Control of Streaming Multimedia Data in Remote Labs

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Abstract: - Over the past few years the Semantic Web has grown into a useful tool in Web-based learning and teaching. Many believe that if the information on the Web were harnessed and processed into useful knowledge, its benefits to the field of education would be unlimited. This seeks to enhance the effectiveness of the Web in teaching and research technology by creating an educational knowledge base, an associated digital library and web-accessible software for its creation and maintenance. Also, this paper discusses the initial components of an advanced software system Streaming Multimedia Data that is a major outcome of this research. The system, called the Advanced Knowledge Acquisition and Dissemination System (AKADS), uses Semantic Web technologies and a hybrid set of software components. Knowledge based components are under development to make this knowledge available to teachers, students, and researchers via the Web. The contribution of this research is that it demonstrates a practical use of the evolving technologies of the Semantic Web, while creating a valuable educational resource.

Key-Words: - Semantic, Streaming, Web, LAN, TCP, HTTP, Multimedia, Data.

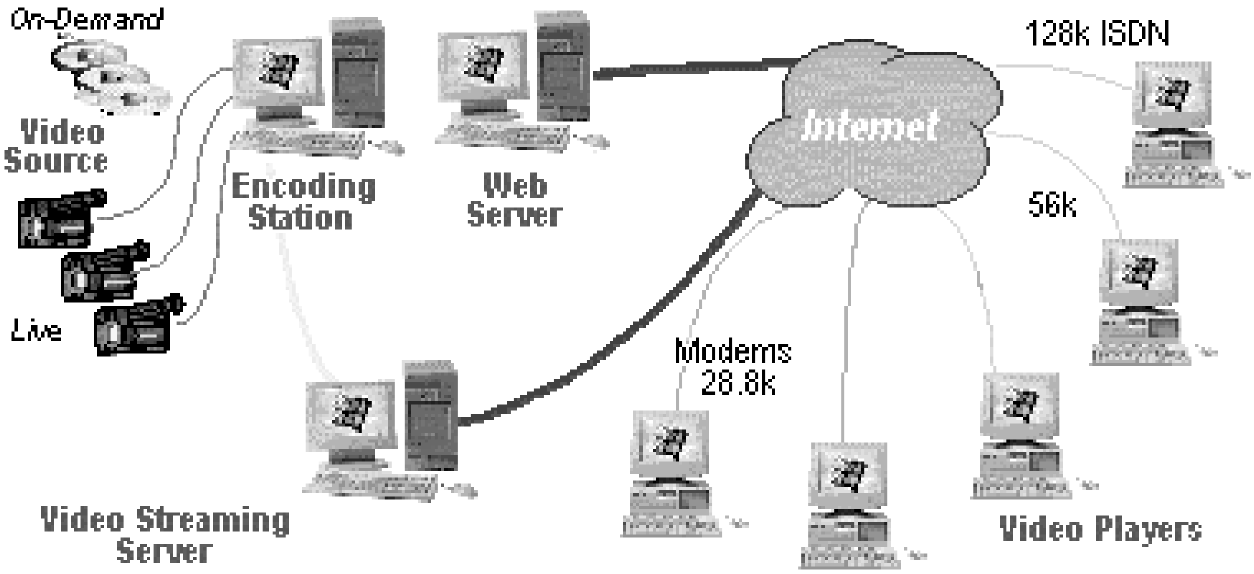
1 Introduction

In the emerging Information Society the goal of diffusing Information and Communication Technology (ICT) skills in the population is a priority for all governments, and a focus in industry and commerce and the educational system. As a result of the growing pressures to improve the ICT skills of whole populations, within the educational systems of all countries there have been substantial moves to provide PCs and Internet access in schools and universities, and to promote their use in the curricula. Moreover, universities are experimenting with effective methods for developing ICT skills and knowledge in their graduates to make them better fitted for employment, both at the start of their careers and in the longer term. Even if the data reported are not yet definitive, since the distance learning and certification activities are still ongoing, preliminary analysis has demonstrated a satisfactory response to this new learning method by the students involved. Full participation in the classroom lessons and frequent access to the on-line teaching material were recorded, and assiduous study yielded good results during the certification phase.

2 Streaming multimedia data

Typically, when accessing multimedia data across a network, a user had to wait for the entire file to be transferred before they could use the information. Streaming, however, allows a user to see or hear the information as it arrives without having to wait. Streaming technology offers a significant improvement over the download-and-play approach to multimedia file distribution, because it allows the data to be delivered to the client as a continuous flow with minimal delay before playback can begin. The multimedia data arrives is briefly buffered before being played, and is then discarded. It is never actually stored on the users' computer. Users benefit by experiencing instant playback without the frustration of having to wait for the entire data to be downloaded before they can determine whether it meets their needs or interests. In most cases, this download process took a long time, and was impractical for widespread acceptance.

Streaming is a server/client technology that allows live or pre-recorded data to be broadcast in *real time*, opening up the network for traditional multimedia applications such as news, education, training, entertainment, advertising, and a host of other uses. Thus, streaming enables the Internet or company Intranet as a new broadcast medium for audio and video. The Video Source is typically one or more streams of analogue video. It can come from cameras, DVD players or VCRs. These video sources will have an analogue video connection to the Encoding Station. It is common for live broadcasts to connect the cameras to video production and editing equipment, before being passed on to the Encoding Station.



The Encoding Station is a computer that captures and typically, encodes both the audio and video *live*, directly into the required streaming format. The most common systems used for encoding are Windows® XP or Windows® 2000 workstations equipped with audio and video capture cards. These systems must have the computational power to encode one or more audio and video streams either in software or via a hardware codec. The use of a good capture card is critical in achieving these high rates with good picture quality. To meet this criteria, Winnov have developed their own ASIC's (Application Specific Integrated Circuit) that are optimized for managing data transfers to the PCI bus. In fact, the latest cards also have SDRAM on board to create an *Elastic Frame Buffer* that holds a digitized frame until the PCI bus is prepared to receive it. This is particularly important when there is high bus traffic and/or multiple cards in the same PC. Without it, you could experience pixels drops that would degrade the video quality. In contrast, most other vendors, including Osprey use the Conexant PCI I/F chip for managing data transfers to the PCI bus. The Encoding Station, which needs to be near the Video Source, sends the compressed audio/video streams on to the Video Streaming Server (typically via a LAN using UDP/ TCP protocol). Individual compressed streams can vary from 20 Kbps (Kilobits/second) to 500 Kbps or more. The connection between the Encoding Station and the Video Streaming Server must be able to accommodate the total of the bandwidths of the individual streams and must be a clear and reliable connection. The Video Streaming Server is responsible for delivering compressed video to each individual request for a particular video stream. This is usually handled by one of the commercial streaming media software packages such as RealNetworks® RealSystem™ or Microsoft® Windows Media™ Technologies. The bandwidth connection to the Video Streaming Server must accommodate the total bandwidth of all the requests for a video stream, unlike the Encoding Station, which must only accommodate one copy of each. As a result, the Video Streaming Server usually has a direct connection to a very high bandwidth line. For example, if there were 100 requests for a video stream compressed at 28.8 Kbps, the server would require at least a 3 Mbps connection. The Encoding Station and the Video Streaming Server can be one single system. However, unless *hardware encoding* is used, this would typically be for a situations requiring limited performance (e.g. a single input stream and a small number of viewer requests). Even so, it would still require a fairly high-performance system. It is much more common to have two separate systems.

The WebServer for video streaming is in no way different from other Web Servers. The web site merely contains a URL link to the Video Streaming Server - one for every available video stream. Typically this is an icon on the web page to be selected. A Video Player application is required to decode the specific video stream received by the system requesting the stream over the Internet (or corporate Intranet). The most popular current video streaming applications are RealNetworks® RealSystem™ and Microsoft® Windows Media™ Technologies. Both of these require downloading a corresponding Video Player application such as RealOne™ Player or Windows Media™ Player; but both of these are *free*. There are other video streaming applications that are implemented in such a way as to include the player in the stream and no download is required.

2.1. Unicast versus IP Multicast

There are two key streaming delivery techniques: unicast and multicast. 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. 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. To use IP multicast on a network, the network routers must support the IP Multicast protocol. Most routers now handle multicast.

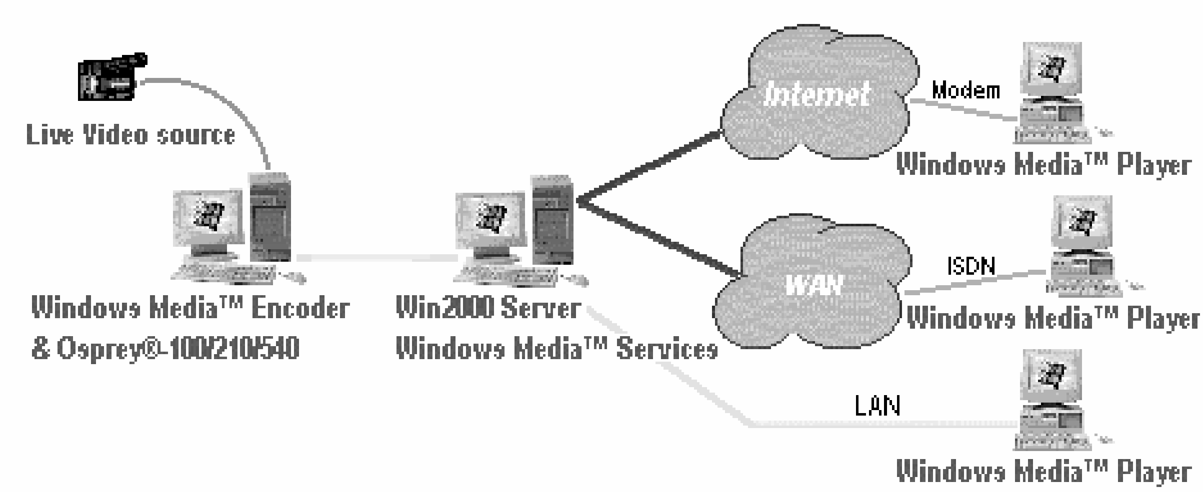
2.2. Internet Protocols

There are several internet protocols available for streaming data, TCP, UDP, RTP, RTSP, MMS & HTTP. Generally, each configures the data into packets, with each packet having a 'header' that identifies its contents. The protocol used is usually determined by the need to have reliable or unreliable communications. TCP is a reliable protocol designed for transmitting alphanumeric data; it can stop and correct itself when data is lost. This protocol is used to guarantee sequenced, error-free transmission, but its very nature can cause delays and reduced throughput. This can be especially annoying when streaming audio and video. User Datagram Protocol (UDP) within the IP stack, is by contrast, an unreliable protocol in which data is lost in preference to maintaining the flow. Real-Time Protocol (RTP) was developed by the Internet Engineering Task Force (IETF) to handle streaming audio and video and uses IP Multicast. RTP is a derivative of UDP in which a time-stamp and sequence number is added to the packet header. This extra information allows the receiving client to reorder out of sequence packets, discard duplicates and synchronize audio and video after an initial buffering period. Real-Time Control Protocol (RTCP) is used to control RTP. With RealServer™, RealNetworks introduced as its primary server protocol the RealTime Streaming Protocol (RTSP); an open, standards-based protocol for multimedia streaming. To use this prot0col, URLs that point to media clips on a RealServer™ begin with rtsp://.

With Windows Media™ Technologies, Microsoft introduced Microsoft Media™ Server (MMS) as its primary server protocol. MMS protocol has both a data delivery mechanism to ensure that packets reach the client and a control mechanism to handle client requests such as Stop/Play. MMS includes both Microsoft Media Server protocol/ UDP (MMSU) and Microsoft Media Server protocol/ TCP (MMST) as subsets to explicitly request the stream to use UDP or TCP respectively. Media Stream Broadcast Distribution (MSBD) protocol was used to transfer streams from the Windows Media™ Encoder to the Windows Media™ Server or between servers. However, Windows Media™ Encoder 7 and later versions no longer supports MSBD and uses HTTP instead. URLs that point to media clips on a Windows Media™ Server usually begin with mms://. Hyper Text Transport Protocol (HTTP) is the slowest of the protocols and is used by Internet Web Servers. HTTP is transparent to some older firewalls and can bypass security in such cases. Unlike RTSP and MMS that can serve the stream at a steady bitrate, HTTP would just serve the stream as fast as it could hence it is better to have separate web and streaming servers

2.3. Streaming Video with Microsoft® Windows Media™ Technologies

Microsoft® Windows Media™ 9 Series is the latest multimedia streaming service that enables Internet Service Providers and organisations to deliver high quality audio and video at every available bandwidth across the Internet or corporate Intranet. Tightly integrated with other Microsoft products; Windows Media™ 9 Series provides the foundation for building multimedia rich applications that can provide news and entertainment, deliver corporate communications or sell goods and services.
Windows Media™ 9 Series includes several streaming media components that are based around the Windows® Server 2003 operating system and are capable of efficiently encoding a 640x480 video window at 30 frames per second over a 1 Mbps stream. These components can be functionality grouped into Windows Media™ 9 Series Encoder to *Create* the .wmv file, Windows Media™ 9 Series Services to *Serve* the .wmv stream and Windows Media™ 9 Series Player to *View* the .wmv stream.
There is also a Windows Media™ SDK that includes a collection of tools and utilities to assist producers in the creation, distribution and playback of .wmv files. Windows Media™ 9 Series Encoder includes features such as De-interlacing and Inverse Telecine that improve the image quality.

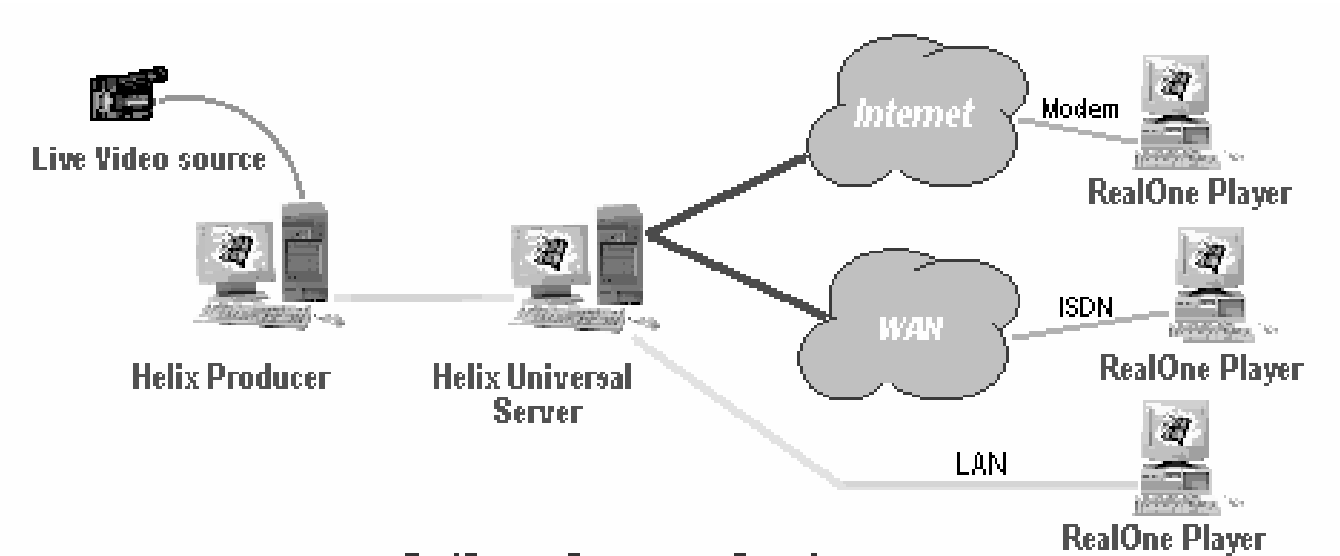


Microsoft® Windows Media™ Technologies Components

By supporting De-interlacing the flicker on computer monitors is reduced when displaying video that comes from interlaced sources such as NTSC systems. Inverse Telecine support improves the quality when encoding sources from film. Typically, video that originated at 24 fps would be padded with extra frames to achieve 30 fps. Windows Media™ 9 Series Encoder can intelligently extract the original 24 fps and encode at just 24 fps, eliminating this extra padding and its corresponding artefacts.

2.4. Streaming Video with RealSystem™

RealNetworks® video streaming software, RealSystem™, includes Helix™ Producer to *Create* the .rm stream, Helix™ Universal Server to *Serve* the .rm stream and RealOne™ Player to *View* the .rm stream. The general sequence of events with video streaming is to first *Create* the video, then *Serve* the stream to the end-user who *Plays* it. Currently, there are several derivatives of RealNetworks® video encoding software available to *Create* the video content. The latest versions is Helix™ Producer and Helix™ Producer Plus, both of which are supported under Windows 2000, XP and Server 2003 as well as Linux. Also available is RealSystem RealProducer Plus 8.51 for Solaris. RealVideo® 9 contains several new and updated features that help to improve the image quality and performance. Two Pass Analysis looks at the individual frames of the source file prior to encoding. It then redistributes bits to give an improvement in the video quality of the encoded file. This will increase encode processing time. Variable Bitrate Encoding increases the quality by varying the playback bitrate, allocating more bandwidth to high-action scenes and less to low-action. A De-Interlace Filter removes artefacts from interlaced video such as NTSC or PAL sources. An Inverse Telecine Filter removes the padding frames or fields from NTSC films that were originally converted from 24 fps to 30 fps in a 3:2 pulldown process. Black Level Correction adjusts the contrast of the input to make blacks appear darker and whites lighter. This filter can be used to restore the saturation levels in 'washed-out' video. Forward Error Correction resends additional packets at preset intervals to ensure against lost packets whilst Loss Protection sends additional information in each packet to enable players to reconstruct missing frames if packets are lost.



RealSystem Component Overview

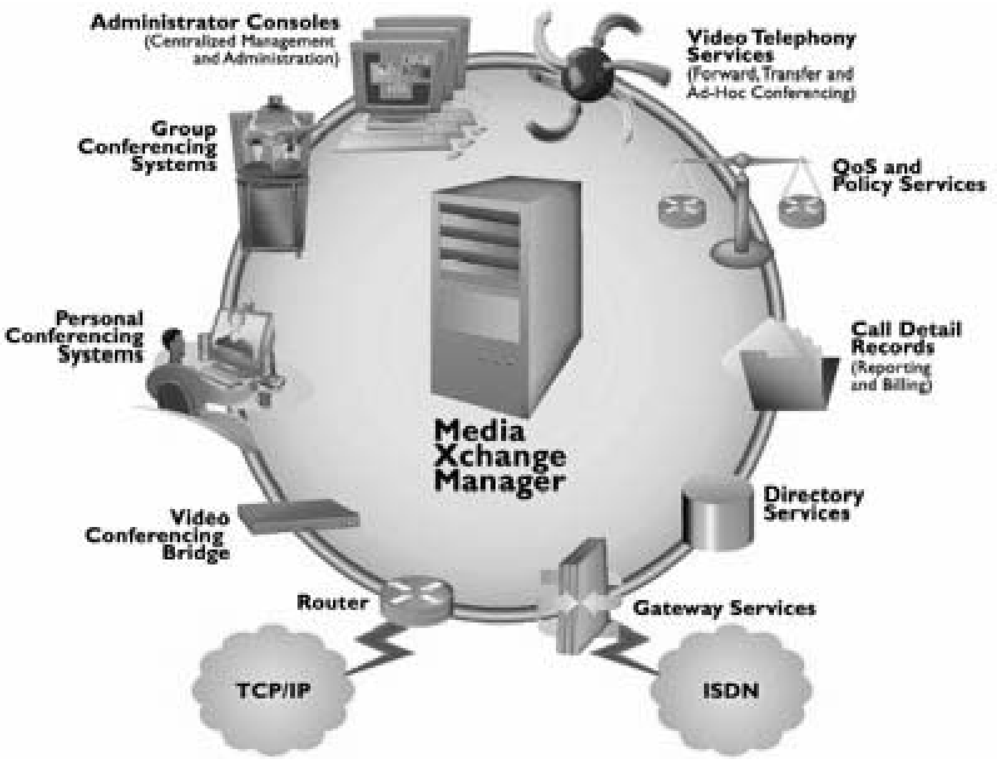
For Linux based PC's, Helix™ Producer can use the Osprey®-100 video only capture card to encode in software.

3 Remote Lab using VCON Media Xchange Manager™ (MXM)

MXM is a software based IP Video PBX that runs on Windows® 2000 or Windows® 2003 Server. From a remote console, an authorised user has the ability to perform centralised management, configuration and administrative functions across multiple zones. MXM supports a combination of both VCON and non-VCON endpoints as well as a wide range of MCU's, Gateways and other networking equipment. With video telephony services such as Call Forward, Call Transfer, Call Pickup, Ad-Hoc Conferencing and Hunting Groups, MXM provides the functions that make Video Conferencing as simple as making a telephone call.

Options include VCON Conference Bridge (VCB), a scalable software MCU that can also stream the conference as an IP multicast; IPNexus Secure Instant Messaging that can launch a Video Conference from with the IM session; Moderator, a web based manager for scheduling both point-to-point and multipoint conferences; Report and Billing Tool that provides system activity and Call Detail Reports (CDR) and SecureConnect, a Firewall Proxy and Encryption Server:

-Supports a Mixed Multi-Vendor environment
-H.323 Gatekeeper and SIP Proxy
-Call Forward and Call Transfer
-Roaming profiles for users
-Highly scalable - up to 5000 users per Zone
-Up to 150 Neighbour Zones
-vPoint HD Software Client Endpoints
-SecureConnect Firewall Proxy
-VCON Conference Bridge (VCB)
-IPNexus Secure Instant Messaging
-Report and Billing Tool
-Web based Moderator for scheduling
-Remote Endpoint Configuration
-Remote Call Initiation and Termination
-Simplified Outbound Gateway dialling
-Advanced Directory Services



4 Conclusion

These technologies come under the generic term, "Rich Media" and include Video Conferencing, Application and Data Sharing, Interactive Whiteboards, Video Streaming and Secure Instant Messaging, plus communications equipment such as Multipoint Conferencing Units (MCU) and Gateways that allow users to connect and share information over networks including LAN, WAN, ISDN, ADSL and the Internet. The skills involved include the appropriate advice, consultancy, supply, installation and training to enable customers to derive maximum potential from their investment.

In today's global economy companies and their partners operate numerous different hardware and software systems. The products and skills bring to the marketplace addresses this, providing secure solutions that allow project teams to communicate anytime, anywhere, independently of which hardware or software systems they may be using.

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