The Public Network, Making it work for Digital Video And Entertainment Production.

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This paper is designed to assist broadcasters, motion picture and television production management, and information technologists, understand the issues surrounding moving content between facilities across town or across the oceans.

The digital television revolution has created confusion and opportunity. TV broadcasting has taking on a new meaning to us all. Television is now digital, interactive and will be delivered to your home via methods different than before. The range of digital video, and entertainment products now include, direct satellite, digital terrestrial broadcasting, DVD, and Internet Broadcasting. All of these new sources and distribution technologies are based on digital.

The demand for content or software as it is called in the consumer electronics business has created new centers of content creation. The number of people required to produce the thousands of hours of content consumed each week is rising quickly. No longer can you count on all of your team or resources being in the same city or state. The Internet and high-speed modems have fostered the idea of collaborating across the "Ether," and it works. Video-conferencing that was the sole domain of large corporations is now available to everyone who owns a PC and an Internet connection. Broadband communication is the current wave of technology to provide new and exciting tools for the creative professional.

Local Area Network or LANs and Wide Area Networks or WANs, are familiar too many of you. You may already have a LAN in your facility or company. If you have multiple offices you may be on a WAN or if you use the Internet you are connected to a WAN. Choices for LANs include, Ethernet 10/100, FDDI copper and fiber, Token Ring, HIPPI, FDDI, ATM and Fibre Channel. Most networks deployed today utilize the Open Systems Interconnect (OSI) standard.

Open System Interconnect

	Number	Name
OSI Layer	7	Application
	6	Presentation
	5	Session
	4	Transport
	3	Network
	2	Link
	1	Physical

Figure 1

WAN choices include ATM, SONET, dedicated video services and the Internet. When considering a network for the transport of video or Production traffic you need take into consideration the bandwidth requirements, the suitability of a network service to your requirements as well as the availability and reliability of the network. The requirements might include; how scaleable is the network, types of traffic they support, standards compliance, ability to interconnect with existing technologies and often the most important component - cost.

As we examine the world of public transport today we find a plethora of transport services being used to move information from one location to another. Some are primarily used as static point to point services, others are more functional in a switched environment and some are application specific.

Following are some of the network options available today to transport voice, data, video and videoconferencing bit streams between locations either locally or around the world:

DS-1 (or DS-1) – 1.544 megabits per second

Integrated Services Digital Network (ISDN)

Basic Rate Interface – BRI (2x 64 kilobits per second)

Primary Rate Interface – PRI (1.544 megabits per second)

Digital Subscriber Line – DSL

Cable Modems

SONET (Synchronous Optical Network) & Asynchronous Transfer Mode (running over SONET)

DS3 – 45 megabit per second (Asynchronous traffic mapped onto SONET)

OC3c – 155 megabits per second

OC12c – 622 megabits per second

TV-1™ and Advanced Broadcast Video Service™

DS-1

Primarily a copper based service that has the capability to move information across the network at 1.544 megabits per second. This was the first truly digital facility offered by the Telco's and is still the most commonly installed transport facility. It is used by end users to connect facilities together for transport of voice, data and video. These services are primarily a point to point link with one source and one sink. Typically DS-1 service is billed at a fixed monthly fee based on mileage with no usage charges. It has become the transport of choice to connect to an ISP for "high-speed" Internet access by many end users.

Integrated Services Digital Network (ISDN)

ISDN is also a copper based service that has the ability to move bits across the network at speeds from 64 kilobits per second (Basic Rate Interface – BRI) up to 1.544 megabits per second (Primary Rate Interface – PRI). ISDN allows the aggregation or bonding of channels together in increments of 64 kilobits per second. Typically ISDN is used for data transmissions or video teleconferencing at speeds of 384 kbps or 1.544 mbps. It can also be used for voice telephone calls or Internet access. ISDN is a switched connection with costs based on called location and connect time.

Digital Subscriber Line **-** DSL

DSL is also a copper based service and comes in a number of flavors such as Asymmetrical or ADSL, Symmetrical or S-DSL, and High Speed or H-DSL. The primary differences between DSL types are the differing transmission speeds in the transmit and receive directions on the service. A typical ADSL has a 384 kbps transmission from the end-user to the network and DS-1, 1.544 mbps form the network to the end user. A SDSL service provides the same amount of data flow in both directions and an HDSL line has the capability to go at speeds up to DS3 or 45 mbps. The DSL service is primarily used to allow higher speed Internet access by small and medium sized business and residential users while simultaneously allowing voice traffic over existing Telco copper lines.

Cable Modems

Cable modems are similar to a DSL connection except that a cable company provides the connection over their existing distribution plant. The initial offering of the service is primarily for residential customers for Internet access in an asymmetrical mode similar to an ADSL line. Cable modems also have support voice traffic and the simultaneous delivery of cable television services.

SONET & ATM
DS3 – 45 megabit per second (Asynchronous Traffic mapped on SONET)
OC3c – 155 megabit per second
OC12c – 622 megabit per second

These services are delivered to end users via fiber optic cable rather than copper. These services belong to the class of transport called SONET, **S**ynchronous **O**ptical **Net**work. They are similar to the workhorse DS-1 service in that they can be provisioned as a point to point service and support voice, data and video streams. However, these services can support switched traffic when they are provide the lower transport layer for ATM or Asynchronous Transport Mode services. Using the ATM data link layer 2 protocol, traffic on these high-speed links can be dynamically configured to any speed and to any other ATM user on the network. The service can be used for Internet access, voice, data, and video transport. Typically SONET services today are offered as fixed monthly billing. ATM service is also being offered at a fixed monthly rate by carriers, but the direction of ATM services is to offer a usage option based on bandwidth and Quality of Service (QoS).

TV-1
Advanced Broadcast Video Service - ABVS™

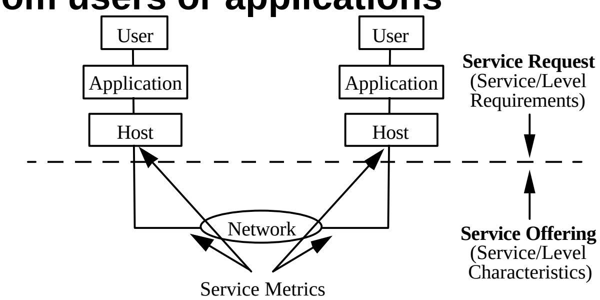
These services are also delivered to end users via fiber optic cable rather than copper. They are used for the transport of NTSC, PAL, or SMPTE 259M video in a compressed or uncompressed format. TV-1 is the original Telco standard for delivery of broadcast video over a wide area network. An uncompressed signal is delivered locally to the receiver in the uncompressed, original format. ABVS™ service takes the NTSC, PAL or SMPTE 259M signal and compresses the signal to be transported over the WAN as a DS3, or 45 mbps signal. These services are single function overlay networks and do not support voice or data transport. The connections are able to be switched today via human intervention at a central operating center. The two video transport services are typically charged on a fixed monthly fee with a cost per switch in destination. When switching over long distances a usage fee is often applied by the long distance carrier such as Vyvyx™

As you can see, traditionally separate networks have been built for each type of service: voice, data, and video. This approach is expensive to carriers and users alike since each service requires the Telco to install and maintain a separate set of dedicated personnel, tools, training and networking technology. For the end user separate networks normally mandate separate access fees and rates for each service plus the costs of implementing multiple technologies at the user locations.

One network today has all the Quality of Service standards defined to meet the needs for a Mulitservice Network. That network is ATM or Asynchronous Transfer Mode.

Services and Quality-of-Service

Services in networks are sets of performance and functional characteristics, tied to service requests from users or applications



Quality-of-Service (QoS) is instrumenting the network to make network resources available to meet these service requests

Quality of Service (QoS) is a term that has taken on many definitions in the last year. Originally, QoS was a term used by the ATM (Asynchronous Transfer Mode) community to define a series of parameters that, when combined together describe the ability to support many different types of traffic with very precise controls on factors such as latency, loss and jitter. In the interim, a number of other networking technologies have also begun to use the term QoS with a slightly different definition of the term. Specifically, LAN technologies such as Ethernet, and WAN technologies embedded in the Internet based on TCP\IP (Transmission Control Protocol\Internet Protocol), are seeking to implement a method to control and manage the vast growth in traffic that are run over the Internet and data networks today.

QoS Characteristics

- QoS Contract Established at Beginning of a Connection
  - Cell Loss Ratio: How many cells have been discarded because of excessive latency
  - Cell Transfer Delay: Node to Node transmission Delay. Inclusive of queing+switching+routing delays.
  - Cell Delay Variation Tolerance: Amount of allowable delay between cells. Implemented to smooth inter-cell arrival time and control jitter.

QoS then is a way for you the network user to gain a bit more control over the transport of your information – data or video, over a network. Specifically, today's the focus is on LAN and WAN networks based on the ATM protocol since those networks are both common to WAN service providers and to LAN networks – offering a seamless end-end control over the network QoS for the video transport application.

ATM as a technology is a result of the quest by public telephone networks to merge the disparate single-purpose networks that had grown up since WW II into an infrastructure that would allow then to support traditional voice telephony, data transport and video over a common infrastructure. One driving motive on this was the need to move off of the first-generation digital transport networks that had come about in the mid – 1960's…the familiar DS-1 and DS-3 services. These technologies are digital, but lack many operational features required by a more competitive, deregulated telephone industry such as add\drop capabilities for signals and the ability to interoperate on a worldwide basis. With these requirements, Synchronous Optical Networking, or SONET technology was created. SONET is now a worldwide standard for transport in public networks, acting something like the concrete used to construct a roadway - the basic foundation for everything that will be transported over that roadway. ATM is the protocol that runs over SONET in order to offer services such as QoS to user applications.

The real question though is – What is QoS and why should you care? Very succinctly, a user cares about QoS because modern WAN's involve a tradeoff in cost versus performance. The traditional video transport services were based on a dedicated connection between two points using a DS-3, 45mb\s connection and codecs to compress the video signal. Modern communication networks are based on switching traffic – voice or data and video on a common infrastructure – SONET based, that increases the utilization on the network over a dedicated line configuration. QoS then becomes the essential factor used to allow a carrier and user to discriminate among different traffic types. QoS provides the control mechanisms for accomplishing this purpose. Specifically, ATM-based networks allow precise control over latency, loss and jitter. ATM networking also defines a number of Classes of Service that provide a variety of inherent QoS values and which are designed to support specific traffic types, including real-time uncompressed SMPTE 259 and 292 video.

Class of Service	ATM UNI 3.1	ATM UNI 4.0	Service Description
CBR	Yes	Yes	TDM digital public/private line services required for voice and Video (Broadcast Video)
VBR-rt	Yes	Yes	Intended for packet switched video and audio in teleconferencing and multimedia (non Broadcast)
VBR-nrt	Yes	Yes	Traditional "data" applications such as frame relay, packet switching and IP
UBR	Yes	Yes	"Background" applications with built in cell tolerant retransmission schemes e.g. file transfers, Email
ABR	No	Yes	Allows flow control mechanisms for applications with high cell delay tolerance and low cell loss requirements, e.g. LAN interconnect and interworking

Figure 2

These separate Classes of Service are designed to specifically meet the needs of individual traffic types. For example, voice is very time sensitive, so real time service is critical. Video is also a time sensitive service that requires a Constant Bit Rate or CBR with tight control over loss, latency and jitter. Internet traffic and File transfers, which cover most of the data traffic we encounter can fluctuate in requirements and performance, commonly use the Unassigned Bit Rate or UBR service.

Network Choices for broadband communications: IP, ATM and others

Public Networks are built around standards. Those standards have defined bandwidths to meet the requirements of the voice and data traffic they were designed for. The Entertainment industry, Broadcasting, and the Tele-production industry have their own unique bandwidth requirements. The chart below shows some of those bandwidths for our commonly used Digital VTR formats.

Avialable Bandwidth

compared to

VTR/DDR Requirements

Carrier Standard/ Bandwidths	I/O Format Bandwidth	Internal Recorded Bandwidth Mb/s
DS-3 45 Megabits Per Sec.		***Sony SX @ 18*** DVC Pro @ 25 Tek PDR 400 @ 25 Tek PDR 300 @ 24 MPEG2 4:2:0 @ 4-15 MPEG2 4:2:2 @ 4-50
OC-3 155 Megabits Per Sec.	ITU 601 could fit into OC-3 with some light compression	DVC Pro 50 @ 50 TEK PDR 300 @ 48 Tek PDR 200@ 48 Dig. BetaCam @100 ***HDCAM @ 140*** ***DVCPRO100 @ 100***
OC-12 622 Megabits Per Sec.	**ITU 601 I/O (Used on most digital VTR's and DDRs)**	***D-1 @ 170*** ***D-5 @ 235***
OC-48 2.4 Gigabits Per Sec. OC-192 10 Gigabits Per Sec. (40 Gigabits/Sec with DWDM)		HDTV uncompressed @1.5 **Gb/s**

Figure 3

The most common transmission connection today is DS-3 at 45 megabits per second. DS-3 is normally a copper connection. OC-3 at 155 megabits per second is becoming more popular with the introduction of more fiber in our cities. As you can see by the two charts the compressed VTR’s and Disk Recorders, fit into the DS-3 45 megabit per second bandwidth. Unfortunately our studio infrastructure is built around the ITI 601 270 Mbs standard.

Transporting video has the same requirements as live video itself. Live video transport technology must meet or exceed the real-time requirements. Professional video is defined as constant frame rate 30fps (29.97fps) for NTSC (525) and 25fps for PAL (625), HD can be either 29.97 or 30fps.

Internet video and distance learning allows users to take advantage of the broad range of performance the IP network protocol provides. These services can take advantage of ISDN or the Internet so that users at all levels can use "video". However, professional video for content creation or contribution must maintain real-time with very high standards of quality and minimal loss.

What is real-time? A simple description of real-time vs. non real-time is offered here. A real-time transport system must be able to provide data reliably and on time, hence the real-time definition. The concept of real-time is always interesting to explain to our colleagues from the computer world. When you talk about storing video on a computer most people rely on the computer's Operating System (OS), to manage the disk drives and files. Most modern operating systems guarantee your data because they tell the disk to write the data and then do some form of checking to see that the data is recorded perfectly. This is fine when saving your word processing files. In the case of storing real-time video the requirements are different. Real-time video does not offer any chance for a re-try. Shall we ask the football player to redo the kick since we had a bad write?

A VTR does not include such a checking or retry mechanism. If it were possible the tape would have to move forward and backwards in the event that the frames were not recorded. The VTR would also have to do this in a very short time since the video input is constant. Analog VTR's have drop out compensation to deal with anomalies in the recording surface of the tape, scratches or dirt. Digital VTR's utilize tape as well, and subsequently suffer from the same potential problems. Digital VTR's incorporate advanced protocols and digital techniques for error correction and concealment. Reed Solomon and Huffman encoding are common techniques for dealing with the potential errors encountered in a digital VTR. Digital Video Disk recorders may incorporate these re-try mechanisms when you give your recorder greater bandwidth rates than the incoming video or you compress the video and you have time to re-try the write via buffering.

Error protection and concealment must be designed into your transport system due to the fact that the video and audio may travel thousands of miles. The request for a re-try may require a huge buffer to keep up, if it were even possible. Most video transport systems deployed today are only unidirectional; Bi-directional is twice as expensive. Bi-directional also may be impractical in the case of Satellite. Your video may travel across copper connections, fiber optic links, satellite and microwave. Each path has its unique transport challenges. In designing a transport interface consideration must be made to deal with each systems limitations.

All networks that are based on packets will have some form of errors. Understanding those potential errors and designing to overcome them is critical. Some terms are important to understand these include Cell Delay Variation, Cell Loss and cell miss-insertions.

Video is a continuous stream. Each digital video format has its own data each transport system also has its own packet or transport scheme. Whatever transport stream you choose the video must be divided up into cells or packets. Each system has its own issues related to delays, miss-insertions, and cell discards. The cells may need to be reconstructed, repaired, reordered at the destination.

Cell Delay Variation, or the random difference between cell in the same path is described in milliseconds.

Video steams need to be limited to a few nanoseconds. A transport system must over come six orders of magnitude to accomplish its goal. Of course this is a critical design goal.

Cell Discard or Cell Loss, is what happens when a cell is lost along the way to your destination. A high quality network like ATM exhibits a cell loss at about one per 10 million (10^{-7}). That would translate to about a lost cell every 3-5 minutes. As described before we do not have a chance to retransmit that lost cell.

Cell miss-insertion can easily be handled either at the switch or at the end-receiving unit.

Physical Layer issues to consider, fiber optic network based systems like SONET and SDH networks may exhibit sporadic bit errors in the range of one in 10 billion (10^{-10}). Even with these low rates a lost cell would occur every 8 minutes. Depending on adaptation type used, one ATM cell lost is 3000 bits lost using ATM Adaptation Layer 5 (AAL-5), or 380 bits lost (AAL-1, no FEC*) With AAL-1 FEC enabled, nothing is lost. Even with a high quality network running at 10^{-10}, a video feed could suffer an error every eight minutes, at a rate of 10^{-7} the error rate is 2 per second. In television that has been compressed every single-bit error may show. A video compressed using MPEG 4:2:2 Profile Main Level, at 20 Mb/s, has a cell rate of approximately fifty thousand cells per second. DS-3 Circuits can be even noisier.

Below is a comparison of Video transported using AAL5 which has been described by the ATM Forum as the appropriate choice for non broadcast video. The ATM Forum and ITU have described AAL-1 as the choice adaptation layer for professional broadcast video.

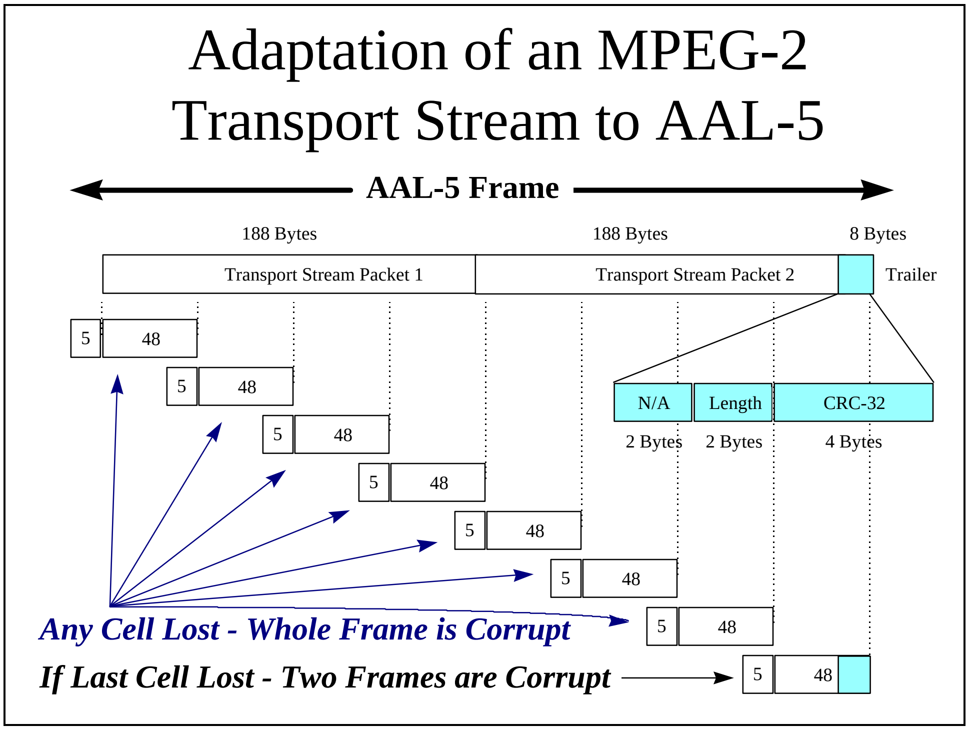
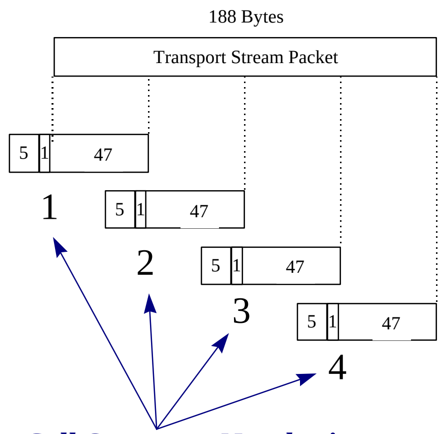


Figure 4
AAL-5 ATM Adaptation Layer

Adaptation of an MPEG-2 Transport Stream to AAL-1



Cell Sequence Numbering

Identifies Lost and Misinserted Cells

The video transport industry organization, Video Services Forum, (you may want to contact them at their web site: http://www.vidtrans.org) has relied on DS-3 circuits which may be based on fiber or copper. These circuits may exhibit more errors.

Due to the standards and cost of bandwidth of our public networks and due to the fact that our video is already compressed lost bits may show in the video. As we discussed earlier error correction and concealment methods are required to assure that video arrives at the destination intact and without losses. Our transport system should not introduce any additional artifacts. Our Digital Recorders have their own artifacts and when we are trying to transport the data/video the best possible techniques must be used.

Several companies are working on the standardization of the pre-compressed video information to eliminate additional cycles of coding and decoding. Transport of SDTI mapped into ATM is under standardization as we speak. This provides an efficient method of getting video across the public network without much loss. All the transport issues and error correction play the same whether you are compressing the video using MPEG or transporting the pre-compressed video.

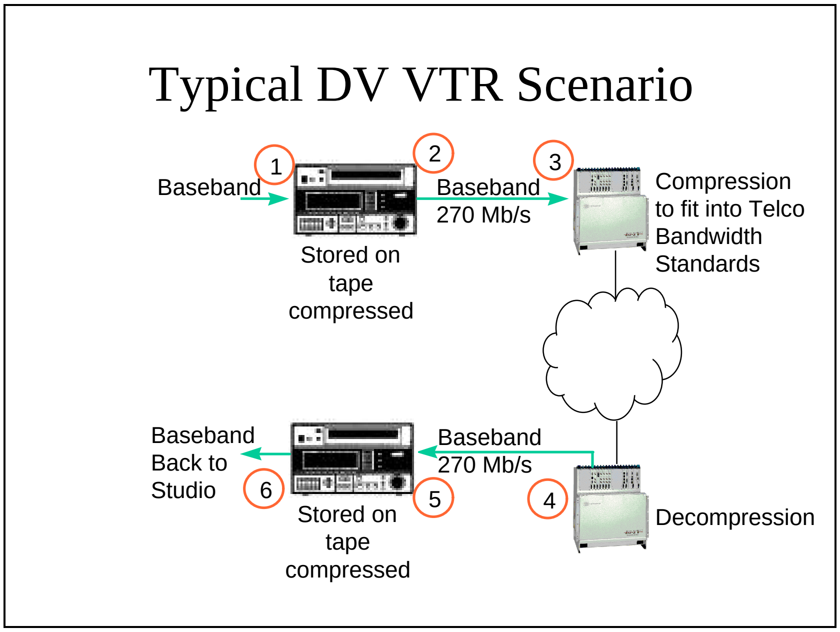


Figure 4
Each red circle represents either an encode or decode cycle

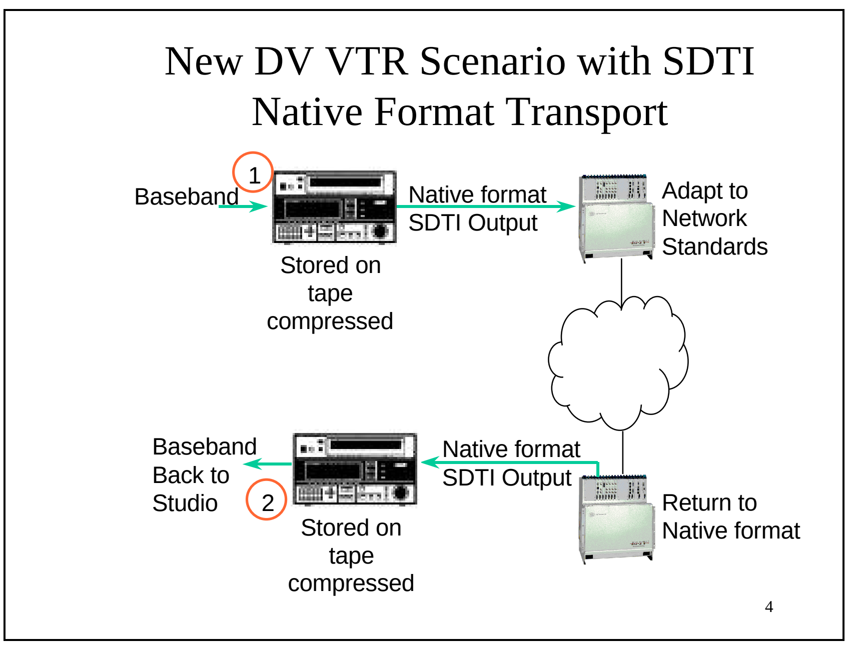


Figure 5
Transporting the SDTI Data eliminates several Cycles

This is a much more efficient method of transporting since the native data format is maintained and no additional compression or data rate reduction is utilized.

Today we can transport video over public and private ATM networks. Real-time Broadcast Video has been tested and utilized using terrestrial fiber, satellites, microwave and hybrid networks of all of the above. Standard definition digital television and high definition TV have been successfully demonstrated, with commercial success. Picture quality ranging from review quality four to ten megabits per second, and full broadcast quality of 50 megabits per second 4:2:2 profile main level MPEG 2.

Transporting video over these networks is not only possible over dedicated private line or private circuits, but can be accomplished using dial up services. Network access devices provide real-time video transport using MPEG encoding to fit our 270 Mb/s studio signals into our Network standard bandwidths.

Multiple live video feeds, High Definition TV and non real-time file transfers have been demonstrated.

Lastly, we need to consider the requirement to manage all of this technology coherently and efficiently. From the beginning, ATM has an attraction because of the ability for a VTR or some other digital-video device to utilize a local ATM network and pass that data across to an ATM-based WAN with full QoS control from source to sink. The ability to enable this kind of control makes full use of ATM's QoS capabilities (Defined in the ATM Forum specification af-tm-0056_000), which can be a complex undertaking for the video engineer. The true requirement is for a management capability by an intermediary application to enable the users to automate three important functions:

- Service Configuration Management (QoS)
- Service Scheduling
- Resource Accounting & Billing

The power of a switched networking technology like ATM is in the ability to dynamically reconfigure the connections without having to involve an outside entity such as the Telco or a specialty provider of video services. On top of this is the inherent capability to ensure a byte-level QoS from output of the source device to the input of the receiving device. The final function has not been applied to the WAN services from carriers to this point, but is imminent in availability – that is billing based on actual usage, moving the costs of WAN networking to the customer and away from the production house or other entity that is the source of the material.

This is a key component in the overall acceptance of the Public Network convergence in the Entertainment industry. Up until now, the company producing the material had to absorb a very high fixed monthly cost for a specialized network connection and then attempt to recoup some portion of that via client chargeback. In situations where there is a very high volume – Advertising Agencies or Broadcast outlets, this is possible, but the more common situation is to go without the WAN service or have the connection installed only for specific, high-value projects due to the high fixed continuing costs. By consolidating all WAN traffic, video, voice, data\Internet, over a single WAN connection, the costs can be spread to the other in-house requirements and a intermediary application can then combine control and billing for services.

The key component then is the combination of the digital technology employed in today's production environment with the ability to control the network Quality of Service. This enables the WAN transport on an ad hoc basis of production content in near-original format and to bill the Client for the incremental use on a project basis that justifies the WAN connectivity in the first place.

Network Services Middleware

- Middle Layer Software that is Tailored for End-to-End Solutions
  - Services Configuration Management
  - Service Scheduling
  - Resource Accounting
  - Quality of Service Guarantees
- Provides “Service Convergence”: Convergence Of Multimedia Data Over a Common Network Infrastructure

Why would a switched service be valuable to our industry? The previous model of dedicated leased lines meant end users had to pay for a leased line whether they used it or not since the older video transport technologies (prior to ATM) did not allow occasional use easily. Network Service Providers (NSP or telcos) could not offer to all customer these high speed connections due to the limited capacity of their dedicated video networks. Now with switched services, it is possible for a NSP's to sign up more subscribers’ share the cost across multiple users. End user pays small monthly subscription fee, and then pays for the calls. With lots of subscribers on the network more subscribers are enticed to join, because it means more companies can communicate with each other. The more users there are the more subscribers… and on it goes. The more users the lower the cost of service can be, because the NSP can recover its cost faster and afford to sign up more subscribers and expand the network.

End users benefit because with lots of companies on the service the easier it is to justify joining the service, because the customers you want to reach are on the service, and so are the vendors who you may want to hire.

What is on the Horizon? As you can as the number of subscribers goes up so does the business opportunity. Small companies can afford to be "on line", fostering creative boutiques. Broadcasters can out source production, editing. Local news casters can sell stories to the networks or any organization who may be interested, due to the fact that they would have two way communications unlike today with only one way satellite services.

The possibilities are exciting…