The Convergence of Networking and Broadcasting

By Randy Conrod

In the last tutorial in this series, we discussed the serial digital interface (SDI) and how things like audio and error detection and handling (EDH) signals can be embedded in serial video signals. In this tutorial, a comparison of broadcasting and networking technologies is made and the convergence of technologies that broadcasters and telephone companies (telcos) are looking at for use now and in the future is discussed.

Let's start with a brief overview of communications. The first communications were digital. Morse code was used to send and receive messages using dots and dashes. The next step was the analog telephone, which allowed two-way voice communications. With the analog telephone came some of the earliest networking concepts that have evolved until today, such as permanent circuits, switched circuits, and virtual circuits.

Permanent circuits use a hard piece of copper wire, much like a connection to a doorbell. Switched circuits are circuits that are set up at the time of the call, such as when a telephone call is made on a modern switched telephone circuit system.

Virtual circuits are the order of the day where no actual hard physical path exists. There are two types of virtual circuits: a switched virtual circuit (SVC) and a permanent virtual circuit (PVC). A switched virtual circuit is set up at the time of the call and may, in fact, change its routing during the time of the call transparently to the user. Permanent virtual circuits are set up at the time of the call and remain in place until the call is completed.

Networking Overview

Computer networking allows the interconnection of data devices. These would include applications such as computers to terminals, computers to computers, and computers to file servers. Computer networking requires error-free communications. This is normally achieved by using a protocol that transports the data over the network and provides a means of checking for and correcting errors as they may occur.

Several types of networks exist. Local area networks (LANs) are networks that connect a small or medium number of computers within an office or a small physical area environment. A campus area network (CAN) is often used to connect two buildings or more within a very small geographical area. Metropolitan area networks (MAN) are used to interconnect facilities within a metropolitan or city-sized area. Wide area networks (WANs) are used to interconnect major centers over large geographical areas.

Several different formats for computer networking exist. Ethernet is the most popular commonly deployed network (Fig. 1). Ethernet is a shared media network, which means that everybody is talking on the same piece of wire and therefore only one person can talk at a time; everyone else has to listen and wait. Only the station to which the message is addressed is interested in the message. Everyone else has to wait their turn to get access to the wire. Gaining access to the communications wire is random and often prone to collisions when traffic levels increase. To help solve this problem, Ethernet networks can be segmented by use of a bridge or a router that provides isolation (Fig. 2). Traffic between computers from one side of the bridge is not heard by the network on the other side of the bridge unless the message is destined for that network, thus reducing congestion.

Another type of network is fiber distributed data interfaced, or FDDI (Fig. 3). FDDI connects the stations in a ring configuration, so that each station is connected to the stations on either side of it. A data token is passed from station to station. When in possession of this token, the station has permission to transmit data onto the network. It then transmits its data and the token down the network to the next station. In accepting the token,

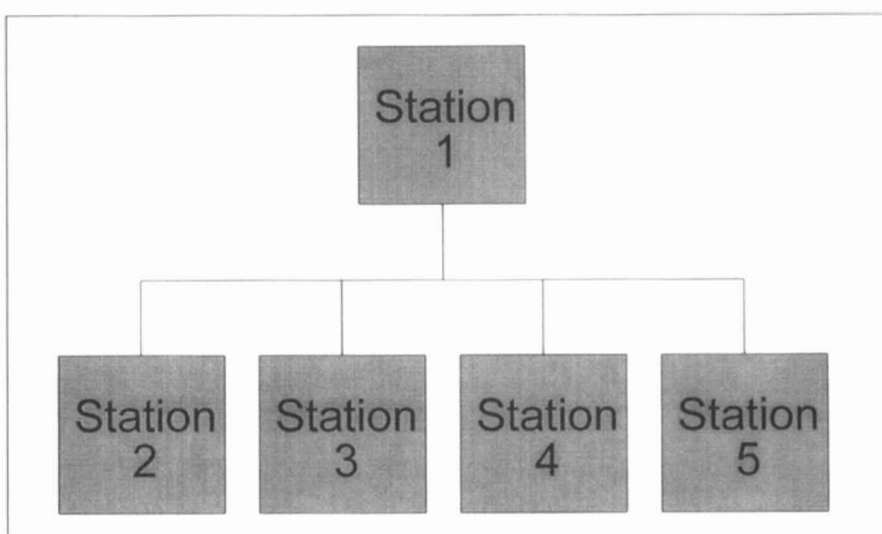

Figure 1. Block diagram of Ethernet.

This is the eighth in a series of tutorial articles by this author (and his colleagues). Randy Conrod is with Leitch Technology International, Inc., North York, Ont., Canada M3B 3J5.

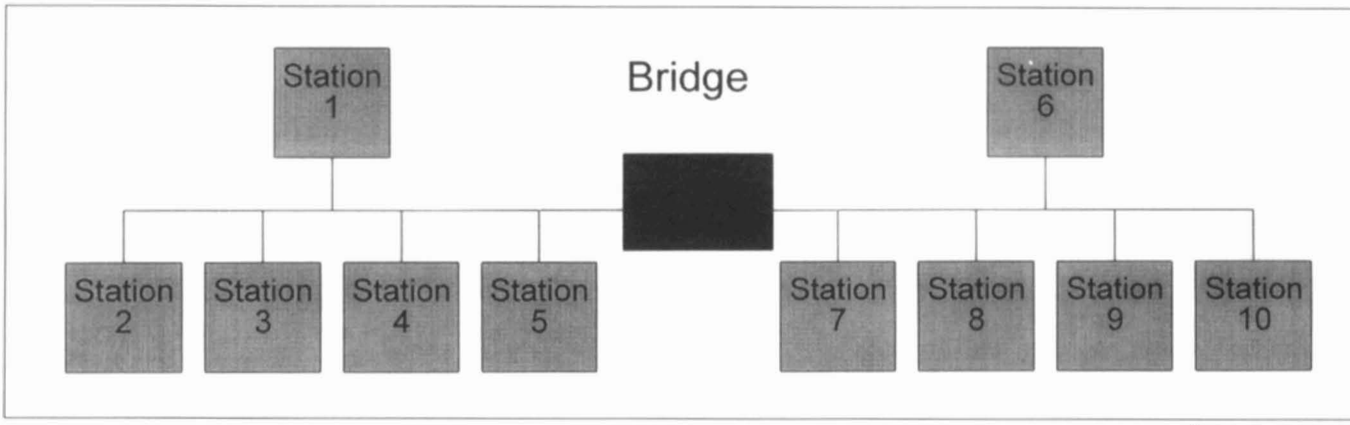

Figure 2. Ethernet network segmentation.

the station passes on the received data as well as its own data, if any. Receiving stations remove the data from the network and do not pass it on, thus opening up network time for subsequent transmissions.

Other types of networks exist, as well. Token ring is a traditional form of networking and is available up to 16 Mbits/sec. Ethernet, mentioned earlier, runs normally at 10 Mbits/sec. Fast Ethernet is a more modern variant running at 100 Mbits/sec. Switched Ethernet is a means of interconnecting Ethernet nodes through a switch so that each channel has full bandwidth available to it, providing it has a unique path to the destination it is communicating with.

Another network that is popular in some applications, including a few broadcast applications, is Arcnet.

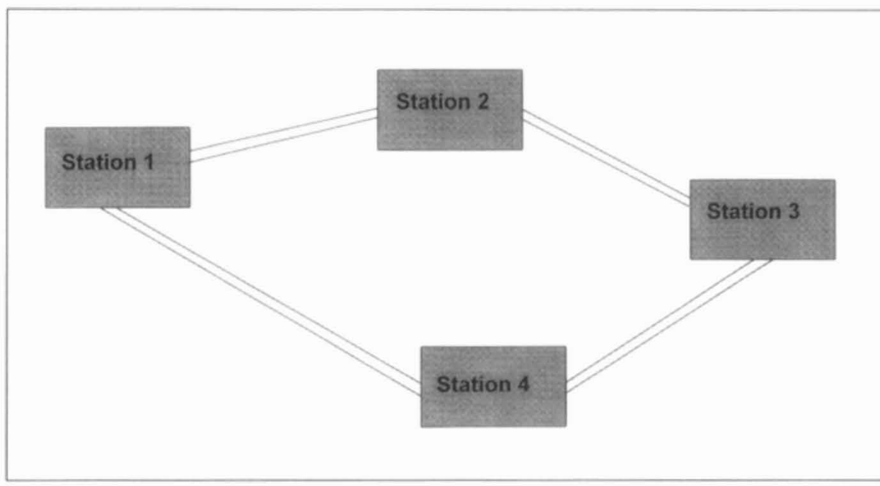

Figure 3. Fiber distributed data interface (FDDI).

- **DS0**
  - one telephone call, 64 kb/s
- **DS1** (T1)
  - 24 DS0s, 1.544 Mb/s
- **DS3** (T3)
  - 28 DS1s, 672 DS0s, 44.763 Mb/s

Figure 4. Telco transport hierarchy.

Computer Network Operation

Computer network operation has some requirements. Acknowledged data traffic is a requirement where the sender must be able to ensure that it knows that the data it sent was received by the receiver. Message broadcasting is required where one station is able to transmit a message to all stations on the network at one time rather than having to individually transport the message to each. Message routing is required so that stations interconnected by several networks can communicate over the various networks by means of the message transversing devices, which know where to route it. Call setup and teardown is done per message in these computer networks, and the data rates between users tend to be very variable. In some cases, a large program file may be downloaded to an individual workstation, and at another time a single keystroke may be traveling across the network to a remote computer. These are known as nondeterministic networks.

These nondeterministic networks tend to be "bursty" and non-real-time. Traffic may be switched, routed, and stored as required in going through these networks, accepting that the data is delayed as a result of being buffered. The problem with these networks for broadcasters is that voice and video can't be delayed; near-real-time performance is required.

Overview of Telcos

A quick look at the telcos is in order when discussing networking and telecommunications. Traditionally, they provide an analog interface to the subscriber; this has been the mainstay of their business to date. High-speed digital networks were only used by the telcos as their own internal backbones between toll offices. Bandwidth available from the telcos is low compared to their telephone network. Telcos use hierarchical multiplexing and demultiplexing to get many telephone calls onto one high-speed digital backbone and must be able to extract them wherever required to complete the circuit.

DS0 is one telephone call and has a bandwidth of 64 kbits/sec. DS1, or T1, is 24 DS0s and has a bandwidth of 1.544 Mbits/sec. DS3, or T3, is 28 DS1s, or 672 DS0s, with a bandwidth of 44.763 Mbits/sec (Fig. 4). Beyond DS3, formats exist but they are not standardized or intercompatible. Because of this hierarchical structure, it becomes difficult and expensive to multiplex and demultiplex telephone calls.

Telephone network operation is somewhat different from computer network operation; network voice traffic is not acknowledged traffic. The data, generally a human voice, is simply sent down the network. Although the line may be prone to noisy conditions, the communications remain robust because there is an intelligent receiver at the other end. Telephone connections are point-to-point. Broadcasting is not allowed because it is not required. Call routing is carried out on telephone calls when a telephone number is dialed; the call is routed at that time. Another difference is that this is a very deterministic network in that the data rate between users is fixed.

When a telephone call is placed, a DS0 channel at 64 kbits/sec is set up in each direction. This provides the necessary bandwidth and is reserved entirely throughout the duration of the call. This is one of the keys of the telcos — the bandwidth is allocated at the time of the call and the entire bandwidth is available for the duration of the call. Huge amounts of bandwidth are wasted during the pauses in normal speech. There are bits available that somebody else could have used when there is silence during a conversation.

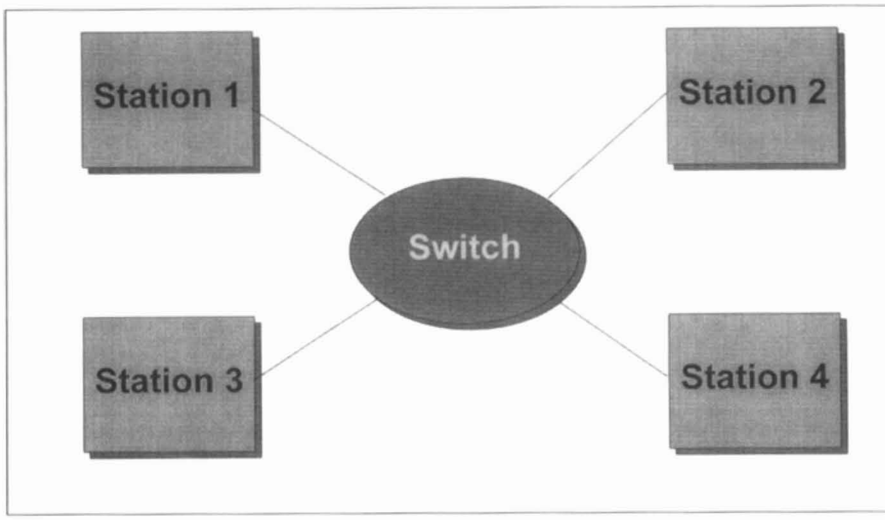

Figure 5. The ATM switch.

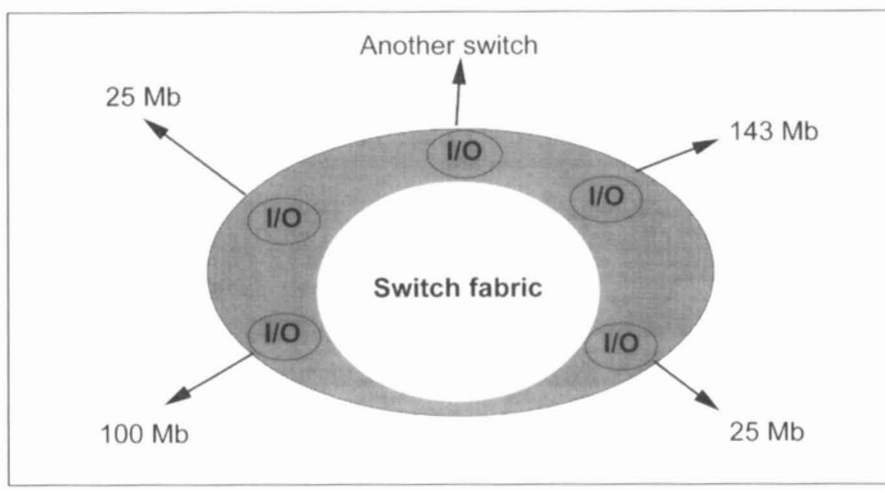

Figure 6. Switching fabric and operation.

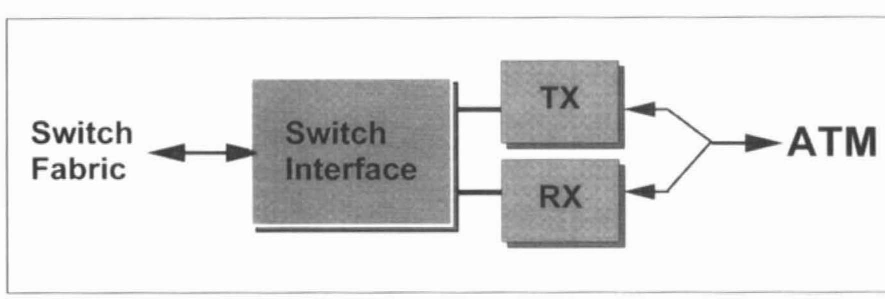

Figure 7. Switch I/O module.

What is Broadcasting?

The signal requirements of broadcasters are as follows: baseband analog video and audio; digital technologies, such as AES/EBU digital audio; digital composite video; digital component video; serial digital video with embedded audio; and data with growing requirements for various compressed services. Broadcasters need guaranteed quality signals, real-time transport of program material, signal routing, and greater than real-time transport.

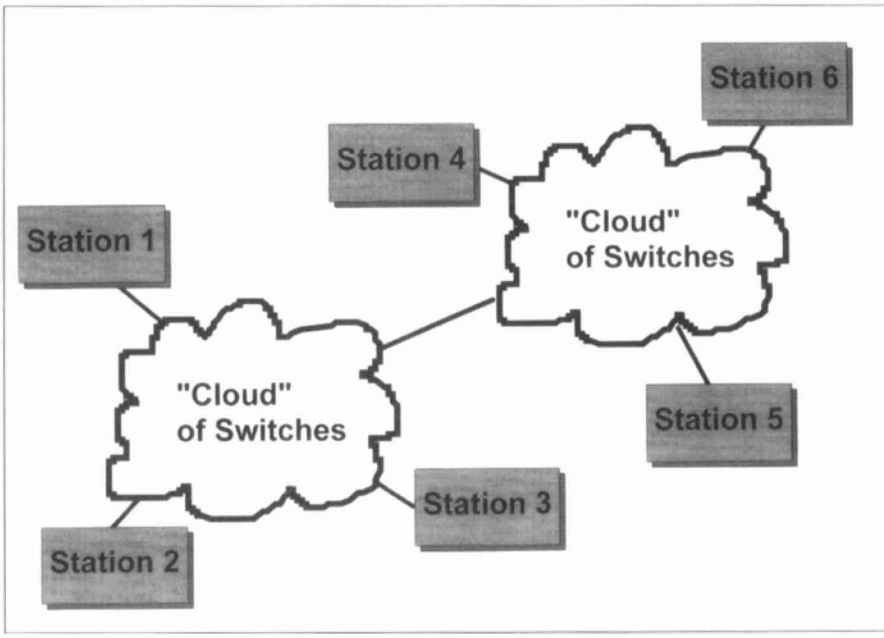

Figure 8. ATM network topology.

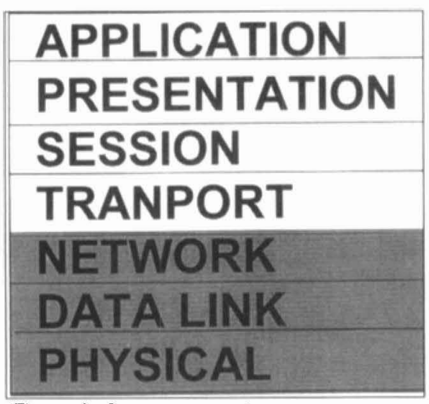

Figure 9. Open systems interconnect.

Traditionally, broadcasters use path-based interconnection. These are connections that run hard-connected in a permanent circuit fashion from one location to another. Sometimes they run one way — such as in the case of a studio whose output signal feeds to a master control facility. Sometimes they run both ways, such as in a post-production facility where signals would be acquired from the plant system, as well as distributed back into it. Computer graphics suites and audio mixdown rooms would be examples of this.

Broadcasters have various types of data to transport on a communications network, such as program video and program audio with multiple channels thereof, machine control for remote control operation of various production machinery, and ancillary data services that would be useful in the teleproduction process.

The signal transport mechanism must provide a low error rate for broadcast data. There must also be a transport mechanism and inherent ability for acknowledged data service to be provided over the same transport channel. High data rates are necessary to accommodate high bandwidth and high-resolution video. The data rates must also be scalable; in the case of high compression perhaps MPEG (Motion Pictures Experts Group). Extremely low data rates must be accommodated with similar ease.

The broadcaster's network must have the ability to route many signals to a number of places. Broadcasters' traditional production styles are based upon the use of routing switchers, which provide the ability to route many signals to many places and that follow the production flow. This includes the ability to map signals one to one across the network, as well as one to many. An example of this might be a source of color bars in the master control facility in a television plant. The routing switcher must provide control over the paths by a number of users to many pieces of equipment on a continuously changing basis as resources are reallocated. In a television facility, these interconnections must be nonblocking — you can never have a busy signal.

Greater than real-time transport is a very important requirement for broadcasters. The present media must go on and off videotape at real time, so dub times, play times, and all transfer times take as long as the program to move. With the advances in data compression including such things as JPEG (Joint Photographers Experts Group), MPEG, wavelets, and fractals; as well as the increasingly faster transport layers available in computer networks, greater than real-time video and audio transport becomes a reality. Storage equipment, such as digital tape drives, can be used to retrieve material onto a disk server for real-time playout. The material can be acquired at many, many times real time. At some point in the future, 100 times real time will be possible, allowing smaller disk buffers to be required and near-instant access to vast amounts of material.

Why Not Use Existing Technologies?

Why not use existing technologies for broadcasting networks? Present networks, such as shared media networks, have low throughput by virtue of their shared nature. Ring networks are a fixed structure and only transport data at a fixed rate around the ring. In addition, internetwork connections using routing cause delays, which in real-time television are unacceptable.

At present the telcos cannot achieve the desired bandwidths with their existing communications networks. These are designed for telephone calls and are not especially suited for the type of needs that broadcasters have.

Asynchronous Transfer Mode (ATM)

ATM is a technology that answers the problems for both broadcasters and telcos. Telcos want to be able to easily assemble and disassemble traffic on and off their networks; this would mean that less switching equipment would be required to accomplish this task. ATM transports any kind of computer or voice data and can be easily switched in and out of the network by virtue of the nature of the ATM

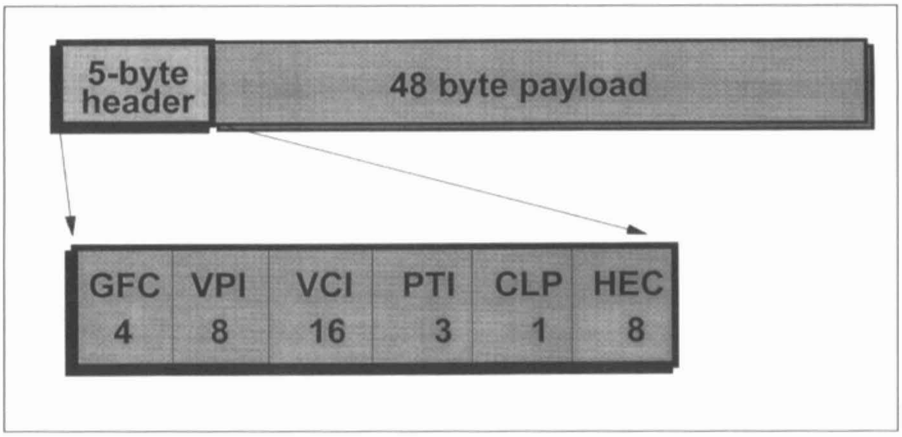

Figure 10. ATM cell structure.

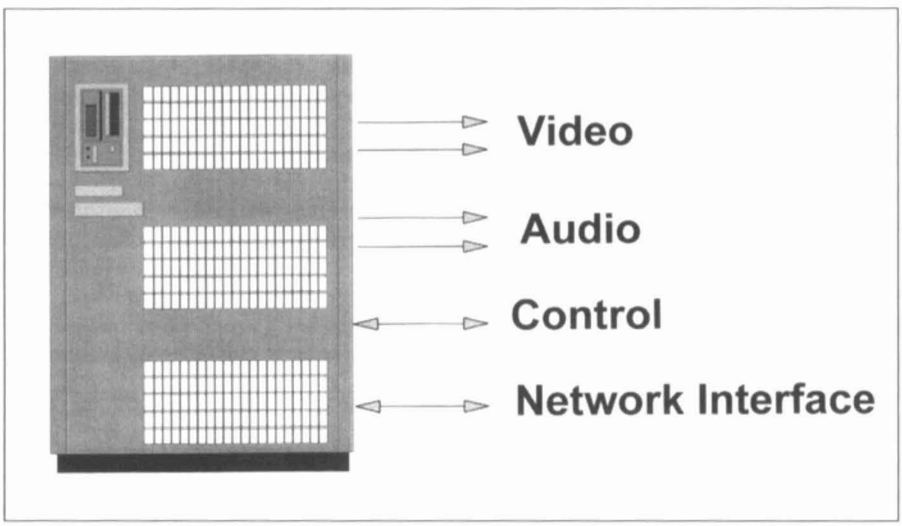

Figure 11. File servers.

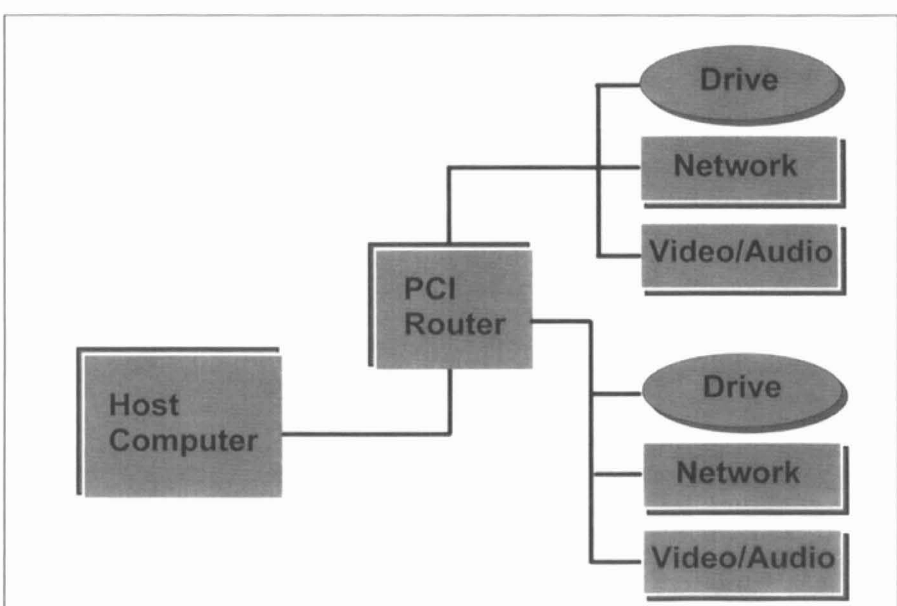

Figure 12. Multi-PCI bus server architecture.

cells. This means the telcos can maximize the use of their resources and sell your unused bandwidth when you pause during a conversation on the telephone. Other types of data would be transferred during the pauses in the conversation.

ATM is a small cell-based transport scheme. It is based on a 53-byte basic cell. Although this cell seems small, the simplistic addressing and routing schemes allow for great flexibility. It has a switch-based physical topology, where every user of the network has a unique connection to a switching device that will, in real time and on the fly, route traffic to the destination port of the switch without any significant delays. ATM provides switched and permanent circuits, meeting the requirements of both telcos and broadcasters.

ATM is available at many speeds. It is fast and provides 25 Mbits/sec, 45 Mbits/sec, 100 Mbits/sec, 155 Mbits/sec, 622 Mbits/sec, 1.2 Gbits/sec, and 2.5 Gbits/sec.

Presently available for ATM are workstation adapters that plug into a computer. These adapters are now available for most buses and have drivers for operating systems. They come today in 25 Mbits/sec, 45 Mbits/sec, 100 Mbits/sec, and 155 Mbits/sec, and 622 Mbits/sec will be available in the not-too-distant future for server connections. Not all computers, however, can sustain the data rates that these cards allow. Presently they are used heavily in LAN extension, where 10 or 12 Ethernet ports are multiplexed onto a single ATM channel for a run down a fiber to another box that has 10 or 12 Ethernet ports for use in a large hub.

The ATM switch is a device that can have input/output (I/O) modules to connect to various other networks or computers at differing data rates (Fig. 5). The only thing that is of interest to the switch is the fact that 53-byte cells are arriving and leaving its switching fabric, or "guts." The switch must sustain rates coming from different channels. It contains an internal mechanism for forwarding the cells, on the fly, to be routed to the correct outgoing port (Fig. 6).

The ATM switch I/O module provides two-way communications between the ATM transport and the switching fabric (Fig. 7). ATM network topology allows for stations such as workstations, hard disk arrays, file servers, studios, TV stations, etc., to be connected. The signal is sent into a cloud of switches that provide the interconnection between devices (Fig. 8).

ATM provides the bottom three layers of the ISO-OSI (International Standards Organization - Open Systems Interconnect) standard (Fig. 9). This is a seven-layer model for telecommunications networks. Of these ATM levels, the lowest (level one) provides the definition of the physical and transceiver interface. Level two provides the data link layer, which includes a media access scheme, framing, encode and decode functions, and error detection. Level three is the network layer. This layer has the logical addressing and provides media-independent interfaces. In addition, it provides unicast, multicast, broadcast, and routing functions.

ATM lies at the bottom of the Telco B-ISDN (Broadcast Integrated Services Digital Network) model. All B-ISDN services ride on an ATM core. The use of ATM adaptation layers allows for various applications above the first three layers.

What allows the ATM cell to be so "switchable" is its structure (Fig. 10). A 5-byte header contains channel and path addresses. The switch knows about these channels and paths and can strip this data very simply off the cell and make a very quick decision in hardware and pipe the cell to the appropriate output port. The header contains two vital addresses out of the six fields used in the header. The virtual path indicator (VPI) is the pipe

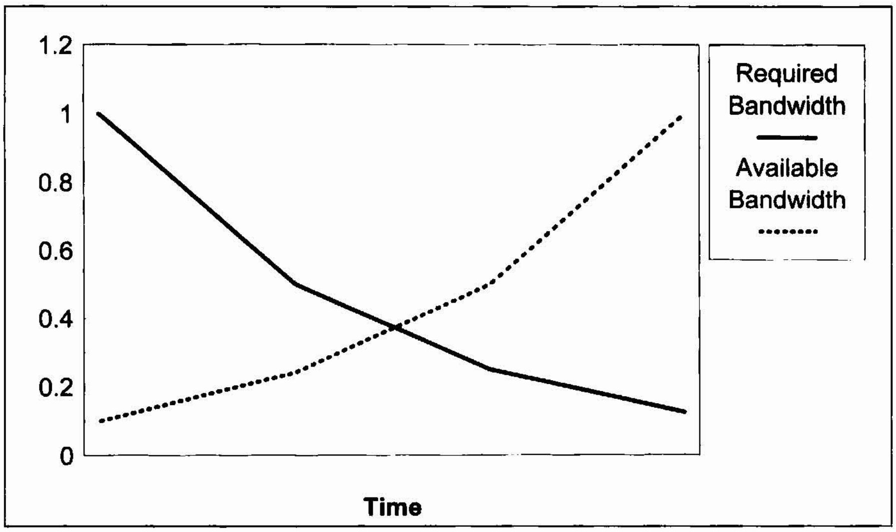

Figure 13. Data compression and network bandwidth.

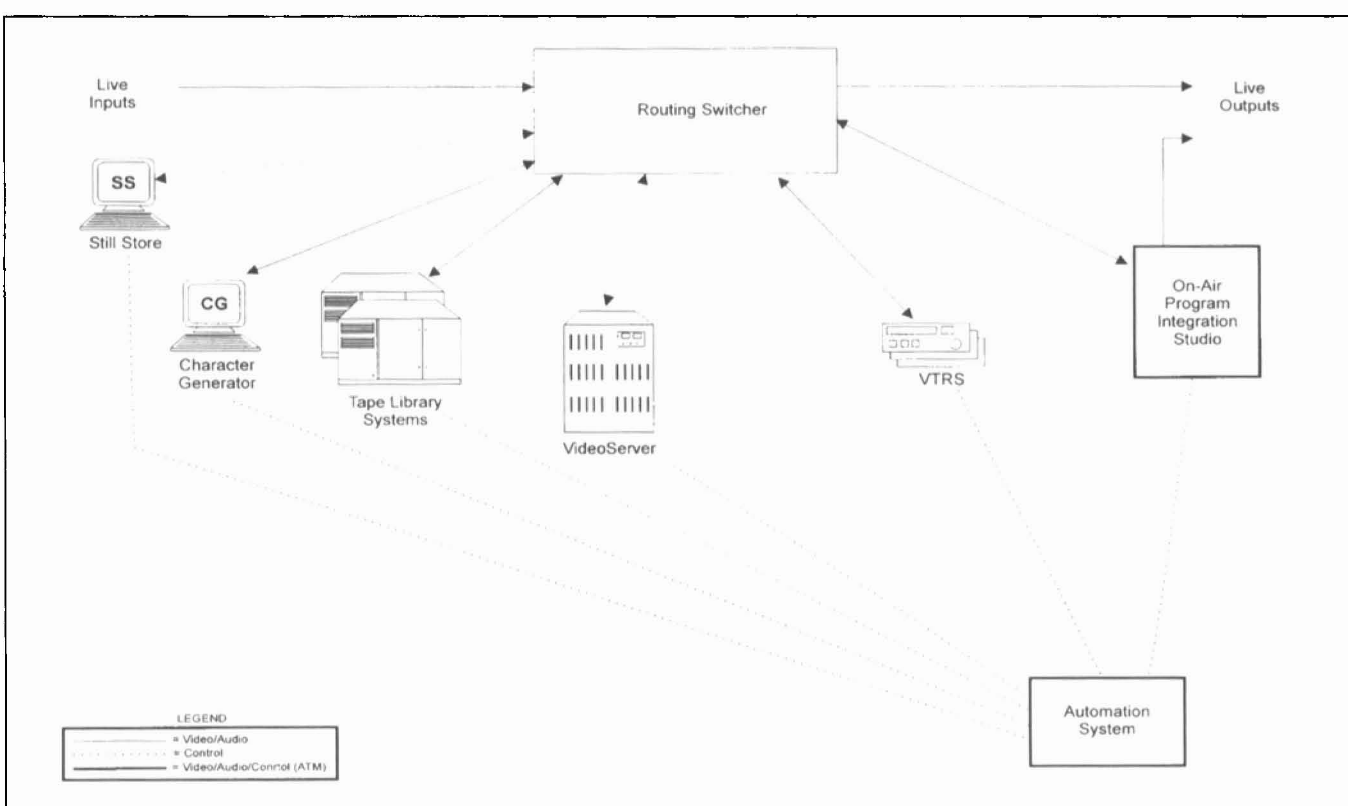

Figure 14. Traditional presentation system with video server.

down which the data is traveling on the network. The virtual channel indicator (VCI) is the address that indicates which channel within that pipe this data represents. The cell loss priority (CLP) is an interesting bit for computer networks. This bit, when set, means that this cell can be lost, if necessary, due to congestion along the network. This data is a type that can be lost without consequence, or data that contains a protocol that will allow resolution of this loss.

ATM will run on various physical layers; it will run over synchronous optical networks (SONET) in telco networks. Fiber or coaxial cable can be used. It will also run with coding such as 4b/5b and 8b/9b over fiber, twisted-pair cable, or coaxial cable in computer networks. The 4b/5b and 8b/9b coding techniques are used to get the signal on and off the wire in computer networks.

File Servers

A file server is a computer containing a motherboard for system control (Fig. 11). It has an I/O device providing video and audio interfaces, as well as a control interface and internal storage devices, which are usually disk drives. Optical or steaming tape drives are included for archiving purposes or near-line access (as opposed to on-line or off-line). A network interface provides communications with other devices. At this time, there is a bottleneck in network interfaces using Ethernet. File servers are intended to take streams of video and audio in and out and sometimes apply compression and decompression as required. File servers using today's technology have three problems.

The first of these problems is sustaining the 22 Mbit/sec data (parallel format with no sync information) required for uncompressed D-1. The small computer systems interconnect (SCSI) bus used at this time will only support 20 Mbits/sec. Dual drives or compression can be used to sustain D-1 data rates.

The second problem with file servers at this time is the fact that network interfaces have a very low speed. As such, the data can only be moved in the background at much less than real-time speed.

The third problem with file servers is "bus bondage." This is the phenomenon that occurs once a single hit on the internal bus occurs, followed by another hit and then another. As each hit occurs the resources are diminished, especially if a network connection is used. Depending on the amount of traffic on the bus between the various media and I/O cards, bandwidth can quickly evaporate. Higher integration of function on a card, such as network and video/audio I/O, will help alleviate this problem.

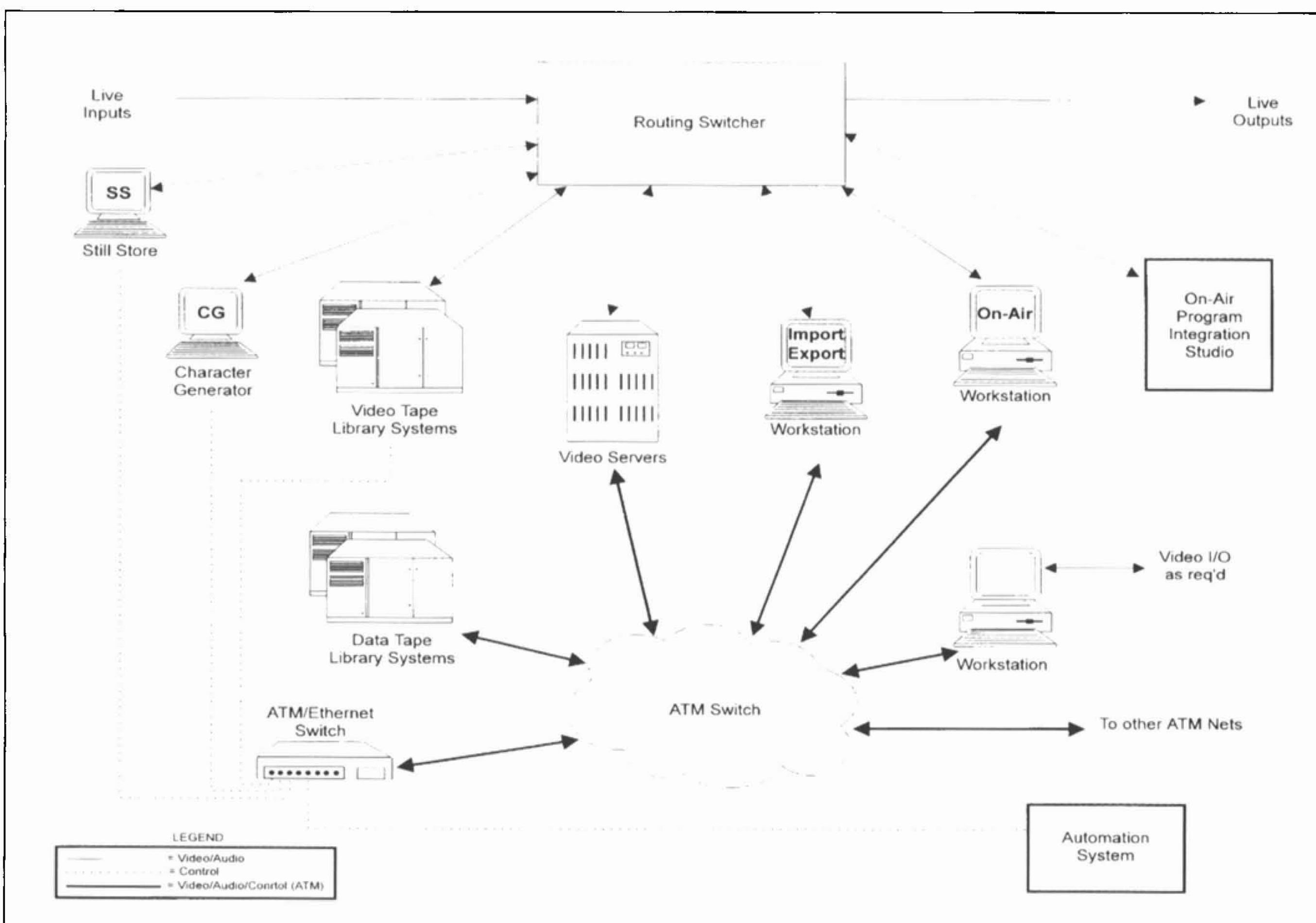

Figure 15. Hybrid video/ATM presentation system.

Peripheral Connection Interface (CPI)

The PCI bus is a better computer bus and is the "hot" bus at this time. It can sustain data rates of 132 Mbits/sec. A multimaster architecture allows for many devices to take control of the bus, as they require, to transfer data along the bus. This is good for building broadcast equipment because network interface cards, disk access cards, and video/audio I/O cards can, as required, initiate their own use of the bus. The use of multiple PCI buses in a host computer requires a PCI router (Fig. 12).

Data Compression

Broadcasters traditionally use real-time data for historical reasons because the signal comes out of the camera in real time, is recorded and played back in real time, and is transmitted in real time. Several types of data compression can be considered: data-only versus real-time compression, numerical, perceptual, and some future possibilities.

Data-only compression involves taking only the picture data from the video signal and transporting it in contiguous data blocks, without transporting the sync intervals. This would result in a 20 to 25% reduction when combined with encapsulated and transport-reduced AES/EBU. It is a fixed data reduction and so is completely lossless.

Numerical compression would involve pure crunching of numbers and would be like a PKZIP for media. It would provide a variable data reduction of some questionable effectiveness but a lossless performance.

Perceptual compression is the wave of the future. This involves compressing data so that data lost is not visible to the viewer. Techniques include compression formats such as JPEG, MPEG, and MPEG-2. This type of data compression is available in fixed or variable-rate reductions. This type of compression is lossy; in other words, data going into the compression device is somewhat different from what is coming out of the decompression device. Future possibilities for compression involve picture interpretation, wavelets, fractals, real-time texture mapping, and real-time rendering.

When compression and decompression are used within a file server, things operate in a different manner. All data in and out of the file server is compressed onto the internal bus and decompressed as it comes off the internal bus. The data on the internal bus (such as the faster PCI bus) and

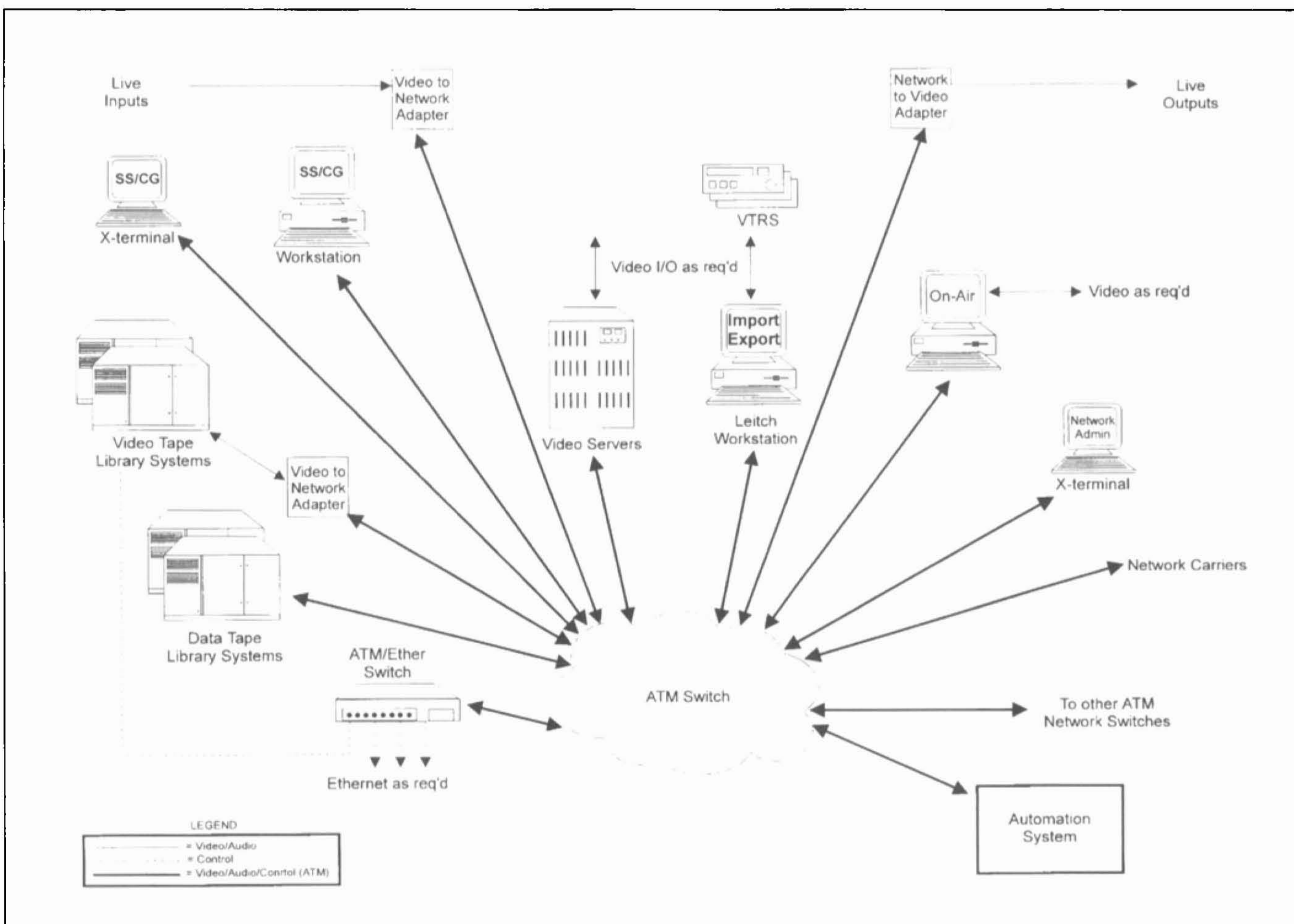

Figure 16. ATM-based presentation system.

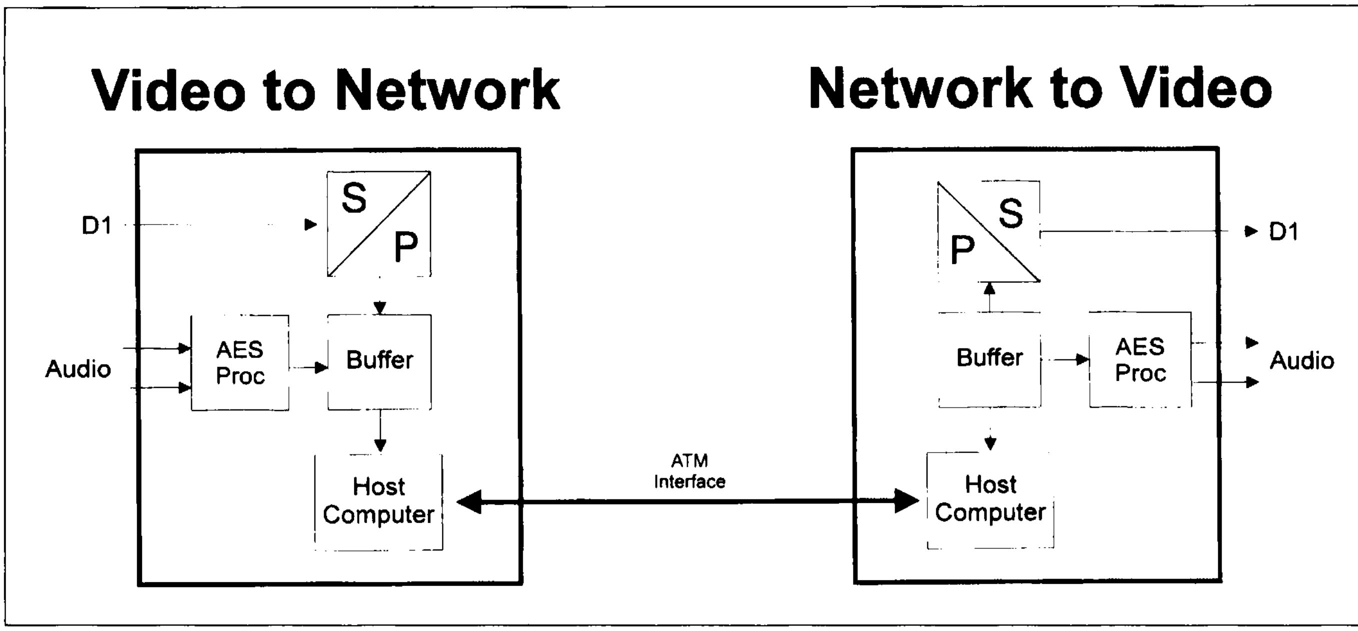

Figure 17. Converters required for ATM-based presentation system.

ATM network is real time or greater than real time.

As compression improves and network bandwidth goes up, the improvements are compounded. In the next 18 months to two years, compression and network technologies will allow bandwidths sufficient for extensive broadcast use. In the near future, technology of this type will allow for the use of file servers extensively in the teleproduction and broadcast processes (Fig. 13).

ATM and Broadcasting

With ATM transporting television media data, routing switchers are replaced by a "cloud" of ATM switches. Control is performed by a routing agent that is essentially a switch control and cell-routing scheme. No physical routing switcher is needed.

The routing agent manages bandwidth resources, manages permanent and switched connections, and ensures that selected sources are available to destinations, as required. Routing is accomplished by cell addressing and the cells are distributed by ATM switches. Cell replication and re-addressing is done as required by the system.

In a traditional presentation system with a file server, all signals are hard wired. Video and audio may be separate or embedded, control is via dedicated links, and file servers are I/O devices (Fig. 14) .

In a hybrid video/ATM presentation system, an ATM switch, or "cloud," is added. The traditional equipment still feeds in and out of the routing switcher, but things like data-type library systems, workstations, and video servers, as well as other network connections like Ethernet, feed in and out of the ATM switch (Fig. 15).

In an ATM-based presentation system, all devices feed in and out of the ATM switch. Video and audio are converted in and out of the ATM switch, as required. Functional items like switchers, character generators, and still and clip stores become routing agents. Interconnection with other networks and long distance/distributed teleproduction becomes possible (Fig. 16).

Video-to-network and network-to-video converters will be required for an ATM-based presentation system in the television station of the future (Fig. 17).

ATM would be useful for applications such as monitor walls, incoming conversions, and outgoing feeds — and no timing is required!

ATM Forum

The ATM Forum sets standards for electrical and optical interfaces, as well as media access and formats. It is also developing adaptation layers. The Forum was established as a result of industry impatience with the CCITT (International Telegraph and Telephone Consultative Committee). It is now the accepted standards body for ATM networking and reports to the CCITT.

Who's the Big Winner?

So, who is the big winner using this technology? Telephone companies have the switching infrastructure to make use of this technology. Cable companies have the bandwidth to the subscriber. Possibly an unknown company may "pass on the left and scoop the market." Users will have a choice between competitive bandwidth providers and can use a standardized data interface format.

Do broadcasters see any of this soon? Three factors will push this technology forward: the advance of multimedia, the acceptance of compression, and the divergence of markets. Don't hold your breath — it may take some time. Watch the telephone companies, watch the server manufacturers, and watch for networked teleproduction.

Acknowledgment

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