 years.

The delivery of digital video and audio programs requires the capability to do broadcasting and selective multicasting efficiently. The interactive applications that the future cable networks will provide will be based on multimedia information streams that will have real time constraints. The largest fraction of the future broadband traffic will be due to real time voice and video streams. It will be necessary to provide performance bounds for bandwidth, jitter, latency and loss parameters, as well as synchronisation between media streams related by an application in a given session.

The potential for IPng to provide a universal inter - networking solution is a very attractive possibility, but there are many hurdles to be overcome. One of these is that a new deployment of IPng threatens the existing network investments that business has made and the other is that business users actually buy applications -- not networking technologies. Some of the the aims of IPng development relevant to multimedia are set out below:-

Two aspects are worth mentioning. First, the quality of service parameters are not known ahead of time, and hence the network will have to include flexible capabilities for defining these parameters. For instance, MPEG-2 packetised video might have to be described differently than G.721 PCM packetised voice, although both data streams are real time traffic channels.

Network media speeds are constantly increasing. It is essential that the Internet switching elements (routers) be able to keep up with the media speeds. A proper IPng router should be capable of routing IPng traffic over links at speeds that are capable of fully utilising an ATM switch on the link.

Processing of the IPng header, and subsequent headers (such as the transport header), can be made more efficient by aligning fields on their natural boundaries and making header lengths integral multiples of typical word lengths (32, 64, and 128 bits have been suggested) in order to preserve alignment in following headers. Optimising the header's fields and lengths only for today's processors may not be sufficient for the long term. Processor word and cache-line lengths, and memory widths are constantly increasing.

There are now many different LAN, MAN, and WAN media, with individual link speeds ranging from a ones-of-bits per second to hundreds of gigabits per second. There will be multiple-access and point-to-point links on a switched and permanent basis. At a minimum, media running at 500 gigabits per second will be commonly available within 10 years. Switched circuits include both "permanent" connections such as X.25 and Frame Relay services and "temporary" types of dial up connections similar to today's SLIP and dial up PPP services, and perhaps, ATM SVCs. Any IPng will need to operate over ATM. However, IPng still must be able to operate over other, more "traditional" network media. A host on an ATM network must be able to inter - operate with a host on another, non-ATM, medium.

Multicasting has been used with a limited degree of success to support audio and video broadcasts. Tests at ULC used DVI video compression with a data rate of up to 600 kbps and achieved a frame rate of up to 5 frames per second. Tests of H.261 video, also from ULC encountered delays of up to 12 seconds on the IP network. Some of this delay could be buffered out, raising the average delay. The conclusions were that slow TCP error recovery mechanism was inappropriate, and the UDP protocol may give better results.

On mixed protocol networks IPv4 currently uses the local media broadcast address to multicast to all IP hosts. This method is detrimental to other protocol traffic on a network. The ability to restrict the range of a multicast to specific networks is also important. Currently, large-scale multicasts are routed manually through the Internet. User configurable Multicast Addressing is vital to support future applications such as remote conferencing.

For many reasons, such as accounting, security and multimedia, it is desirable to treat different packets differently in the network. For example, multimedia is now on our desktop and will be an essential part of future networking. Multimedia applications need to acquire differing grades of network service, for voice, video, file transfer, etc. It is essential that this service information be propagated around the network. To support multimedia features will be needed such as policy-based routing, flows, resource reservation, type-of-service and quality-of-service .

9.3. Internet Protocols and ATM

An example of the issues in implementing this requirement is proposed IPng support for ATM. Current IP does not provide much support for a quality of service specification and provides no support for the specification of link level performance needs by an application directly. This is due to the fact that only a single type of link level performance is available with link technologies like Ethernet. As a result, all applications over IP receive the same level of link service.

ATM is a link level technology which provides the potential capability for applications at the TCP level to map to a single ATM virtual circuit for transport across an ATM network(s) customised to the network performance and traffic requirements for that application. The future Internet will be comprised of both conventional and "sophisticated" link technologies. The "sophisticated" features of link layers like ATM need to be incorporated into an internet where data travels not only across an ATM network but also several other existing LAN and WAN technologies. ATM allows for each logical channel to have a customisable set of performance and quality of service characteristics. Hence a single ATM link level media appears like an array of link level technologies each with customisable characteristics.

There are several parameters required to map ATM services from a higher level service like IPng [Brazdziunas94]. These ATM parameters can be categorised as: addressing parameters, connection QOS - related parameters, connection management information, and ATM virtual circuit identifier. The first three categories provide support for ATM signalling. The last parameter, a connection identifier that maps IPng packets to ATM virtual circuits, provides support for an ATM virtual circuit per application when the end-to-end connection travels across an ATM subnetwork(s) (this does not assume that ATM is the only type of subnetwork that this connection travels across).

An ATM virtual circuit is established based upon a user's traffic characteristics and network performance objectives. These characteristics which include delay and throughput requirements can only be defined by the application level (at the transport level or above) as opposed to the inter - networking (IPng) level. For instance, a file transfer application transferring a 100 Mbyte file has very different link level performance requirements than a video application. The former requires a high throughput and low error rate connection whereas the latter requires a guaranteed bit rate. Applications will be responsible for reserving the required type of connection from the ATM link.

9.4. ITU-T Standards

The ITU-T is working on a the I.374 standard 'Network Capabilities to Support Multimedia Services'. This will cover service control, connection management, service management, and multimedia interaction for networks carrying multimedia traffic. Multimedia application programming interfaces (APIs) will need to be able to interface to this standard. The capabilities being considered for this standard [CCITT92] are listed below.

Service Control includes:-
Call Set-up which cover the establishment of a call; and
Call Release which releases a call and all media connections
Establish Connection establishes a link between two or more users
Join adds more users to a multipoint configuration
Leave enables users to leave a multipoint conference call
Disconnect completely releases a call
Allocate adds another medium to a call
De-allocate removes a medium from a call

Connection Management

- Capability to control virtual circuit connections (for ATM)
- Support of point to point or multipoint and broadcast configurations
- Change of media from within a call by all parties, e.g. ATM to ISDN
- Negotiation of Quality of Service
- Reconfiguration of a multi-party call

Allowing different media to be used with different users in a multi-party call, e.g.. an audio call with one user and a data call with another.

Service Management

This includes aspects of synchronisation. Different information types may experience different delays through the network. Issues to be addressed in the standard include:
Differential time delay between media carried on separate virtual or physical channels.
Inter - channel synchronisation
Inter - working between different coding schemes, e.g. audio coding schemes
Support for signalling through the network between users

Multimedia Interaction

This includes the multiplexing of different media into a single stream onto physical or virtual channels and the ability to change the bandwidth allocated to different media such as audio and video from within a call.

Further issues are raised for network operators by multimedia. Two important ones are charging mechanisms for multimedia services and performance limits.

10. MULTIMEDIA PROGRAMMING INTERFACES

10.1. Design Issues

There are conflicting realities to consider when designing a common multimedia architecture:-

- It is desirable to have a single architecture or protocol stack for multimedia and networks.
- In practice there will always be more than one protocol stack.
- Even within a single successful architecture forces of evolution will lead to periods of multiple protocols.

The fact that the Internet was based on a single, common, virtual network service (IP) allowed a ubiquitous underlying communication infrastructure to develop upon which a set of services could be provided to the user. This also allowed for a large market to develop for applications which were built upon the underlying communications.

When there is a single common layer the selection of applications becomes the province of the end-user rather than the intermediate network provider such as a telecommunications, cable operator or computer services department. By having this common underlying infrastructure, users are able to select their desired/required application services based on their unique needs, with assurance that the intermediate networking service will support their communication requirements.

Designers of a common multimedia architecture face three problems :-

> At what level in the protocol stack should common interfaces be provided? Should the network level be the common point, able to handle many different requirements, or should a common transport layer be able to request the service of many networks?
>
> Secondly any common architecture will have to recognise the existence and be able to operate with other architecture sets. For instance it is conceivable that the MMCF reference model may have to inter-operate with a Microsoft or IBM model, according to how the industry politics play out.
>
> Thirdly each architecture will have to share resources at some point in the chain of memory, processing, network requests, and transport of packets on the network.

Achieving inter - operability and resource sharing is difficult. For example, sharing bandwidth on a link may not work effectively if one protocol suite backs off in its demands and the other does not. Inter - operability and resource sharing both require co-operation between the various developers and users. The process of co-operation is a dynamic one, when it works. Attempts to achieve inter - operability and resource sharing may bring the multiple architectures into some level of harmonisation, even if it is just to simplify the problems of inter - operability and sharing. Together with the normal process of evolution, there may then be lead to changes in one of the architectures, as well as the other suites. Thus, the need for new technologies leads to a natural process of diversion. The process of harmonisation leads to conversion.

The Internet community have decided that there should be single next generation Internet (IPng) protocol and are developing methods to ease the transition from IPv4 to IPng. The intention is to promote different approaches at the applications layer and let the users market decide which one is best for their needs. The Internet community have therefore staked a claim to the network layer, and it will be up to multimedia users and developers to decide if a similar common layer or architecture below multimedia applications is of benefit to users and achievable.

10.2. Multimedia Communication Models

In fact the Multimedia Communications Forum (MMCF) are developing a reference model [Zakowski94] for multimedia architecture to allow easy application development for independent software producers. The first result of this work will be Transport Services Interface which will be a type of multimedia WinSocket specification, able to negotiate bandwidth, delay, priority and other quality of service parameters required by multimedia. AT&T, Intel, Motorola, National Semiconductor, and Seimens are some of the companies involved. The MMCF work is intended to complement the work of the Interactive Multimedia Association which has initiated a compatibility project to develop solutions to multimedia cross platform compatibility issues.

The objectives of the MMCF Architecture Reference Model are to provide:-

End to end networked multimedia communication solutions independent of applications and transport technologies

Extensible Application Programming Interfaces and protocol infrastructure to support end to end multi-vendor inter-operability

Support horizontal and vertical integration of applications.

Support distributed networking and computing architecture.

Easy application development for independent software developers.

The MMCF have developed a concept of *Middleware* as part of the Multimedia Architecture Reference Model. Middleware is a suite of applications, functions and programming interfaces that reside above the transport level in the OSI stack, but below the application level. The interface from the Middleware domain to the application and transport levels will be via application programming interfaces (APIs). It is not clear if the OSI presentation and session layers will be part of this architecture.

The intention is that through the use of these application programming interfaces and middleware 'software' that flexible, inter-operable multimedia communication applications can be built. Control of Quality of service requests and negotiation will be part of these interfaces. The MMCF has already proposed the multimedia transport level interface (TS1), and further work will continue during 1995.

Intel is also part of a Personal Conferencing Work Group along with AT&T, DEC, Hewlett Packard, Compaq, Compression Labs, NEC, Novell and Lotus, which are about to release a Personal Conferencing Specification. Part of this specification may include specifications for converting H.320 video conferencing data into a format which can be transported on a LAN.

Both the French and German ISDN communities have submitted outline proposals is being put to the European Telecommunications Standards Institute (ETSI) for a multimedia programming interface appropriate for ISDN and other wide area networks. Initial inspection indicates that neither proposal yet tackles the problems of multi - network multimedia communications.

Applications which are unable to work with similar applications from a different manufacturer will not reach the widest market. Even Microsoft appreciates this fact. Of course manufacturers will try to drive standardisation efforts to their own advantage, but users do have a voice. The standardisation of video conferencing around the H.320 series of documents, desktop conferencing with the T.120 series, and electronic mail with X.400 and SMPT MIME extensions are examples.

10.3. First Implementations

Implementation of TS1 is expected by the end of 1995. The API primitives include:-

Administration - e.g. socket, bin, close, register

Communications Session - e.g. listen, look, connect, multicast, join

Information Transfer - e.g. receive, send

Supplementary - e.g. hold, retrieve, conference, add, drop, transfer

Quality of Service - e.g. priority, message size, bit rate, burst rate, delay, delay variation, cost, configuration, symmetry, reliability, type of service, security, negotiation.

IBM has designed a Lakes Architecture for Collaborative Networking [Aldred94]. This is meant to be an open platform for personal video conferencing. However the package includes the IBM Person to Person application. Lakes was not designed around audio or video applications but claims to be capable of handling both. Lakes features are outlined below:-

Multimedia Communications Logical channels between applications, quality of service, synchronisation and data conversion, and call management.

Device Independence For audio, video, clipboard, display and capture

Resource Manager Management of communications and collaborating applications

Programming Interface For interface to file systems and network directory services

Lakes also claims to inter - operate with the H.320 and T.120 video conferencing / multi party conferencing standards. Lakes is probably the first of several attempts that will be made to create tools to handle multimedia communications.

11. NETWORKED MULTIMEDIA APPLICATION DESIGN

11.1. Real Time Multimedia Applications

A number of real time applications are available for use over networks based on computers, which are capable using audio or video. The two networks over which applications have been developed td date include ISDN, Local Area Networks, and Local Area Networks connected over high speed ATM or SMDS connections. The table below summarises the most important applications with examples.

Application	ISDN	LAN's
Desk Top Conferencing	Proprietary Desktop conferencing systems	wb... an Internet white board tool, proprietary software.
Video Conferencing	H.320 video conferencing	vat ... AVisual audio internet tool, ivs... video conferencing software
Video Mail	Based on H.320 video conferencing	X.400 and MIME electronic mail with video content.
Remote Image viewing and manipulation	Medical applications, eg X-Rays	Medical Applications. Multicast of JPEG Satellite images
Information Kiosks	Proprietary systems	
Distance Learning	File Server Access	File Server Access

11.1.1. Desk Top Conferencing

Desk Top Conferencing is a means of working with a remote user on their computer running the same application. Typically a drawing or spreadsheet will be transmitted via the ISDN line. A voice conversation between the two users can be held at the same time. Within desk top conferencing remote control of a computer is possible. Since these applications must reproduce the screen of a PC on another, the speed of the connection between the two computers is important. Products are available from Fujitsu for LAN and ISDN use and IBM (Person to Person). Desktop conferencing is also available over some of the video conferencing products such as the Olivetti PCC.

The standardisation of multimedia conferencing using the T.120 series of International Telecommunication Union standards is now widely accepted. Products based on these standards are likely to appear on the market next year from several manufacturers. The applicability of the T.120 standards will not be limited to WAN or ISDN networks, but will be appropriate to Local Area Networks too.

The T.120 standards will apply to terminal with audio, audio and interactive video, or interactive graphics or all three. It will support point to point and multi-point conferencing. ISDN is the initial focus. Work is underway to define ATM support. Areas to be standardised include still image, annotation, application sharing, conference control, and multi-point conferencing. The standard set will be largely defined in early 1995.

11.1.2. Video Conferencing or Videophone products

Until recently the only implementations of video over the telephone network have been poor quality video phones or very expensive video conferencing for executives.

A selection of video conferencing products is listed below:

VidiMac for the Apple Mac based on the Planet ISDN card, which uses motion JPEG techniques.

IBM, Fujitsu, and Olivetti have video conferencing products based on the BT VC8000 PC card. The IBM product is called Screen Call, the Fujitsu is called Team Vision, and the Olivetti is called PCC. All three use the services of the VC8000 card in different ways. The common feature is support for H.320 video conferencing. Support is available in all three for file transfer, whiteboards, and remote control. All retail for around the £3000 mark.

Intel have released a product called ProShare, which is very competitive, but at present will not work to the H.320 standard.

Northern Telecom have a product called Visit 2.0 which runs on a PC or Mac. It uses external ISDN Terminal adapters and audio transmission need to be via an independently set up telephone call.

Invision have made proprietary non H.320 video conferencing available for LAN connections running LAN protocols such as TCP/IP. Frame rates range from 1 to 20 frames per second with corresponding data rates fro 64 to 512 kbps.

Picturetel Live PCS100 is one of the more expensive PC based products, but manages excellent quality through good implementations and extensions to the H.320 standard. Full CIF pictures are available. At QCIF resolution of 7.5 frames per second are available.

The Olivetti PCC based on the BT VC8000 PC card provides QCIF at 15 frames per second, audio, file transfer, whiteboard, remote application control, remote form entry, image capture and transfer, and a text chat mode. All of these applications are programmable and should users should use this feature to customise the screen interface, which is based on standard Windows 3.1 menu bars and buttons. Interfaces are available for video and audio from other sources. All of these facilities are programmable and customisable. H.320 standards are supported. T.120 conferencing will be available in 1995. Price is about £3,500.

Most of these products will provide ISDN applications such as file transfer, but the additional facility of a quarter screen video picture of the caller will be available in colour. Compression techniques are used to improve the quality of moving pictures. Sadly these some products use different compression methods so they cannot communicate with each other. But with manufacturers moving to the H.320 series of compression standards, at least for video phone communications this situation may improve. Other standards e.g. MPEG are be better for straight video broadcast.

11.1.3. Video Mail

Video mail can be delivered via X.400 mail, or Internet mail with MIME extensions for file attachments. Obviously transmission times can be longer. An appropriate viewer (hardware or software) is then needed to replay the mail.

An alternative approach has been taken by Olivetti, who will release in 1995 extensions to the PCC video conferencing system. These extensions will enable video and voice messages to be left on a PC for viewing by a local or remote user connected over ISDN.

11.1.4. Image Viewing

An image database is considered to be a useful application for attachment to existing databases, or for standalone use. Products have been design for both general use and access via ISDN. Most products are designed not as stand alone image viewers but as part of an information system or kiosk for tourism or specialist applications. An example of the latter is a system from On Demand Information which markets a database for the building industry, accessible over ISDN.

11.1.5. Information Kiosks

Information kiosks are starting to appear with multimedia features. Information kiosks have commercial applications in tourism, government information and education. Extensive multimedia material can be held on local hard discs and updated at regular intervals, manually or via a network connection such as ISDN. The Olivetti PCC video conferencing system allows video conferencing to be incorporated in the design of an information kiosk. All the functions of the system can be programmed from within a high level language such as C++ , Toolbook or Visual Basic.

11.1.6. Distance Learning

Distance learning applications can held on a network file server. They can then be accessed from the local LAN or by ISDN. The high speed of ISDN means that the remote connection of PC's to LANs is now viable. The response of a distance learning application running over ISDN instead of a local LAN will depend on its design and the amount of visual material. transfer of audio material will also take time and require a remote workstation set up for audio playback.

A remote access connection over ISDN to a LAN can be set up in three ways:

(a) Setting up a serial synchronous or asynchronous connection using terminal adapters to link the PC into a conventional LAN bridge or router.

(b) Placing an ISDN card in the remote PC and a similar card in a dedicated PC which is also connected to the LAN via an Ethernet card. The latter then acts as a dedicated gateway and is sometimes sold as a dedicated box.

(c) Placing an ISDN card in the PC and a similar card in a server for the LAN

All of these methods require appropriate software to be run on the PC and gateway/server. LAN access is available for the Novell IPX and TCP/IP protocols, for Ethernet and Token Ring LANs.

The costs of such access range from £1000 to £5000 at the central site, and from £300 to £600 at a remote location, excluding the costs of ISDN provision and calls. Reliable systems, reasonably easy to set up, are available. Further information is available below in section 11.2.2.

Distance learning can take many forms as pioneered by the Open University in the UK. The development in the communications technology and computer systems has resulted in an increase interest towards the support of this mode of study using multimedia. One such project [An93] was conducted at the California State University at Chico. The trials aimed to assess the viability of using Integrated Services Digital Networks (ISDN) Basic Rate Interface (BRI) (128Kbits/s) for transmission of voice, data, and video. The distance learning trial was divided into several phases. Phase I involved a point-to-point trial between two locations. Phase II demonstrated the interlocated access transport connectivity. While phase III involved the use of networked-based bridging equipment to accommodate simultaneous instruction to three separate classes. A list of the components and hardware and software configuration can be found in reference [An93].

The evaluation of the experiment has found the following:

Audio quality is very important. Even though the video, still image, and graphics media are available, the overall perception of the service is greatly affected by the quality of the voice communications.

For distance learning applications, the ISDN bandwidth of 112/128 kb/s delivers adequate real-time video quality.

Animated graphics, audio, and two-way video are effective ways to instruct students in remote classes [An93].

11.2. Non-Real Time Multimedia Applications

A number of non-real time multimedia applications are available for use over networks which can support audio or video. The table below summarises the most important applications. Some of these have also been listed in section 11.1, because they have real time and non-real time characteristics, and are not further discussed.

Application	ISDN	LAN's
File Transfer	Various standards	Internet or OSI standards
Information Services with multimedia content	Proprietary or via World Wide Web (WWW)	Any networkable information sytem or WWW technology
Multimedia Mail	Based on H.320 video conferencing	X.400 and MIME electronic mail with video content.
Image Databases	Proprietary systems	File Server access
Information Kiosks	Proprietary systems	
Distance Learning	Remote access to file server	File Server Access

11.2.1. File Transfer

For LANs the de facto file transfer standard is the TCP/IP application - FTP. There are various approaches to file transfer on ISDN.
Existing file transfer protocols can be used. These may be XModem, and YModem in the case of asynchronous transfers or the Internet FTP as used over LANs and WANS. Either asynchronous communications software such as Crosstalk, Procomm or LAN TCP/IP applications such as FTP are needed.
Or file transfer protocols specifically written for ISDN can be used, using proprietary protocols or the emerging Eurofile standard. Data rates of about 1 Mbyte per minute are obtainable on an ISDN2 line.

11.2.2. World Wide Web

Information provision is increasingly moving away from text to graphical interfaces with a multimedia content. The World Wide Web is an example. The World Wide Web has rapidly gained interest through its client implementations such as Mosaic. The delivery of multimedia requires a widespread network capable of delivering at high data rates. Paradoxically, on the Internet, the embryonic 'Super Highway' many thousands of users have to compete to use links which are rarely above 2 Mbps in capacity.

The interesting feature of ISDN for World Wide Web users is that an ISDN channel offers at least 64 kbps which is dedicated to one user. ISDN in the narrow band form is the most widely available access and delivery medium available. Using remote LAN access products running TCP/IP, it is also relatively easy to implement.

ISDN is seen by many in the industry as the ramp through which multimedia networking will gain acceptance. The installed base of ISDN is growing rapidly (30,000 line per month in Germany). ISDN is able to provide connections throughout the world. In Europe there are 'Euro-ISDN' agreements between operators. In the USA the use of the Web is driving the growth of ISDN in some states. Although ISDN could be cheaper, particularly in the UK, it is likely to be cheaper than ATM connections and more widespread in availability for a long time. ISDN will be the feeder network for broadband ISDN based on ATM standards. The idea that multimedia can only be delivered on broadband networks is erroneous as the assertion that only a Macintosh can deliver multimedia.

There are a number of issues in ISDN access to the World Wide Web. The relatively high cost of ISDN in the UK means that the user or organisation must carefully assess the cost effectiveness of their use of the Web over ISDN and participation with other organisations via LAN to LAN connections. The design of Web pages should make economic use of the ISDN call charging regime. This means that large images may need to be avoided, the size of a Web page in bytes can be dimensioned to fit within a call charging unit, and if your client can down load second level hyper-links response times would improve.

There are still major problems of software and hardware compatibility between different ISDN products. The cost of these products is still high but falling. ISDN access to World Wide Web servers on a LAN can be implemented in two ways, routing and bridging.

A Bridge is used to connect two different LAN's that use the same LAN protocol such as Novell IPX or TCP/IP. The bridge acts as an address filter, picking up packets from one LAN and passing on those packets intended for the other LAN. A bridge does not modify the packets or add anything to them. A bridge operates at the data link, Level 2 of the OSI model. A bridge uses the Media Access Control level addresses on LAN adapters to direct packets.

A Router is used to connect two networks that may not use the same LAN protocol. A router uses an inter-networking protocol which is understood by other routers and machines connected to each network. A router operates at the Network, Level 3 of the OSI model. A router uses an addressing scheme such as the Internet address scheme to direct packets.
ISDN connected LAN's can use either bridges or routers. Remote access to these LAN's is achieved by enabling the remote workstation to pretend that it is directly connected to the LAN.

Both routers and bridges need to provide basic functions to ensure reliability and security of data. The use of a bridge or router over ISDN also requires some additional mechanisms to reduce the cost of making unnecessary calls on the ISDN. At present this is implemented by 'spoofing' or fooling a LAN that wishes to send these packets to a remote LAN into thinking that these packets have actually been transmitted.

To connect LAN's which are closely coupled and within one organisation a bridge has some advantages and may give better performance. To connect LAN's operating between different organisations a router enables more effective management of addressing schemes and security.

The most efficient way to interconnect LAN's or remotely attach a workstation via a dial up connection such as ISDN is by use of the Point to Point Protocol.

At the end of 1994 about 27 suppliers had a bridge or routing product for ISDN. There are also a shareware PC based , and UNIX routers which may be able to be used over the ISDN. Some companies also provide a multiple serial line solution for access to a central host. The results of a survey by the Manchester ISDN Partnership [MIP95] indicate the following :-

Performance The use of ISDN to connect LANs and enable remote access to LANs and LAN based services is feasible and offers bit rates to users of up to 128 kbps, which is adequate for many sorts of computer based 'multimedia' sessions.

Costs The running cost of LAN to LAN and remote access connections would be considerably reduced if BT implemented a per second charging scheme. The cost to remote users, of hardware and software is still too high for widespread use of the equipment by small companies. Cheaper ISDN products are available on the German market, but are only just beginning to appear in the UK.

Standardisation The implementation of the de-facto ISDN channel bonding standard is necessary for remote users requiring high performance. Standardisation of asynchronous use of Terminal Adapters above the speed of 19.2 kbps would enable a greater degree of high speed inter-working with ISDN routers equipped with 115 kbps Terminal adapters.

Inter-operability The availability of remote workstation synchronous access via PC cards to a an ISDN LAN router using PPP software would enable some degree of inter-working for remote users of different routers.

The implementation of remote access using TCP/IP protocols by router manufacturers is often tied to on software companies TCP/IP 'stack'. Router manufactures need to provide more options for users to use their existing TCP/IP software, which may be different from that used by the manufacturers. An inter - operability test facility for ISDN routers using PPP and remote access solutions would be beneficial to users. There are already limited claims of inter-operability by AVM with other products.

The most promising products are, in alphabetical order :-

ACOTECH ISDN for Workgroups, AVM Multiple Protocol Router, with Netways remote access. EU-Systems Maxpro Multiple Protocol Router, Jaguar Nile ISDN Router, KNX ISIS Bridge and remote access, Sonix Arpegio Ethernet ISDN Bridge, Spider Networks

Pico and Mezza with remote access using Spider Integrator, Telebit Netblazer.

11.2.3. Multimedia Mail

Multimedia mail can be delivered via X.400 mail, or Internet mail with MIME (Multipurpose Internet Mail Extensions) extensions for file attachments. Obviously transmission times depend on the network capacity An appropriate viewer (hardware or software) is then needed to replay the mail. MIME is a way of transferring multiple objects in a single electronic mail. These objects can be text, images with a JPEG format, 8 bit PCM audio, MPEG video, application specific data, or postscript files.

11.3. Networking Costs

Costs of implementing LAN technology and ISDN can very roughly be compared, excluding the costs of central LAN resources such as file servers :-

Network	Capital cost per user	Cost per MByte
LANs	£200	? very low
ISDN, Basic Rate	£800	£0.05 to £0.10

The cost of broadband circuits such as SMDS and ATM are an unknown quantity. Most of these systems are still at the trial stage. However the cost of sending information over a WAN is likely to remain high. The cost of provision of large bandwidth on LANs will also rise as users adopt multimedia. This will force users to choose the most economical access mode for multimedia.

11.4. Design of Networked Multimedia Applications

Multimedia can be transferred in three distinct ways, by a file transfer, as electronic mail, or in real time. Each is appropriate in different situations.

A file transfer can be initiated manually or automatically at any time. If done in advance of the anticipated time of use of the application, quite large files can be transferred. ISDN can typically transfer about 1 MByte of data per minute, which means that all but the largest multimedia applications, and large sections of video can be comfortably transferred in reasonable times.

Electronic mail systems are increasingly multimedia aware. Both X.400 and the Internet MIME standards can deliver audio, images, pictures and video. The recipient of the mail will of course need to have software integrated into the mail system which can recognise these data types and play them. Electronic mail can take longer to deliver than a file transfer, but has the advantage of being able to be used at any time in the absence of the participants.

A real time multimedia connection is the most demanding multimedia access mode. File transfer and electronic mail do not require a sophisticated, high capacity, or isochronous network to travel over. But real time multimedia demands the correct information at the right time. Users expecting real time pictures or sound are intolerant of delays in reception. Response times need to be of the order of one second. Delays in audio need to be under 150 milli-seconds to avoid echo.

Currently a lot of the media coverage of multimedia is oriented towards the possibilities of real time video applications such as video on demand, video conferencing etc., but it is likely to be applications such as multimedia electronic messaging and transfer of multimedia material between businesses that will become the workhorses of multimedia. In education non-real time use of multimedia material will also prove easier to set-up, and cost effective.

11.5. Design Guidelines

As with any computer application or teaching material creators of multimedia applications need to analyse the aims, objectives and environment in which the application will be used. Video or audio incorporated into a tutorial must have added value over and above over types of presentation, whether it be text, diagrams or photos. In other words the learning outcomes and type of understanding that is desired to create in the user of a multimedia application must be clearly identified. At an early stage in the design process the mechanism that will be used to evaluate the success of the application needs to be identified and a decision made about possible incorporation of testing or feedback mechanisms in the application.

A series of questions can be posed. The reader can no doubt add more.

Question. Is real time interaction with another human, machine or application really needed?

Result If real time interaction is a requirement the nature of the interaction must be established. A simple yes /no or mouse movement is not so network demanding as a video phone conversation. The response times will also need defining.

Question. What are the criteria for access and response times for different activities to be transacted between the user and the application.

Result Users may not require instantaneous access to an application. For instance downloading of software off a file server to a class of students could take place while a lecturer is introducing the application.

Question. What is the length of time over which the application will be used?

Result. Over a long session there will be opportunities for information to be fetched over a network as a background task. This reduces the requirement on the network and the application.

Question. Will the user or student be using the application on a scheduled or timetabled basis?

Result If the student or user is scheduled to use an application, then the material can be made available locally using a timed download.

Question. How many simultaneous users of the application are expected?

Result. A class of 10 students all trying to video conference with a tutor will quickly bring an Ethernet network to its knees. Is the use of multicasting techniques appropriate. Do all users have to perform the same tasks at the same instant?

Question. Over which networks will the application be fetched or used?

Result The network type available will constrain the use of some applications, such as video.

Question. What is the capacity of the network at the likely time of use?

Result Network capacity varies. A university LAN will be exceptionally busy during the middle of the day. If your application expects data which must travel on that network then delays can be expected.

Question. Are there network usage cost considerations, either for hardware or running costs?

Result Do not design a multimedia application largely based on the use of video conferencing over ISDN and expect a small telephone bill.

Question. Which computing platforms, networks and protocols will the application use?

Result. You will reach the widest audience or market if your application is cross platform, able to use more than one type of network and can be configured for different protocols.

Question. Can existing applications be used ? e.g. an e-mail or slide presentation.

Result No need to design your own?

Question. Can recognised standards be used in design the software.
Result If recognised standards for information and communication are used your material will be re-usable and accessible by others.

Question. Can the package be designed in a modular form?
Result A large multimedia package if broken into several modules may be more efficiently transportable across a network.
Question. How much application data and executable code needs to be down loaded?
Result The load on a network can be reduced and response times improved if file sizes are kept small.
Question. Can remote users be given regular updates of the package or data?
Result The application can be given a new look or improved facilities. This is particularly relevant to remote information kiosk design.

There are a few common rules linking the questions above. These are :-

Do not use multimedia unnecessarily
Choose your media types carefully
Keep the application small
Do not put more information over the the network than needed.
Hold information locally where possible
Consider the capabilities of the network before the implementation stage
Try to anticipate the needs of a user for information and responses
Try to use standards and integrate standard applications into your design.

Examples

You have recorded a series of video clips for students to view and manipulate. Can they be downloaded on request, in sequence by file transfer from a server to save on local storage and network bandwidth?

You are considering using video conferencing. Will a video messaging facility suffice? Can the same aims be accomplished using voice and still image transmission?

In an information kiosk design can updates of information be kept to text only.

You are choosing an authoring platform. Can the platform accept common media formats, and access data or applications and outside of its immediate environment?

12. FUTURE TRENDS

The evolution of communication networks from the current mix of analogue and digital access to full digital high capacity or broadband systems will occur. What is not clear is the exact path likely to be followed. Important networking issues will arise. With the availability at some sites of ATM technology, the use of FDDI for local networks at some institutions, and a large installed base of twisted pair Ethernet or Token Ring networks the migration path is not clear.
Access to ISDN, the availability of ISDN on PABX switches for internal use, the use of Metropolitan are networks to carry voice telephony and data traffic and the use of LANs for telephony all throw up important issues of network integration.

12.1. Broadband Network Services

B-ISDN (Broadband ISDN) with its ATM technology, going through the process of standardisation, is likely to be the most important candidate for the broadband multiservice network of the future. Much effort, worldwide, is being concentrated on building an understanding, through pilots projects, of this new technology. One of the main achievements of the ATM technology is its flexibility. It does not differentiate between various information characteristics, nor does it deal directly with end user time characteristics.
Users may not be satisfied with a low level direct access to the service offered by the ATM layer. To remedy this point an ATM (AAL) adaptation layer has been proposed. This layer will sit outside the pure ATM network and will provide the functionality not provided by the ATM network. These services are summarised below.

12.1.1. ATM Adaptation Layer Services

Currently there are four types of adaptation protocols progressing through the standardisation bodies: ITU-T, ETSI, and ATM forum. These are AAL-1, AAL-2, AAL-3/4 and AAL-5. Each AAL type is aimed at supporting specific communication requirements such as Connection Oriented, real-time issues, Variable Bit Rate etc.

The AAL-1 [ETS93c, ETS93e] offers a service that accepts Service Data Units (SDUs) a t a fixed clock rate for transmission over the network and delivers them at the same clock rate in a Connection Oriented mode, also called isochronous service. The basic characteristic of the AAL-1 is the ability to offer a Constant Bit Rate (CBR) service. In addition it performs some basic error control mechanisms are performed which include sequence numbering in order to detect lost or mis-inserted ATM cells and Forward Error Correction (FEC). There are, currently, 3 services by AAL-1:

- The CBR circuit emulation service support the transmission of isochronous digital information. The CBR Circuit Emulation service provides two options: asynchronous circuit transport unstructured signals (2 Mbits/s or 34 Mbits/s); and synchronous circuit transport of 64kbits time slots.

- Video and Voice-band Signal Transport Services. The main differences between the CBR Circuit Emulation (CE) and these two services are the SDU size, one bit for CBR CE and one octet for the video and Voice-band Signal transport, and that for the Video Signal transport a specific FEC mechanism is foreseen. It is as yet undecided in how to handle voice signals in ATM cell stream with the problem of partly filled cells.

The AAL-2 protocol type should support Connection Oriented Variable Bit Rate (VBR) traffic and is intended to support transmission of VBR video codecs signal as an example. However, the specification process is at its early stages.
The AAL-3/4 protocol [ETS94b] offers a connection oriented service, not including any timing aspects, and is intended to support general data transmission applications. The basic functionality of the AAL-3/4 is SDU delimiting and multiplexing of higher layer SDUs on the ATM connection. From its incept the AAL-3/4 has been seen a provider of a Connectionless (CL) data service over the Connection Oriented ATM. Since it is an architectural requirement that the AAl should not deal with network layer issues, an additional layer is expected above the AAL-3/4 to offer a CL service. This CL protocol layer mainly provides the addressing functionalities in order to offer a Broadband Connectionless Data Service (BCDS) in ITU-T terminology or CBDS/SMDs in ETSI terminology. Therefore it makes no sense to consider the AAL type-3/4 service on its own.

The prime objective of the AAL-5 [ETS93d] is high speed transmission with reduced overhead. It is a compromise between overhead and functionality. The strong wish to develop a high speed AAL protocol type, which should support existing protocols, was the reason behind the specification of a reduced AAAL protocol type. From a service point of view AAL-3/4 and AAL-5 offer the same layer functionality. The main differences between these two protocol types are: the AAL-5 performs minimum error control mechanisms in comparison to the AAL-3/4; they perform different mechanisms for SDU delimiting; and the AAL-5 does not offer a higher layer SDU multiplexing capability.

Currently AAL-5 offers a service for the transport of B-ISDN signalling information, and a Frame Relay service. AAL-5 is being considered in the ATM for possible use to transport real-time multimedia information.
The following B-ISDN bearer services are currently under standardisation: Virtual Path (VP) ATM bearer service, Broadband Connection Oriented Bearer Service, Connectionless Broadband Data Service (CBDS), and Frame Relay service.

12.2. Future Teleservices

A number of other service (standards) are being considered by broadband standard organisations ITU-T, ETSI and ATM Forum.

12.2.1. ITU-T and ETSI Services

Tele-service	ITU-T Rec.	ETSI
Broadband Video Telephony Services	F.722	DE/NA-010029
Broadband Videotex Services	F.310	-
Broadband Video Conference Services	F.732	DE/NA-010030
Broadband TV Distribution Services	F.821	-
Broadband HDTV Distribution Services	F.822	-
Multimedia Distribution Services	F.MDS	-
Multimedia Delivery Services	F.MDV	-

The specifications of these services within ITU-T and ETSI are at their very early stages and still incomplete. A number of new activities have also been started within ETSI/NA5 and ETSI/NA1 in order to derive a specification for a “Best Effort Service” (DE/NA-010031) and a “Video on Demand” (VOD) service. The Best Effort Service is mainly aimed at applications running over interconnected LANs with a variable bit rate.

12.2.2. ATM Forum Services

ATM Forum Standardisation of teleservices are also at an early stage. The following services for Audio Visual Multimedia Service (AMS) have been identified [Coppo94]:

- Audio visual Service and Multimedia on desktop: Conversational service with information types such as moving pictures, sound, and data. This service is equivalent to ITU-T Rec. F.22 “Broadband Video Telephony Service”.
- Real-Time Transport Service: The objective is the identification of a set of generic requirements for a transport service that support real-time delivery of digitally encoded video or audio streams [Patel94].
- Video Conferencing: Conversational service with information types such as moving Pictures Sound and Data. This service should be equivalent to the ITU-T Rec. F.732 “Broadband Video Conference Service”.
- Audio & Data: Conversational service with information types of sound and data.
- Video on Demand: Retrieval service with information types that include moving pictures and sound.
- Broadcast Video Service: Distribution services without user presentation control. Service include moving pictures and sound. This service should be equivalent to the ITU-T Rec. F.821 and F.822 Broadband TV/HDTV distribution services.
- Best Effort Service is in line with the same work undertaken by ETSI/NA5 mentioned earlier.

12.3. State of the Art Projects

The rest of this report describes a number of projects which point towards the future in multimedia networking. A number of these, as we shall see, have working prototypes that highlight what can be achieved in the immediate future. Their collective output provide a good resume of the issues and possible paths for the provision of multimedia services over computer networks. This list of projects is not exhaustive or a description of the best efforts but rather a representative sample.

BERKOM

The BERKOM project [Butscher91; Domann88] was started in 1985 in order to identify and prototype services and applications for the future B-ISDN. BERKOM has strong links to the German PTT. The main results so far have been the identification and definition of a communication architecture with a number of broadband services. These are multi-vendor and multi-network environments for a Multimedia Collaboration service and a Multimedia Mail service. The BERKOM Multimedia Collaboration service [Altenhofen93] allows audio-visual conversions that consist of directory service, conference management and applications sharing. The BERKOM Multimedia teleservice [Blum93] supports multimedia message transfers in a heterogeneous environment. In particular, it provides a framework for the presentation of multimedia messages and their compositions, sending and receptions.

RACE/CIO

CIO is a RACE project (R2060) with the main objective to specify and implement advanced multimedia teleservices on various end systems, which include Multimedia Mail and a Joint Viewing and Teleoperation Service (JVTOS) [Dermler92].
JVTOS has been developed in order to share information between users using X Window applications. It allows for a distributed display of graphical data as well as remote control for , say, joint editing. In addition, JVTOS offers the facility of a picturephone for direct audio and video communication which can be used independently of X-Windows applications.
The Multimedia Messaging Service is built around X 400 and X 500 which have been extended to support multimedia data (such as audio and video data). In contrast to JVTOS this service has been developed as stand alone application and is available for a restricted set of platforms.

TINA-C

There is much interest in the research community for the flexible introduction of new services, the management of these services as well as the provision of integrated platforms which provide a pre-defined quality of service (QoS) [Carew94, Carew95, Leopold92]. In addition, there is a wide interest in the management of the services and integration of the network infrastructure. One such an initiative is by the Telecommunications Information Networking Architecture Consortium (TINA-C) [Natarajan92]. The TINA-C consortium consists of the telecommunications operators, telecommunications and computer vendors was in formed in 11992. The objective of TINA-C is to provide and architecture that is based on Open Distributed Computing (ODP) technology [ISO91]. This architecture framework follows the ODP view points which address the semantics of information and the information processing activities in a system. The architecture addresses, so far, two main issues: a service session model, and the service management model.

A service session model provides the concepts for establishing, using and releasing sessions. A session in this framework is not unlike a “call” in the telecommunications world. The service management model deals with subscriptions, fault, performance, accounting, security, and configuration. To provide multimedia teleservices developers will need to use TINA-C Distributed Processing Environments. It is, as yet, unclear which of the existing technologies, ANSAware, OSF DCE/DME, OMG CORBA, are to be used to build this support environment.

Sequoia 2000

The Sequoia 2000 [Stonebraker92] project was sponsored by the University of California and Digital Equipment Corporation (DEC) and by a consortium of industrial and Government Agencies. The Sequoia 2000 network provides the communication infrastructure for global change researchers and computer scientists involved in the project. Sequoia scientists require networks which support real-time scientific visualisation and video conferencing applications as well as high-speed data delivery services for the very large data that characterise global change applications. To satisfy these requirements, Sequoia researchers are investigating methods for providing both real-time and best effort services for voice, video and data delivery on the Sequoia network. The Sequoia network provides services to the California Department of Water Services, UC Davis, UC Berkeley, UC Santa Barbara, UCLA, the San Diego Supercomputer Center, UC San Diego, and the Scripps institute of Oceanography. The infrastructure consists of FDDI rings for local distribution with private T1 (1.54 Mbps) leased lines for wide-area services. A number of DECstations 5000/240 general-purpose workstations interconnect the FDDI and T1 links and serve as network routers. The researchers use scientific workstations to load, browse and query objects such as satellite weather maps and global climate modelling data. Additionally, many Sequoia workstations support network transmission of digitally-encoded audio and video streams. Several workstations use DEC’s J-Video and J300 hardware compression/decompression cards with live video and audio capture features. These cards are linked to cameras, speakers and microphones to cater for multimedia requirements.

Again just as the OSI stack has been found to be inadequate, the Internet protocols, the most widely used protocols in research environments, have also been found to be incapable of catering for some of the real-time requirements of multimedia data. This realisation has led to the development of a new suite of protocols called TENET after the TENET group [Ferrari92] of the University of California at Berkeley. Thus the Internet Protocols (IP, UDP, TCP) are used for best effort data delivery, while the Tenet protocols are under investigation as possible solution for real-time requirements.

The TENET suite of protocols provides real-time or guaranteed performance communication services in an internetworking environment and adopts a connection

oriented and reservation-based architecture. The Tenet suite of protocols is divided into data delivery and control protocols. The data delivery protocols include the Real-Time Internet Protocol (RTIP) [Zhang92] at the network layer, and the Real-Time Message Transport Protocol (RMTP) [Zhang93] and the Continuous Media Transport Protocol (CMTP) [Wolfinger91] at the transport layer. The control protocol is called the Real-Time Channel Administration Protocol (RCAP) [Banerjea91], which performs the channel establishment, status reporting, and closing down.
It has been found that the Sequoia infrastructure with it T1 links is prone to congestion due to its competitive multimedia workloads leading to the degradation in the audio and video quality. These T1 links are due to be updated to T3 (45 Mbps) links, however, it is anticipated that the new infrastructure will also be congested due to the rapid growth of the offered load on the Sequoia network.

North Carolina Information Highway

A good example of what the information superhighway will provide to future users is highlighted by the North Carolina Information Highway (NCIH) [Patterson94]. The NCIH is the first wide scale public deployment of ATM technology. In its initial architecture users are connected via 155 Mb/s to twelve ATM switching systems. The NCIH became operational in August 1994 and provide links to around 50 Universities, community colleges, schools, hospitals, prisons, and government facilities. It is expected that 100 more sites will be connected in 1995 and around 3000 more sites to be connected over the next decade.

The NCIH services [Grovenstein] include ATM Cell Relay, SMDS, and Circuit Emulation. The ATM Cell Relay is a connection oriented service that provides high-speed and low delay transfer. All other services in the NCIH are built on top of ATM Cell Relay. Initially, only permanent virtual circuits but with future plans for ATM switched virtual circuits. In addition, a constant bit rate (CBR) and variable bit rate (VBR) are also supported. The Switched Multimegabit Data Service (SMDS) is a connectionless data service with packets of up to 9,188 bytes in addition to a header that contains a source and destination address. The header information is used to segment it into ATM cells for transport. The ATM Adaptation Layer Type 3/4 (AAL 3/4) that occupies 4 bytes out of 48 bytes ATM cell. The circuit emulation is needed to provide backwards compatibility to existing telecommunications network that operate at the rates of 64 kbs/s 1.5 Mb/s and 45 Mb/s.

The typical NCIH architecture consists of :
(1) A LAN linking to a router and then a 1.5 Mb/s SMDS link to an ATM service Mux.
(2) and/or a video input to a switched 45 Mb/s link to the ATM multiplexer.
(3) An ATM/OC-3c (155 Mb/s), with an upgrade path to 622 Mb/s or even 2.4 Gb/s link from the ATM service Mux to the ATM switch. The ATM switch output can then be linked to various other components, including a SONET network for interexchange carriers.

The video connection costs are likely to go down when the MPEG-2 operating at 6 Mb/s (as opposed to 45 Mb/s) are released. It is expected that multiple streams of 6 Mb/s signals will be multiplexed at the customer's site ATM Mux.
A number of applications have been specifically designed to exploit the high bandwidth offered the data highway. The first one to implemented was that of distance learning. A number of configurations are possible but the most demanding is that of video links between classrooms. Other trials included the VISTAnet and MICA (Medical Information Communications Application) [Bruwer94] which enabled real-time multi-dimensional imaging for health care applications. For example, radiation therapy and planning through the provision of real-time X-ray consultation and teleconferencing.

Worksatations

Instead of the augmented multipurpose workstations of the Sequoia project another path of investigation has been the development of new integrated workstation environments consisting of hardware, software, and communications protocols for multimedia support. This path of investigation is highlighted through the description of the Olivetti's Medusa environment [Wray94] and MIT VuNet project [Adam94].

The Medusa project at Olivetti research aims to provide a networked multimedia environment in which many streams of multimedia data are active simultaneously. Medusa software environment consists of a number of active objects or modules that represent cameras, displays, format converters etc. Applications are then build on top by grouping together these objects. This approach results in a heavy biased and architecture that is based on the underlying hardware. The Medusa hardware is made of a collection of ATM direct peripherals, cameras, audio, systems, multimedia storage servers, LCD displays and televisions. The authors refer to the different components as bricks to convey the analogy with building and how they are built. Each of the direct peripherals are build around an ARM processor from Advanced Risc Machines in Cambridge, which directly connected to an input or output device of the ATM network. These processors a microkernel which has been specifically designed fro networked, embedded, real-time systems. This is different from a traditional workstation where there is no direct connection for the separate components.

The Medusa integration of networked components is taken further by the VuNet approach. VuNet desk area expands throughout the network with small switches are distributed all over. This leads to the multimedia components usually found in the workstation to be spread out over the network. The authors claim that through this approach real-time processing is enhanced. This processing include real-time stationary filtering, motion detection, shot change detection, edge filtering, and blue-screening. One of the unknown in this approach is the intensity of computation in these distributed resources, in addition to the complex management of many devices. The management in synchronisation for cooperation could lead to performance problems for example in trying to achieve a consistent state.

Protocols Research

The increase in interest and the need to develop multimedia products and networking has resulted in a number of initiatives for the development of high speed protocols that could cater for the high bandwidth, real-time characteristics, and synchronisation requirements. Most of these developments stem from the realisation that current networking protocols cannot meet these requirements. For example, the RACE project 2060, OSI-95 had for goal the development of protocols that could replace the inadequacies of the OSI stack. The OSI'95 consortium with Lancaster University as UK partner has identified a number of requirements for the support of distributed multimedia systems. These include: transport protocol support [Shepherd92] with synchronisation, orchestration and mechanisms for different rate control protocols. The project has also identified the need for integration to support user requirements and the quality of service [Leopold92] and proposed a framework for integration between ODP and OSI [Leopold93]. Further information on this and other related research such as multimedia storage and retrieval over the network for education can be obtained from Prof D. Hutchison at Lancaster University [Hutchison 94].

Another effort at the development of new protocols is that of the Real-Time Transport Protocol (RTP) [Schulzrinne93]. RTP combines the tasks of application, presentation, session and transport layer in a single protocol. This results in an improved performance but at the expense of the layering principles that have been established by the OSI Model and standardisation processes. RTP has been used to provide some MBone service such as vat and ivs.

An area which has received much research interest is that of multicasting. The ability to multicast is the basis for collaborative work and thus one of the main applications of the multimedia capability. Multicast Transport Protocol (MTP) [Armstrong92] provides a reliable transport service on top of the network layer and with a multicast facility. In the case of ST-II [Topolcic90] we have a guaranteed end to end bandwidth and delay (thus suited for multimedia) together with multicast support. The contention for resources and the wide area coverage of the potential applications, for example, a conference can lead to congestion and thus the need for routing. One solution has been the Distance Vector Multicast Routing Protocol (DVMRP) [Waitzman88]. DVMRP is an experimental routing protocol for internet multicasting. It forms the basis for MBone and as such has achieved some popularity. The problems in efficiency due to for, example to truncated broadcasting has resulted in a new protocol: Protocol Independent Multicast (PIM) [Deering94]. PIM approach to multicasting routing is to operate in two different modes: dense and sparse mode to cater for situations where nodes are clustered closely together or widely and sparsely distributed.

12.4. Summary

12.4.1. Network Evolution in the Short Term

While TV networks have existed for sometime, data networks are only just beginning to approach their capabilities. Distribution of multimedia in the same way that TV is distributed is simply not possible on existing data networks. These networks have relied on the telephone system, and a few high capacity links between centres. The 'Information superhighway' is all about distribution of multimedia. The problem is one of speed and cost. If multimedia contains video, then a VHS type display can be made in real time if the transmission link can carry 1.5 Mbps of data. At present this can only be provided by :-

- Cable TV companies which have suitable equipment on their system
- BT. by a fixed dedicated connection costing from about $2000 upwards.
- BT. via the telephone system when and if the government allows competition with the cable TV companies.
- The academic network linking some universities called Super JANET
- A local area network

So for the present most of us will have to use a low end alternative - ISDN and an Ethernet LAN. Ethernet can support less then 10 Mbps and ISDN can transmit data at 128 kbits/sec. The ISDN is adequate for live person to person video conferencing, and data transfer for many LAN applications.

There are two schools of thought on the use of multimedia over LANs. One is that ways can be found to transport multimedia over traditional LANs. The other looks to new LAN and WAN technologies, and argues that conventional LANs are unsuited to multimedia.

The reality is that some real time multimedia applications will continue to suffer performance restrictions when operating over LANs, but some applications will become available. Multimedia applications which are less demanding of bandwidth and which do not operate in isochronous mode will be more easily carried over traditional LANs.
To provide for high capacity on local and wide are networks there will be a period of considerable competition between switched Ethernet, isochronous (Iso-ethernet) Ethernet and local ATM solutions. Suppliers will attempt to provide equipment and adapters at prices which will encourage users to upgrade their LAN infrastructure.

For wide area networking narrow band ISDN will play an important role for some time. ISDN connections will replace the provision of analogue PSTN lines, enabling data calls to be placed worldwide. Initially there will be applications developed for ISDN use, which will then incorporate Iso-ethernet and local ATM interfaces. The ISDN network and ATM network will operate alongside each other, but once the ability to connect through from an ISDN point to an ATM end user appears the networks will begin to converge. ISDN applications and connections will serve as feeder applications for the higher speed ATM backbone.

LAN emulation over ATM will assist high speed connections between networks. As experience of ATM use develops and local ATM becomes more widespread applications designed for ATM use by users on with a PC or Mac type platform will become available. For users in the academic community the twin criteria of cost and speed will continue to be determining factors.

12.4.2. Future Networks

The development of computer communications has been led by the very different environments of wide area networks and local area network. Wide-area networking is high cost, low speed, high error rate, large delay. The design of the switch was dictated by the need to manage and allocate the bandwidth effectively. Both conventional circuit switching and packet switching are difficult to implement at higher data rates. In most current designs for local area networks, where bandwidth is not expensive, simplified addressing and switching has been employed at the expense of the effective use of the bandwidth. In particular, networks such as Ethernet use broadcast as the normal distribution method, which essentially eliminates the need for a switching element.

Fibre optics has made high-speed information transmission possible. The problem is how distribute or switch this informati on when the intelligence needed to make switching decisions is operating in comparatively low speed electronics. For optical switching to be possible the switch must use very simple switching logic, require very little storage and operate on packets of a significant size.
For example, at a gigabit, a 576 byte packet takes roughly 5 microseconds to be received so a packet switch must act extremely fast to avoid being the dominant delay in packet times. Moreover, the storage time for the packet in a conventional store and forward implementation also becomes a significant component of the delay. Thus, for packet switching to remain attractive in this environment, it appears necessary to increase the size of packets (or switch on packet groups), or specify the route at source.

For circuit switching to be efficient at high speeds, it must provide very fast circuit set-up and tear-down to support the bursty nature of most computer communication. For long distance routes this is difficult because the propagation delay is greater than the transmission time at high data rates.

The choice of switching technology determines its performance, its charging policies, and even its effective capabilities. As an example of the latter, a circuit-switched network may not provide strong multicasting support. Since high speed networks will be built from point-to-point fibre links that do not naturally provide multicast/broadcast it is difficult to predict whether multicasting can be provided as part of a network protocol, or should be supported by higher layers only.
In the future, the host may see the network as a message-passing system, or as memory

[Leiner88]. At the same time, the network may use classic packets, wavelength division, or space division switching. Future network protocols will need to provide a secure connection independent of the networks for applications to use.

Hitherto the concept of layering has allowed dramatic variation in network technologies without requiring the complete re - implementation of applications. Unfortunately, the layer interface designs are all organised around the idea of commands and responses plus an error indicator. For example, the TCP layer provides the user with commands to set up or close a TCP connection and commands to send and receive data. The user may well "know" whether they are using a file transfer service or a video call, but can't tell the TCP. The underlying network may "know" congestion is limiting the throughput to levels too small for acceptable video, but it also can't tell the TCP implementation.

The implementation of Quality of Service requests flowing in one direction from application to network are a partial solution. It would be more useful if a two way dialogue could be established between applications and the network(s) in a dynamic fashion. For instance if a World Wide Web application finds it is running very slowly over a congested trans-atlantic connection it could request a dial up ISDN connection as an alternative route or a higher class of connection from an ATM switch. Implicit in managing this request is the presence of a more controlled allocation of resources. When data calls are mixed with new services such as video streams unless these are separately recognised and controlled, there is little reason to believe that effective service can be delivered unless the network is very lightly loaded.

As networks get faster, bottlenecks move into the host. The speed of the asynchronous port on a PC is a classic example, when connected to ISDN at 64 or 128 kbps. Network adapters have a crucial role to play. The future network adapter may be viewed as a memory interconnect, tying the memory in one host to another. The integration of the network adapter into the operating system with the the transport level will be implemented largely, if not entirely, in the network adapter, would provide the host with reliable memory-to-memory transfer at memory speeds with a minimum of interrupt processing, bus overhead and packet processing.

13. References

[Adam94] Adam, J.F., Houh, H., Ismert, M., and Tennenhouse, D., *“Media-Intensive Data Communications in a Desk-Area Network”*, IEEE Communications Magazine, vol. 32(8), pp. 60-67, August 1994.

[Aldred94] Aldred, B.K., “*Lakes Real Time Multimedia Communications and T.120”*, European ISDN User Forum, Copenhagen, November, 1994

[Altenhofen93] Altenhofen, M., et al. *“The Berkom Multimedia Collaboration Service”*, Working Document Release 2.0, Berkom, June 1993.

[An93] An, C., Barcelo, B.T., Inkrott, J.A., Snowdon, A.R., and Trojniar, K.J., *“A Multimedia Distance Learning Trial Using ISDN BRI”*, AT&T Technical Journal, vol. 72(1), pp. 15-21, January/February 1993.

[Armstrong92] Armstrong, S., Freier, A., and Marzullo, K., *“Multicast Transport Protocol”*, RFC 1301, February 1992.

[Banerjea91] Banerjea, A., and Mah, B., *“The Real-Time Channel Administration Protocol”*, Proceedings of the 2nd International Workshop on Network and Operating System Support for Digital Audio and Video, Heidelberg, Germany, pp. 160-170, November 1991.

[Bellcore92] “*HDSL and ADSl: Giving New Life to Copper”*, Bellcore Exchange, March/April 1992.

[Blum93] Blum, C., et al., *“The Berkom Multimedia Mail Service”*, Working Document Release 2.2, Berkom, May 1993.

[Brazdziunas94] Brazdziunas,C., "*IPng Support for ATM Services*", RFC 1680, 1994.

[Bruwer94] Bruwer, W.A., Loop, J.D., Symon, J.A., and Thompson, B.G., *“VISTAnet and MICA: Medical Applications Leading to the NCIH”*, IEEE Network, Vol. 8(6), November/December 1994.

[BT94] BT, “*Installation notes for ISDN2 lines*” , British Telecom, 1994.

[Butscher91] Butscher, B., and Popescu-Zeletin, R., *“Berkom a B-ISDN Pilot Project”*, 3rd IFIP Confrence , Berlin, Germany 1991.

[Carew94] Carew, M, Merabti, M., and Whiteley, K., *“Towards A Unified Approach to Supporting Multimedia Applications in Open Distributed Processing Environements”*, Communication

Networks Symposium, Manchester, 11-12 July 1994.

[Carew95] Carew, M, Merabti, M., and Whiteley, K., *"Multimedia Support for Distributed Systems"*, IEEE International Conference On Communications, Seattle, Washington, June 18-22, 1995.

[CCITT92] "*Network Capabilities to Support Mulimedia Service"*, CCITT Framework Recomendation I.374, June 1992.

[Clyne94] Clyne, L., "*SuperJANET Update*", Computer Networks for Research in Europe Vol. 25, Supplement 3, 1994.

[Coppo94] Coppo, M., and Ullio, M., *"Service Identification in the SAA Group"*, Draft Contribution ATM Forum/94-0159, ATM Forum Technical Committee Working Group, March 1994.

[CT95] "*Last Ditch Entry Muddles ATM Race*", Communications Networks, p. 25, January 1995.

[Deering94] Deering, S., Estrin, D., Farinacci, D., Jacobson, V., Liu, C-G., and Wei, L., *"Protocol Independent Multicast: Motivation and Architecture"*, IETF Internet Draft, March 1994.

[Dermler92] Dermler, G., and Froitzheim, K., *"JVTOS - A Reference Model for a New Multimedia Service"*, 4th IFIP Conference on High Performance Networking, Liège, Belgium, 14-18 December 1992.

[Domann88] BERKOM, *"Test Network and B-ISDN/CATV Concept"*, Electronics Communication, 62(3), 1988.

[EIUF94] *European ISDN Users Forum* , Copenhagen, November, 1994.

[ETS94] ETSI/NA5, *"B-ISDN ATM Adaptation Layer (AAL) specification, type 3/4"*, Draft ETS prETS 300 349 (DE/NA-52618), ETSI, April 1994.

[ETS93c] ETSI/NA5, *"B-ISDN ATM Adaptation Layer (AAL) Specification, type 1"*, Draft ETS prETS 300 353 (DE/NA-52617) ETSI, September 1993.

[ETS93d] ETSI/NA5, *"B-ISDN ATM Adaptation Layer (AAL) Specification, type 5"*, Draft ETS DE/NA-52619, ETSI, September 1993.

[ETS93e] ETSI/NA5, "Optionality Aspects of ATM Adaptation Layer type 1", Technical Report, ETSI/NA5, November 1993.

[Ferrari92] Ferrari, D., Banerjea, A., and Zhang, H., *"Network Suuport for Multimedia: A Discussion of the Tenet Approach"*, Technical Report TR-92-072, International Computer Science Institute, Berkeley, California, October 1992.

[Griffin94] "*C-PHONE*", Specification, Griffin Sight & Sound, London ,1994.

[Grovenstein94] Grovenstein, L.W., Pittman, C., Simpson, J.H., and Spears, D.R., *"NCIH Services, Architecture, and Implementation"*, IEEE Network, Vol. 8(6), November/December 1994.

[Hutchison94] Hutchison, D., Department of Computing, Engineering Building, Lancaster University, Lancaster LA1 4YR, UK.

[ISO10] "*Compression Standard for continuous tone still images"*, ISO Standard JTC1/SC2/WG10.

[ISO11] "*Coding of Moving Pictures and associated audio for Digital Storage Media at up to 1.5 Mbps*", ISO Standard JTC1/SC29/WG11.

[ISO91] ISO, *"Basic Reference Model of Open Distributed Processing - Part 1: Overview"*, Draft X.901, ISO 10746-1, ISO/IEC JTC1/SC21/WG7, November 1991.

[ITU-T93] ITU-T, "Multimedia Distribution Services", Draft Recommendation F.MDV, ITU-T/SG1, ITU-T Report COM I-R 54-E, Annex 1, 1993.

[Janet93] "*Network News, SuperJANET Special Edition*", No. 40, UKERNA, November\1993.

[King93] King, T., "*SMDS, The Switched Multimegabit Data Service - A Technical Review*", Proc. of Networkshop 21, Birmingham, UKERNA, March 1993.

[LAN95] "*100 Mbps Etherne*t", p 117, LAN Magazine, February 1995.

[Leiner88] Leiner, B., Editor, "*Critical Issues in High Bandwidth Networking"*, Gigabit Working Group, RFC 1077, November 1988.

[Leopold92] Leopold, H., Campbell, A., Hutchison, D., and Singer, N., *"Towards an Integrated Quality of Service Architecture (QoS-A) for Distributed Multimedia Communications"*, 4th IFIP Conference on High Performance Networking, Liège, Belgium, 14-18 December 1992.

[Leopold93] Leopold, H., Coulson, G., Frimpong-Ansah, K. Hutchison, D., and Singer, N., *"The Evolving Relationship Between OSI and ODP in the New Communications Environment"*, 2nd Int. Conf. on Broadband Islands, Athens, Greece, 14-16 June 1993.

[MCF] *Multimedia Communications Forum,* Suite 201 - 931 Brunette Avenue, Coquitlam, BC, Canada, V3K 6T5, E-mail : 72630.107@compuserve.com

[Midwinte94] Midwinter, J.E., "*The Threat of Optical Communications*", Electronics and Communications, Vol 6(1), IEE London, February 1994.

[Minoli93] Minoli, D., "*First, Second and Next Generation LANs*", McGraw Hill, New York, 1993.

[Minoli94] Minoli, D., and Keinath, R., "*Distributed Multimedia Through Broadband Communication Services*", Artech House, London , p. 131, 1994.

[MIP95] "*Manchester ISDN Partnership Final Report*", Centre for Employment Research, Manchester Metropolitan University, Manchester, February 1995.

[Natarajan92] Natarajan, N., and Slawsky, G., *"A Framework Architecture for Multimedia Information Networks"*, IEEE Communications Magazine, pp. 97-104, February 1992.

[Nelson94] Nelson, T.H., "*Publishing in the Point and Click Universe"*, Proc. National Convergence Symposium '94, Sydney, Australia.

[Ovum94] "*Euro-ISDN Handbook*", Ovum publication, 1994, available from Ovum, Telephone 0171 255 2670.

[Patel94] Patel, A., *"Outline for the SAA AMS Baseline Text Document"*, Draft Contribution ATM/94-0279R1, ATM Forum, Munich, Germany, May 1994.

[Patterson94] Smith Patterson, J., and Smith, W. L., *"The North Carlina Information Highway"*, IEEE Network, Vol. 8(6), November/December 1994.

[Rame93] Rame, R.C., "*100 Mbps Ethernet (CSMA/CD) for UTP Wiring*", COMNET3 Conference Proceedings, 1993.

[Schindler94] Schindler, S.R., et al., "*Desktop Videoconferencing with EURO.VISION*", TELES, Draft Version, Berlin , 1994.

[Schulzrinne93] Schulzrinne, H., and Casner, S., "RTP: A Transport for Real-Time Applications", IETF Internet-Draft, October 1993.

[Shepher92] Shepherd, D., Hutchison, D., Garcia, F., and Coulson, G., *"Protocol Support for Distributed Multimedia Applications"*, Computer Communications, vol 15(6), July/August 1992.

[Stonebraker92] Stonebraker, M., *"An Overview of the Sequoia 2000 Project"*, Proceedings of COMPCOM 92, San Francisco, CA, February 1992.

[Topolcic92] Topolcic, C., ed., *"Experimental Internet Stream Protocol: Version 2 (ST-II)"*, RFC 1190, October 1990.

[Waitzman88] Waitzman, D., Partridge, C., and Deering S., *"Distance Vector Multicast Routing Protocol"*, RFC 1075, November 1988.

[Wolfinger91] Wolfinger, B., and Moran, M., *"A Continuous Media Data Transport Service and Protocol for Real-Time Communications in High Speed Networks"*, Proceedings of the 2nd International Workshop on Network and Operating System Support for Digital Audio and Video, Heidelberg, Germany, pp. 171-182, November 1991.

[Wray94] Wray, S., Glauert, T., and Gopper, A., *"Networked Multimedia: The Medusa Environment"*, IEEE Multimedia, vol. 1(4), pp. 54-63, Winter 1994.

[Zakowski94] Zakowski, W., "*A Reference Model for Multimedia Communications"*, European ISDN User Forum, Copenhagen, November, 1994.

[Zhang92] Zhang, H., Verma, D., and Ferrari, D., *"Design and Implementation of the Real-Time Internet Protocol"*, Proceedings of the IEEE Workshop on the Architecture and Implementation oh High Performance Communication Subsystems, Tucson, Arizona, February 1992.

[Zhang93] Zhang, H., *"Service Disciplines for Integrated Services Packet-Switching Networks"*, PhD Thesis, UCB/CSD-94-788, University of California at Berkeley, November 1993.

BIBLIOGRAPHY

"*Distributed Multimedia Through Broadband Communication Services*", D. Minoli and R. Keinath, Artech House, London ,1994, p. 131.

"*Multimedia in Higher Education: Portablity and Networking*", Workshop Report for the Advisory Group on Computer Graphics, Technical Report No. 24 December 1993

"A Survey of Distributed Multimedia Research, Standards and Products", ed. C.J. Adie January 1993 (RARE Technical Report 5)

"Network Access to Multimedia Information", C.J. Adie January 1993 (RARE Project OBR(93)015) August 1993.