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Implications of Implementing HDTV Over Digital Subscriber Line Networks

Thomas J. Lantier

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Implications of Implementing HDTV Over Digital Subscriber Line Networks

By

Thomas J. Lantier

Thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Information Technology

Rochester Institute of Technology

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April 26th, 2006
Rochester Institute of Technology

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Abstract

This thesis addresses the different challenges a telecommunications company would face when trying to implement an HDTV video service over a Digital Subscriber Line (DSL) connection. Each challenge is discussed in detail and a technology, protocol, or method is suggested to overcome that particular challenge.

One of the biggest challenges is creating a network architecture that can provide enough bandwidth to support video over a network that was originally designed for voice traffic. The majority of the network connections to a customer premises in a telephony network consists of a copper pair. This type of connection is not optimal for high bandwidth services. This limitation can be overcome using Gigabit Ethernet (GE) over fiber in the core part of the network and VDSL2 in the access part of the network. For the purposes of this document, the core portion of the network is considered to be an area equal to several counties or approximately 50 miles in radius. The core network starts at the primary central office (CO) and spreads out to central offices in suburbs and small towns. The primary central office is a central point in the telecom operator’s network. Large trunks are propagated from the primary central office to smaller central offices making up the core network. The access portion of the network is considered to be an area within a suburb or small town from the central office to a subscriber’s home.

Appendix A, located on page 60, contains a network diagram illustrating the scope of each of the different portions of the network. Considerations must also be given for the internal network to the residence such as category 5 (Cat5) cable or higher grade and network equipment that can provide up to 30 Megabits per second (Mbps) connections or throughput.
The equipment in the telecommunications network also plays a part in meeting the challenge of 30 Mbps bandwidth. GE switches should be used with single mode fiber optic cable in the core part of the network. Digital Subscriber Line Access Multiplexers (DSLAM) with the capability to filter Internet Group Management Protocol (IGMP) messages should be used in the access part of the network to facilitate bandwidth utilization. Placement of this equipment and how the data is aggregated is another issue to consider when implementing HDTV service.

Another major challenge facing the implementation of HDTV over DSL networks is controlling quality of service (QoS) throughout the network. Class of Service (CoS) and Differentiated Services (DiffServ) is a method of QoS that would enable video packets to have a higher priority and less delay than other data packets. The consumer could have data, video, and voice traffic all over the same DSL connection. Data, video and voice packets would need to have a different priority in order to maintain appropriate QoS levels for each service.

The use of advanced technology in video encoding will be essential to the success of the video service. MPEG-2, MPEG-4, and Windows Media 9 are just a few of the video encoding technologies that could be used to reduce the necessary bandwidth for HDTV. The advancement of this technology is essential to allow telecommunications providers to offer HDTV.

Another challenge for the telecom operator concerns the security of the network and service after implementation. Theft of service will be another area that the telecomm operator will be forced to resolve. The cable operators currently face this issue and lose millions of dollars in revenue. Authentication, IP filtering and MAC address blocking are a few possible solutions to this problem.
1 Introduction

Communication providers are constantly trying to break into new markets to increase revenue using their existing network infrastructure. Wireless providers are implementing data and video capabilities on mobile phones; cable providers are using voice over IP (VoIP) to offer voice service using the coaxial network and telecommunication companies are trying to offer video over their existing copper network. “It’s probably an understatement to say that the telco market is getting pretty competitive.” (Bray, 2003) “The traditional local exchange carriers (LECs) are being squeezed both for access to the local loop and also by alternative access providers.” (Bray, 2003) There are many hurdles to overcome to make a network that was originally designed only for voice traffic to handle a bandwidth intensive service like IP video. The bandwidth necessary to handle multiple streams of video and data to one location could be upwards of 20 to 40 Mbps. Utilizing newer technologies such as GE for Wide Area Network transport, superior video-encoding schemes with better data compression, advanced DSL technology, and bonded copper pairs, the telecommunication provider can implement HDTV over Digital Subscriber Line (DSL) networks.

This document will discuss the major challenges involved in implementing HDTV service over a DSL network and suggest methods to use to overcome each challenge. The major challenges that will be covered include the core and access network architecture, equipment for that network, necessary bandwidth to support video services, quality of service requirements, choosing a video encoding scheme, choosing a DSL protocol, and security concerns. A diagram of the overall network architecture is shown in Appendix A. The core network and access
network sections are labeled in this diagram along with the central office equipment and an example of the network in a subscriber’s home.

2 Hypothesis

My assertion is that to implement high definition television (HDTV) over copper and fiber telecommunications networks, the telecom operator should use H.264 MPEG-4 part10 (AVC) video encoding, Gigabit Ethernet as a wide area network (WAN) transport, IP video for video delivery, VDSL2 over bonded copper pairs and Digital Subscriber Line Access Multiplexers (DSLAM) with the capability to filter Internet Group Management Protocol (IGMP) messages. The telecom operator should also use multicast communications for commonly viewed video and use unicast communications for video on demand (VOD) service. Using these advanced encoding technologies and communications techniques the telecom operator will be able to provide a video service over a copper pairs to the home.

It would be much easier for a telecommunications company to implement video services over a total fiber optic network because of the increased bandwidth and low repair rate. However, the expense of running new fiber optic cable to each residence would be prohibitively expensive not to mention upgrading neighborhoods that have underground utilities would be a logistical nightmare. Telecommunications companies have billions of dollars invested in the current copper infrastructure. These companies will want to use their existing copper network to avoid the expense of running new fiber optic cables. They will need to use many new technologies such as GE or Metro Ethernet, MPEG-4 video encoding for transmitting data, and VDSL over bonded copper pairs to achieve the bandwidth necessary. I plan on discussing each of these areas and why the telecommunications carriers should implement HDTV over DSL networks using each of the methods I have referenced.
3 Core Network Design

The network architecture of a DSL network that is going to be carrying HDTV packets must be designed to avoid bottlenecks and packet delay at all costs. A service like HDTV over DSL networks will only be successful if it offers the same or better quality picture and other enhanced features like VOD and an electronic program guide (EPG). To implement these enhanced features, an ultra high-speed core network must be designed and implemented. The only choice for the telecom operators is to use fiber optic cable instead of copper and use it deeper into the network. “Fiber has been driven deeper into the network to remote nodes, allowing customers who previously could not be reached to be serviced.” (Labbe, 2005) For the fastest connections and the biggest distance between central offices or network equipment single mode fiber optic cable should be deployed.

Single-mode fiber gives you a higher transmission rate and up to 50 times more distance than multimode, but it also costs more. Single-mode fiber has a much smaller core than multimode. The small core and single light-wave virtually eliminate any distortion that could result from overlapping light pulses, providing the least signal attenuation and the highest transmission speeds of any fiber cable type. (Fiber Optics)

Most telecomm operators have implemented an asynchronous transport mode (ATM) backbone in their core network. ATM has worked well for voice and Internet traffic because it is a dedicated connection switching technology and it organizes data into 53 byte cells. ATM is also implemented using hardware as opposed to software so faster switching speeds are possible. The standard bit rates for ATM are 155 Mbps or 622 Mbps, but can go up to 10 Gbps using Synchronous Optical Network (SONET) and several other technologies. “ATM’s incumbency in
telco networks is based on its strengths in controlling jitter and delay at low speeds, but in today’s network, which mixes high-rate video, voice, and data delivery, new solutions are required.” (Labbe, 2005)

3.1 Comparing Gigabit Ethernet to ATM

An alternative to using ATM as the transport technology is to use GE. Ethernet can be used as a transport technology and has many advantages over ATM. Some of these advantages are easier management, scalability, lower cost, less inherent overhead and data encapsulation to and from TCP/IP. “GE provides the capacity, simplicity, interoperability with TCP/IP, and feature suite required for successful deployment of bandwidth-hungry IPTV and entertainment services.” (Labbe, 2005) GE and ATM will be compared on ease of management, scalability, cost, data encapsulation, and inherent overhead in the following paragraphs. When Ethernet is used in a typical LAN environment, a distance limitation must be considered between each node of the network. Depending on the type of cable being used in the network and the speed of the Ethernet connection, the distance between each node in the network could be a troubling factor when using half-duplex and CSMA/CD. Full-duplex communications over separate fiber cables should be implemented when using GE as a transport protocol to eliminate the possibility of a packet collision. This method allows for greater distance between two nodes and expands the distance limitation between each node. Using single mode fiber optic cable and specific wavelength lasers, the distance between nodes can be stretched out as far as 62 miles. The table in Figure 1 shows the type of Ethernet connection, the wavelength, the type of fiber cable used and the cable distance.
The ATM transport technology has an inherently high overhead per cell when compared to the amount of data transmitted. For every 53 byte cell transmitted, 5 bytes are used for a header, which translates to a 10.6% overhead. This 10 percent overhead is also known as the “cell tax”. In addition to the “cell tax” there are three other headers or trailers added to the ATM transmission called the ATM Adaptation Layer (AAL).

The ATM adaptation layer adds overhead that supports the quality of service needs of an ATM service category, like CBR or nrt-VBR. AAL5 is the most commonly used AAL type in an ATM implementation. An AAL5 service data unit (SDU) is defined as the layer-three datagram plus the optional Logical Link Control/Subnetwork Access Protocol (LLC/SNAP) header. An AAL5 PDU is defined as the AAL5 SDU plus variable-length padding and the eight-byte AAL5 trailer. (Cisco Systems, 2005)

The LLC/SNAP header and the AAL5 trailer are both 8 bytes and are specified in RFC 1483. The AAL5 PDU is variable length because of the 47 bytes of available padding to make
the full ATM cell of 48 bytes. Figure 2 shows an example of an AAL5 type cell encapsulating an IP packet.

![Figure 2 AAL5 cell in an IP packet](image)

(Cisco Systems, 2005)

ATM overhead can consume as much as 25 percent of a circuits bandwidth. “To calculate this total, consider the three average sizes of Internet packets 1) 64 bytes for control messages, 2) 1500 bytes for file transfers 3) 256 bytes for all other traffic.” (Cisco Systems, 2005) The average of all Internet packets is closer to 250 bytes. Now I will presume that some of the overhead is predictable and some is variable. The table in Figure 3 shows a chart that demonstrates the different types.

![Figure 3 ATM headers and trailer size and type](image)

(Cisco Systems, 2005)
Now I will use the chart above to calculate the percentage of overhead for a 64 byte packet, a 256 byte packet, and a 1500 bytes packet.

AAL5SNAP encapsulation for 64 byte packet:

8+8+16=32 or 50 percent "AAL5" overhead + 10 percent cell tax = >60 percent overall overhead (Cisco Systems, 2005)

AAL5SNAP encapsulation for a 256 byte packet:

8+8+16=32 or 12.5 percent "AAL5" overhead + 10 percent cell tax = >22.5 percent overall overhead (Cisco Systems, 2005)

AAL5SNAP encapsulation for a 1500 byte packet:

8+8+12=28 or 1.9 percent "AAL5" overhead + 10 percent cell tax = >11.9 percent overall overhead (Cisco Systems, 2005)

The average of the overhead for the three packet sizes is 31.5%.

Telecomm operators cannot afford a 31.5% percent overhead charge on their core network to deliver HDTV. GE is a much more efficient transport method and the packets do not have to be translated or encapsulated to the DSL access network.

Figure 4 is an illustration of the difference between the transmission path of an ATM network and the transmission path of an Ethernet network.
The overhead for an Ethernet frame is usually about 10 percent depending on the size of the datagram. The figure of 10 percent overhead is calculated from the following Ethernet packet structure as described in the IEEE 802.3 specification.

**Figure 5 Ethernet Packet Header**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Preamble (7-bytes)</td>
<td></td>
</tr>
<tr>
<td>Start Frame Delimiter (1-byte)</td>
<td></td>
</tr>
<tr>
<td>Dest. MAC Address (6-bytes)</td>
<td></td>
</tr>
<tr>
<td>Source MAC Address (6-bytes)</td>
<td></td>
</tr>
<tr>
<td>Length / Type (2-bytes)</td>
<td></td>
</tr>
<tr>
<td>MAC Client Data (0-n bytes)</td>
<td></td>
</tr>
<tr>
<td>Pad (0-p bytes)</td>
<td></td>
</tr>
<tr>
<td>Frame Check Sequence (4-bytes)</td>
<td></td>
</tr>
</tbody>
</table>

(TechFest, 1999)
In addition to the minimum 26 bytes of overhead, a 96-bit time ‘Interpacket Gap’ is required between each packet as specified in the Ethernet standard. This ‘Interpacket Gap’ cannot be used to transmit data and is considered overhead as well. As an example, if an Ethernet frame is sent with the minimum size datagram of 46 bytes the overall frame size is 84 bytes with 38 bytes of overhead. That datagram calculates out to be 45 percent overhead. This amount of overhead is clearly too much for implementing HDTV over DSL networks and more than the 31.5 percent overhead that ATM produces. This is obviously a worse case scenario and the average size of a frame sent over an Ethernet network is more likely to be about 250 bytes. Now I will use the same size packets that I used to calculate packet overhead in an ATM network.

Ethernet overhead on a 64 byte packet:

\[
\text{26 bytes} + 12 \text{ bytes} = 38 \text{ bytes or 60 percent overhead}
\]

Ethernet overhead on a 256 byte packet:

\[
\text{26 bytes} + 12 \text{ bytes} = 38 \text{ bytes or 15 percent overhead}
\]

Ethernet overhead on a 1500 byte packet:

\[
\text{26 Bytes} + 12 \text{ bytes} = 38 \text{ bytes or 2.5 percent overhead}
\]

The average overhead for the three packets sizes = 25.8 percent

The 25.8 percent overhead is better than the 31.5 percent of ATM, but still not acceptable. When the speed of the Ethernet network is increased to Gigabit or even 10 Gigabit, the overhead ratio can drastically change. When the speed of the network changes so does the time for the 96-bit time ‘Interframe Gap’. “The minimum interframe gap is 96 bit times, which is 9.6 microseconds for 10 Mb/s Ethernet, 960 nanoseconds for 100 Mb/s Ethernet, and 96 nanoseconds for 1 Gb/s Ethernet.” (TechFest, 1999) In order for a GE network to send the required interframe gap, the
size of the gap would be .96 bits. Obviously one bit would not be sent but one byte would have to be sent as the gap. Now using the average of the three packet sizes from above, and 27 bytes of overhead instead of 38, the percentage comes out to be just under 18.1 percent, increasing the connection efficiency to almost 82 percent. The frame size of the datagram for sending HDTV packets over an Ethernet network is more likely to be closer to the maximum of 1518 bytes or even 9000 bytes if jumbo frames are implemented. Switch support for jumbo frames can be a boon to enterprise networks because packet size is more than six times the maximum allowable standard Ethernet frame, reducing overhead processing on the switches. “Ultimately, jumbo frames are ideal for making large-scale file transfer more efficient.” (Network World, 2002) Using a 9000 byte frame, the overhead percentage would be less than one percent. This is a theoretical maximum efficiency and will probably never be achieved in practice but it becomes obvious that GE can be a much more efficient transport technology.

Cost of hardware is another attribute to use when comparing ATM to GE. Ethernet as a technology has been used for over 20 years and in that time the equipment has become a commodity. Even GE cards are relatively cheap compared to other connection cards such as T1 or ATM cards. When a telecomm operator or enterprise is upgrading their connections to a higher bandwidth, the associated hardware costs should be considered. As stated by Cisco systems, “another caveat is associated with the cost in upgrading ATM connections to higher bandwidth levels. These upgrades routinely involve replacing fixed-rate interface hardware and are typically an expensive process.” (Cisco Systems, 2005)

“Ethernet has repeatedly demonstrated cost advantages over every other networking technology, including ATM, SONET, and frame relay. This has been proven repeatedly over the years as Ethernet has consistently been enhanced to deliver new levels of bandwidth at record-
ATM was once thought to be the networking choice of the future and that it would eventually replace Ethernet all the way to the desktop. Disagreements among the ATM standards body members delayed the standard from being completed in a timely manner. This delay gave Ethernet a chance to increase speed from 10Mbps to 100Mbps to 1000 Mbps and improve QoS to compete with ATM. ATM never became a commodity and therefore prices on ATM equipment stayed higher when compared to other networking technologies like Ethernet.

Gigabit Ethernet transport links are typically a fraction of the cost of ATM uplinks of similar capacity. Low complexity directly translates to lower cost and without a doubt, Ethernet offers lower operations, provisioning, management, and maintenance costs than ATM, as well as less expensive education and training, deployment, problem determination and resolution, ongoing operational support, and test equipment. Ethernet chipsets are also less costly. (Labbe, 2005)

Figure 6 shows a comparison of equipment cost between ATM and Ethernet. The ATM column shows a line card for an OC-3 and an OC-12 connection. Each of these cards has only four ports on them. The GE column shows the cost of one 24-port switch and one Gigabit Interface Converter (GBIC). The cost of the OC-3 line card is equal to the entire GE switch with a GBIC. The GE switch could be used at 100 Mbps or any speed up to 1000 Mbps. If the ATM link needed to be upgraded to an OC-12, the line card would have to be replaced at the cost of $25000.

<table>
<thead>
<tr>
<th>ATM Desc</th>
<th>Price</th>
<th>GE Desc</th>
<th>Price</th>
</tr>
</thead>
<tbody>
<tr>
<td>CBX 500 1 Slot OC3/STM4 ATM</td>
<td>$8750</td>
<td>Catalyst 3750 12 SFP Standard Multilayer Image</td>
<td>$7,995</td>
</tr>
<tr>
<td>CBX 500 1 Slot OC12/STM4 ATM</td>
<td>$25000</td>
<td>GE SFP, LC connector SX transceiver</td>
<td>$500</td>
</tr>
</tbody>
</table>
Another disadvantage of using ATM is the complicated manner in which the network devices, virtual circuits, and interfaces are managed. “ATM edge devices are usually more feature robust than other technologies so their configurations are by nature more complicated. As a result, complicated router configurations are typical, increasing the time necessary to tune configurations and manage their performance.” (Cisco Systems, 2005) An ATM connection has to transmit a mixture of data about the ATM system like QoS and traffic information through the network. At each node in the ATM network, resources have to be verified to see if the connection can be accepted. Point-to-point virtual circuits need to be manually configured which also takes more time and complexity. All of these management tasks make the ATM signaling system more complex and harder to manage.

Ethernet is much simpler to manage when compared to ATM. “Switched Ethernet operates on a point-to-point optical circuit over dedicated fibers or over shared fiber with a wave-division multiplexed (WDM) overlay.” (World Wide Packets) Simply plugging the RJ-45 connector into the Ethernet switch can set up an Ethernet connection and the spanning-tree-protocol takes over the configuration. Other complex configuration parameters might have to take place at a router for configuring the speed of the connection or the QoS parameters but it is still less complicated than an ATM connection.

IP broadcast and multicast will be two critical communication methods used to implement HDTV over DSL networks. Ethernet has the ability to use both broadcast and multicast while ATM does not. Ethernet networks are connection-less as opposed to ATM which is connection oriented. This requires that a connection be established before any data can be sent. This poses a problem for ATM when trying to use broadcast or multicast
communications. In order for ATM to provide broadcast or multicast service, a point-to-point virtual channel connection would have to set up to each of the receiving nodes. Using this method, the ATM network would have to send the same message to each node one at a time to replicate the same functionality that Ethernet can perform using broadcast or multicast. The attempt to make ATM like a LAN was named LANE (Local Access Network Emulation) and it was not favorably received with network administrators.

LANE's complexity (for example, creating a broadcast environment out of a virtual circuit technology) didn't appeal to network administrators accustomed to Ethernet's simplicity. And ATM's Multi-Protocol over ATM routing (which builds on LANE) never appealed to administrators experienced with traditional routers. (Passmore, 1999)

Multicast routing is an essential part of implementing HDTV services and will be covered in more detail later in this paper.

Ethernet also has an advantage over ATM when it comes to scalability. The bit rates that ATM is capable of operating at are speeds of 155 Mbps and 622 Mbps as well as 2.5 Gbps and 10 Gbps in the backbone. These pre-specified bit rates come from the optical carrier (OC) connections that make up an ATM network. Below is a small table representing the different bit rates and the corresponding OC level.
Figure 7 OC levels and corresponding bit rates

<table>
<thead>
<tr>
<th>OC Level</th>
<th>Bit rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>STS-3/OC-3</td>
<td>155.520 Mbps</td>
</tr>
<tr>
<td>STS-12/OC-12</td>
<td>622.080 Mbps</td>
</tr>
<tr>
<td>STS-48/OC-48</td>
<td>2.488 Gbps</td>
</tr>
<tr>
<td>STS-192/OC192</td>
<td>9.952 Gbps</td>
</tr>
</tbody>
</table>

These are the pre-specified speeds that ATM can operate at over optical connections. ATM links cannot be customized to different speeds even if the bandwidth is not necessary. This causes wasted bandwidth throughout the network. Ethernet links can be throttled to the speed that is necessary down to one Mbps at a time if needed. Control can be customized at the switch port so that bandwidth can be used more efficiently. Using wave division multiplexing, 10 GE is possible and has become an approved standard as a viable protocol for use in the core network. This proves that ATM does not have a speed advantage over Ethernet.

Ethernet has many advantages over ATM including easier management, scalability, cost, less inherent overhead, and data translation to and from TCP/IP.

Ethernet WANs have several advantages. They can plug into enterprise-class Gigabit Ethernet switches or routers, eliminating the need for channel service units and other equipment required for traditional ATM/Sonet services. A native Ethernet design is easier to support, the equipment costs less than that of an ATM/Sonet network, bandwidth can be provisioned more quickly and efficiently (in increments as small as 1M bit/sec.), and the networks can scale to 10G bit/sec. But the biggest benefit is cost
savings. "The Ethernet solution tends to be 50% cheaper than Sonet," says analyst Jay
Pultz at Stamford, Conn.-based Gartner Inc. (Mitchell, 2003)

Many telecomm operators and enterprises have realized the benefits of using Ethernet in their
transport networks and for more reasons than just cost. “Telcos now recognize that a new-
generation access network with Gigabit Ethernet transport is an extremely viable architecture for
delivering high-bandwidth IPTV services – and for more reasons than simply beating ATM on
cost.” (Labbe, 2005)

3.2 Core Network Equipment

The typical DSL core network equipment consists of a Broadband-Remote Access Server
(B-RAS) along with the previously mentioned ATM equipment. I have already asserted why the
ATM links and equipment should be replaced by GE links and equipment, but IP video routers
should also replace the B-RAS. A B-RAS currently has many functions in a DSL network such
as subscriber authentication and billing, IP address assignment for the subscribers DSL modem,
dynamic binding to virtual routing domains, and service advertisement. According to Laurel
Networks the load on the typical B-RAS makes it one of the most unreliable devices in the DSL
network. The telecom operator will not be able to tolerate outages that affect the HDTV service
offering like some data outages are tolerated today. The general public has become accustomed
to consistent and reliable television service and if the HDTV over DSL service is faulty, the
subscribers will not accept such error prone service. The B-RAS also tends to support lower rate
interfaces like ATM over an OC3 and does not support multiple GE ports necessary to
implement highly reliable HDTV service.
The solution is to take the B-RAS out of the network and replace its functionality with a device built to be more reliable and highly available. The main replacement would be an IP video router. An IP video router is specifically designed to transmit video packets over an IP network. The IP video router is built for high reliability, wire speed throughput and is generally less expensive than the B-RAS. The functionality that the IP video router would provide is only IP address assignment. The rest of the B-RAS functionality such as dynamic discovery of set top boxes (STB’s), provisioning of the HDTV service, authentication, and network configuration of the STB should be moved to the DSLAM.
4 Access Network Design

The overall design of a telecom operator’s network will change to incorporate the HDTV service. Where it is economically feasible, fiber optic cable will replace copper cable all the way to the customer premises. The areas that are not deemed economically feasible for fiber to the home will have fiber run to a node in the vicinity and copper will still be used from that node. This area from the CO to the home is considered to be the access network. DSL technology will be used to send and receive data in this portion of the network. Several DSL standards have been developed and each standard has different bandwidth capabilities and methods for transmitting data. Currently the two DSL standards with the highest capable bandwidth are named very high bit rate DSL (VDSL) and asynchronous DSL (ADSL). Each of these standards has distinct speed versus distance limitations, different ways of handling error correction and different methods for downstream data multiplexing. The following sections will compare the two types of DSL standards and recommend the best one to use when implementing HDTV.

4.0.1 Asynchronous Digital Subscriber Line

According to the DSL forum website ADSL is the most commonly implemented type of DSL in the world. The word ‘asynchronous’ in ADSL describes the division of bandwidth between downstream to the user and upstream to the node or DSL equipment. ADSL is faster in the downstream direction of the connection than in the upstream direction. “An ADSL circuit connects an ADSL modem on each end of a twisted-pair telephone line, creating three information channels -- a high speed downstream channel, a medium speed duplex channel, depending on the implementation of the ADSL architecture, and a POTS (Plain Old Telephone Service) or an ISDN channel.” (DSL Forum) Basic ADSL downstream speeds range from 300...
Kbps to 9 Mbps and upstream speeds range from 16 Kbps to 630 Kbps. Later versions of ADSL named ADSL2 and ADSL2+ have speeds that can reach 24 Mbps downstream and 2 Mbps upstream. These numbers go down with increasing distance from the CO or remote node because of signal attenuation, but it is still a huge improvement over basic ADSL.

Transmitting digital compressed video on a DSL connection creates a challenge for implementing an error detection and correction method. Digital compressed video is a real time signal so the data packets must be sent without error before the viewing device receives the packets. Therefore, normal link layer or network layer error control measures that are usually found in data communication protocols cannot be used. ADSL employs a method called Reed-Solomon forward error correction (FEC) that greatly reduces the errors caused by impulse noise. The Reed-Solomon method uses a mathematical formula to correct data errors at the receiving end of the transmission path instead of requesting a resend of the corrupt data. To correct the data at the receiving end, the Reed-Solomon encoder manipulates the data packet into a parity packet and appends the parity packet to the end of the data packet. The parity packet is then compared to the data packet to detect errors. The entire data and parity packet is called the Reed-Solomon code word. This error correction method can correct half the amount of bytes of the total number of bytes in the parity packet. An example of this method is shown below.

The encoder takes \( k \) data symbols of \( s \) bits each and adds parity symbols to make an \( n \) symbol codeword. There are \( n-k \) parity symbols of \( s \) bits each. A Reed-Solomon decoder can correct up to \( t \) symbols that contain errors in a codeword, where \( 2t = n-k \).

The following diagram shows a typical Reed-Solomon codeword (this is known as a Systematic code because the data is left unchanged and the parity symbols are appended):
Example: A popular Reed-Solomon code is RS(255,223) with 8-bit symbols. Each codeword contains 255 code word bytes, of which 223 bytes are data and 32 bytes are parity. For this code:

\[ n = 255, \quad k = 223, \quad s = 8 \]

\[ 2t = 32, \quad t = 16 \]

The decoder can correct any 16-symbol errors in the code word: i.e. errors in up to 16 bytes anywhere in the codeword can be automatically corrected.

(4i2i, 2004)

Using this method of error correction, ADSL is able to transmit digital compressed video and maintain a quality picture.

To be able to send data upstream and downstream, ADSL must divide the frequency of the copper line into multiple channels. To create multiple channels, ADSL modems divide the available bandwidth of a telephone line in one of two ways -- Frequency Division Multiplexing (FDM) or Echo Cancellation. (DSL Forum) FDM uses one band to send upstream data and one band to send downstream data. Time division multiplexing is used to divide the downstream path into multiple high-speed channels and multiple low speed channels. The upstream path also uses multiplexing to create parallel low speed channels. Echo Cancellation implements data transmission by assigning the upstream band to overlap the downstream band and then using local echo cancellation. Both FDM and Echo Cancellation techniques separate off a 4 kHz section for analog telephone service and ISDN in the DC area of the spectrum. Figure 8 shows a chart of the two kinds of frequency division used in ADSL.
The ADSL modem categorizes the entire data stream created by multiplexing downstream channels, duplex channels, and maintenance channels together into blocks, and attaches an error correction code to each block. On the receiving end of the transmission, FEC is used to correct errors that occur during transmission up to the limits of the Reed-Solomon code and the length of the block. “The unit may, at the users option, also create superblocks by interleaving data within subblocks; this allows the receiver to correct any combination of errors within a specific span of bits.” (DSL Forum) Using the superblock method assists in better error correction and enables the transmission of data and video.

ADSL is capable of using two types of data modulation for encoding the electrical signals over a local-loop. The first type of data modulation is named Carrierless Amplitude and Phase (CAP). CAP is a well-known and relatively well-understood modulation technology because of its similarity to Quadrature amplitude phase modulation (QAM). “Although CAP is well-understood and relatively inexpensive, some argue that it is difficult to scale because it is a single-carrier modulation technique and is susceptible to narrowband interference.” (Cisco Systems, 2005)

The second type of data modulation is named Discrete Multitone (DMT). DMT uses multiple carriers and is capable of more speed than CAP. It is for this reason that the American
National Standards Institute (ANSI) committee approved DMT as a standard for DSL deployment.

This standard (DMT) calls for 256 subbands of 4 KHz each, thereby occupying 1.024 GHz. Each subband can be modulated with QAM 64 for clean subbands, down to QPSK. If each of the subbands can support QAM-64 modulation, then the forward channel supports 6.1 Mbps. On the return path are 32 subbands, with a potential for 1.5 Mbps. (Cisco Systems, 2005)

There are advantages and disadvantages for both CAP and DMT. CAP is a single-carrier technique that uses one wide band, which can make it difficult to scale and susceptible to narrow-band interference. Whereas DMT uses multiple carrier types and many narrow band channels resulting in greater speeds to the users modem. However, the high number of multiple bands increases the amount of power demanded compared to CAP. The 256 channels that DMT uses to modulate data consume 5 watts per channel. If hundreds or thousands of transceivers are in one central office, the heat dissipation requirements greatly increase thereby increasing costs. Another disadvantage of CAP modulation is the use of using one wide band because it creates a larger dynamic range when transmitting the signal. Adaptive equalization must be used to compensate for attenuation and phase error, which requires that the modems learn line characteristics and do so by sending probes and looking at the return signals. “The equalizer then knows how it must amplify signals to get a nice, flat frequency response.” (Cisco Systems, 2005) DMT does not need to use adaptive equalization because noise characteristics do not fluctuate over the 4-kHz subband.

After comparing the two modulation types most ADSL and VDSL deployments use DMT. “DMT is the most widely deployed DSL line code, with more than 36 million ADSL
lines deployed worldwide at the end of 2002, according to a study by Point Topic.” (STMicroelectronics, 2003) By deploying DMT for line coding, telecom operators have chosen higher speeds than increased costs.

Although ADSL is the most commonly used type of DSL around the world, it does not meet the bandwidth requirements for sending voice, video and data over a copper loop. The VDSL standard was created to leverage the ADSL design and increase the possible throughput over a copper loop so voice, video and data could be sent reliably over the same twisted pair.

### 4.0.2 Very High Bit Rate Digital Subscriber Line (VDSL)

The International Telecommunications Union (ITU) approved the standard for VDSL, or the latest version VDSL2, in 2005. This standard is based on work done by the T1.E1.4 working group and the IEEE 802.3ah Ethernet in the First Mile (EFM) task force. The two standards bodies chose DMT instead of CAP or QAM for line coding when deploying VDSL2. VDSL2 is very similar to ADSL because it uses advanced transmission techniques as well as FEC and DMT for line coding.

ADSL employs advanced transmission techniques and forward error correction to realize data rates from 1.5 to 9 Mbps over twisted-pair ranging to 18,000 feet; VDSL employs the same advanced transmission techniques and forward error correction to realize data rates from 13 to 55 Mbps over twisted pair ranging to 4500 feet. (VDSL Forum, 2005)

VDSL is able to realize data rates that are nearly ten times greater than ADSL because of the shorter distances from the CO to the user but also because ADSL has to deal with bigger dynamic ranges. The diagram below shows the difference for distance and speed between ADSL2+, VDSL1, and VDSL2.

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Figure 9 VDSL2 performance graph

VDSL2 Performance

(DSL Forum, 2005)

Figure 9 shows that VDSL2 can reach up to 50 Mbps at 3300 feet from the CO, dropping to 26 Mbps at 4900 feet from the CO. This is the bandwidth needed to implement HDTV over DSL networks, especially with more than one television set in the premises watching different programming and other services transmitting and receiving at the same time, such as voice and data. Figure 9 also shows VDSL1 with a much shorter reach and ADSL2+ with a lower possible bandwidth when compared to VDSL2.

VDSL2 is able to transmit at the higher bit rate and for longer distances because it uses a higher Mhz band, trellis coding, Generic Convolutional Interleaving, and echo-cancellation. The higher 30 Mhz band allows the signal to travel faster than the 12Mhz VDSL1 signal or the 2.2 MHz ADSL2+ frequency. The higher the frequency used the faster the signal can travel but the
signal will attenuate sooner. Trellis coding is very similar to Quadrature Amplitude Modulation (QAM) except it has an extra redundant bit added. QAM divides the bit stream into four-bit sections but Trellis adds a fifth or quint-bit to the transmission. The value of the quint-bit is calculated from the value of the data bits. The problem with a QAM system is the four bits used to check the validity of the received data can be altered by distortion during transmission. If one of those bits is closer to an adjacent point instead of the anticipated point after transmission, an error can occur and misidentification of a bit or bits. “By adding a redundant bit to each quadbit, trellis-coded modulation increases the amount of information used to identify each bit pattern and thereby reduces the number of possible matches.” (Forouzan, 2001) A Trellis-encoded signal is less susceptible to distortion because of line noise. This allows the VDSL2 signal to travel further and tolerate more line noise. The Generic Convolutional Interleaving uses a technique called interleaving that is used to overcome a situation when a burst of noise corrupts more bits in one block of data than can be corrected by the error correction algorithm. The combination of interleaving and de-interleaving overcomes this problem by spreading the error bursts. “After de-interleaving, the errors due to an impulse will be distributed to a small number of bytes of several RS blocks instead of many consecutive bytes of a small number of blocks.” (Toumpakaris, 2003) Interleaving also protects VDSL2 transmissions from noise allowing the signal to travel further without distortion. Echo cancellation was described earlier and enables the VDSL signal to travel further because any echo of the data signal would be cancelled out. Without the use of the echo cancellation, the VDSL signal could be compromised causing interference in the HDTV picture.

After comparing possible transmission speed, distance from the CO, and error correction techniques, VDSL2 is the best choice for the access portion of the network. VDSL2 will enable
the telecom operator to deliver HDTV service to the user with enough bandwidth for multiple television sets viewing at the same time along with voice and data transmissions.

### 4.0.3 Bonded Copper Pairs

Another technique the telecom operator should use in the access portion of the network is called **DSL bonding**. “DSL ‘Bonding’, is the process of splitting a single bit stream into multiple bit streams for transport over multiple DSL lines, and then reassembling it into the original bit stream at the receiver.” (Telcordia Technologies, 2005) A bonding compliant hardware device or software is required to be installed at both ends of the connection to accept the second pair of lines and to reassemble the bit stream.

The increase in competition for voice services in the residential market and the decrease in use of a second dial up line has left an abundance of unused copper pairs in the telecom network. The telecom operator could make use of those extra pairs by bonding them together to get higher data rates in conjunction with VDSL2, while still being 4900 ft from the customer home. There is an obvious need for more bandwidth to enable voice, video and data but not all customers have access to fiber. “That leaves some basic alternatives to drive more bandwidth: use advanced forms of very-high-data-rate DSL (VDSL) and VDSL2; bond existing copper pairs; or, in the most outward-looking scenario, bond copper pairs running advanced VDSL.” (Barthold, 2005)

The common protocol used for bonding copper pairs is called **PPP Multilink Protocol (MP)**. MP is an extended version of Point-to-Point Protocol (PPP) and has the ability to bond two or more concurrent parallel connections. The result of the combined links is
bandwidth that is close to the sum of the original two connections. MP is a non-proprietary TCP/IP standard defined in RFC 1990. Some overhead is added to the data stream when MP divides the packets into fragments. “MP transmits each individual packet or fragment along the first available link, along with extra information to enable the receiving end to recombine the fragments into a single packet for onward routing.” (Vicomsoft, 2003) Figure 10 shows an example of a PPP data stream and the additional data to recombine them at the receiving end.

Figure 10 PPP datastream

PPP Packets contain information used to recombine and sequence them.

(Vicomsoft, 2003)

MP has the benefit that a single TCP/IP connection can take advantage of the increased bandwidth. An FTP download could complete twice as fast and the client or server would not be aware that the MP connection is in the middle of the transmission. Likewise, an HDTV connection could also take advantage of an MP connection without ever knowing it exists in the middle of the network.

Referring to Figure 9, the VDSL2 data rate at 4000 feet could be doubled to greater than 50 Mbps using DSL bonding or reach greater distances from the node and maintain the 25 Mbps rate.
Bonding may cost-effectively allow Local Exchange Carriers (LECs) to achieve aggregate line rates necessary to deliver multi-channel IP video service to a widely base of customers over copper by selectively bonding two pairs of ADSL2+ or VDSL on the longer lines in a serving area. (Telcordia Technologies, 2005)

4.1 Access network Equipment

4.1.1 Digital Subscriber Line Access Multiplexers (DSLAM) and IP Multicast

A Digital Subscriber Line Access Multiplexer (DSLAM) is a network device that transmits and receives data to and from a DSL modem located at the customer premises. In the transmitting process the DSLAM multiplexes the data into a single stream and then transmits it over the core network connection. In the receiving process, the DSLAM de-multiplexes the data streams and propagates the data to each individual DSL connection. The DSLAM is the piece of equipment that separates the voice and data network and modulates the DSL signal to the higher Megahertz range without interfering with the lower Megahertz range voice traffic. The abilities and the location of the DSLAM will have to change to accommodate the needs of providing HDTV service over DSL networks.

The new HDTV DSL network architecture needs to include moving the DSLAM closer to the customer premises to shorten the local loop and enable VDSL2 to have the shortened distance to provide higher bit rates. At this time most of the DSLAM’s have been located in the central office, which has created local loops of 18,000 feet or less. The DSLAM will have to be moved outside of the central office to remote pads and cabinets. Even with compression, delivering video services to the home requires an order of magnitude more bandwidth. “Since
DSL rates increase only as the DSL loop length decreases, DSLAM’S must be placed closer to residential subscribers to service their video bandwidth needs.” (Laurel Networks, 2005) This will enable the distance from the customer premises to the DSLAM to be within acceptable limits for VDSL2 to provide enough bandwidth for HDTV service.

One of the advantages of sending video over an IP network is the ability to multicast only the subscribed channels to each of the customers (STB’s). Instead of broadcasting all channels like the current cable system does, an IP video system can save bandwidth by sending only the necessary channels and sending one copy of each data stream in the core portion of the network. By pushing the IP multicast functionality out to the remote DSLAM’s and closer to the customer premises, the core network load is reduced. In the current DSL network architecture the DSLAM does not support the multicast communication method. The DSLAM would need to incorporate IP multicast as its communication with the DSL modem.

The first commercial deployments of video over DSL have relied on replication from deep in the network at the IP multicast router level or via ATM point-to-multipoint PVC’s (Permanent Virtual Circuit) in the DSLAM’s. However, most of today’s ATM-based DSLAM’s are not able to support multicast or the necessary protocols, such as IGMP (Internet Group Management Protocol) snooping, to make multicast delivery practical. (Laurel Networks, 2005)

In essence, the ATM upstream equipment would have to send multiple copies of the same data stream. This situation becomes a typical scalability problem and the ATM DSLAM would deplete its core network bandwidth, most likely an OC3 or OC12, long before it ran out of DSL ports. This is a common problem among first generation DSLAM’s because the core network
technology that is most prevalent in today’s telecom networks is ATM, and therefore the equipment had to interface with the ATM network as well.

The use of IP multicast is an important method of transmitting the HDTV packets and saving bandwidth through the core and access portions of the network. A single copy of the HDTV stream can be transmitted through the core network to the aggregation point, most likely an enterprise size Ethernet switch. Then the Ethernet switch copies the packets to the DSLAM and the DSLAM can use IGMP to multicast the stream only to the STB’s that are requesting that channel. This saves a great amount of bandwidth over broadcasting to every STB even if the STB isn’t requesting the channel. Figure 11 shows an example of the multicast transmission of a HDTV stream. Notice that only one video stream is sent in the core network and then only three of the four STB’s receive the video stream.

Figure 11 Multicast Transmission

![Multicast Transmission Diagram](image-url)
The telecom operator must use DSLAM’s that incorporate the use of IP multicast or face bandwidth issues in the core and access portions of the network. The use of ATM DSLAM’s would be a critical error for the telecom operator.

4.1.2 Video on Demand Servers and IP Unicast

The HDTV service will offer a VOD service to its subscribers to view content whenever the subscriber would like. This service would use IP Unicast to transmit video from the video server to one subscriber’s STB only. This could have major impact on the network if all subscribers decided to watch VOD content at the same time. That is why the VOD servers should be placed in the CO’s so the unicast transmission only affects the access portion of the network and most likely just one DSLAM.

Lets assume that the core portion of the network supports 10 DSLAM’s. Each DSLAM has 256 ports that can support two STB’s for a total of 512 possible HD streams. If all of those STB’s were used at the same time to watch HDTV consuming 6 Mbps channel, the total necessary bandwidth would be over 30 Gigabits. By moving the VOD servers to the CO the impact on the core portion of the network is reduced.

4.2 Internet Group Management Protocol (IGMP)

The key element to using IP multicast in the access network is the use of Internet Group Management Protocol (IGMP). IGMP is based on the Internet Engineering Task Force (IETF) standard RFC 3376. IGMP is considered a Network Layer protocol (Layer 3) within the OSI model and provides a way for a device on a network to request its desire to receive video traffic

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and to periodically have the router poll STB’s for current status. IGMP plays an important part in the implementation of HDTV over DSL networks because it allows each STB to send a report to the DSLAM to receive a certain video stream or channel. “The set top box sends a join (also referred to as an IGMP report), which permits the IP multicast switch to begin forwarding the video traffic to the set top box.” (Bray, 2003)

The status of a STB is called a state and the host router polls the STB for the current state. If a STB is equipped with more than one decoder it can have more than one state or act like multiple hosts. The table below defines the host states. A STB can be in one of three possible states in regards to any single IP multicast group (video stream) on any single network interface.

<table>
<thead>
<tr>
<th>State</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Non-Member</td>
<td>The STB does not belong to the group on the interface. Initial state for all memberships on all network interfaces.</td>
</tr>
<tr>
<td>Delaying Member</td>
<td>STB belongs to the group on the interface and has a report delay timer running for that membership.</td>
</tr>
<tr>
<td>Idle Member</td>
<td>Host belongs to the group on the interface and does not have a report delay timer running for that membership.</td>
</tr>
</tbody>
</table>

(Deering, 1989)

An IGMP state can be changed by six different significant events and these events are shown in Figure 13. These appear as IGMP messages as defined in RFC 1112.
Figure 13 IGMP significant events

<table>
<thead>
<tr>
<th>Event</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join Group</td>
<td>Happens when a STB decides to join the group, or video stream, on the</td>
</tr>
<tr>
<td></td>
<td>interface. May only occur if the STB is in the Non-Member state.</td>
</tr>
<tr>
<td>Leave Group</td>
<td>Occurs when the STB decides to leave the group on the interface. Only</td>
</tr>
<tr>
<td></td>
<td>happens when in the delaying member state and Idle member states.</td>
</tr>
<tr>
<td>Query Receive</td>
<td>Occurs when the STB receives either a valid General Membership Query</td>
</tr>
<tr>
<td></td>
<td>message or a valid Group-Specific Membership Query message.</td>
</tr>
<tr>
<td>Report Received</td>
<td>Occurs when the host receives a valid IGMP Membership Report message.</td>
</tr>
<tr>
<td>Report Received, continued</td>
<td>A Membership report only applies to the membership in the group identified</td>
</tr>
<tr>
<td></td>
<td>by the Membership report, on the interface from which it is received. It is</td>
</tr>
<tr>
<td></td>
<td>ignored for memberships in the non-member or idle states.</td>
</tr>
<tr>
<td>Timer Expired</td>
<td>Occurs when the report delay timer for the group on the interface expires.</td>
</tr>
<tr>
<td></td>
<td>Only occurs in the delaying member state.</td>
</tr>
</tbody>
</table>

(Deering, 1989)

Taking into account the three states and the significant 6 events that can change those states, the following diagram reflects a quick summary of the different message exchanges. In figure 14, each state transition is labeled and with the event that causes the transition, and in parentheses, any actions taken in transition.
(Deering, 1989)

The all-hosts group (address 224.0.0.1) is handled as a special case. The host starts in Idle Member state for that group on every interface, never transitions to another state, and never sends a report for that group.

A basic ‘channel change’ would consist of an IGMP ‘Leave’ immediately followed by an IGMP ‘Join’. The ‘channel change’ operation should be considered a priority for network traffic. Any delay in the IGMP ‘Leave’ or ‘Join’ events would slow the ‘channel change’ and make the user experience painful and frustrating. An important function of the DSLAM for IGMP will be to turn off the multicast streams instantly when a user changes the channel. Otherwise, the aggregation switch will be sending more than one stream of data to a single VDSL line, and flood the line with unnecessary packets from a channel that has been switched from already. This is also an important factor for keeping the channel changing function quick and user acceptable.
The IGMP protocol can be considered a timed polling protocol because the DSLAM must poll the STB about every 30 seconds and the STB sends an IGMP report in response. Upon receiving the report the multicast switch continues to send the data stream. The DSLAM is required to monitor communications between the multicast layer 3 switch and the STB. This process is called IGMP snooping and it is also used to direct which video streams are transmitted to a particular subscribers STB. The vital scaling issue associated with IGMP snooping is that all messages must pass through the DSLAM and be processed by the layer 3 aggregation switch. The layer 3 switch can become a bottleneck in the network trying to respond to all the IGMP reports sent from the STB.

A typical DSLAM has 256 ports and each port supports two STB’s. A single upstream link from the DSLAM to the aggregation switch can produce up to 512 IGMP report messages every 30 seconds. This scenario can be taken a step further and assume the aggregation switch supports up to ten DSLAM’S and receives 5,120 IGMP report messages every 30 seconds. Filters at the DSLAM should deal with this enormous message response from the STB’s. “The DSLAM has to control, filter, and interrogate the IGMP report messages – aligning them in a logical order and removing redundant packet information – and then transmit a condensed IGMP report list to the Layer 3 aggregation switch.” (Bray, 2003) To the aggregation switch, there appears to be only one STB per requested video stream. This type of advanced IGMP snooping allows for greater scalability over the access or aggregation portion of the network. DSLAM’s with this advanced type of IGMP snooping have now relieved the constraint from the aggregation switch.

IGMP is an important protocol to use in the access portion of the telecom network when trying to deploy HDTV over copper lines. The use of IGMP and multicasting will enable the
telecom operator to save crucial bandwidth in the core and access portions of the network and allow the aggregation switch to handle multiple streams from multiple DSLAM’S. The use of the IP DSLAM instead of an ATM DSLAM is also a critical piece to implementing HDTV over the copper DSL network. The IGMP snooping function in the DSLAM is a key element to reducing the overhead on the aggregation switch from 1000’s of STB’s.

4.3 Home Media Gateways

The final part of the network to deliver HDTV over DSL networks is the local area network in the customers home. If the HDTV packets are transmitted through the core and access portions of the network at high bandwidth speeds and without error, then the LAN at the customers home must be sufficient to carrier those packets as well. Otherwise the quality of the HDTV service will suffer and all previous steps taken to deliver quality data are moot.

The cabling in the customer premises must be at least CAT5e twisted pair. CAT5e cable is capable of transmitting data at speeds of up to 100 Mbps and should be sufficient to carry HDTV packets and any other packets around the customer premises. The other choice for residential cabling would be fiber optic cable. Multimode fiber optic cable is capable of transmitting at speeds over 100 Mbps and is less expensive than singlemode fiber optic cable. If a new home is being built then fiber optic cable should be implemented to prepare for future services and needs of the customer.

There are currently two implemented architectures to propagate data around the customer premises and both involve different equipment. There are other proposed methods for data
transmission of voice, video and data services like a wireless solution, but the specifications and equipment are not yet available.

The first type of architecture would be considered a decentralized method where each television in the premises has its own STB. The HDTV signal would be transmitted to a network interface device (NID) and then a separate cable would be run to each of the television sets and personal computers within the customer premises. A telephone splitter would be used at the NID for traditional analog telephone service. This method is more like a mesh network and is very wasteful when considering all the extra cabling that would have to be installed throughout the premises. An example of this type of implementation is shown on the left side of figure 15.

The other type of architecture would be considered more of a centralized method of distributing the data around the premises with one STB, or in this case a home media gateway. Again, the HDTV signal would be transmitted to the NID and then a single VDSL2 line would go to the home media gateway. A pots splitter would still be used for analog telephone service. The home media gateway could have multiple decoders to provide service for more than one television set. Coaxial ports could be used to make use of an existing coaxial network within the premises. The home media gateway would also have an RJ-45 port for data delivery to personal computers or other network equipment like a wireless router. If a wireless router was not used an additional Cat 5e cable would have to be run from the home media gateway to the PC or router. At this time wireless transmission is not recommended for propagation of the HDTV signals to other television sets in the home. “HD video can't be transferred through the bundled 802.11b or 802.11g wireless network connections, as the bandwidth required is too high.” (Shah, 2005) Plans for the 802.11n wireless protocol, which is expected to be capable of speeds up to
10 times that of the 802.11g, should be adequate for use with HDTV signals. This centralized type of implementation is shown on the right side of figure 15.

Another possible means of transmitting data packets to personal computers around the home is over the existing power lines. In this centralized type of solution the media gateway/server uses the homes existing coaxial network to distribute the HD packets to televisions and the existing power lines to distribute data. The HomePlug Powerline Alliance has standardized the Powerline Communications in the HomePlug 1.0 specification. “The HomePlug 1.0 specification, with a PHY rate up to 14 Mbps, was approved in 2001 and has since been implemented by a number of equipment manufacturers offering an array of compatible home networking products.” (Intellon, 2005) The next standard for the HomePlug connection was called HomePlug AV and was approved in 2004. The standard specifies a 200 Mbps bit rate and is intended to offer enough bandwidth for digital music, multiple HDTV streams, Internet traffic and other data needs. Security and quality of service were considered during the drafting of the HomePlug AV standard.

Security is built-in and turned on to support privacy and Digital Rights Management concerns, and most importantly, only HomePlug AV technology has designed-in Quality-of-Service that ensures a consumer receives a great experience no matter what they're using HomePlug AV technology for. (Reeber, 2004)
As shown in the previous paragraphs there are many choices for distributing HDTV packets throughout the home. Choosing one distribution method over another could depend on the existing network in the home. Overall I would suggest using a centralized method for data distribution with Cat5e cable for data and the existing coaxial cable for HDTV signals, if it already exists. For future implementations a centralized architecture is still suggested plus using a STB or media gateway with an 802.11n wireless router incorporated. The wireless router makes it easier to move television sets or computers to different areas of the house without adding more Cat5e or coaxial cable runs. The 802.11n standard should provide enough bandwidth for all data uses including multiple streams of HDTV. Additional STB’s would be needed at each television set to receive the 802.11n signal. However, the need for a wireless
signal with a laptop computer already exists and the laptop could then be a portable TV receiving HDTV signals anywhere inside or outside the home.
5 IP Video vs. Radio Frequency Overlay

The telecom operator can choose between two mechanisms when delivering video services, IP video and Radio Frequency (RF) Overlay. Each mechanism has its advantages and disadvantages when transmitting through a telecom network.

RF overlay generates a network capable of providing analog broadcast, digital and HD broadcast, and VOD services. The first advantage of using RF overlay is the economies of scale that are gained because the cable industry uses RF overlay so the equipment, including the RF headend, are readily available and relatively inexpensive. The video signal would be transported on a separate fiber network compared to the VOD traffic, and then to the central office and modulated to the home using VDSL2. A STB is still needed to decode the signals when they arrive at the subscribers home. RF overlay is ideal for broadcasting video content where each subscriber receives the same content. However, when transmitting targeted content delivery or VOD, RF overlay is not the best mechanism. “With the RF overlay model, all the video content must reside on the customer’s set-top box, using all the bandwidth going into the home.” (Fuller, 2005)

To implement IP video, the video signal is converted to IP data streams and sent over the Ethernet WAN links. When the packets arrive at the home, the STB converts the packets back to video signals that supply the TV. The first advantage of IP video is that it offers other enhanced features that RF overlay cannot. “For example, you could integrate telephony with video, being able to see caller ID on your TV as you’re watching it. From a video standpoint, IP gives you the ability to do things like change camera angles on a video on demand system.” (Fuller, 2005) The second advantage of IP video is that the data is easily transmitted over the GE WAN links without conversion. The RF signal needs to be demodulated and the data needs to be
encapsulated into IP packets to be transmitted on the GE links. When comparing RF overlay against IP video for delivering HDTV, IP video offers more features and only uses one network as opposed to two separate networks. The IP video equipment is more expensive because RF overlay has been around longer and used by the cable companies but the price of IP video equipment will come down in the future.
6 Quality of Service Techniques

The quality of service (QoS) for an HDTV service is a critical component to meeting the subscriber’s expectations. Video is more sensitive to jitter, delay and packet loss than data and about the same as voice. If the level of QoS is lacking or insufficient, the result will be a substandard HDTV service. As mentioned throughout this document, sufficient bandwidth to support HDTV service as well as data and possibly voice is the first and most crucial part of QoS for HDTV service. The next QoS concern is to not allow a video packet to be dropped. Dropping a video packet would cause an “artifact” in the picture quality, or a black square, that is very obvious and not acceptable. The third concern is avoiding jitter and delay in the HDTV signal. The results of both of these problems would also mean artifacts in the HDTV picture quality. “While QoS is often handled by network layer transport devices, assigning the correct priority level in the IP packet header is the responsibility of the sending ends including IP streamers, video servers and STB’s.” (Broadband Network Systems, 2005)

To effectively implement QoS in a network that is transmitting voice, data, and HDTV packets, it is necessary to identify which packets belong to which application. Identifying the packets is a process that is usually done by the network in a process called packet marking. Packet marking can be accomplished at the VLAN level by using the Class of Service (COS) field. Class of service is a way of organizing data in a network by grouping similar types of traffic together and treating each category as a class within its own level of service precedence. Some of the advantages of using CoS are its easy management and its ability to scale as a network grows in volume and structure. CoS is based on the Institute of Electrical and Electronic Engineers (IEEE) 802.1p standard which is actually an extension of the IEEE 802.1q standard. The two standards work in tandem with each other to mark Ethernet frames with a
certain priority. CoS has three main technologies to choose from; 802.1p Layer 2 tagging, Type of Service (ToS), and Differentiated Services (DiffServ). 802.1p tagging and Type of Service offer only simple priority tagging that is not sufficient for all the types of traffic that would be transmitted by the telecom operators network. The most critical portion of the network for QoS would be from the DSLAM to the subscriber’s home because all traffic travels over the same copper pairs. In the WAN portion of the network, the different data can be separated out into voice, video and data on separate fiber links so there is less contention between the three services. The fiber link that carries the video traffic should still use 802.1Q and 802.1P to classify the traffic as high priority for the multicast real-time viewing stream. The other possible streams such as propagating data to the VOD server would take a secondary priority, as they would not be affecting the direct viewing quality of the subscriber.

"Differentiated Services architecture is composed of a number of functional elements implemented in network nodes, including a small set of per-hop forwarding behaviors, packet classification functions, and traffic conditioning functions including metering, marking, shaping, and policing." (Network Dictionary, 2004) A traffic conditioner might not contain all four elements depending on the implementation. Differentiated Services operates at the IP layer and each of the IP packets are marked in a six-bit Differentiated Services field with a value known as the Differentiated Services Code Point (DSCP). The other two bits of the 1 byte DS field are currently not used. The DSCP value specifies a “per hop behavior” (PHB) for a given set of packet travel rules. The PHB describes the service level of the packet in terms of bandwidth, queuing theory, and discarding the packet. One of 64 possible forwarding behaviors is assigned to a packet from the value of the DSCP. The DSCP value goes through one logical classifier and
three logical traffic conditioners called the meter, the marker, and the shaper/dropper. Figure 16 shows a diagram of a traffic conditioner.

**Figure 16 Traffic Conditioner Flow Diagram**

![Traffic Conditioner Flow Diagram](image)

The traffic classifier selects packets based on one or more fields in the packet header.

There are two types of classifiers: the Behavioral Aggregate (BA) and the Multi-Field (MF). The BA classifier classifies the packet based on the DS codepoint only. The MF (Multi-Field) classifier selects packets based on the value of a combination of one or more header fields, such as source address, destination address, DS field, protocol ID, source port and destination port numbers, and other information such as incoming interface. (Blake, 1998)

The classifier is used to guide the packet to one or more conditioners.

The meter is the first conditioner and it measures the properties of a packet that the classifier set against a traffic profile preset in the Traffic Conditioning Agreement (TCA). The state of the meter in regards to a single packet can affect whether the marker or shaper/dropper takes action. The marker conditioner is the second in line and it sets the codepoint in the DS field adding the packet to a specific DS behavior total. “The marker may be configured to mark...
all packets which are steered to it to a single codepoint, or may be configured to mark a packet to one of a set of codepoints used to select a PHB in a PHB group, according to the state of a meter." (Blake, 1998) If a packet already has a codepoint and the marker changes that codepoint, the packet has been "remarked". The third conditioner is called the shaper/dropper. The shaper delays packets to ensure the stream is in compliance with the traffic profile. The shaper has a limited size buffer, and if the buffer runs out of space the packets may be discarded. The dropper is similar to the shaper because it holds a traffic stream in compliance with a traffic profile and drops packets if necessary. This process is known as "policing" the traffic stream. "The dropper can be implemented as a special case of a shaper by setting the buffer size to zero (or a few) packets. “ (Blake, 1998)

Using Differentiated Services for QoS in the access portion of the network, the telecom operator has the necessary number of priority levels to control all the different types of traffic going in and out the home. In a scenario where the subscriber has all three services of VoIP, IPTV, and broadband internet, the VoIP and IPTV service would get the highest priority with VoIP getting the lowest allowable latency. The broadband data packets would get the lowest level of priority because the data stream can tolerate the most delay. Differentiated Services might also be used in the core portion of the network on the video links to give priority to real time video instead of VOD packets. The VOD packets are stored in a VOD server in the central office so delaying or dropping them would not impact the subscriber’s service. Giving priority to the real-time HDTV packets would help to ensure the best throughput for those packets and a quality service.
7 Video Encoding

One of the critical technology improvements that make HDTV over DSL networks possible is the advanced video-codec technology like H.264 MPEG-4 part 10 Advanced Video Compression (AVC), RealVideo 9, On2 and Windows Media 9. Each of these video-codec formats provides increased compression rates for video coding reducing the required bandwidth by 50 percent over the MPEG-2 video-codec format.

RealVideo 9, On2, and Windows Media 9 are all proprietary codec formats based on the open standards developed by the International Telecommunication Union (ITU) and the Motion Picture Experts Group (MPEG). Proprietary versions of open standards don’t always integrate well with other software or hardware and could be difficult when trying to implement a new service. The MPEG-4 is an open standard that has many parts and profiles. The standard classifies architecture and coding methods to present compressed multimedia content in an interactive setting. The H.264 MPEG-4 part 10 AVC is a standard that was created as a joint venture between the ITU and the MPEG. Figure 17 shows the development of the open standards and the creation of the joint standard.
(Bray, 2005)

The MPEG-4 standard includes the AVC codec for video and the Advanced Audio Compression AAC codec for audio. These two codecs are the most advanced in their field and allow services like HDTV over DSL networks to be possible. The AVC codec can be implemented with different profiles that change the complexity as well as the compression performance of the codec.

The Baseline profile was created for low delay applications such as videophones and mobile phones. The Extended profile is intended for streaming video, the Main profile was created for interlaced video applications, such as broadcast TV and DVD’s. The Main profile uses the most tools and therefore gives the best compression. The AVC codec is not that different from the MPEG2 standard and actually uses many of the same concepts but improves on them. “Using a combination of all the new compression tools in AVC with the video
techniques developed for MPEG-2 (such as preprocessing and stat muxing). AVC's Main Profile will offer 40 percent to 50 percent gains in compression efficiency over today's MPEG-2. This applies for both Standard Definition (SD) and High Definition (HD) video.” (Bray, 2005) Figure 18 shows a comparison of MPEG-2 coded video and H.264 MPEG-4 part 10 AVC coded video and the Mbps bandwidth consumed by each codec in relation to the type of video.

Figure 18 Video encoding bandwidth comparison

<table>
<thead>
<tr>
<th></th>
<th>MPEG-2</th>
<th>H.264 MPEG-4 part10 AVC</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Standard Definition TV</strong></td>
<td>3.2 – 3.8 Mbps</td>
<td>&lt; 2 Mbps</td>
</tr>
<tr>
<td>Mid-Range MPEG-2 encoders</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Standard Definition TV</strong></td>
<td>1.8 – 2.8 Mbps</td>
<td>&lt; 1.4 Mbps</td>
</tr>
<tr>
<td>High-End MPEG-2 encoders</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>High Definition TV</strong></td>
<td>12 – 19 Mbps</td>
<td>5 – 6 Mbps</td>
</tr>
</tbody>
</table>

(Weiner, 2004)

The Mid-Range and High-End encoders refer to how the signal is collected. The quality and use of pre-processing, video analysis, and statistical multiplexing all play a part in the difference between mid-range and high-end encoders. Without the use of pre-processing, video analysis, and statistical multiplexing the H.264 MPEG-4 part 10 AVC would not be as efficient and therefore the increased size of the packets would have a major impact on the necessary bandwidth in the DSL network.

The H.264 MPEG-4 part10 AVC codec offer huge improvements over MPEG-2 in compression efficiency, providing high-definition video at 5-6 Mbps. H.264 MPEG-4 part10 AVC is also an open industry standard and not a proprietary adaptation of that standard so there shouldn’t be any problems when integrating with other software or hardware. The increased compression efficiency of H.264 MPEG-4 part10 AVC is significant enough to allow telecom operators to offer HDTV service over a DSL network.
8 Security Measures

HDTV over DSL networks is an IP enabled service and therefore the telecom operator must take precautions to protect against normal IP network security vulnerabilities, as well as theft of service issues that have plagued the cable industry for years. Digital Rights Management (DRM) is a huge concern for the movie industry with the implementation of digital IP video services. Hollywood will not provide content to a telecom operator that does not have sufficient security measures in place. Some of the techniques that could be used to provide security are firewalls in the STB’s, MAC address blocking, IP filtering, and authorization at the access level.

The first rule of network security is to secure the perimeter of the network. In this case the intruders or a hacker may be in the network already as a subscriber. However, a firewall built into the STB will help prevent attacks on other subscriber’s home networks. The firewall should be turned on as a default when sent to the subscriber’s home, but it should also be configurable by the subscriber because the STB is now the gateway to the Internet for data as well as video.

The firewall in the STB is a good start for protecting the subscribers network against malicious attacks but IP spoofing is another concern raised when discussing IP video services. “Hackers can spoof an IPTV network by trying to pretend they are a headend and flowing content down to your set-top box that might not be appropriate.” (Sullivan, 2005) The use of a public key infrastructure (PKI) system could prevent this IP spoofing type of attack. The PKI system uses a pair of electronic keys to share information across a public network. The STB’s could be pre-populated with the public key and the IP head end could own the private key. The IP video content could then be sent encrypted with the private key. The STB should only play content that is encrypted with the private key matching its own public key. To go one step
further, this should all be done without interaction from the subscriber and the keys should be updated periodically in case a hacker did manage to compromise the integrity of the PKI system. The new public keys could be sent using IP multicast to STB’s with MAC addresses that are registered with the telecom operator. This PKI implementation would also help protect against man-in-the-middle attacks.

MAC address blocking could be another useful security tool to prevent unauthorized access to IP video content. Address Resolution Protocol (ARP) request packets could be blocked by keeping an Access Control List of authorized STB’s and their MAC addresses. This technique would not hold up very well against an experienced hacker that has the ability to spoof a MAC address but it would prevent an inexperienced person from stealing a STB and illegally trying to access the HDTV services.

Authorization at the access level could be used as another security measure. A personal identification number (PIN) could be required to use the STB for viewing video. Separate PIN’s could be used to access different groups of channels giving parents control over what their children watch. Smart cards have also been suggested for use at the access level. A smart card containing a microchip could be preprogrammed with access codes that are required to use the STB for viewing. A small problem with this scenario is the smart card could be lost or stolen. A company named Widevine has invented another similar technique to smart cards.

Rather than relying on traditional smart cards used in conditional access systems (CAS) the Widevine “virtual smart card” system transfers highly secure “conditional access kernels” to set-tops and other devices that are equipped with AES (Advanced Encryption Standard) 128-bit hardware decrypters, resulting in an environment where security can be
renewed on a system-wide basis through software downloads whenever a service provider or other content source chooses. (ScreenPlays, 2005)

The biggest concern in the movie industry and therefore the telecom industry is DRM of the VOD content. The movie industry does not want the general public to be able to download newer movies and replicate them to other devices to share or sell. If the telecom operator does not implement a DRM system they will withhold content. “Once one person has managed to pirate a piece of content, it can go from one user to a million users in a very short period of time.” (Sullivan, 2005) The ideal situation for the movie industry would be to have the decryption keys for the content separate from the content. Using this method the decryption keys could be stored on the STB so that only that STB could view the content.

The most important aspect to keep in mind when planning a security method for the HDTV service is to make sure the method is updateable and manageable. When given enough time any security method can be broken so it is best when the security measures are flexible enough to change and adapt to the newest techniques and methods. The telecom operators could learn from the cable industry that has been trying for years to prevent theft of service. They should also leverage their experience as an ISP and the security issues that come with an IP network and the Internet.
9 Thesis Conclusion

The implementation of HDTV over DSL networks is very complex involving many factors that could prevent a quality service from being deployed. Quality of the data being delivered is the key to a successful implementation. The principal elements that affect the quality of the data are sufficient bandwidth throughout all portions of the network and the capturing and encoding of the data from the headend.

Sufficient bandwidth can be provided in the core portion of the network by using GE instead of ATM. GE can be used as a transport technology and it has many advantages over ATM such as easier management, scalability, cost, less inherent overhead, and data translation to and from TCP/IP. GE is much easier to manage than ATM and GE is more widely used so knowledge of the protocol is more common. Scalability is better with GE because the software can be used to control bandwidth in 1 Mbps increments. The cost of GE is much lower than ATM because GE is more widely used and therefore economies of scale help reduce costs. GE is more interoperable with TCP/IP so there is less translation between protocols than with ATM. IP broadcast and multicast will be two critical communication methods used to implement HDTV over DSL networks. Ethernet has the ability to use both broadcast and multicast while ATM does not. Ethernet networks are connection-less as opposed to ATM that is connection oriented. This requires that a connection be established before any data can be sent. This poses a great problem for ATM when trying to use broadcast or multicast communications. In order for ATM to provide broadcast or multicast service a point-to-point virtual channel connection would have to be set up to each of the receiving nodes. The biggest reason to use GE over ATM is the inherent overhead that comes with ATM. The 5 byte ATM header and additional headers and trailer of the AAL5 can have an overhead charge as much as 31.5 percent. This much
overhead cannot be tolerated on a network that requires maximum bandwidth and the best quality of service. GE should be used as the core transport protocol.

The network equipment in the core portion of the network, namely the B-RAS, needs to be replaced by the IP video router to provide a more reliable service. The load on the typical B-RAS makes it one of the most unreliable devices in the DSL network. The telecom operator will not be able to tolerate outages that affect the HDTV service offering like some data outages are tolerated today. The IP video router is specifically made for IP video transmission and should be used for dynamic discovery of STB’s, provisioning of the HDTV service, authentication, IP address assignment, and network configuration of the STB.

The access portion of the network will need to be upgraded to use VDSL2 and possibly bonded copper pairs. VDSL2 is able to transmit at a higher bit rate and for longer distances because it uses a higher Mhz band, trellis coding, Generic Convolutional Interleaving, and echo-cancellation. The VDSL2 standard provides for higher bit rates at longer distances, making HDTV service over DSL networks possible. Bonding copper pairs theoretically doubles the bandwidth allowing for VDSL2 to maintain higher bit rates and greater distances between the DSLAM and the subscriber’s home. Using Multilink Protocol as the bonding technology, the VDSL2 signal can be multiplexed to use more than one copper pair and deliver 30 Mbps from distances of 4900 feet or greater.

IGMP should be used in the access portion of the DSL network as a key element of IP multicast routing. IGMP provides a way for a device on a network to request its desire to receive video traffic and to periodically have the router poll STB’s for current status. IGMP plays an important part in the implementation of HDTV over DSL networks because it allows each STB to send a report to the access router indicating the desire to receive a certain video stream or

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channel. IGMP needs to be used in the access portion of the DSL network to enable diligent video stream controlling communication between the STB and the aggregate router. The crucial bandwidth saver in the access portion of the network involves IGMP and the DSLAM.

The equipment in the access portion of the DSL network, specifically the DSLAM, will need to incorporate more functionality. IGMP snooping is one of the critical functions that will enable HDTV service because it conserves bandwidth in the access portion of the network. The DSLAM is required to monitor the communication between the multicast layer 3 switch and the STB so it can direct which video streams are transmitted to a particular subscriber’s STB. The vital scaling issue associated with IGMP snooping is that all messages must pass through the DSLAM and be processed by the layer 3 aggregation switch. A layer 3 switch may become a bottleneck in the network when trying to respond to all the IGMP reports sent from the STB. The DSLAM can filter and remove the redundant packets sent from the 1000’s of STB’s sending IGMP report messages transmitted every 30 seconds to the aggregate switch. The DSLAM needs to be IGMP aware to enable the filter functionality and save on bandwidth in the access portion of the network. The DSLAM will also need to be moved closer to the subscriber’s home to shorten the length of the local loop. This will aid in the use of the VDSL2 protocol and help keep the busiest part of the network as short as possible. The telecom operator must use DSLAM’s that incorporate the use of IP multicast or face bandwidth issues in the core and access portions of the network. The use of ATM DSLAM’s would be a critical error for the telecom operator.

The internal network in the customer’s home has become an important part of the network architecture and overall quality of service for HDTV over DSL networks. If the HDTV packets are transmitted through the core and access portions of the network at high bandwidth
speeds and without error then the internal network at the customers home must be sufficient to carry those packets as well. When choosing between the two types of network architectures that exist for IP video service, I suggest using the centralized architecture. Bringing the DSL service to one device, the home media gateway, and propagating the data and video services from that device is less costly and involves less wiring than the decentralized architecture. Cat5E twisted pair cable should be used for data propagation and existing coaxial cable for HDTV service to other television sets within the house. A wireless LAN might be able to be used once the 802.11n specification is approved and STB’s are sold with an integrated wireless router. Until the 802.11n specification is approved a wireless LAN does not have sufficient bandwidth to support HDTV service.

The telecom operator also has to choose between IP video and RF overlay mechanisms when delivering video service. RF overlay has the advantage of being cheaper because it has been around longer and used by the cable industry. RF overlay is ideal for broadcasting video content where each subscriber receives the same content. However, VOD content must reside on the STB using up all the available bandwidth. IP video offers other features that RF overlay cannot like changing camera angles during a show. IP video contains less overhead when transmitted on an IP network than RF overlay because RF overlay has to be demodulated and then converted to IP packets. The IP video equipment is comparatively more expensive because RF overlay has been around longer but the price of IP video equipment will come down in the future. IP video is the mechanism to use because it offers the end user more features and transports more efficiently on an IP network.

New advancements in video encoding technology and compression have been one of the main factors that allow HDTV service over DSL networks to be feasible. Lower sizes of HDTV
video packets created with RealVideo 9, On2, Windows Media 9 and H.264 MPEG-4 part 10
AVC encoders require less bandwidth while still maintaining the HD quality of service. I
suggest using the H.264 MPEG-4 part 10 AVC encoder because it creates HDTV packets at that
require only 5-6 Mbps and the other encoders are proprietary standards encoders based off the
open MPEG-4 standard. Proprietary standards don’t always integrate well with other hardware
and software and there isn’t any advantage in packet size.

Quality of service (QoS) for an HDTV service is a critical component to meeting the
subscriber’s expectations and is the main focus of all the other measures implemented in the
DSL network. Video is more sensitive to jitter, delay and packet loss than data and about the
same sensitivity as voice. The next QoS concern is to disallow a video packet to be dropped.
Dropping a video packet would cause an “artifact” in the picture quality, or a black square, that
is very obvious and not acceptable. The third concern is avoiding jitter and delay in the HDTV
signal. Assigning the correct priority level in the IP packet header for video packets is the way
to ensure jitter, delay and packet loss are at a minimum. The best way to mark packet priority is
to use the Class of Service standard. CoS has three main technologies to choose from: 802.1p
Layer 2 tagging, Type of Service (ToS), and Differentiated Services (DiffServ). 802.1p tagging
and ToS offer only simple priority tagging that is not sufficient for all the types of traffic that
would be transmitted by the telecom operators network. DiffServ offers 64 levels of priority and
configurable options like per-hop forwarding behaviors, packet classification functions, and
traffic conditioning functions including metering, marking, shaping, and policing. DiffServ
should be used in the access portion of the network where the traffic is most congested because
all services are transmitted on the one copper local loop. 802.1p tagging and ToS could still be
used in the core portion of the network on the video links to ensure real time video traffic gets priority over other traffic like VOD traffic that is going to be stored on the VOD server.

Security should be an integral part of the network architecture for the telecom operator. Not only to avoid theft of service but also to protect subscribers from hackers and tampering with STB functionality. Since HDTV over DSL is on an IP network, attacks like denial of service, packet sniffing, snooping and man-in-the-middle are possible. To help avoid such attacks I suggest using a firewall on the STB to protect against intrusion. I also suggest the use of a PKI system so content could only be sent from the headend of the HDTV network. The public key that resides on the STB should be updateable by the telecom operator so keys can change in case they have been compromised. MAC address blocking is another method that could be used to help prevent theft of service. This method would not prevent an experienced hacker from spoofing a MAC address but it would help prevent an average user from stealing a STB and trying to use it to receive HDTV service. Access level security like a PIN number could also be used to stop theft of service. The PIN number could also be used as a filter for parents trying to monitor the content a child has access to. To protect digital rights of VOD content and appease the movie industry, encryption keys could be used to encrypt the HDTV packets. The ideal situation would be to store the encryption keys on the STB so the content could only be viewed using a STB and not copied to view on a personal computer. The most important aspect to put into practice when incorporating the security measures is to make sure the system is updateable. Given enough time and effort any security can be broken. It is important to be able to change the PKI keys, or data encryption for DRM protection so the system is a moving target.

The implementation of HDTV over DSL networks is possible using the right technology and network architecture. The telecom operator will be able to provide the necessary 30 Mbps of
bandwidth to support an HDTV service by implementing single mode fiber optic cable and GE in the core network along with VDSL2 and bonded copper pairs in the access network. Implementing IP multicast for daily normal content viewing and unicast for VOD viewing will assist in keeping the required load at a minimum on the entire network. DSLAM’s with the capability to filter Internet Group Management Protocol (IGMP) will assist in keeping the access network from being over burdened with IGMP report messages. Using H.264 MPEG-4 part10 (AVC) video encoding and IP video for video delivery, the size of the video packets can be minimized down to a size that would require 5 – 6 Mbps per HD channel. Using all of these techniques the telecom operator will be able to deliver a high definition video service over a copper local loop to the subscriber’s home.

9.1 Future Considerations

One area that could be considered for follow up is to see how some of the major telecom operators implemented video services in their networks. It is already known that AT&T (formerly SBC) and Bell South are choosing a fiber to the node approach and then using ADSL2+ over the copper local loop. Verizon is choosing to run fiber to the home in their approach to implementing a video service. Many analysts feel that the cost of running fiber to the home is going to bankrupt Verizon. It would be interesting to research whether AT&T and Bell South’s implementation were successful and if they can provide a quality high definition service and not just standard definition or if it is truly necessary to run fiber to the home. AT&T also plans to implement IP video using 40 IP video hubs spread out in the network and 140 IP office servers instead of one centralized IP video head end. This decentralized architecture has implications for data distribution and the way multicasting is set up through out the network.
Another subject that could be researched is the comparison of HDTV over DSL networks to HDTV over WiMax. "The WiMax protocol comes from the IEEE 802.16a standard and is capable of delivering up to 100 Mbps in a 20 Mhz channel, broadband connectivity to the home, business, and "WiFi hotspots" over distances as great as 30 miles." (Broadcast Engineering, 2003) If the WiMax protocol can deliver the bandwidth advertised and to the distances advertised this could be a big threat to HDTV over DSL networks. Obviously, the implementation of a tower every 30 miles or so would make it easy to reach a large number of subscribers. The towers would need to be connected to a distribution network to propagate the IP video content. What would that network architecture look like? What type of interference could cause a quality of service problem in the WiMax implementation? How would VOD work with such a service and what would the home network or STB be like? Could HDTV over DSL survive in the same area that HDTV over WiMax was implemented?
Figure 19 Network Diagram
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