Delivering video services over IP networks

Huda Al-Habsi

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Master’s Thesis

Title

Delivering video services over IP networks

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April 15th, 2004

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If we knew what we were doing, it wouldn’t be called research, would it?
Albert Einstein (1879-1955)
Abstract

The main goal pursued in this Thesis is to contribute towards the design and development of an end-to-end solution/system that would assist in reliable, consistence, less packet-loss delivery of high-quality video signals of pre-recorded presentations, training lectures, live events such as seminars over standard IP networks.

This Thesis will focus on the existing Internet Service Provider, Oman Telecommunications Company (Omantel) and its best delivery of high-bandwidth data such as video to its Local and regional offices and departments over IP networks.

This video-over-IP system aims to accumulate the technical scientific knowledge required to be able to offer high-quality video, which is fully scalable over IP networks. It aims to convert this knowledge into experimental prototypes, which, after the Thesis, can be developed into an integrated generic environment for Video-over-IP service development and content production.

The objective is to initially define the functionality of content Services that can be incorporated into the operations of Oman telecommunications company networks. Then define the functional characteristics and system requirements necessary for the deployment of content streaming services over Omantel IP based networks.

The design of this system would be combined with streaming high-quality video, while maintaining scalability and bandwidth efficiencies required for large-scale enterprise deployment. The design would encompass various components that are needed to capture, store and deliver streaming video to desktops. It will investigate on what is required to deliver quality video over Omantel IP networks and will recommend the actual products and solutions for achieving the end result.
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1. Introduction

Oman Telecommunications Company (OmanTel) was formed in the year 1970. OmanTel was entrusted with the task of developing the communication infrastructure of the fledgling country. Omantel has grown in these years from a mere 100 hundred line then to now connecting over 2 million lines across the breadth and length of Oman today. The technology the company has implemented has changed from the antiquated manual type exchange to the new generation high speed switching centers that dot the country. The growth has seen the employee strength of the Omantel skyrocket. The growth of Omantel is in such a way that the technology applied changes very fast. This requires constant upgrading of the skill set of its employees. Further more new recruits need to be trained the operational rules and regulation of the company. Being a vast country, combined with the natural geography of the terrain, it has seen the country develop in pockets that are distant from each other. Omantel forms a pivotal part of the renaissance taking place in the Sultanate of Oman. Serving the nation and its people has always taken precedence, with updating and improving customer services and training attaining tremendous importance. Employing and investing in state-of-the-art technology, Omantel has always tried to get the best of the world leading the nation towards the digital environment. With Global Networks Services (GNS), Digital Data Network, GSM and the Internet, the Sultanate of Oman has established a vital link with the rest of the world, opening up new horizons of investment.

Omantel believes in empowering its people being an important asset of the organization. Empowering people means empowering the company helping it to grow and thrive in today's competitive marketplace. Employees play a major role in its success. Empowering employees takes many forms, but for the purpose of this Thesis I'll concentrate on one section of the electronic Services – the video content services.

Generally speaking, Omantel internal electronic services empower employees with email services, providing each employee with a Max of 6MB of email server storage capacity. Employees are also provided with Browsing services, ftp and Intranet departmental resources and content. Omantel constantly tries to improve its internal communication services. The latest internal electronic services that was introduced lately was to track down the internal memos sent across units and departments. So far, it has proven to be an effective tool that assists administrators/secretaries identify the status of each memo and the action taken as a response to that memo.

Knowledge is power and it is vital to empower employees with knowledge through various channels. Omantel, as any telecom service provider, focuses on training and building up employees’ knowledge and awareness of new trends in the IT and Telecommunication industry. As a result, Omantel holds hundreds of seminars/ events and training presentations yearly. Some seminars are exclusive for top executives, General Managers, Managers and
Section Heads. Others are open invitation but usually the event venue cannot accommodate more than 60 people, besides it’s impossible for all Omantel employees to be present in a single seminar, unless it’s a weekend and the event will take place in one of the biggest reserved auditorium/ball rooms in a Hotel. The actual alternative would be to arrange these seminars/presentations to be held in a sequence/timely manner and to be attended by selected employees and based on monthly turns.

Omantel is finding the cost of rotating the people for training and back to production as time consuming and prohibitively expensive.

Introducing video-over-IP services would improve and enhance corporate communications in the organization through a client/server based application that would deliver (over IP-based network) live or on-demand content to Omantel’s employees’ desktops. This service will therefore allow Omantel employees to have the opportunity to watch and listen to important meetings in a live or pre-recorded form 24 hours a day directly at the comfort of their own desktops. This will certainly reduce traveling expenditures for employees/managers that are located distances from the headquarters/ the event venue.

This service will help in the delivery of important company information in a secure and timely manner. In addition, it will assist in extending the reach of corporate meetings or announcements beyond the boardroom, to the entire company or to a selected departments or groups. The proposed Omantel internal communications system will allow CEOs and executives to immediately and securely communicate with employees - anytime, anywhere. Therefore, the Key benefit of this system includes delivering critical information immediately, Making content available anytime, anywhere which will allow effective communication within the organization and will contribute towards training new recruits and upgrade the knowledge base of the existing staff.

The proposed video services is considered to be one of the most effective electronic solution for this problem. The proposed content delivery services would ensure a wide accessibility for training and seminar videos. Any event that takes place in Omantel is normally recorded for future retrieval. These Videos are usually kept in storage rooms for years until they’re requested for. Most videos have big values that will benefit employees for year and years to come. Therefore, it’s important to make those videos widely accessible for everyone to benefit.

This proposed video delivery solution will facilitate 2 services: Streaming live content and streaming on-demand content. Depending on the nature of the video, a decision will be made to what could be accessed by all employees and what be restricted to only decision makers and people in power.

This system is proposed to be deployed in the 6 Omantel Major Offices, 5 in the Capital (North) and 1 in the South Of Oman.
Two of the major offices are located within 1KM distance from each other. The 3rd is located 6KM from the first 2. The other 2 are located 15KM from the 3rd office and 2KM from each other. The last Main Office is located in the South Of Oman (Salalah) 2000KM from Muscat.

The company’s long distance switching division has implemented high speed ATM switches from Alcatel (Formerly New Bridge) and is providing high bandwidth links up to 30 Mbps to its 6 main offices. These ATM Switches are connected to the 6 Core Switches in the 6 locations (Cisco 6509) via STM1 (Fiber). Each of the 6 Core switches is connected via (Fast Ethernet) to the 6 Office’s LAN access Switch (Cisco 3500) that provides a 100Mbps Connection speed to each office (See Figure 1.2).

This thesis will look into the possibility of utilizing these high bandwidth lines to deliver video based training and seminars to the employees in the 5 Main Offices within the Capital Muscat (North) and to the Main Office in Salalah (South).

Figure 1.1 illustrates how this system would cover 6 Major Omantel branches in 6 locations in Oman:

- The Capital-Muscat (North) – 5 Main Offices including: TCC (Tower), Training Center, Technical Building, Burj Al-khuwair, Head-quarter’s Building.
- The City Of Salalah (South) – 1 Main Office.

Figure 1.1: Map of Oman
OmanTel Data Network Design

Figure 1.2: Omantel Data Network Design
2. The Internet Protocol (IP)

Packet-switched networks have evolved for over 30 years to form the basis of advanced data communications networks today. "Packet-switching initially provided the network environment needed to handle bursty, terminal-to-host data traffic over the analog telephone network. The emergence of the world-wide-web (WWW) as the user interface to 'cyberspace' firmly established IP as the de facto standard." (McDysan, 2000, p22). The Internet Protocol (IP) has enabled a global network between an endless variety of systems and transmission media. Around the world email exchange and web browsing are a part of everyday life for work, study and play. And by all indications other networks--phone, radio, and television--are also converging on IP to leverage its ubiquity and flexibility. With these new networks come new applications and more new users.

One reason for IP's tremendous success is its simplicity. The fundamental design principle for IP was derived from putting "smarts" in the ends of the network, the source and destination network host, leaving the network "core" dumb. IP routers at intersections throughout the network need do little more than check the destination IP address against a forwarding table to determine the "next hop" for an IP datagram. If the queue for the next hop is long, the datagram may be delayed. If the queue is full or unavailable, an IP router is allowed to drop a datagram. In other words, this protocol/layer provides a connectionless, unreliable packet based delivery service. It can be described as connectionless because packets are treated independently of all others. The service is unreliable because there is no guarantee of delivery. Packets may be silently dropped, duplicated or delayed and may arrive out of order. All attempts to deliver a packet will be made, with unreliability only caused by hardware faults or exhausted resources. Routers are allowed to discard IP datagrams en route, without notice to sender or receiver. IP relies on upper-level transports (e.g. Transmission Control Protocol TCP) to keep track of datagrams, and retransmit as necessary. And "these 'reliability' mechanisms can only assure data delivery; neither IP nor its high-level protocols can ensure timely delivery or provide any guarantees about data throughput" (McDysan, 2000, p24). IP provides what is called a "best-effort" service. It can make no guarantees about when data will arrive, or how much it can deliver.

The most widely used version of IP today is Internet Protocol Version 4 (IPv4). However, IP Version 6 is also beginning to be supported. IPv6 provides for much longer addresses (128 bits addresses) compared to IPv4 (32 bits addresses) and therefore for the possibility of many more Internet users. IPv6 includes the capabilities of IPv4 and any server that can support IPv6 packets can also support IPv4 packets.

IPv4 has two main disadvantages over IPv6 with respect to end-to-end transmission performance. First, the IPv4 header includes a checksum which must be computed by each
intervening node on a per packet basis. The designers of IPv6 resigned this time consuming and costly processing mechanism because most-higher level protocols have their own checksum control mechanism. Second, according to the IPv4 specifications [11], each router along the transmission path must process the variable length Option Field, again, on a per packet basis even though an option might be effectively used only by the end hosts. In IPv6, the new concept of an “ordered” linked list of Extension Headers ensures that routers process only the options necessary for correct operation. Hence, IPv6 resolves both problems; therefore, we conclude that IPv6 has the capability to reduce end-to-end delay due to faster packet processing in each single node along a given route.

3. Best-effort services and Real-time Data Traffic Requirements

The greater part of the traffic on the Internet, today, is non-real-time data. Real-time data traffic will come to dominate non-real-time as new services become available. Some of the new and emerging services, which are moving onto existing networks, and which are currently available are as follows:

- IP Telephony
- Video Distribution & frame synching
- Transaction processing
- Data acquisition and Control
- Virtual Reality
- Video teleconferencing
- B2B
- B2C

The Internet as well as IP networks architectures currently have one class of service normally referred to as “best effort.” This service is typified by first-come, first serve scheduling at each hop in the network. In this model, the highest guarantee the network provides is reliable data delivery using protocols like TCP etc. This is adequate for traditional data applications like electronic email, Telnet, File transfer (e.g. FTP), World-Wide-Web (WWW) access, etc. However, for real-time traffic such as voice and video (like video teleconferencing, Video streaming, video-on-demand and virtual reality), Internet as well as highly utilized networks have performed well only across unloaded portions of the network.

These Bandwidth-hungry applications generate large volumes of network traffic. This traffic is not network friendly in the sense that it does not "back-off" in the face of congestion. Because of the potential impact of this type of traffic on network resources, network administrators are prohibiting or limiting the deployment of multimedia applications on their networks. In
addition, these bandwidth consuming applications are sensitive to the quality of service they receive from the network. In particular, their treatment in the traditional manner by trying to ensure correct and fair delivery by trading off delay is not acceptable.

If we take one example, someone at home browsing the web, we see that in general a 56K Baud modem is generally sufficient to maintain desired data transfer rates. Web sites with large amounts of complex graphics make the data rate less than desirable. On average probably 10k to 20k baud is sufficient for the hours of browsing, e-mailing etc. This has changed since video took over. This data rate is over a thousand times greater than the average today. Clearly, many existing infrastructures are not going to be adequate to handle the increased load.

As a result of the tremendous growth of the Internet/ IP networks, IP's weaknesses are showing. "Increasing the available bandwidth to avoid congested networks is the obvious solution. But the problem is more than a simple capacity issue. The issue is that not only has traffic increased in volume, it has also changed in nature." (McDysan, 2000, p36)

As discussed earlier, this limitation has not been a problem for traditional Internet/ corporate networks applications like web, email, file transfer, and the like. But the new breed of applications, including audio and video streaming, demand high data throughput capacity (bandwidth) and have low-latency requirements when used in two-way communications (i.e. conferencing and telephony). Public and private IP Networks are also being used increasingly for delivery of mission-critical information that cannot tolerate unpredictable losses.

Network services can be categorized as best-effort, connectionless services or reliable connection- oriented services. Best-effort delivery describes a network service in which the network does not provide any special features to recover lost or corrupted packets. These services are instead provided by end systems. By removing the need to provide these services, the network operates more efficiently. For example, the postal service delivers letters using a best-effort delivery approach. It's hard to know whether a letter has been delivered or not. However, by paying extra for a delivery confirmation receipt, which requires that the carrier get a signature from the recipient and return it to the sender.

In the Internet protocol suite, IP is a best-effort service and TCP is a reliable service. Because IP provided basic packet delivery services without guarantees, it is called a best-effort delivery service. It does its best to deliver packets to the destination, but takes no steps to recover packets that are lost or misdirected. TCP, on the other hand, implements flow controls, acknowledgements, and retransmissions of lost or corrupted packets. This split in services "decentralizes" the network and moves the responsibility for reliable delivery to end systems. TCP is an end-to-end transport protocol, meaning that it runs in end systems, not the network. The services offered by TCP include the following:
Flow-control mechanisms control packet flow so that a sender does not transmit more packets than a receiver can process. Flow controls are necessary because senders and receivers are often unmatched in capacity and processing power. A receiver might not be able to process packets at the same speed as the sender. If buffers fill, packets are dropped. The goal of flow-control mechanisms is to prevent dropped packets that must be retransmitted. Flow controls are also used in the data link layer to control flow between devices that are directly connected. In contrast, TCP controls flow between devices that may be connected across a multi-hop routed network.

Reliable delivery mechanisms provide a way for a receiving system to acknowledge that it has received a packet, and a way for the sender to know that it must retransmit a lost or corrupted packet.

Congestion control mechanisms allow network systems to detect network congestion (a condition in which there is more traffic on the network than can be handled by the network or network devices) and throttle back their transmission to alleviate the congestion. Congestion occurs on busy networks. When it occurs, end systems and the network must work together to minimize the congestion. In contrast, flow controls are used between end systems. A receiver uses flow controls to signal to the sender that it is overloaded. The sender then throttles back or stops its transmission.

In the TCP/IP protocol suite, TCP provides guaranteed services while IP which is a protocol of the network layer provides best-effort delivery (Figure 4). “TCP (connection-oriented) is a reliable data delivery service that end systems use to recover packets that are dropped in the network due to congestion, or that are dropped at the end system itself due to overflowing buffers.” (Stevens, 1993, p223).

“During development, TCP/IP protocol suite designers realized a need for timeliness rather than accuracy.” (Stevens, 1993, p223) In other words, speed was more important than packet recovery. In real-time voice or video transfers, a few lost packets are tolerable. Recovering them creates excessive overhead that reduces performance. To accommodate this type of traffic, Transport layer Protocol was reorganized into TCP, and UDP (User Datagram Protocol). The basic addressing and packet-forwarding services in the network layer are done by IP. TCP and UDP are in the transport layer on top of IP (Figure 3.1) “Both use IP's services, but UDP is a stripped-down version of TCP that provides applications with access to IP's best-effort services. Applications go through UDP when they don't need TCP's services.” (Stevens, 1993, p143)
<table>
<thead>
<tr>
<th>Layer</th>
<th>Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application Layer</td>
<td>FTP, Telnet, SMTP</td>
</tr>
<tr>
<td>Presentation Layer</td>
<td>TCP, UDP</td>
</tr>
<tr>
<td>Session Layer</td>
<td>TCP, UDP</td>
</tr>
<tr>
<td>Transport Layer</td>
<td>TCP, UDP</td>
</tr>
<tr>
<td>Network Layer</td>
<td>IP</td>
</tr>
<tr>
<td>Data Link Layer</td>
<td>LLC</td>
</tr>
<tr>
<td>Physical Layer</td>
<td>Ethernet / Ring / RS-232 / RS-422 / RS-449 / RS-530 / V.35</td>
</tr>
</tbody>
</table>

1 Application / Service-Oriented Layers
2 Transport Layer / Delivery and Verification Services
3 Communication / Network-Oriented Layers

**Figure 3.1**: OSI 7-layers Functionalities

With best-effort services, packet discard is acceptable because recovery is handled by other services. Figure 3.2 illustrates where discards may occur. In the lower physical and data link layers, frames may be corrupted and dropped. In the network layer, congested routers drop packets.
In summary, IP packets are sent without establishing a connection first. If a router receives a packet, reads the IP layer information and can not figure out where to send it based on its rules, then the router will drop the packet. When dropping occurs, the source node has no notice that a packet no longer exists. Therefore, a higher layer or protocol must take control of maintaining a connection and keeping track of packets. Additionally, higher-layer protocols must also perform flow control, since connectionless services typically on a best effort basis without notion of bandwidth allocation.

Real-time Data Traffic requires network services beyond the simple "best-effort" service that IP delivers. Connectionless services do not guarantee packet delivery therefore, applications rely on higher-level protocols (e.g., TCP) to perform the end-to-end error detection/correction.

On congested links, best-effort service queuing delays will adversely affect real-time traffic. This does not mean that best-effort service cannot support real-time traffic - merely that congested best-effort links seriously degrade the service provided. For such congested links, a better-best-effort service is desirable. One method to reach beyond the best-effort service is by classifying real-time traffic separately from non-real-time traffic and giving real-time traffic priority treatment to ensure that real-time traffic sees minimum delays. Non-real-time TCP traffic tends to be elastic in its bandwidth requirements, and will then tend to fill any remaining bandwidth. This could be achieved by QoS.
4. Quality of Service (QoS): Overview

Quality of Service (QoS) refers to the “capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies” (McDysan,2000,p77). The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. Also important is making sure that providing priority for one or more flows does not make other flows fail. QoS offers applications a bandwidth-level guarantee, hence enabling network administrators to control the impact of these applications on the network.

In a QoS-enabled network, applications that need constant, high levels of bandwidth and low levels of latency, jitter, and loss (e.g., high-quality streaming video) can request and receive a guarantee for those network resources. Network Administrators can assign traffic that is less demanding (e.g., email) a lower priority.

Mixing real-time video services with non-real-time data services over the same network creates service quality problems that must be addressed. Without technology to guarantee the Quality of Service (QoS) of these new multimedia networks, non-real-time data traffic will interrupt video streams, and multiple video streams will interfere with each other. The result would be poor video/audio quality, long delays, and lost data packets resulting in poor system response.

“The starting point for providing QoS in any network is to control and avoid congestion.” (McDysan,2000,p278). The obvious solution to improve QoS is to overprovision network capacity and upgrade to the most efficient networking equipment. This is often a practical solution in the private network environment, but not for private WAN links. Another solution is to classify traffic into various priorities and place the highest priority traffic in queues that get better service. This is how bandwidth is divided up in packet-switched networks. Higher-level queues get to send more packets, and so get a higher percentage of the bandwidth.

Service providers have been reluctant to implement QoS across their networks because of the management and logistics problems. QoS features must also be set up from one end of a network to another, and that is often difficult to accomplish. QoS levels must be negotiated with every switch and router along a path. Still, QoS is getting easier to manage, and, in some cases, it is the only way to optimize network bandwidth. QoS focuses on the following parameters:

- **Throughput**: data transfer rate.
Packet loss: When a shared network gets busy, queues in routers and other network devices can fill and start dropping packets.

Latency: This is the delay in the time it takes a packet to cross a network. Packets may be held up in queues, on slow links, or because of congestion. The more networking devices a packet crosses, the bigger the delay.

Jitter: Delay that is variable and difficult to interpret.

Of course, the range, location, and ownership of the network will make a big difference in how QoS is applied. An enterprise may wish to install QoS on its own intranet to support voice and video. QoS may also be applied to the LAN/WAN gateway to ensure that private WAN links or VPNs are appropriately loaded and provide quality service for inter-company Video Streaming, for example.

4.1 Service Levels: IP Versus ATM

The Internet is a connectionless packet-switching network, meaning that without any special QoS provisions, all services are best effort. In contrast, leased lines and ATM naturally support QoS because they deliver data in a predictable way. “Leased lines such as T1 circuit use TDM (time division multiplexing), which provides fixed-size repeating slots for data. ATM uses fixed-size cells and has built-in traffic engineering parameters to ensure QoS.” (McDysan,2000,p82).

Obtaining QoS in IP networks is not so easy, primarily for the following reasons:

- The architecture is routed, meaning that packets may take different paths, which produces unpredictable delays.
- IP is connectionless, that is, it does not have virtual circuit capabilities that could be used to allocate and guarantee bandwidth.
- IP uses variable-size packets, which makes traffic patterns unpredictable.
- Packets from many sources traverse shared links and may burst into routers, causing congestion; packet drops; retransmission; and, ultimately, excessive delay that is unsuitable for real-time traffic.

A typical LAN/WAN interface is “an aggregation point where traffic from many sources inside the network comes together for transmission over the WAN link. If the WAN link has insufficient bandwidth, congestion will occur.” (McDysan,2000,p256).

The following describes the various techniques that may be used to provide QoS on the enterprise networks as well as the Internet. Some of these solutions provide only partial QoS, but are required to provide higher levels of service.
• **Congestion management:** Schemes that actively work to prevent congestion from occurring.

• **Classification and queuing techniques:** Classifying Traffic according to service levels. Queues exist for each service level, and the highest priority queues are serviced first.

• **Bandwidth reservation techniques:** Bandwidth is reserved in the network to ensure packet delivery.

• **Packet tagging and label switching:** Packets are tagged with identifiers that specify a delivery path across a network of switches. The paths can be engineered to provide QoS.

4.2 Congestion Management Techniques

Managing network congestion is a critical part of any QoS scheme. As mentioned earlier, TCP uses some congestion control mechanism. The technique relies on dropped packets. When a packet is dropped, the receiver fails to acknowledge receipt to the sender. The sender assumes that the receiver or the network must be congested and scales back its transmission rates. This reduces the congestion problem temporarily. The sender will eventually start to scale up its transmissions and the process may repeat.

Packets are dropped because a router queue is full or because a network device is using a congestion avoidance scheme, such as RED (random early detection). "RED monitors queues to determine when they are getting full enough that they might overflow. It then drops packets in advance to signal senders that they should slow down. Fewer packets are dropped in this scheme.” (McDysan, 2000, p301). The problem with RED is that it relies on dropping packets to signal congestion.

"Traffic shaping is a technique that "smoothes out" the flow of packets coming from upstream sources so that downstream nodes are not overwhelmed by bursts of traffic.” (McDysan, 2000, p107). An upstream node may be a host, or a network device that has a higher data rate than the downstream network. At the same time, some hosts with priority requirements may be allowed to burst traffic under certain conditions, such as when the network is not busy. "A traffic shaper is basically a regulated queue that takes uneven and/or bursty flows of packets and outputs them in a steady predictable stream so that the network is not overwhelmed with traffic." (McDysan, 2000, p107).

4.3 Classification, Admission, and Tagging

Any QoS scheme involves guaranteeing service levels to traffic flows. In a world of infinite bandwidth, all flows could be handled equally. But networks are still bandwidth limited and congestion problems occur due to improper network design. Therefore, "traffic must be classified-and, in some cases, tagged-so that downstream devices know what to do with it” (McDysan, 2000, p99). Basic classification techniques are as follows:
Classifying incoming traffic using various techniques, such as “sniffing” the MAC address, source and destination IP address, well-known TCP/UDP port numbers, application information at layer 7, such as cookies and other information.

If a flow is requesting a particular service, use admission controls to either accept or reject the flow. “Admission controls help enforce administrative policies, as well as provide accounting and administrative reporting.” (McDysan, 2002, p278)

Schedule the packets into appropriate queues and manage the queues in a way that ensures that each queue gets an appropriate level of service for its class.

Classification requires administrative decisions about how traffic should be classified and where it should be tagged (McDysan, 2002, p99). Administrators might classify traffic based on whether it is best effort and suitable for discard, real-time voice and video, network controls (e.g., OSPF messages), or mission critical.

4.4 IETF QoS Solutions

The IETF has been working to define Internet QoS models for many years. The primary QoS techniques developed by the IETF are Int-Serv (Integrated Services), Diff-Serv (Differentiated Services), and MPLS (Multi-protocol Label Switching), as described next.

This section will focus briefly one of the Standards used in QoS and is recommended to be explored and tested after the implementation of the proposed Streaming Services solution.

- Integrated Services (Int-Serv)
  This is a model for providing QoS on the Internet and intranets. The intention of Int-Serv designers was to set aside some portion of network bandwidth for traffic such as real-time voice and video that required low delay, low jitter (variable delay), and guaranteed bandwidth. The Int-Serv Working Group developed RSVP (Resource Reservation Protocol), a signaling mechanism to specify QoS requirements across a network. “Int-Serv has scalability problems and it was too difficult to deploy on the Internet. However, RSVP is used in enterprise networks, and its control mechanism for setting up bandwidth across a network is being used in new ways with MPLS.” (McDysan, 2000, p70).

- Differentiated Services (Diff-Serv)
  Diff-Serv classifies and marks packets so that they receive a specific per-hop forwarding at network devices along a route. The important part is that Diff-Serv does the work at the edge so that network devices only need to get involved in properly queuing and forwarding packets. “Diff-Serv works at the IP level to provide QoS based on IP ToS settings.” (McDysan, 2000, p72). Diff-Serv is perhaps the best choice for signaling QoS levels available today.
• **MPLS (Multiprotocol Label Switching)**

MPLS is a protocol, designed primarily for Internet core networks, that is meant to provide bandwidth management and quality of service for IP and other protocols. "Control of core network resources is accomplished by building LSPs (label switched paths) across networks and rapidly forwarding IP packets across the network through these paths.” (McDysan, 2000, p.358). By labeling packets with an indicator of the LSP they are to traverse, it is possible to eliminate the overhead of inspecting packets at every network device along the way. LSPs are similar to virtual circuits in ATM and frame relay networks, and traffic engineering approaches can be used to create LSP that delivers a required level of service.

The final pieces of the QoS picture are policies, policy services, and policy signaling protocols. Most of the QoS systems just described use policy systems to keep track of how network users and network devices can access network resources. A defining feature of a policy system is that it works across a large network and provides policy information to appropriate devices with that network.

A policy architecture consists of the following components, which primarily manage the rules that govern how network resources may be used by specific users, applications, or systems. When rules are specified and programmed into policy systems, they are known as policies.[11]

- **Policy clients** Network devices that process network traffic such as switches and routers running various queuing algorithms. Policy clients query policy servers to obtain rules about how traffic should be handled.
- **Policy servers** This is the central authority that interprets network policies and distributes them to policy clients.
- **Policy information system** The information about who or what can use network resources is stored in some type of database, usually a directory services database.

This architecture allows network administrators to specify policies for individuals, applications, and systems in a single place—the policy information system. The policy server then uses protocols such as LDAP (Lightweight Directory Access Protocol) or SQL to obtain this information and form policies that can be distributed to policy clients. Policy clients talk to policy servers via network protocols such as COPS (Common Open Policy Service) and SNMP (Simple Network Management Protocol). “COPS is an intra-domain mechanism for allocating bandwidth resources and it is being adapted for use in establishing policy associated with a Diff-Serv-capable networks.” [13]
5. Video-Over-IP / IP Streaming Video

Legacy video signals are based on analog technology. They were carried via expensive transmission circuits. Through advancements in digital video compression, composite audio and video signals can now be carried over typical network circuits both on the LAN and across the WAN, and even over the Internet. Video-over-IP is a new and emerging technology that combines switched packet networking with streaming video. “IP-based streaming video is poised to become an enterprise application for distributing sales, marketing and training across Intranets, Extranets and VPNs.” (Menin, 2002, p52)

IP Streaming Video allows video signals to be captured, digitized, streamed and managed over IP networks.

The first step is the capturing of the video content. This can be accomplished via several means. The content is then processed, compressed, stored and edited. The content can either be “live” or pre-recorded and stored. These transmissions can then be sent via the network to either one or several stations for viewing singularly or simultaneously. The viewing station will need to have the hardware and the software to view the delivered content.

When an analog signal is converted to a digital signal (as in video or voice transmissions) the process is completed by what is known as sampling. Sampling, as the name implies, refers to “taking samples of the signal at various times per second (the sampling rate) within a sampling depth (bits per sample). The greater the sampling, the larger the file will be.” (Mack, 2002, p196)

The same technique is used for video, although a bit more complex. The difference here is that what is now being transmitted is a raster image in picture elements also known as pixels. The MPEG standard uses what is called “lossy” compression. “That is much of the image is “lost” but not enough to diminish comprehension by the human eye as the human brain fills in the gaps.” (Menin, 2002, p113) The video is sampled in segments of the video. The first frame (the index frame) is transmitted entirely and the remaining frames transmit changes as compared to the initial index frame. The greater the compression, the greater the “lossiness” of the frame will be.

“In a congested network, samples can be received out of sequence and a phenomenon known as Pixilation occurs.”[16] Pixilation happens when the pixels seem out of place when compared to the original index frame and the image is skewed. There are two ways to compress the feed. One is to lower the resolution and the other is through reduction in the sampling rate. When compressed, the feed will consume fewer resources, but obviously there is a trade-off between compression and video quality.
Video presentations can be grouped into three categories: Video Broadcasting, Video on Demand, and Video Conferencing. Of the three, only video conferencing is full duplex, the others are essentially one way transmissions. These new business tools bring disparate offices together on one enterprise and are being deployed rapidly.

Video broadcast over IP is a network-based one-way transmission of video file content. The endpoint is merely a passive viewer with no control over the session. Video broadcast can be either Unicast or Multicast from the server. In a Unicast configuration, the transmission is replicated by the server for each endpoint viewer. In a Multicast configuration, the same signal is sent over the network as one transmission, but to multiple endpoints or, simply, a group of users.

This technology is being implemented in corporate environments as a means to distribute training, presentations, meeting minutes and speeches. It is also being utilized by universities, continuing education or technical education centers, broadcasters, webcast providers, just to name a few. "There are three factors to determine how much bandwidth this technology will require: the number of users, their bandwidth to the server, and the length of the presentation or video." (Mack, 2002, p32)

The integration of these video and IP has lead to several questions as to the measurements that determine the quality of a Video-over-IP stream. There are different fundamental properties that define the quality of a Video-over-IP stream.

Unlike data transfers over IP, streaming video quality is measured live and at the end-point. "Quality end-point video is not solely a function of network bandwidth nor is it solely a function of MPEG-2. In fact many of the issues that surround quality end-point video are a combination of both the MPEG-2 quality and the level of deterministic IP packet delivery of the network." [17] Unlike data traffic which measures quality by speed of reliable throughput with little attention to the nature of the payload, video (as voice) demands more from network transport.

Networks designed to carry streaming video must account for the payload they carry. Furthermore "the type of MPEG-2 stream being transported effects the minimum and maximum boundary characteristics that the network packet delivery and the overall system must conform to for quality streaming video at the end-point." [17]

The quality of a particular streamed video experience is clearly a subjective judgment; there are no universal metrics. However, as mentioned earlier, there are many factors that determine that experience, the most obvious of which is the effective bandwidth of the connection between the viewer and the server of that content. Other than the network related measures like propagation delay and latency, it depends on other factors related to content itself which include:

Thesis
Format. Although each format can typically be used at a wide variety of bitrates, different formats will give different results at different rates and for different types of content.

Codec. For some formats, such as MPEG, there’s more than one codec to choose from. Choosing a different (sometimes more expensive) codec will sometimes allow better quality at a given bitrate and with other parameters equal, usually at the cost of requiring more processor power and memory during the encoding.

Source material quality. Clearly if quality is compromised in the source it won’t be so good after impression. If the source is analogue, the first stage in encoding is usually to transfer it to uncompressed high definition digital video, which gives a digital master with the maximum amount of information in it which can be copied and re-copied without loss.

Encoding software. Encoding tools will usually provide a wide range of parameters to tweak during the encoding; some tools help in specifying whether the video is action, interviews, landscapes etc, selecting the output streaming technology (e.g. ISDN) etc. The Codec as well as the quality basically determines the transmission delay which in turn affects the end-to-end delay and latency.

The process of encoding, if done right, involves a certain amount of trial and error since the right parameters are very much dependent on the type of video being encoded. “For example, A kung-fu movie needs to be encoded at the highest frame rate available or it will look jerky - again it is possible to reduce the frame rate of a talking head without being noticed by the viewer, which can save a lot of bandwidth for better picture quality.” (Mack, 2002, p271)

5.1 Streaming

In order to avoid latency, video data needs to be available continuously and in the proper sequence without interruption. Until fairly recently, video had to be downloaded entirely to the computer before it could be played. With streaming, the file remains on the server. The initial part is copied to a buffer on the computer and then, after a short delay, starts to play and continues as the rest of the file is being pulled down.

“Streaming provides a steady method of delivery controlled by interaction between the computer and the server. The server regulates the stream according to network congestion and thereby optimizes the presentation on the client’s computer.” (Menin, 2002, p17)

The non-streaming method used which is Download and Play/ Website Embedded video/Audio, which is a process whereby a user must first download the entire media file before playing it cannot be used for live broadcasts, however it is often a good way to deliver high quality media content over any bandwidth (although download time may be a problem when delivering large files over lower bandwidths). So the disadvantages of this method are that a storage space would be required and the download times can be
very long particularly with dial-up modems/ highly-utilized networks. They are however the easiest to implement as no special 'server' is required.

This alternative type of delivering video to the end-user desk-tops is not recommended for Omantel's proposed video-end-to-end delivery solution because it is not desired to leave a copy of these training videos on any user's desktops for copy-right reasons and for the confidentiality nature of these training videos. Section 4.6 will explain the Distribution control in more details.

There are two key streaming delivery techniques that is recommended to be used by Omantel streaming Services: unicast for on-demand content and multicast for Live-content. Unicast refers to networking in which computers establish two-way, point-to-point connections. Most networks operate in this fashion, users request a file, and a server sends the file to those clients only. When streaming multimedia over a network, the advantage to unicast is that the client computer can communicate with the computer supplying the multimedia stream. “The disadvantage of unicast is that each client that connects to the server receives a separate stream, which rapidly uses up network bandwidth.” (Menin, 2002, p187)

On the other hand, IP Multicast refers to the networking technique in which one computer sends a single copy of the data over the network and many computers receive that data. Unlike a broadcast, routers can control where a multicast travels on the network. When streaming multimedia over the network, the advantage to multicasting is that only a single copy of the data is sent across the network, which preserves network bandwidth. “The disadvantage to multicasting is that it is connectionless; clients have no control over the streams they receive.” (Menin, 2002, p188) To use IP multicast on a network, the network routers must support the IP Multicast protocol and most routers now handle multicast. More details on IP multicast will be explored in Section 7.

5.2 Types of Streaming

There are two basic types of streaming media: http (Hyper Text Transfer Protocol) and rtp (Realtime Transfer Protocol)

1. http (Hyper Text Transfer Protocol)

Is used for pre-recorded streaming only. In addition, it requires downloading the entire file to the hard disk, though it might start playing partway through the download (progressive download).

"http is already used by all Web servers to store and transmit ordinary text and graphics files on the Web. And from the producer's point of view, there's no added effort because it could be used without the added management requirements and expense of server-side streaming software." (Microsoft Corp, 1999, p107) Although
this technique is not well-suited for high-volume sites serving numerous simultaneous streams, many smaller Web sites can benefit tremendously from this simple and inexpensive approach. The HTTP-based approach does not allow for live streaming audio or video presentations because complete files must be stored on the Web server before they can be accessed. Finally, “HTTP does not make efficient use of server resources, and as a result doesn’t perform well under heavy server loads.” (Microsoft Corp, 1999, p260) (Section 5.7 will explain this in much more details)

2. RTP (Realtime Transfer Protocol)

RTP uses both IP and Ethernet as a delivery mechanism, while controlling the problems that these layers arouse. It can be viewed as the big brother, controlling everything and keeping order. It firstly packages the application data into IP Packets and ships them to a router.

Furthermore, “RTP is used for real-time streaming. Movie packets are sent in real time, so that a one-minute movie is sent over the network in one minute. The packets are time-stamped, so they can be displayed in time-synchronized order.” [14]

Because packets are sent in real time, RTP streaming works with live content in addition to pre-recorded movies (on-demand content) and can even carry a mixture of the two.

RTP can be used for live transmission and multicast, this allows the user to view long movies or continuous transmissions without having to store more than a few seconds of data locally. Using RTP transmission under RTSP control, a user can skip to any point in a movie on a server without downloading the intervening material. [15] In other words, RTP buffers only a few seconds of the media stream; the whole file is not saved on our hard disk. Real-time streaming / true streaming send data to the desktop continuously, but do not download the entire file. Some streaming servers maintain a constant conversation with the client in order to determine how much bandwidth the user can support. Based on this information, the server adjusts the data-stream accordingly and sends just enough video to the client. The streaming player on the client machine plays the video in real-time.

Real-time streaming is generally best for longer pre-recorded clips or live events. A dedicated server is used to deliver video in real time. The server, client and the respective streaming software manages to maintain the delay and jitter to acceptable values.

“RTSP takes advantage of streaming which breaks data into packets sized according to the bandwidth available between client and server. When the client has received enough packets, the user’s software can be playing one packet, decompressing another, and downloading the third.” [15] This enables the user to listen or view the real-time file almost immediately, and without downloading the entire media file. This applies to live data feeds as well as stored clips.

- Provides for on-demand access of multimedia items such as stored real-time audio/video files, live real-time feeds, or stored non-real-time items.
- Allows interoperability between client-server multimedia products from multiple vendors.
- Provides for control and delivery of real-time media and associated events between a media server and large numbers of media clients.
- Addresses key concerns of Internet content-providers and users—quality of service, efficiency of delivery, rights management, and measurement. It also provides a underpinning for developing the richest possible streaming multimedia applications.

RTSP (Real Time Streaming Protocol) is an Open Standard protocol for streaming media, used by the majority of streaming vendors, including RealNetworks, Apple, and Microsoft. "It is important for network administrators not to block RTSP protocols and connections in their firewalls." (Mack, 2002, p425)

5.3 Advantages of Real-time Streaming

There are many advantages to direct streaming, as opposed to simply posting a movie to a web site and having people download it.

- **Instant play**
  With streaming, people can see media play right away—there are no lengthy downloads.

- **Live events**
  Streaming is the only way to distribute live events such as seminars and training events.

- **Long-form media**
  Streaming media are not limited to file sizes that make a reasonable download. "Long-form media such as feature films and concerts that would make multi-gigabyte downloads can stream effortlessly." (Menin, 2002, p17)

- **Unicasting, Broadcasting and Multicasting**

- **Random access**
  Viewers can pause, fast forward, or otherwise interact with prerecorded movies and play only the parts they want. It allows the user to skip ahead.

- **Distribution control**
  "Streaming allows you to maintain control over the distribution and copyright of your media. Anyone can download a movie, alter it, and redistribute it
themselves, but it’s much harder to redistribute the contents of a stream.” (Menin, 2002, p79). When the client saves the streaming movie, all it is saving is the URL of the stream and some user settings. The actual data is never copied.

- Uses no space on viewer’s hard disk and doesn’t leave a copy of the movie on the viewer’s hard disk and prevents pirating it.

- Never uses more bandwidth than it needs. That’s when the proper Streaming server is in place to determine how much bandwidth the user can support and based on this information, the server adjusts the data-stream accordingly and sends just enough video to the client.

5.4 Disadvantages of Real-time Streaming

- If the user’s connection isn’t fast enough, or the network experiences congestion, frames may be dropped in order to preserve the real-time playback. Lost packets are gone for good; movie always loses some data (though some data is almost always lost over the Internet – over a LAN, there is normally no data loss). In other words, “sound quality and stream may be affected by low speed or inconsistent Network connections.” (Menin, 2002, p18)
- Movie breaks up if data rate exceeds connection speed.
- Can be stopped by firewalls or NAT (Network Address Translation)
- High cost of server software.
- Requires a preconfigured server.

5.5 Streaming Architectures & Codecs

As far as a computer is concerned, the two most important aspects of working with streaming media are:

- “Synchronizing, managing and playing media” (Mack, 2002, p78)
- “Making both the video and audio components small enough to play properly on a computer or through a network connection while storing the file in a manageable amount of space” (Mack, 2002, p78)

To effectively handle these tasks, two special types of technology were developed: streaming architectures and Codecs.

Architectures provide the overall structure and synchronization for media delivery. Codecs are the smaller encoding components that fit within an architecture. For example, QuickTime, Network’s RealSystem and Windows Media are the 3 major architectures today, Sorenson Video and MPEG-4 are video Codecs, and QDesign, RealAudio and WMA are audio Codecs.
Multimedia architectures address the first issue of handling and synchronizing digital video files. Architectures are often called formats, which is misleading. Formats are the file structures an architecture creates with its Codecs, such as a QuickTime Movie or a Windows Media Video. Besides, an architecture is much more than just a format. For example, QuickTime controls how streaming media is handled by the computer, including file conversions, how movies are displayed on the screen, and much more. “Although the various architectures have a lot in common, there are also quite a few differences between them. Some are dedicated to playback via the web, others are better for CD-ROM and many work best on a specific range of computers.” (Mack, 2002 ,p88)

The following section will look at the digital video architectures that have come to dominate the online market. They include QuickTime – originally developed by Apple, Real Video – a product of Real Networks, and Windows Media – the Microsoft version. After the comparison, we will explore the different digital video formats and compare them to MPEG.

5.5.1 RealNetworks

RealNetworks’ products are fully SMIL (Synchronized Multimedia Integration Language) compatible, and are designed for various types of content to be integrated together within the player. There are two sides to this approach, one being that no matter what operating system the user views content on it will appear and operate the same way within the player. “The disadvantage of this approach is that content is limited to the file compatibility of RealOne (the new RealPlayer). For example, you cannot currently include Microsoft PowerPoint presentations in their original format. They must be converted to images such as JPEG or GIF.” (Mack, 2002 ,p86)

RealNetworks have recently launched their ninth platform version Helix. Helix is the new media delivery system from RealNetworks. It consists of several components, the encoder Helix Producer, the Helix Universal Server, and Helix Universal Gateway. “The basic versions of these are free but very limited in their capabilities.” (Menin,2002 ,p148)

Helix Producer Basic will encode the recorded content into a streaming format but limits the multiple bit-rate encoding to three connection speeds. Helix Producer not only encodes the recording into RealVideo format but it can also create synchronised multimedia presentations for playback within the RealOne player. "Helix Producer also has the ability to capture and broadcast directly from a FireWire source, which keeps the quality high and the connection of devices simple." [25] Helix Producer Plus is an enhanced version that offers more encoding features.

Helix Universal Server is the universal digital media delivery platform from Real. With industry leading performance, integrated content distribution, Web Services support, and native delivery of RealMedia, Windows Media, QuickTime, and MPEG 4, Helix
Universal Server offers a complete range of solutions and features for deployers of digital media.\[^{24}\]

Helix Gateway Server Offers Universal media delivery and caching capabilities in one integrated solution. Containing all the functionality of the Internet server, plus an integrated universal media cache to reduce bandwidth usage, this server is the ideal edge solution for Service Provider distributed deployments.\[^{26}\]

### Advantages

- Helix Producer Basic version is free with good support on operating systems
- The most widely used streaming format
- Helix Universal Server can stream all the three main formats (Windows Media, QuickTime, RealVideo)
- Excellent quality at modem streaming rates. For most situations, the best quality streaming available
- SMIL support

### Disadvantages

- Cost: Extended encoding features (e.g. Helix Producer Plus) must be paid for
- Slow updates to the format

#### 5.5.2 Windows Media

At the other end of the scale, Windows Media Player is designed to be embedded and controlled from other sources, such as an HTML page, or Java/VBScript commands. "The main advantage of this approach is the ability to control and link all web-compatible sources of media. This is extended considerably by all browsers having support for plug-in modules which extend their capabilities and compatibilities." (Menin, 2002, p148) Of course there is a downside. As this software is designed by Microsoft, it is intended to work at its best with other Microsoft products. This limits its compatibility with other operating systems and non Microsoft browsers. Microsoft's free encoding software is Windows Media Encoder 9 Series that allows converting both live and pre-recorded audio/video into Windows Media format for live and on-demand delivery. "Windows Media Services 9 Series is one of the most powerful streaming media servers available in the market." (Microsoft Corp, 1999, p15)

Windows Media Producer is free from Microsoft and it uses a wizard to walk you through the creation of single-speed and multi-speed ASF (Advanced streaming format) files. Windows Media offers scalability through "Intelligent Streaming"\[^{38}\]. Meaning that each file can contain up to five video tracks. At the beginning of playback, the Media Player and the Windows Media Server communicate to pick the video track which best matches the viewer's connection.
Advantages

- Free: All Microsoft developed video applications are freely available and unrestricted in use
- Good integration abilities, such as HTML embedding and Visual Basic support
- Encoder application is wizard-based for ease of use
- Overall high quality video and audio from 128Kbps upwards

Disadvantages

- Poor audio quality when streaming with video at modem speeds
- Users require Microsoft’s Internet Explorer to view embedded media
- Streams can only be viewed on Microsoft Windows or Macintosh computers
- Needs a plug-in to be compatible with Netscape
- Requires a Microsoft Windows Server to host streams

5.5.3 Apple QuickTime

"Apple QuickTime has the ability to display various synchronized media within its player through SMIL, and also be controlled externally when embedded into a web page." (Mack, 2002, p351). However, it does not have the same level of functionality as RealNetworks’ player or Windows Media Player.

QuickTime Pro also provides simple editing, basic effects, and import/export of many types of media. Sorenson is Apple’s Encoding server that allows for very high quality video, indistinguishable from its source in most cases; “the downside of this is the file size for the stream is usually far larger than either Real or Microsoft and would require a substantial amount of bandwidth to stream.” (Menin, 2002, p154)

QuickTime Streaming Server is designed for Mac OS X Server, but it’s also available as an open source server called Darwin Streaming Server. Versions are available for Linux, Solaris and Windows NT/2000. In addition, there is the QuickTime Streaming Server 4 which extends its support for standards by adding MPEG-4, and MP3 to its palette of capabilities.

One major benefit for QuickTime when it is used in its native Macintosh environment or on a Windows based PC is its compatibility within numerous types of web browser. (Table 5.1 summarizes each player’s compatibility with systems and file formats.)

Advantages

- Extremely high quality video reproduction at a fast connection speed
- Player can view a wide range of media formats
- SMIL support
- Support 3D objects

**Disadvantages**

- Requires a bought-in package (e.g. QuickTime Pro, Adobe Premiere) to create
- Apple MOV files on PC systems
- "Poor quality footage below 1000Kbps when real-time streaming" (Menin, 2002, p155)

**Table 5.1: Architecture Comparison**

<table>
<thead>
<tr>
<th>Player</th>
<th>Compatibility</th>
<th>File Formats</th>
<th>Browser Compatibility</th>
<th>Availability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Windows Media Player</td>
<td>All Windows Mac OS 8.1/ 9.x Mac OSX Pocket PC Solaris HP HandHeld Csio Plam Compact Plam</td>
<td>ASF, WMV, WMA WAV, AVI, MPEG-1 MPEG-2 MP3 MPEG-4</td>
<td>Internet Explorer 5.0 +, Netscape 4 + (Additional plug-in needed), Mozilla (with Netscape plug-in)</td>
<td><a href="http://www.windowsmedia.com">www.windowsmedia.com</a> Free</td>
</tr>
<tr>
<td>RealOne</td>
<td>All Windows Linux Mac OSX Pocket PC Solaris</td>
<td>RAM, RA, RealPix RealText MPEG-1 MPEG-2 MPEG-4 AVI, SMIL MP3</td>
<td>Internet Explorer Netscape, Mozilla Opera, Various Linux Browsers</td>
<td><a href="http://www.realnetworks.com">www.realnetworks.com</a> Free Option for plus version at cost</td>
</tr>
<tr>
<td>Quicktime</td>
<td>All Windows All Macintosh</td>
<td>MOV MPEG-1 MPEG-3 MPEG-4 AVI, SMIL + others</td>
<td>Internet Explorer, Netscape, Mozilla</td>
<td><a href="http://www.apple.com/quicktime">www.apple.com/quicktime</a> Free Option for Pro version at cost</td>
</tr>
</tbody>
</table>
5.6 Digital Video Formats \[31\]

There are various high quality file formats that could be used to store videos. The appropriate choice is a compromise of quality versus file size. **Table 5.2** shows the main examples digital file format.

**Table 5.2: Digital Video Formats** (Menin, 2002, p82)

<table>
<thead>
<tr>
<th>Format</th>
<th>Advantage</th>
<th>Disadvantage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Uncompressed AVI</td>
<td>Perfect reproduction of digital video and sound (Microsoft)</td>
<td>Requires huge amounts of hard drive space – 12GBs per hour of footage.</td>
</tr>
<tr>
<td>Uncompressed MOV</td>
<td>Perfect reproduction digital video and sound (Apple)</td>
<td>Requires huge amounts of hard drive space – 12GBs per hour of footage.</td>
</tr>
<tr>
<td>MPEG-1</td>
<td>Small file size, compatible with most computer systems.</td>
<td>Poor quality reproduction in comparison to other formats.</td>
</tr>
<tr>
<td>MPEG-2</td>
<td>Very high quality reproduction, as used on DVD. Manageable file size at 9GBs for 1 hour at the highest bit-rate.</td>
<td>Although the bit-rate can be adjusted, MPEG-2 video quality does suffer as a result. It is designed for one purpose.</td>
</tr>
<tr>
<td>MPEG-4</td>
<td>Best all-round. Depending on bit-rate (150Kbps to 15000Kbps) it can produce almost uncompressed AVI quality at much smaller file sizes. Audio can also be compressed with little or no loss.</td>
<td>MPEG-4 is implemented only in a few solutions as Apple (QuickTime 6) and also the open source DivX. Both are very high quality and highly configurable.</td>
</tr>
</tbody>
</table>

All of the file formats shown in **Table 5.2** can be exported using standard editing packages, such as Adobe Premiere. There are no Licensing implications when using these file formats for storage and backup. The emerging standard is MPEG-4 as it proves to be the best all round format for real-time streaming because of the requirement for the low bandwidth and can be edited with Adobe Premiere as an AVI.

5.6.1 MPEG (Motion Picture Experts Group)

MPEG is not a video architecture, nor is it exactly a codec. It is a standard file format and set of compression algorithms which were (and continue to be) designed by the Motion Pictures Experts Group for audio and video compression. “MPEG-1 and MPEG-2
are widely used in the television industry (most digital TV is MPEG-2 at high to very high bit-rates)". (Mack,2002,p86)

The most current version in wide use today is MPEG 4 which is designed to degrade gracefully with varying traffic conditions.

Features of MPEG 4:

- As a standard which specifies a data model for video and audio compression, "MPEG is truly platform independent." (Mack,2002,p86). In fact, the MPEG 4 standard is expected to be applied to wireless video and handheld applications.
- Based in the QuickTime format, MPEG 4 goes beyond just compression to include animated sprites and interactivity.
- In addition to high quality video, MPEG 4 provides compressed CD quality audio.
- MPEG 4 compression algorithms work by selecting a series of key frames. Other frames are described according to how much they change between the key frames.

In Summary, true streaming media requires a specialized server such as RealServer, WindowsMedia Server or the QuickTime Streaming Server. Content is then delivered and viewed in real-time, whether it originates from a live event or an archived media file. Each of the mentioned architectures uses several tools to create the streamable files. The video architectures we have seen has both strengths and weaknesses as developers use it to show video on CD-ROM, DVD, and the Internet. QuickTime, has a long history as a multiplatform and multifunctional performer. It offers developers of CD/DVD and Internet applications full download, and progressive and real time streaming video. Windows Media (Windows Video) offers developers high quality full download or streaming images with or without a dedicated server. Real System is designed for web delivery for streaming video alone, offering the users a range of options for ensuring that the quality of the streaming video and audio remain constant in spite of network traffic. The main goal that Omantel’s implemented architecture is to produce an MPEG1 /MPEG2 quality videos.

The context of all three platforms is likely to change dramatically over the next few years as the latest MPEG compression formats and algorithms make the codecs that encode the video more standard. Last but not least, there is no one ‘best” architecture; choosing among them depends on the organization needs.

5.7 Content Streaming Methods: Web Server vs. Streaming Media Server

As audio and video streaming over the Internet has become more popular, two primary methods for streaming content have emerged. The first method is the Web server approach, in which a standard Web server is used to supply data to the client. The second method is the streaming media server approach, in which a specialized
streaming server delivers the data to the client. Both methods have advantages and disadvantages that will be discussed as well as the way each method works.

As previously mentioned, audio and video on the Web was primarily a download-and-play technology. The entire media file had to be downloaded first before it could play. “But because media files are usually very large and take a long time to download, the only content found on the Web was short 30-second clips—often even shorter. Even these files could take 20 minutes or longer to download.” (Mack, 2002, p92)

In the case of audio and video files that stream, it starts playing almost immediately, while the data is being sent, without having to wait for the whole file to download.

As a Comparative Analysis, “the differences between the Web server and streaming media server solutions translate into clear distinctions in both ease of implementation, ease of management, and quality of user experience.” (Menin, 2002, p12). For the remainder of this Section, the comparison will be between a generic Web server and Streaming media server.

5.7.1 Streaming with a Web Server: the Advantages

There is really only one major advantage to streaming with a Web server rather than with a streaming media server—utilizing existing infrastructure. Because the Web server approach uses only the standard Web server --that presumably already exists in the organization—no new software infrastructure need be installed or managed. The Streaming Media server approach, on the other hand, requires the content producer and/or the systems administration staff to install and manage additional server software. This can result in incremental training and staffing costs to learn and manage the Streaming Media Server environment.

On the other hand, it is important to note that “the increased load that Web server-based streaming puts on existing Web server infrastructure often results in the need for additional Web server hardware to service the client requests.” (Menin, 2002, p14). So, choosing Web server streaming over a dedicated streaming media server based on hardware cost alone usually does not result in any financial savings.

5.7.2 Streaming with a Streaming Media Server: the Advantages

Designed specifically for the task of delivering live or on-demand streaming media, a Streaming Media Server such as, for example, Windows Media server, offers many advantages over standard Web server including:
**More Efficient Network Throughput.** One of the main advantages of Streaming Media Server is the ability to use UDP, the specialized protocol optimized for live and on-demand streaming. “The TCP-based transfer used in Web server streaming is designed to repeatedly drive the slowest network link to packet loss.” (Stevens, 1993, p255) This wastes bandwidth by: (i) re-transmitting data to replace the lost packets; and (ii) under-utilizing the network link while re-estimating the throughput that can be supported by the network connection.

The UDP protocol allows higher bandwidth to be delivered to the client (resulting in better video quality), even when assuming the same network connectivity between server and client and the same level of congestion on the network. By having a specialized streaming media server, we know what rate the data is going to be consumed, based on headers of the compressed media file. “The Streaming Media Server sends data to the client only at this required bit-rate, and it doesn’t drive the bottleneck network link to loss. Thus the network throughput is better, resulting in better quality audio and video for the client.” (Mack, 2002, p31)

**Better Audio and Video Quality to the User.** Better network throughput is only one of several ways that a Streaming Server delivers superior audio and video quality for users. Here are two more examples:

- Because the Streaming Media Server and the Client Media Player remain in contact throughout the play interval, the server can dynamically respond to client feedback. “If network congestion is allowing only 22 Kbps of data to reach the client (instead of 58 Kbps), the server can decide to retain the audio quality but slightly lower the frame rate of the video stream so that it doesn't overshoot the available 22 Kbps.” (Mack, 2002, p32) This ability is not possible with the Web server approach. In a Web server scenario, with no feedback from the client and no ability to dynamically prioritize audio over video, the audio/video delivered by a Web server would be stop-and-go at the client, causing “re-buffering” delays”[20]. In contrast, the Streaming Media Server provides a continuous, smooth stream with barely perceptible changes in video frame rate during periods of network congestion.

- Streaming with a Streaming Media Server takes advantage of UDP's inherent higher priority over HTTP traffic to give the streaming audio and video data higher priority than file and Web page transfers. This increases the likelihood of uninterrupted viewing.

**Support for Advanced Features.** “The Streaming Media Server approach supports such advanced features as detailed reporting of streams played, VCR
controls (seek, fast-forward, rewind), live video delivery, and delivery of multiple streams to the client.”(Mack,2002,p420) With Web server streaming such features, if they are even possible, are difficult to implement and inefficient to support.

- **Cost Effective Scalability to Large Number of Users.**
  In the early days of streaming media, deployments often needed to serve only a small number of users simultaneously, making Web server streaming an adequate solution. But as delivery of audio and video has increased, sites often serve hundreds or thousands of simultaneous users. In these situations, two key capabilities of the Streaming Media Server provide increasing advantages over a Web server:

  - **Specialization.** In the Web server approach, the Web server is used to deliver the media files to the client. Web servers, however, are optimized for delivering lots of small HTML files, not large media files. With high volumes of file requests, “a Streaming Media Server greatly improves performance by optimizing how media files are read from the disk, buffered in main memory, and streamed onto the network.”(Menin,2002, p12) A Streaming Media Server can easily improve scalability than a Web server.

  - **Multicast Support.** One way to get live or stored audio and video to large audiences with minimum network congestion is to use multicast networking technology. Multicast allows a single media stream to be played simultaneously by multiple clients, drastically reducing bandwidth use. Only specially designed streaming media servers, such as a Windows Media server, have this capability.

- **Protection of Content Copyright.**
  Because Web server streaming creates a local cached copy of every media file played, there is no way to prevent end users from copying the files to a personal directory for later viewing. For Omantel’s internal use, this is not desired because of the copyright issues & the confidentiality of the videos. With a Streaming Media Server, users can only stream data and are prevented from downloading the file directly to their hard disk. As data packets are received over the network, they are delivered directly to the client application with no easy way for the end user to intervene and make a copy.

- **Multiple Delivery Options.**
  “With a Streaming Media Server, the media may be streamed with the optimal UDP or Multicast protocols when possible, and streamed with the TCP protocol
when necessary.” (Mack, 2002, p36) This enables corporate users to view Internet content without compromising firewall security and ensures that all users on all networks can access all streaming media content. The Streaming Media Server such as Windows Media server implements its own version of the HTTP protocol to enable streaming through a firewall or proxy server.

In summary, most Streaming Media Servers support four different protocol configurations, each offering specific benefits.

1. **UDP** – As detailed in the Streaming Media Server section, UDP provides the most efficient network throughput and can have a very positive impact on the user (player) experience. The only downside to UDP is that many network administrators close their firewalls to UDP traffic, limiting the potential audience of UDP-based streams.

2. **TCP** – As detailed in the Web server section, TCP provides an adequate, though not necessarily efficient, protocol for delivering streaming media content from a server to a client. As customers often open the TCP ports in their firewalls, Streaming Media Server can use the TCP protocol enable streaming media to flow through these firewalls, which often block UDP traffic. TCP is not suitable for real-time applications.

3. **HTTP + TCP** – The Streaming Media Server can also support HTTP-based control commands along with TCP-based data delivery. This combination has the benefit of working with all firewalls that let Web traffic through (port 80) and provides much more control (fast forward, rewind, etc) than a standard Web server, but also adds some overhead to the raw TCP stream that decreases scalability.

4. **Multicast** – The Streaming Media Server can also support IP Multicast protocols to allow very efficient delivery of streaming content to large numbers of users. Multicast enables hundreds or thousands of users to play a single stream, but will only work on networks with Multicast-enabled routers. Multicast is becoming prevalent on corporate networks, but is still very rare on the Internet. UDP & IP (with Multicasting is more suitable for real-time applications.

The Streaming Media Server such as Windows Media server will automatically switch to the appropriate protocol so no client-side configuration is necessary. “The server will initially attempt to transmit files using the optimal UDP or Multicast protocols. If unable, the server will then attempt to send first via the raw TCP protocol, then via TCP with HTTP-based control.” (Mack, 2002, p345)
5.7.3 Conclusion

This section has evaluated the two primary methods for streaming media content to users. The first, the Web server approach, uses a standard Web server and the associated HTTP and TCP protocols to request and deliver the content for the client. The second approach uses a streaming media server specialized to the audio/video-streaming task. The specialization takes many forms, including optimized routines for reading the huge media files from disk, the flexibility to choose any of UDP/TCP/Multicast protocols to deliver data, and the option to exploit continuous contact between client and server to dynamically improve content delivery to the client.

The primary advantage of the Web server approach is that it requires less software component to learn and manage. This method can be an effective first step in developing a streaming solution. Although, this kind of Streaming which is static is often the cheapest way to deliver content on a small scale. Unfortunately, this method cannot be used for live streaming and does not allow for the advanced features of true streaming such as multiple bitrate encoding. In addition, this method enhances the likelihood for time-outs (‘buffering’) and cannot deliver the same amount of simultaneous player connections as true streaming.

In other words, without the specialized server application residing on a web server, media can be delivered by using progressive download. This is sometimes referred to as HTTP streaming because the media is delivered from a basic HTTP server - any server set up to deliver web-pages as well. "HTTP serving is designed to be an error-free delivery process using error correction mechanisms that will retransmit lost data. This process is not conducive to streaming media because it interrupts the playback of such files on the client's computer. With many simultaneous clients, this can also drain the resources of the server.” (Mack,2002,p348)

Longer programs are more conducive to true streaming, since then they are not saved on the client’s computer. There will be a short delay on the client side as the streaming server buffers the content. On steady network connections, no further interruptions will occur during the delivery.

Therefore, to enable live streaming over Omantel networks to the end-users’ desktops and to gain the full functionality of streaming static files, a streaming media server is required. For static files, these should be hosted from a specialized streaming server connected to the internet. This server is usually a standard server hardware but with the necessary streaming server softwares installed. The most popular servers for this are, as mentioned earlier, the QuickTime, RealMedia, and WindowsMedia servers. Placing a static streaming file on a streaming server is the same process as putting a web page on a web server. A single streaming server can be used for both live and static file delivery.
Furthermore, the live stream delivery process is quite different. In this process the encoding software delivers a continuous stream from an encoding computer to the streaming server. This encoding computer will typically have a sound and video card installed and will encode these inputs and deliver this in a continuous stream to the streaming server. Similarly with the static file streaming, the media player connects to the content via the streaming server and hence true streaming is realized. (A more detailed description of the process will be explored in section 6)

In conclusion, the streaming media server approach, has these advantages:

- More efficient use of the network bandwidth.
- Better audio and video quality to the user.
- Advanced features like detailed reporting and multi-stream multimedia content.
- Supports large number of users.
- Multicast capabilities.
- Handles Live Streaming
- Multiple delivery options.
- Content control & copyright protection.

This section concludes that, while Web server streaming can be an effective solution, a streaming server is more efficient and flexible and provides a better user experience. The tradeoffs clearly indicate that, for virtually all providers of streaming media content, the streaming media server approach is the superior solution for OmanTel.
6. End-to-end proposed solution for best delivery of high-Bandwidth applications over IP networks.

6.1 Pre-production Planning

Before embarking on the implementation of the proposed goal, a fundamental structured Plan/framework has to be put in place that will clearly identify the physical and procedural processes required for the implementation of streaming media service. This framework will assist in reliable, consistence, scalable, less packet-loss delivery of high quality live/pre-recorded videos over Omantel IP-based networks.

Creating any video project begins with a vision inspired by the perceived need or desire. In order to organize what is needed to be communicated, it is essential to prepare some type of a project definition and include project specifications, such as minimum system requirements needed by viewers of the movie. The project definition will be the master plan and will help in the actual creation and preparation of the movie’s elements. “A storyboard is highly recommended to keep the story organized. And a script is needed to include with the video as closed captioning. This provides complete accessibility by the audience.” (Mack, 2002, p105) Once the script and storyboard are completed, only then it is time to assemble the hardware that is needed for production and to begin shooting.

This section will present an integrated framework for media production in which live as well as on-demand content will be authored and generated using a range of tools and processes.

6.2 Media production process: The Setup

From the time a video is shot until the end user views the stream, the bits and bytes forming the streaming media are subjected to several different pieces of hardware and software. It is transformed from a moving picture on one end into a series of ones and zeros. The information is processed by several software programs and transferred through the network to the end users computer in the form of streaming video.

The following figure shows a simple block diagram of the theory.
6.2.1 Creating Content

The Actual Content Creation will start with the following step:

1. **Recording equipment** for Video & audio (VCR, Camcorder).

   Streaming media begins with recorded content. As with all things, the better the recording equipment, the better the end results so it is important to start the production process with the highest-quality video and audio that can be offered.

   Recording has to be planned carefully because this will save hours and hours of editing later. Digital camcorders will even allow to record digitized media. Although this is somewhat convenient because digitizing will already be done, it will allow more control and will end up with superior results by recording the audio and/or video and digitizing that recording. This is because the format of the media recorded by digitizing equipment (digital camcorder) may not necessarily be optimal and further manipulation of the digitized media may result in unacceptable degradation.

   This process starts with the output of the VCR/ Camcorder fed into video and audio capture cards located in the encoding computer. This could be done by following the video and Audio cards’ instructions. If a digital camera is used, a FireWire capture card will be needed. (See Figure 6.2)

   Video content can be created with a camcorder or VCR. The type of film and the filming technique will greatly affect the clarity of the resulting video stream.
The basic equipment required for production includes:

- The script, as explained earlier.
- A video camera (digital or analog);
- A Camera stand (Tripod);
- Lighting equipment (if necessary);
- Additional sound equipment, such as an external microphone that can be plugged into the video camera;
- Enough videotapes for recording.

To ensure Best quality for streaming clips, it is essential to keep in mind that Quality starts at the source, therefore, a high-quality video and audio input for the Producer/Encoder is required to create high-quality streaming clips.

![Figure 6.2: Camcorder/ VCR connectivity with the Encoding Computer](image)

### 6.2.2 Managing and distributing content

**1. The Encoding Computer**

Video & Audio Capturing and encoding for streaming are accomplished on a dedicated workstation which is called the Encoding Computer.
The Encoding computer is used to capture, edit and encode live audio or video signals into a codec which is then broadcasted to the streaming server; it will have the capabilities of streaming through a network.

The Encoding computer, as previously mentioned, has a video and audio capture card installed. There are several brands of commercially available Video Capture Cards (VCCs). Some cards allow multiple inputs to the single card. There must be enough capture ports to accommodate all the streams that are expected to be captured simultaneously. It should also have a video editing program and a broadcasting software. The input to the capture process can be analog or digital, composite or component, over firewire. It all depends on the allocated budget and choice of capture cards and systems. The digitizing process depends on the video capture software program used. There is a wide variety of cheap or free software available with the ability to capture. Also, many cameras include their own free capture software. Capture software usually has the ability to capture to AVI or MOV. These formats can then be converted to a streaming format using the target player's encoding software.

“Encoding has 2 purposes, Compression for Delivery and saving in a suitable video file format to be played back by the chosen player” (Mack, 2002, p38). As mentioned earlier, different multimedia architectures offer different features and options, and store data in different file formats. QuickTime, RealVideo, and Windows Media are examples of streaming media architectures (refer to section 5). In addition, encoding a video depends on 2 things: What media player will be used (eg. Windows media, RealOne media etc) and What Network connection speed will be available. There are systems today capable of encoding four to six simultaneous streams. Some Encoding solutions such as RealNetworks (SureStream) and Windows Media (Intelligent Streaming), as mentioned earlier, offers the opportunity to encode just once for multiple connection speeds (multi bit rate). In other words, they are capable of packaging different bit rates into one stream for more efficient delivery. This is as an alternative to producing individual encodings for each speed and letting the user choose.

a. Video and Audio Editing Software

The following is a compiled review (in alphabetical order) of some of the more popular free or relatively cheap digital video editing packages.

- **Adobe Premiere** has long been the industry standard for the Macintosh and PC video market, due to its combination of cost and capabilities. Premiere can take beginners through the steps but there is an advanced option for more versatility. Premiere also includes an audio mixer and 20 audio filters to help process audio.
- **AIST MovieXone 1.0** is a free complete package for video production including tools for editing video, animation, audio. Transitions and effects are also included.

- **Microsoft Windows Media Encoder** is a free production tool that converts both live and pre-recorded audio, video, and computer screen images to Windows Media Format for live and on-demand delivery. It only has limited editing capability such as adjusting sound or colour levels and adding author details.

- **Pinnacle Studio Basic for RealVideo** is a free software download. It helps edit digital video files and export to RealMedia. It supports FireWire cards but limits the resolution to 320x240 pixels and a target audience of only two.

- **Pinnacle Studio DV 7** offers standard DV editing: trimming and arranging the clips, adding titles, scene transitions, music, narration, special effects and more. It comes with a FireWire Card and can export to RealVideo8 and Windows Media format.

- **Microsoft Producer for PowerPoint 2002** is a free software download. It has limited editing capabilities but it does simplify the process of making rich-media presentations.

- **Flickerfree Video Framer for RealVideo** allows digitizing and editing video using a multi-track timeline and transition effects. It can create basic RealVideo directly or export the video as an AVI.

- **ULead VideoStudio 6.0** is quite an intuitive interface that allows scrolling titles, 3D effects, filters, motion overlays and extensive audio options. It is good for beginners as it takes them through the steps but can be restrictive for the more advanced.

- **Vegas Video LE 3.0** by Sonic Foundry is a free, streamlined version of Vegas Video 3.0, Sonic Foundry’s professional digital video and audio editing system. It allows mixing audio and video using multiple tracks, perform fast edits, apply effects and transitions, and hear and see the results in real-time.

- **Sound Forge 6** by Sonic Foundry is perhaps the industry standard audio editor. It includes a set of audio processes, tools and effects for recording
and manipulating audio. There are many other free, shareware and low-cost
alternatives available on the web.

Generally speaking, for media production, one needs to learn the editing hardware
and software thoroughly, paying close attention to the manufacturers’
recommendations for producing high-quality media files.

b. Compression

To accommodate bandwidth limits, content should be compressed because only
compressed clips stream well. Compression is vital for video that is delivered via the
web. If the file size is too large, the image and/or sound quality will be very poor if
viewable at all. Uncompressed-clips is not a good streaming format because it
requires a lot of bandwidth for even a small clip. Compression therefore has to
reduce the file size of video and/or audio for more effective delivery out to the user.
Every streaming technology provides its own lossy compression schemes (Codecs).
So the video capture software compresses the video to a codec file such as an .avi,
.qt, or .mpeg file, depending on the program.

Finally, the compressed file is fed into encoding software. The encoder uses a Codec
to convert it from an AVI file/any other compressed format to a streaming format
(i.e., Windows Media .ASF, Real's .RM, Apple's .MOV). Codecs generally work by
removing unnecessary visual and audio data from the file to reduce the amount of
data sent.

c. Common Components of Compression

It's always important to know the target audience when setting up the basic
parameters for video Streaming Services. Regardless of the technology chosen to
deliver the media, it's important to make decisions about the following parameters
prior to encoding:

- Video Bit Rate
- Frame Size
- Frame Rate

Video Bit Rate

The "bit-rate" numbers (ie: 28K, 56K, 100K, etc) indicate "the speed at which the
video data is streamed from the server to your computer." Bit rates determine the
quality of audio and video played over the Network. So if audio or video are encoded
at a higher bit rate, the video will end up with better resolution, smoother playback, greater fidelity, providing the viewer/listener with a better overall experience. So the higher the bit-rate or speed, the better the picture and sound quality.

Media can be encoded at single or multiple bit rates starting from 5 Kbps to more than 1 Mbps. determining the right bit rate, depends on the Target audience. “To compensate for congested networks --especially relevant for true streaming-- the target bit rate is set lower than the ideal bit rate for the connection (e.g. 56kbps modems can only achieve 53kbps). Many encoding technicians will target a bit rate at 34 kbps. For ISDN connections that have a maximum bandwidth of 128k, set your target at 80kbps.”(Mack,2002,p418) (Refer to Table 4)

“When selecting the bit-rate for a certain video, it should either matches or is lower than the speed of the target audience connection.”(Mack,2002,p420)(example: 100K streaming video will not play through a 56K modem).

In Omantel case, the end-users have a fast network connection (100Mbps), so a higher bit-rate could be selected. However the rule-of-thumb is to select a lower bit-rate if the Audience experience intermittent playback.

Frame Size

During the encoding stage there is often the option to specify the frame (window) size of the video. This can also be done in the editing stage. The video frame size is the size of the display window in which the movie plays.

“The best size is highly dependent on data rate, frame rate, codec, source material and personal preference. These factors are all interrelated so experimentation is the best way to find the optimal setting for the project.”(Mack,202,p270)

Reducing the window size reduces the amount of data in the video and so the file size. In other words, “scaling down the frame size is one of the best ways to lower the bit rate requirements”(Mack,2002,p270). It is for this reason that video produced for slower connection speeds usually has a smaller window size than faster connections. “Many software packages offer default frame sizes when you choose different connection speeds.” (Mack,2002,p270)

Like frame rate, choosing an appropriate frame size for any movie has an important effect on video quality. All else being equal, the higher the image size at a given data rate, the lower the resulting image quality will be.
The following table demonstrates a very rough frame size guidelines (Mack, 2002, p270). The best way to find out the best setting is by experiment and certainly by testing results to make sure they will play on the minimum target.

Table 6.1

<table>
<thead>
<tr>
<th>Network Speed</th>
<th>Frame Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>Modem</td>
<td>160x120</td>
</tr>
<tr>
<td>ISDN</td>
<td>192x144</td>
</tr>
<tr>
<td>Broadband</td>
<td>240x180 / 320x240</td>
</tr>
</tbody>
</table>

As shown above, larger frame sizes are generally expected at broadband rates, usually 240x180 or 320x240. In the latter case, if the client allows doubling of the image, it will be right to describe the streaming video as "full-screen." (Of course, 640x480 will not fully consume many computer screens set to 800x600 or higher).

As mentioned earlier, many software packages offer default frame sizes when choosing different connection speeds. Again, experimentation is critical to achieving the best results.

Frame Rate

Frame rates are another important parameter to choose carefully in order to cut down on the bit rate requirements. The video frame rate is how many frames are displayed each second. Higher frame rates produce smoother motion in the movie and create larger files.

"There are a few different factors that normally limit the frame rate and frame size. The most common limitation is the image quality." (Mack, 2002, p271). Larger frame sizes and higher frame rates require more data to maintain acceptable image quality if the data rate of the movie isn't high enough to accommodate the frame size and rate, the image quality will suffer.

"The other common limitation of frame size and rate is the Codec you are using and the CPU of the target machine." (Mack, 2002, p271). Some CPU-intensive Codecs can produce high quality images at high frame sizes and frame rates with reasonable data rates. However, these movies may require very high-end machines to play properly, so testing is critical.

The appropriate frame rates and sized guidelines should be available with the purchased Encoding software. Again, experimentation is the best way to determine the optimal frame rate for the project.
d. Presenting the content

Once capturing, editing and possibly encoding all the material is completed, it is important to think how the video clips will be presented to the audience. As mentioned earlier, there are major products that offer various options to combine video, sound and other resources. Clearly this offers many opportunities to integrate the moving image, not only with slides but discussion boards, quizzes and other web-based resources. These are often delivered via a virtual learning environment (VLE) [51] which often demonstrated using SMIL (Figure 6.3). In this Figure example, the video image is embedded in the left hand window with the slides on the right. Both video and slides are synchronized and both have interactive control buttons. Thus clicking back to slide 10, for example, will bring up the appropriate section of video, or fast forwarding the video will cause the slides to catch up. The user can also navigate using the menu of bookmarks on the left of the video image. This is also synchronized. In the bottom left corner are the communication tools (in this case List, Discussion Board, HyperMail) together with links to other, for example, departmental resources.

![Diagram of video window, presentation window, and link window]

**Figure 6.3:** The layout created by the SMIL code

Technically, this can be from quite simple clip links to quite complex (in design and technical terms) integrated layouts like the one above. These will require varying
degrees of technical skills. A summary of the main options is given in Table 6.2 (Mack, 2002, p585), with a guide to the technologies below.

Table 6.2: Proposed Options for displaying streaming video

<table>
<thead>
<tr>
<th>Content Presentation Options</th>
<th>Tools</th>
</tr>
</thead>
<tbody>
<tr>
<td>A clip and media player controls embedded within a web page</td>
<td>HTML or HTML generators (e.g. FrontPage, Dreamweaver), RealNetworks’ Producer, Windows Media Encoder (WM), Microsoft Producer (WM), Media Cleaner (QT)</td>
</tr>
<tr>
<td>A link to a pop-up clip from a web page (e.g. a video clip or an audio clip)</td>
<td>As above</td>
</tr>
<tr>
<td>A clip with synchronized slides (e.g. Microsoft PowerPoint with video and audio)</td>
<td>HTML, Microsoft Producer (WM), PresenterOne for RealOne (Real), SMIL (Real and QT), HTML + TIME (WM)</td>
</tr>
<tr>
<td>A clip with synchronized slides, hyperlinks and bookmarks (e.g. web addresses or email addresses)</td>
<td>As above</td>
</tr>
<tr>
<td>A clip with synchronized slides, hyperlinks, bookmarks and communication tools (e.g. discussion forums)</td>
<td>HTML/HTML+ TIME, Microsoft Producer with HTML (WM), SMIL 2.0 Code within RealOne Player (Real)</td>
</tr>
<tr>
<td>A clip with synchronized slides, hyperlinks, bookmarks, communication tools and shared software (e.g. a Microsoft Word document)</td>
<td>HTML with Windows Media and NetMeeting ActiveX plugin (WM)</td>
</tr>
</tbody>
</table>

Embedding clips with HTML

The video file could be played from a web page by creating a simple text hyperlink on the page using a text editor or an application like Microsoft FrontPage or Macromedia Dreamweaver. When the user clicks on the hyperlink, the appropriate player opens up as a separate application and the user can view the media file as it streams from the server. The very simplest option is streaming from a normal web server.
Synchronized slides and video

There are several options open to developers for producing more complex, integrated media. One of the most important is “SMIL (Synchronized Multimedia Integration Language) an HTML-like World Wide Web Consortium (W3C) specification markup language which enables simple authoring of multimedia presentations that integrate streaming audio and video with images, text or any other media type” (Mack, 2002, p.588). It does this by dividing the different media components into separate files and streams, then displaying them together on the user’s computer as if they were a single multimedia stream. For example, “SMIL is commonly used for delivering audio and video together with a slide presentation and perhaps links to handouts” (Mack, 2002, p.589). (Figure 5). Current desktop versions of QuickTime player and RealNetworks’ player support SMIL. “Microsoft has developed an alternative, SAMI (Synchronized Accessible Media Interchange), similar to SMIL, but supported only by Microsoft products, including Windows Media Player. “Internet Explorer 5.5 (or later) also supports a protocol called HTML+TIME based on the SMIL 2.0 specification.” (Mack, 2002, p.589). Both SMIL and SAMI are easy-to-learn HTML-like languages, and SMIL and SAMI presentations are often written using a simple text-editor.

SMIL has many advantages and creation possibilities (Mack, 2002, p.590), such as:

- A combination in real-time of many different types of file, such as video, audio, text, images and Macromedia Flash in real-time.

- The original content remains intact. There isn’t a need to edit and merge clips into a single streaming file. To make any changes it’s simply needed to edit the SMIL file.

- A complete control over how the presentation is displayed, i.e. its layout and the timing in which events occur, such as slide changes, audio track change or the start of an animation.

- Different files from different locations could be used. SMIL uses links, much in the same way as HTML, to locate clips or other such files.

- A major advantage is to be able to offer different presentations dependant on users. Also, it is possible to define bandwidth usage, e.g. the more bandwidth the user has, the more complex the presentation could be.

However, the following are other alternatives:
Microsoft Producer for PowerPoint 2002

Windows Media Player does not contain the functionality for displaying rich media from within itself. Instead, "Windows Media Video (WMV) files are embedded into a web page and controlled via HTML. This allows any web-based compatible media to be synchronized with the video/audio file of choice. A disadvantage of this approach is the complexity of creating such presentations." (Microsoft Corp, 1999, p163) To help the user in creating online presentations, Microsoft has released Microsoft Producer for PowerPoint 2002. This free application synchronizes the video media with Microsoft PowerPoint, HTML, images, audio and also other video sources. This is a very capable wizard-based package, although it’ll be required to have Microsoft Windows 2000 or XP and have Office XP installed.

Accordent’s PresenterOne Basic for RealOne

As RealNetworks launch the new Helix platform, RealPresenter is replaced by PresenterOne for RealOne. It is a software that enables creating streaming web-based presentations for the RealOne player. It is used for adding and synchronizing audio and video to Microsoft PowerPoint slides.

Using the wizard it is quite straightforward to add and synchronize audio and video to Microsoft PowerPoint slides. “To use PresenterOne your system will need the free Helix Producer Basic which converts live feeds or existing audio and video files into streaming RealMedia files.” [24]

When time comes for the implementation of this proposed streaming media services, a plan will be written which will clearly identify which software Omantel IT managers wish to use in order to present the streaming training/seminar content.

2. The Video Server (Archive Server)

The video server is the network equipment providing the storage for video program material, which can be requested by the users.

For on-demand content, the encoding server will have to send the compressed video clips to the Video server but for live event streaming, it should directly broadcast it to the Streaming server (if separate). In addition, archiving the material for viewing on demand should be considered when live-encoding.

As required, all Media could be centrally stored in a Video Server (May be an archive server or cluster of servers). “This server has to perform many functions, such as
admission control, request handling, data retrieval, guaranteed stream transmission, and stream encryption.” (Mack, 2002, p415)

The Video Server is required especially when there’s a big volume of video storage required. With a large video archive, there will be a need for a dedicated storage server.

3. Streaming

As previously mentioned, Streaming media can be streamed from either a Web server or a streaming media server, typically at several different speeds. Typically, the higher the speed, the better the quality of the streaming media.

a. Streaming with a Web server

Posting and Hosting
Deploying streaming media content with the Web server approach is actually only a small evolutionary step away from the download-and-play model. After compressing the file into a single “media file” for delivery over a specific network, it is then placed on a standard Web server. Next, a Web page containing the media file’s URL is created and placed on the same Web server. This Web page, when activated, launches the client-side player and downloads the media file. So far, the actions are identical to those in the download-and-play case. The difference lies in how the client functions.

Data Delivery
Unlike the download-and-play client, the streaming client starts playing the audio or video while it is downloading, after only a few seconds wait for buffering (the process of collecting the first part of a media file before playing). “This small backlog of information, or buffer, allows the media to continue playing uninterrupted even during periods of high network congestion. With this delivery method, the client retrieves data as fast as the Web server, network and client will allow.” (Mack, 2002, p31). Only certain media file formats support this type of "progressive playback".

As mentioned earlier, Web server streaming uses the Hyper Text Transport Protocol (HTTP), the standard Web protocol used by all Web servers and Web browsers (such as Microsoft Internet Explorer) for communication between the server and the client. HTTP operates on top of the Transmission Control Protocol (TCP), which handles all the data transfers. Optimized for non-real-time applications such as file transfer and remote log-in, TCP's goal is to maximize the data transfer rate while ensuring overall stability and high throughput of the entire network. To achieve this, TCP first sends data at a low data rate, and then gradually increases the rate until the destination
reports packet loss. TCP then assumes it has hit the bandwidth limit or network congestion, and returns to sending data at a low data rate, then gradually increases, repeating the process. TCP achieves reliable data transfer by re-transmitting lost packets. However, it cannot ensure that all resent packets will arrive at the client in time to be played in the media stream.

b. Streaming with a Streaming Media Server

Posting and Hosting
In the streaming media server approach, the initial steps are similar to the Web server approach, except that the compressed media file is produced and copied to a specialized streaming media server instead of a Web server. Then a Web page with a reference to the media file is placed on a Web server.

“This could be an independent computer or a built-in feature in the Encoding & Editing Computer” (Mack, 2002, p414). Either case, to have a streaming capability, a streaming server software with a Broadcast software should be installed in order to send the captured media to viewers.

Data Delivery
The rest of the streaming media server delivery process differs significantly from the Web server approach. The data is actively and intelligently sent to the client, meaning that “it delivers the content at the exact data rate associated with the compressed audio and video streams” (Mack, 2002, p264). The server and the client stay in close touch during the delivery process, and the streaming media server can respond to any feedback from the client.

As mentioned earlier in section 4, while streaming media servers can use the HTTP/TCP protocols used by Web servers, they can also use specialized protocols such as the User Datagram Protocol (UDP) to greatly improve the streaming experience. Unlike TCP, UDP is a fast, lightweight protocol without any re-transmission functionality. This makes UDP an ideal protocol for transmitting real-time audio and video data, which can tolerate some lost packets. UDP traffic gets higher priority than the TCP traffic on the Internet/IP networks. And instead of the blind retransmission scheme employed by TCP, streaming media servers such as “Microsoft's Windows Media Services use an intelligent retransmission scheme on top of UDP.” (Microsoft Corp, 1999, p86). UDP Resend feature ensures that the server only retransmits lost packets that can be sent to the client in time to get played.

It will be required to decide whether to create multi-streamable files or single streamable files. In either case, it is also important to decide the target speed(s).
If media streaming is intended from the web server, separate files have to be created for each speed; on the other hand, if media streaming is intended from the streaming media server, one file could be created which will include the capability to stream at multiple speeds.

“When creating streaming video, it is important to optimize the file for the speed of the connection of the viewer.” (Mack, 2002, p.269). For example, a staff viewing a streaming media file over a 2MB link would get the best results when that file is optimized for a 2MB delivery speed. (More details in section 5.2.3).

These two functions (production and serving streams) don’t have to reside on the same platform, however, as if money is a limited resource, one high-powered server should be able to adequately perform both functions. The following specifications should be considered for Media Streaming Server Technical Specifications. The following is based on different streaming Products specifications. The specifications also depend on the Chosen architecture requirements:

- Processor Speed (measured in MHz/GHz) – + 500MHz.
- Disk Storage size (Measured in Gigabytes - GB); A RAID system is recommended to be employed because allows smaller drives (e.g. 100GB) to be linked as one drive and mirrored for backup very quickly within the server. This is a huge cost saving when storage is concerned. Also RAID systems have a very fast drive access and throughput, a huge benefit on servers dealing with multiple tasks involving video and other media.
- Memory (measured in megabytes - MB or Gigabytes - GB) – (+256MB RAM is recommended) or + 32GB for enterprise Edition.
- Network Card (measured in Megabits per second - Mbps). A 100MB network- interface card is recommended.
- Video/graphics Card (measured in Megabits per second – Mbps). +32MB RAM is recommended. The graphics card can affect the quality of the captured video.

In Summary, Files made to be streamed over a single speed can be pulled from a normal web server but files created to be streamed over multiple speeds can only be streamed from a streaming media server. This is because the streaming media server is specifically designed to determine the speed of the connection and to stream the file at the optimum speed.
6.2.3 Content Delivery

1. Bandwidth

Effective bandwidth remains the most important determinant of the end user experience. Bandwidth and the capacity of the network are vital for effective video streaming. The streaming server must have enough bandwidth available to cope with the demands made on it or else staff will experience a very poor quality playback i.e. fragmented or jerky images, broken sound, video skipping and possibly total loss of connection.

Streaming media services over the Internet/corporate networks tends to be offered at a variety of bitrates to suit different standard bandwidth options. However, the nature of the internet as well as highly Utilized networks such as corporate networks means that the effective current bandwidth for a user can fluctuate by orders of magnitude during a long viewing experience.

To eliminate bandwidth issues, the following has to be taken into consideration:

- Streaming Server Bandwidth
- How many staff are likely to access the streaming services
- Target Audience Bandwidth – to ensure enough bandwidth is available.
- Server streams have to be encoded to suit the target audience.

As mentioned previously, the IP networks are traditionally based on ‘best effort’ protocols that do not make any promises about the quality of connections between its nodes. This best effort philosophy is one of the reasons it has scaled so well, but is a problem for delivery of streamed media that requires some assurance about quality over an extended period. Information delivered from a source server to an end-user can typically travel over a dozen or more routers; the effective bandwidth will be determined by the lowest effective bandwidth between two of those routers, which is subject to system failures outside the control of either the content provider or the user, as well as traffic congestion from the Network as a whole. The Internet protocols are designed to be dynamic, so that they can find alternative routes in the case of severe congestion or system failure, but these protocols are not instantaneous and in the meantime a streaming video or audio experience can severely degrade.

Streaming media system vendors, and the IETF, have tried to address this problem in a variety of ways. The first step was to develop protocols (such as rtsp, rtp, etc) that would sit on top of the standard internet protocols, and try to compensate for lower-level problems. Initially this was by buffering, in which a quantity of the stream is delivered to the player before streaming begins, which can compensate for short-term fluctuations in bandwidth. The next stage was for
proprietary servers to monitor the effective bandwidth on the connection between the user and the server, and lower and raise the quality of video and audio served accordingly (see Real's SureStream technology). In other words, it is possible to create streaming files which can be streamed over several different speeds (called "SureStream" in RealMedia and "Intelligent streaming" in Windows Media – Refer to section 5). This allows the same file to be viewed at different OmanTel offices. As mentioned in Section 5, files created to be streamed over multiple speeds can only be streamed from a streaming media server; whereas files created to be streamed over a single speed can be streamed from a Web server or a streaming media server. This is because the streaming media server is specifically designed to determine the speed of the connection to the streaming media file, and to stream the file at the optimum speed for the particular connection. Using this feature in the Streaming server clearly changes the user's viewing experience, but in a more graceful way (so that, hopefully, the stream doesn't just stop or break up).

Therefore and as mentioned in the earlier section, when creating the streaming media, it is important to optimize the media for the speed of the connection over which the media is being streamed. In other words, persons viewing a streaming media file over a 100mbps would see the best results if that file is optimized for a 100mbps speed. The reality to true streaming is that the bit rate of the movie must match the bandwidth of the connection or buffering will occur and playback will be interrupted. (Mack, 2002, p244)

Table 6.3 illustrates maximum streaming speeds for common network connections. To reach 28.8 Kbps modems, for example, a video should stream no more than 28 Kbps of data per second. It also illustrates the typical use of each bandwidth.

Table 6.3 (Mack, 2002, p799)

<table>
<thead>
<tr>
<th>Connection Type</th>
<th>Typical Use</th>
<th>Max Bandwidth /Second</th>
<th>Typical Video Bandwidth/ Bit rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>28.8kbps Modem</td>
<td>Quite old</td>
<td>28kbps</td>
<td>16-20kbps</td>
</tr>
<tr>
<td>56kbps Modem</td>
<td>Home users</td>
<td>56kbps</td>
<td>34-37kbps</td>
</tr>
<tr>
<td>ISDN</td>
<td>Small business</td>
<td>64-128kbps</td>
<td>45-80kbps</td>
</tr>
<tr>
<td>256kbps Xdsl/Cable</td>
<td>&quot;Broadband&quot;</td>
<td>256kbps</td>
<td>225kbps</td>
</tr>
<tr>
<td>384kbps xDSL/Cable</td>
<td>&quot;Broadband&quot;</td>
<td>384kbps</td>
<td>350kbps</td>
</tr>
<tr>
<td>512kbps xDSL/Cable</td>
<td>&quot;Broadband&quot;</td>
<td>576kbps</td>
<td>450-500kbps</td>
</tr>
<tr>
<td>10Mbps Ethernet (LAN)</td>
<td>On Campus / Business</td>
<td>10Mbps</td>
<td>9Mbps</td>
</tr>
<tr>
<td>100Mbps Ethernet</td>
<td></td>
<td>100Mbps</td>
<td>90Mbps</td>
</tr>
<tr>
<td>1000Mbps Ethernet</td>
<td></td>
<td>1000Mbps</td>
<td>950Mbps</td>
</tr>
</tbody>
</table>
In summary, to deliver time-based content to any IP-connected desktop, bandwidth needs to be available. Streaming should not consume all the audience’s connection bandwidth. They must always have bandwidth for network overhead, error correction, resending lost data, and so on. Otherwise, they may frequently pause while waiting for more data to arrive. The number of simultaneous streams wished to be served is a function of the total network capacity and the server capacity. Aggregate bandwidth adds up quickly, and infrastructure for a large network is costly. Depending on the size of the target audience and the amount of bandwidth they have available, there must be enough network capacity available to cope with demand because otherwise, with large target audience and not enough bandwidth, streaming server will be flooded and live broadcasts’ quality will be effected.

Omantel should have the facilities for on-demand content storage capacity and bandwidth to house and deliver the content to the users when access is desired.

With the Current Omantel main offices’ Architecture which is based on 30Mbps (ATM) WAN and 100mbps switched architecture (LAN), offering streaming services shouldn’t affect the current existing Network. This effectively creates a dedicated 100-mbps path from every machine back to the streaming server. With this architecture in place, the Streaming video packets intended for one machine will have no impact on other machines’ ability to access the web-server, email server or wide-area network. On the other hand, Omantel streaming services don’t have to create streaming files which can be streamed over several different speeds because all employees are using a 100mbps network-speed through out the 6 main locations. In conclusion, the current network architecture should enable Omantel to produce a reasonably quick and stable streaming services.

In conclusion, based on many case studies, the experience of streaming has shown that it is practicable to deliver multimedia broadcasts across local and wide area networks, providing the end user is connected to the network with a reasonably fast connection such as Fast Ethernet, DSL or cable modem. However, it’s not considered feasible to use a dial-up modem connection to view full motion video streamed broadcasts although it should be adequate for audio only or slide-show presentations.

This proposed Hypothesis/ Theory is based on research and yet to be implemented and only when implemented, Potential Network bottlenecks and problems with this proposed Plan will be diagnosed. Real testing will determine the Network reaction to streaming services.
2. Firewalls, NAT and Security

Firewalls could cause problems for all formats of live Streaming. Therefore, live connection must be setup on the outside of firewalls or through popping a hole in the firewall so a signal could be sent through. Each of the 6 major Omantel offices’ Networks are protected by Firewalls. Normal Web traffic is allowed, special provision may have to be made to allow access to the ports used to receive streaming video. The same applies when serving video to the Internet from inside a firewall.

In addition, the most common issue facing streaming users is NAT (Network Address Translation) devices. These devices can allow multiple machines to use one IP address. It is very important to ensure that NAT devices have automatic configuration options for streaming video/audio.

As mentioned in Section 5, it is possible to specify which protocol options to allow or deny in the Firewall and NAT, for example, it’ll be required to open UDP ports in the Firewalls; and UDP port-forwarding should be enabled through any NAT router. TCP is less efficient than UDP, but will work automatically through Firewalls and NAT routers without any special configuration. HTTP is the least efficient and will always work, but is subject to web proxy cache degradation. In Addition, If Multicasting is enabled, Firewalls must be specially configured to allow multicast traffic.

To secure the proposed system, Omantel streaming services will include password protection, audience size restrictions, and referring URL or IP address blocking.

6.2.4 Viewing Content

1. Playback Software

The main formats in streaming video are currently RealNetworks. RealMedia, Microsoft’s Windows Media, and Apple’s QuickTime. Each of these has different advantages and disadvantages which was discussed earlier in section 5.

As discussed earlier, RealNetworks, Microsoft and Apple use their own proprietary format for encoding and serving media. This situation is the same for the media players. In an ideal world viewing any content is expected on any player. Content generated by one supplier’s encoder is generally only viewable through that
supplier’s player. (However, RealNetworks. new Helix platform promises that any of the three main formats can be streamed from their server.) There may also be a version incompatibility between players and content. For example, content encoded with the latest version of a supplier’s encoder may turn out to be not viewable by earlier versions of the player. In short, a choice has to be made at an early stage as to what player the end-user is intended to use and accordingly develop the compatible material for.

As illustrated in Section 5, the choice of player can have a lot of impact on the experience that the end-users achieve when viewing material. Each player has benefits over the other (refer to Table 5.1). Unfortunately there is no best option as it depends on costs, the material itself, how the end-users are connected and how the material is integrated. (Mack, 2002, p88)

The Seminar/Training Videos could be viewed through particular links within the Intranet webpage. End-users should be provided access to a video server (if exists) and browse through a list of archived video-clips.

2. Hardware

Omantel employees are typically provided with Powerful PCs running Microsoft Windows 2000/ XP with the following Specs:

Pentium 4 CPU 2.40 GHz
512 MB RAM Memory
48x CD ROM
40 GB Hard Disk

6.3 Conclusion

For Omantel streaming services initial implementation Phase, 2 scenarios are recommended.

- **Scenario 1:**

  Obtaining 2 Encoding Machines with streaming Capabilities. One will be used for Live-content Encoding and streaming, and the other will be used for on-demand content Editing, Encoding and streaming. In addition, the on-demand content Encoding machine will be used for video storage. (Figure 6.4) If the second dedicated encoding server couldn’t keep up with the demand, a dedicated video
server will be required based on the convenience and the volume of videos that would be produced.

**Scenario 2:**

Obtaining 2 Encoding Machines with broadcasting capabilities and 1 Streaming Server. One Encoding machine will be used for Live-content Encoding, and the other which will include an Editing software will be used for on-demand content Editing and Encoding. The on-demand content Encoding machine will also be used for video storage. (Figure 6.5) If the second dedicated encoding server couldn’t keep up with the demand, a dedicated video server will be required based on the convenience and the volume of videos that would be produced.

**Figure 6.4**: Recommended Scenario 1 Demonstration

**Figure 6.5**: If the second dedicated encoding server couldn’t keep up with the demand, a dedicated video server will be required based on the convenience and the volume of videos that would be produced.
For both scenarios, a Webpage will be developed with references to on-demand content as well as the live content. A control mechanism will be implemented which will only allow authenticated domain users access to Media Streaming Services. Staff will only be allowed accessing the on-demand content services. Whereas top executives, General Managers, Managers, Section Heads, advisors and Specialists will be able to access live events. This Mechanism is essential to prevent/eliminate Network congestion and bottlenecks problems.

The following figure demonstrates the webpage that will be developed to enable Omantel Staff to access the streaming Media services.
Omantel media streaming services support administrators have to set the servers to stream content at specific Streaming Rates, depending on the network speed (Table 6.3). Omantel's streaming video services should be delivered via a Multicast stream to all recipients. This means that for the live stream, the number of recipients is immaterial; the stream is fed onto the network at (eg.500 kbps) regardless of the number of clients receiving the stream. The on-demand content, however, is a unicast stream. Each Staff requests to view a certain video delivered from a stored file.

Omantel will need to have properly trained staff to manage and support the Streaming service. The support staff should be trained in both content creation and server administration. Their skills required for encoding, managing, and distributing must be available to support Omantel Media streaming services.
7. IP Multicast

An approach to the problem of congestion is to try to reduce the traffic generated by requests for streaming media. This is done through the implementation of IP multicasting which supports a separate stream for each user at the server.

This section provides a simplified introduction to IP Multicasting. An Overview of the basics of implementing IP Multicast for video streaming on Omantel networks will be illustrated briefly. However, Cisco provides a straightforward implementation Guide for IP Multicast.

IP multicast is an ideal alternative to IP broadcast and unicast transmission for Video. Consider a specialized Video Streaming server that transmits packets to 100 employees. Unicasting would require the periodic transmission of 100 identical packets, with many packets traversing the same links. Using broadcasting, all network end stations would receive the packets, regardless of the small recipient group (Figure 7.1). But multicasting sends the same information only once and only to the intended recipients (Figure 7.2). The resulting bandwidth savings and scalability inherent in multicast provide a major benefit to network Administrators.

**Figure 7.1:** Network content channel with Broadcasting.
**Figure 7.2:** Network content channel with multicasting.
As previously explained, the fundamental principle behind IP multicast is efficient handling of group communications within an IP network. This technology extends a single line of communication (normally called the “data stream”) over a network, allowing those users who wish to participate in the data stream activity to do so (if they belong to a specific multicast group). In this configuration, each connection does not require new bandwidth to be consumed; rather, each user connects to the single data stream. The net result is a far more efficient use of network bandwidth for group communications; for example, a group of users wishing to view a multimedia stream will only consume bandwidth equal to one user in a unicast environment.

Multicast packets are UDP. The multicast environment consists of senders and receivers. Any host can send to a group, but only members of the group will receive the message. Cisco Systems has built multicast support into their IOS’s for quite awhile now. There is nothing that we need to do to our present routers, except to turn them on. “The main point to remember is that traditional IP is concerned with where a packet is going. In IP multicast the emphasis is on where the packet came from.” (Menin, 2002, p228)

There are some basic terms used in IP Multicast, especially in Cisco System’s implementation of multicast:

Group- a number of hosts that form a “membership” to receive an IP stream. Membership is dynamic; hosts can join or leave at anytime. There is no restriction to the location or number of hosts. A multicast address is chosen for the receivers in a multicast group. Senders that use that address as the destination address of a datagram reach all members of a group. \(^{52}\)

Source- a router that has a video server on a directly connected subnet, or the server itself. Rendezvous Point- in PIM sparse mode, a router on the network where senders and receivers meet. The Rendezvous Point router keeps a table of available sources and directs requests for services to those sources. It is not important where the RP is physically located, only that it has a reliable connection. \(^{52}\)

A multicast address is chosen for the receivers in a multicast group. Senders (sources) use that address as the destination address of a datagram to reach all members of a group.

Multicast uses Class D addresses: 224.0.0.0 to 239.255.255.255

Protocols Described below are some of the protocols used in Cisco’s implementation of IP multicast. There are many others that are used to cross autonomous systems and on the Internet backbone.

To exploit this technology, these new set of protocols must be implemented in parallel with existing IP unicast networks. These protocols (described below) are the ones that should apply directly to Omantel’s Video streaming Services implementation.
- **IGMP** - Internet Group Management Protocol, used between hosts on a LAN and the routers on that LAN to track the multicast groups of which hosts are members. IGMP is used for hosts to tell directly connected routers about group membership. The information is stored in tables on the router. The process begins when a host sends an IGMP Report saying that it is joining a group. After the initial Join message the router will send periodic queries to the subnet to see if any hosts are active. If a host leaves it will send a Leave message to the router. The router will query to see if there are any other active hosts on the subnet. If it does not receive replies to its queries then the group will time out. [52]

- **PIM** - Protocol-Independent Multicast, versions 1 and 2. Version two is the preferred version. If a router detects version 1 it will make all devices use version 1 until version 1 can be replaced. PIM is used between routers so that they can track which multicast packets to forward to each other and to their directly connected LANs. [52]

Multicast routing protocols also come in two flavors: dense mode and sparse mode. "Dense mode protocols use a source tree and are ideal for LANs where group members are distributed densely throughout the network. Sparse mode protocols use a shared tree, thus offering a more scalable solution for building distribution trees when group members are distributed sparsely on a WAN." [52]

The most popular multicast routing protocol is Protocol Independent Multicast-Sparse Mode (PIM-SM), which is being deployed widely across IP WANs. "PIM-SM uses an explicit "join" model that blocks multicast traffic forwarding unless it is requested; routers must explicitly join a group to receive multicast traffic for that group." [311] This prevents unnecessary flooding of multicast traffic throughout the network, resulting in greater bandwidth efficiency and multicast scalability.

"IM-SM, a shared tree protocol, also "allows switch-over from the rendezvous point-based tree (RPT) to the SPT model if a performance threshold is violated, offering the best of both worlds." [52] A router will change from the RPT to the SPT if the multicast traffic it receives exceeds a predefined latency threshold, providing a quality-of-service mechanism for multicast.

For most local segments operating on switched or shared media, multicast traffic flows with no administrator intervention. However, when IP multicast needs to flow between different network segments, three different objectives need to be met at the network layer of the OSI protocol model.

- The network layer must contain an address of a group of recipients rather than simply one recipient.
- There must be a method for client computers to join the group receiving IP multicast packets. This is accomplished by using the IGMP protocol, which
issues simple commands that regulate the management of IP multicast groups.

The network infrastructure must have a way to communicate IP multicast routing decisions and trees. Ideally this architecture minimizes the amount of IP multicast packets on each branch, meaning, there is only one copy of multicast data on each branch of the IP multicast routing tree. There are multiple ways of creating this tree. There are currently several standards for IP multicast routing. They are Distance Vector Multicast Routing Protocol [53], Multicast Open Shortest Path First [54], and Protocol Independent Multicast protocol. In short, with multicast, a single virtual connection uses no more bandwidth for hundreds of users than it does for a single user.

Furthermore, the proposed video streaming services for Omantel IP networks would support IP unicast for on-demand Content and IP multicast standards for live streaming. Streaming can only run in one of these modes at a time. The Video Streaming administrator controls the particular mode. Both implementations are UDP (connectionless), eliminating TCP ACK (acknowledgement) packets, resulting in better throughput for streaming applications in a high-latency network. If streaming in unicast, a unique client/server connection is required for each connection to the Video Streaming Server. “The net effect is for every n client stations viewing media, the network bandwidth consumed equals n times the bandwidth of a single connection.” (Menin, 2002, p188)

A far more efficient use of bandwidth is that which makes a single stream of media available for every client to tap without requiring a unique connection. This methodology is implemented in multicast routed network configuration.

In addition, streaming multiple unicast sessions is more taxing on the CPU, as the appliance is then responsible for managing the distribution to each client connection. As a result, it becomes necessary to limit the number of simultaneous users under a unicast scheme.

In the unicast model, each user requires a unique connection, resulting in three copies of the same data being streamed over the network. The IP multicast model demonstrates how the data stream can be viewed by an unlimited number of users, consuming only the bandwidth required for the single multimedia stream.
Conclusion

When installing and configuring the Video Streaming Services, what applications are supported will determine whether to choose unicast or multicast.
Unicast is designed to support transmitting data to a single destination to one PC on the network. It is also the method of choice when transmitting secure data, data over a public network such as the Internet, or when viewing appliances through routers that are not configured to enable multicast traffic. However, if it’s required to send data to multiple recipients, unicast can be inefficient because identical data streams are sent to each recipient. In addition, streaming multiple unicast sessions add more load on the CPU especially when unicasting simultaneous streams. Therefore, as mentioned earlier it’ll be important to limit the number of simultaneous users under a unicast scheme. This will be done through testing the service after implementation and identify how much unicasts can the streaming server handle simultaneously. If results show that it can handle for example 10 unicasts simultaneously then a mechanism will be thought of which will only allow 10 streams simultaneously.

On the other hand, multicast is an efficient method for distributing data by transmitting a single data stream to multiple recipients. This method will work especially well for Omantel Live Video Streaming Services across the enterprise networks. Multicasting must be implemented carefully as it has the ability to overwhelm networks. As mentioned earlier, PIM Dense Mode floods a network before pruning itself back. On the other hand, PIM Sparse Mode solves this problem.
8. Conclusion and Future Implementation Plan

Video streaming is a technology that can extend benefits to applications that are already in place in most enterprises. As for other IT expenditure, it is hard to place a monetary value on return on investment. But there are other tangible advantages that video streaming contributes to effective organizational management. When deployed for corporate communications and inter-departmental collaboration, video streaming can improve the knowledge-base of staff. This in turn promotes better working relations and can facilitate greater commitment between colleagues. When applied to training, education, product launches or investor announcements, video streaming facilitates the creation of a central archive of information that can be accessed at a later date. It can reduce the amount of employee travel considerably. Streaming can help companies get their messages out to larger audiences, or a greater number of a narrowly defined group, more quickly and cost effectively.

In the enterprise, networking infrastructure is less of an inhibitor to the adoption of video streaming than is perceived. Most organizations such as Omantel already have high-speed LANs that can support streaming. Multicast stream is recommended to be implemented for live-content streaming. As Omantel gets involved in the implementation and configuration of IP multicast, its deployment should be well tested. Omantel’s networks must be evaluated to see if they can handle the demands of IP Multicast. Multicasting should be turned on all routers in the network. Multicasting as well as other streaming protocols should be enabled in Firewalls and NATs.

Omantel’s on-demand content streaming, however, will be based on unicast-delivery method.

The recommended technology which will be used in streaming services is a streaming server. The server hardware should meet certain specifications. The software part of the server which will include the right codec and the right editing tools will require further investigation. Testing and analyzing different streaming products will be required to identify the best ultimate streaming solution that will best suit Omantel’s streaming needs.

Omantel should offer different means to push content closer to the end user, reducing the number of hops in the network path and therefore reducing latency and the chance of dropped packets. With the combination of a powerful streaming server and better Codecs that use less bandwidth and improved delivery networks, reliable streaming media is set to take off.

Accessibility to Omantel’s streaming content services will be through a special web-site. This web-site will be restricted to Intranet users only and will be blocked for external users, so it will be impossible to access it from home. And only Omantel staff will be authorized to access
the content of this site. Each streaming service (Live and on-Demand content) will have its own specific members/ group identified.

A Further detailed Plan will be developed which will include the optimal streaming server solution (Hardware + Software) that needs to be purchased. In addition, a feasibility study will have to be done analyzing all vendors streaming proposed solutions.

Furthermore, the proposed Video-delivery solution over Omantel IP-networks is planned to be implemented in 2 Phases.

Phase 1:

- The Implementation of Live streaming services which will require a dedicated Streaming server with encoding capabilities.
- The capturing equipments have to be purchased in advance.
- Enabling IP Multicasting through out the organization’s networks.
- Enabling the end-user with the playback software based on the chosen streaming Architecture (eg. Windows Media).

Testing will be required to determine network performance with enabled streaming services. Any problems that might be encountered during phase 1 should be eliminated and solutions should be identified before moving into phase 2.

Phase 2:

- The implementation of on-demand content streaming services will either use the existing streaming server or will require a second dedicated on-demand streaming server with encoding and editing capabilities. The choice of whether to purchase a second streaming server will be based on the performance of the existing streaming server. Either case, this server will have to have a large hard-drive space for video-storage. Else, a dedicated Video server will have to be purchased especially for future larger storage requirement. Other factors that will enforce obtaining a video server is that the streaming server is also responsible for content encoding which is known to be a CPU-intensive operation. This as well as the great demand on the streaming server might cause the streaming server’s performance to be affected or the whole server might go down. That’s why it’s highly recommended to study the streaming products very carefully.
- Enabling the end-user with the playback software based on the chosen streaming Architecture (eg. Windows Media).

Phase 2 should be implemented gradually (i.e department by department). The roll-out should initially start serving a particular department. This will help determine the kind of problems IT managers might face and have them think of alternate solutions to solve the problems. After
solving the encountered problems, then the implementation should cover another department and so on and so forth.

As part of Risk Management, a Streaming server back-up should be available.

Furthermore, the rapid growth in Omantel employees and Omantel Offices will require extending the streaming services as well as other services to the new offices. This will obviously indicate more computer network traffic and an increase in demand for streaming services. This indicates that Network performance is expected to suffer.

To solve this problem and guarantee end to end bandwidth efficiency, QoS Mechanisms have to be considered. As mentioned at the beginning of this Thesis, QoS protocols and Mechanisms, when implemented, provide predictable service during periods of congestion. It is the periods of congestion that are the target of QoS.

To initially improve Omantel's Private Networks' Quality Of Service, it would be required to overprovision the networks’ capacity and upgrade to the most efficient networking equipment. For Omantel's private WAN links which are based on ATM which already has built-in traffic engineering parameters to ensure QoS, however, this has to be further investigated. Another solution which could be implemented in Phase 1 of the Plan, which is classifying real-time traffic separately from non-real-time traffic and giving real-time traffic the highest priority treatment to ensure that real-time traffic sees minimum delays and gets a higher percentage of the Bandwidth. Whereas, Non-real-time traffic will tend to fill any remaining bandwidth.

The Plan is to continue exploring the mentioned QoS mechanisms and protocols and ways to apply them to the existing Omantel IP networks.

In summary, the focus of QoS is based on the implementation issues, the performance parameters have to be experimentally analyzed and that could lead to the further requirements of the existing solution. Therefore, experimentation is the best way to determine the optimal end-to-end Video streaming solution for Omantel.
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