BROADCASTING IN THE CYBERSPACE

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ABSTRACT

Cyberspace is opening new possibilities for administering broadcast services. Audio and Video broadcasting using Internet have become a reality today. Many sites already offer audio / video on demand, including news, interviews, concerts, documentaries and music. This paper examines the current technology for transmission and reception of audio and video through the Internet. The paper further describes the available transport protocols, and a system architecture for audio/video broadcasting through the Cyberspace. The performance expectations and the emerging scenario are also examined.

I. INTRODUCTION

Broadcast technology is undergoing a serious change with the advent of digital compression techniques. MPEG-2 [1,2] has been currently accepted as the world standard for broadcast purposes. Digital Video Broadcasting (DVB) using MPEG-2 standards for Source Coding and Multiplexing has allowed five Standard Definition TV (SDTV) programmes on a 33 MHz transponder with a data rate of 38.1 Mbit/s [3]. Work has already began on a new standard MPEG-4 with facilities for content based accessibility and manipulation, support for interactive communication, object addressing etc. [4]. MPEG-4 standard, with target for a very low bit rate (LBR) between 4800 to 64000 bits per second and sampling dimension upto 176 X 144 X 10 Hz, is expected by November 1998.

The control over encoding parameters viz. video and audio bit rates, distance (M) between reference frames, the distance between intra-coded I frames, number of I frames in a group of picture (GOP), intradistance (N) etc. have been exploited to develop several lossy compression techniques which can provide greater compression [5].

These lossy compression techniques have been used for digital broadcasting through low band-width Cyber-space (INTERNET). Audio and Video including multimedia, animation and virtual reality scenes are already available on many sites.

In this paper, we investigate the technology used for the Broadcasting on Internet, the systems architecture and the methodology adopted, the quality to be expected and the on-going research in this field.

II. CYBERSPACE BROADCAST TECHNOLOGY

The predominant technologies for broadcasting on the web, use buffering, codec (compression/decompression) and stream technologies. Buffering is provided to make up for transmission delays. By allocating portion of memory to store a few packets, usually a dozen or so of audio/video information, the player always finds data to play from buffer rather than waiting for receipt of data from server. Codec technologies compress the data using compression algorithms at the server end and then decompress at the receiving end. Stream technology allows for real time repositioning within a file as well playing files as they are downloaded.

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A. *Audio On The Net*

Until the advent of several lossy compression technique, large size of audio files coupled with the bandwidth limitations of the Internet made it impossible to use the World Wide Web for efficiently and reliably accessing large volumes of archived audio content. This was because an entire audio file had to be downloaded to the machine before the playback could begin. Buffering, codec and stream technologies have made it possible to deliver sounds on the web even using 14.4 Kbps modem in a real time with controls for rewind, forward, pause and playback. The use of audio stream (continuous-delivery) technology permits playing of a single audio packet on receipt. The transfer communication being bidirectional, the player can request the server to send a specific audio packet [6].

Streaming audio technologies are designed to overcome the limited bandwidth of Web: a 14.4/28.8 Kbps modem or 128 Kbps ISDN connection. A 14.4. Kbps modem has a throughput capacity of 1.8 kilobytes/sec, as against the requirement of 176 kilobytes/sec of CD quality audio (97 times the capacity of 14.4 kbps modem). For this reason all the streaming audio technologies compress the data drastically to match the throughput of the Internet connection. While CD quality audio requires a compression of 97:1, several audio codecs start with lower quality for example 8 Khz, 16 bit audio requires a compression of only 8:1.

B. *Video On The Net*

As a medium, video is much more demanding than audio, both technically and aesthetically. Compressing television quality video, whose original bandwidth is about 27 megabytes per second, to a usable 28.8 kbps modem, requires an astounding 7500:1 compression ratio. This extreme compression, achievable only by lossy techniques, causes tremendous distortion in the form of pixelation, blockiness and gross artifacts. Using a 64 Kbps -single line ISDN or 128 Kbps - dual line ISDN, greatly enhances the quality of the video. A high bandwidth network or T-1 connection can play a stored file at full frame rate [7].

C. *Compression Techniques*

Several application specific compression techniques are used for authoring audio and video on Web. Meta Voice compression algorithm is best suited to low band-width (2400 bps) speech broadcasts. LBR (Low Bit Rate) compression algorithm has been found to be suitable to provide crisp and clear speech and music on 28.8 Kbps. Video has two components that affect the overall result: the absolute quality of individual frames and the number of frames displayed per second. The approaches available are high quality low-frame-rate or full motion and degraded frame quality. The lossy compression techniques available are Prediction (Motion Compensation), Frequency Oriented Compression, Importance Oriented Compression, Sub Sampling, Vector Quantization, Hybrid Coding, Fractal Block Coding etc. The lossy techniques used for authoring video on the Web are described below:

1) ***Derivative of MPEG*:** The compression is controlled by selecting the number of I (intra-frame), P (predicted-frame) and B (bidirectional interpolated) frames. For example at lower data rates, only higher-quality I-frames are selected and B and P- frames are completely dropped. The advantage is that a user with high bandwidth network like T-1 can access stored video at full frame rate while a user with 28.8 Kbps modem might receive frames at a slide-show frame rate maintaining good image quality.

The technique has been used to create audio/video streams ranging in bandwidth from 28.8 to 150 kilobytes per second and providing crisp, high quality frames at receiver that often move like a slide show than video on a 28.8 kbps modem.

2) *Wavelet Algorithm* : Characterised by high quality, dynamic bandwidth management and scalability, the compression is achieved by dividing each video frame into multiple layers and then dropping the layers depending upon the availability of bandwidth. In this technique, lower bandwidth results in degradation of picture quality but motion and audio are preserved. Wavelets remove blocking artifacts by the fact that their basic functions are overlapped one another and decay smoothly to zero at their end points. The scheme achieves higher compression ratios for a given picture quality and is simpler to implement to achieve real time performance for video sequences. Due to lack of blocking effects, the errors introduced in the scheme are less visually annoying than for the DCT compressed images.

In wavelet transformation [8], the input signal is splitted into number of bands by applying low pass filter L and high pass filter H in both horizontal and vertical directions. The filter outputs are sub-sampled by a factor of two, generating three orientation selective high pass sub-bands, HH, HL, LH and a low pass sub-band LL. The process is repeated on LL band to generate the next level of decomposition. Four octaves of decomposition leads to 13 sub-bands. The image data is compressed by controlling the transmission of coefficients and number of bits to code them. Each group of transform coefficients are gathered according to their spatial position and quantized. The quantized transform coefficients are thereafter scanned out from lower frequency block to higher frequency block followed by variable length coding. The scanning path is chosen to increase zero run so that the efficiency of entropy coding is improved.

D. *Transport Mechanism*

Internet uses TCP/IP protocol for data delivery which is not efficient in handling continuous time based audio / video. Therefore, the way data is transmitted across the Internet and the protocols used have direct bearing on the overall efficiency, performance and reliability of broadcast application. Flow control mechanism is used to achieve high performance continuous audio / video delivery. In a flow control mechanism, statistical and instantaneous information about network throughput, reliability measurement, bandwidth availability, receiver's current buffer size, packets received by receiver etc. are used to control the video transmission. Flow control can also be used to embed security mechanism, such as copyright protection and encryption, into video transmission. Few of the protocols used for audio /video delivery are described below:

1) *User Datagram Protocol (UDP)* : UDP is a bandwidth-oriented streaming protocol without error correction. UDP transmits small packets at a high priority but does not guarantee packet delivery. A flow control mechanism is used to avoid the saturation of the network.
UDP opts for broadcast efficiency at the risk of quality degradation and is best suited for audio-delivery.

2) *Transmission Control Protocol (TCP)* : TCP uses large packet size for delivery of high volume of data and uses flow control mechanism to ensure fair resource utilization. The guaranteed in-order packet delivery is ensured by re-transmitting of lost or scrambled packets by the server. However, the packets being larger, in case of packet loss, the playback at receiver stops till the packet is received again. For continuous media applications, therefore, a different strategy is adopted. Instead of re-transmitting everything lost, only significant data viz. I-frames in MPEG scheme, is selected for re-transmission [9].

3) *IP Multicasting*: In IP multicasting, a dynamic host group is created and IP datagrams are sent to all the members of the host group. In order to provide a reliable multicast service, the following mechanisms are employed:

a) *Adaptive time-out with exponential back off*: In this technique, the RTT (round trip time) for every datagram is measured, and the packets are re-transmitted in absence of acknowledgement from the receiver. In case no response is received within a predeifined time period, the receiver is taken off the subscription list. Exponential backoff identifies the strategy used for calculating time -outs for re-transmission.

b) ***Packet sequence numbering*:** This allows the listener to discard duplicate packets and take action to prevent what is known as sorcerer's apprentice syndrome (RFC 1123).
IP multicasting is ideal in situations like scheduled programming or live audio where many people are like to tune to the server simultaneously.

III. SYSTEM ARCHITECTURE

The logical architecture of the broadcasting through cyberspace service is illustrated in Fig. 1. The system consists of:

A. *Broadcast Delivery System*

The core of the system is the Broadcast (Media) Server that interfaces with an http (Web) Server [10]. There can be remote audio/video databases with multiple Servers. The processes involved are mentioned below:

1. ***Media File Digitization*:** Since many broadcast system are still analogue based, the audio and video source are digitized using a standard sound card, a video capture card or both. Compression being a garbage-in, garbage out proposition, highest quality signal to the encoder is ensured.

2. ***Media File Encoding*:** The data compression to enable real time delivery over connections as slow as 14.4. KBPS (9.6. KBPS for audio) is achieved by encoding. Encoder software uses lossy compression algorithms to squeeze data into compressed format. The title, author and copyright information is also embedded at this stage to be read by the player.

3. ***Embedding Multimedia into a Web Page*:** A pointer placed into standard HTML tag in the web page links to a token file on the media server. This token file contains type and location data about the media file itself.

4. ***Integration Of Multimedia Files into the Server Environment*:** The web server recognises the multimedia formats by registering the file's extension with the server as a MIME (Multi purpose Internet Mail Extension) type. The

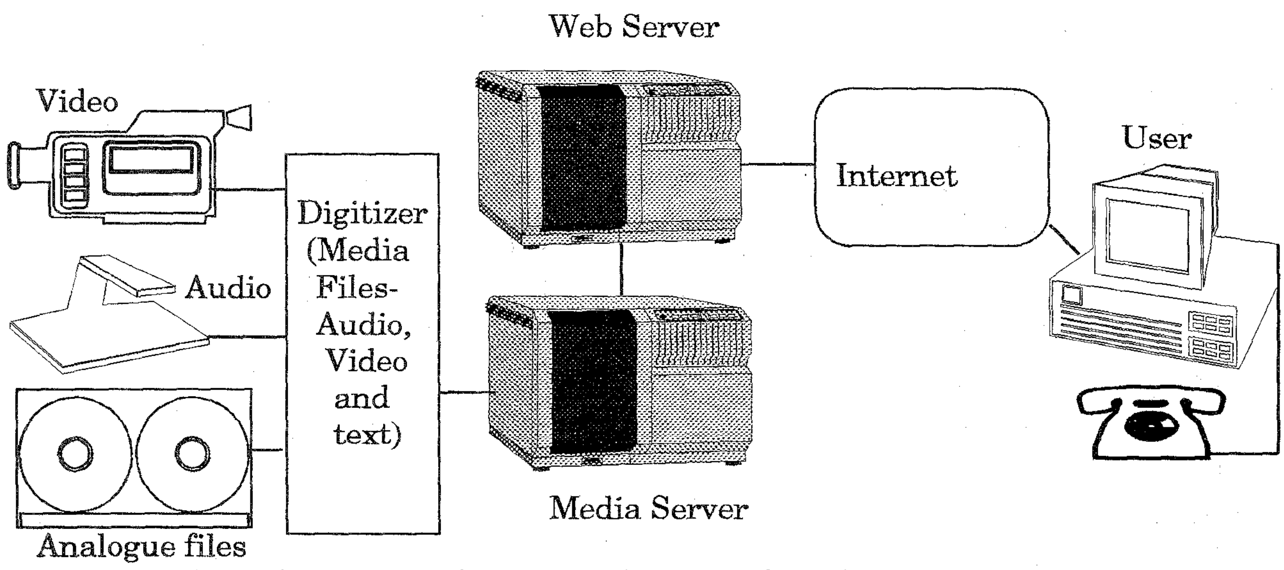

Fig. 1. System architecture for broadcasting in cyberspace

server also contains information such as Protocol Port, maximum connections, connection time-out, security etc.

B. *Receipt System*

The receipt system consists of a Multimedia Personal Computer (MPC) with a sound card, a 14.4 Kbps modem or better connectivity, Direct SLIP or PPP connection to Internet alongwith Browser with media player registered as a helper application. The player offers features such as volume control, fast-forward, stop and resume. When a media link is clicked, the browser sends a request to a media server that returns a token file to the PC. This file requests the browser to spawn the appropriate player. Once the player is running, the player sends the request to the media server which transmits the data to the player. After few seconds of buffering, playback begins.

IV. QUALITY

The listening observation of audio received from a number of Web sites was made at Delhi (India), using a 14.4 Kbps modem. The line connection was mostly at 9600 bps. The quality was found to be of AM broadcast but with breaks. Sometimes the audio was found to be choppy due to packet loss. This packet loss was due to unreliability of transmission media. Using a 14.4 kbps modem for video, new frames were displayed every 10 or 12 seconds. The performance dramatically improved with a faster CPU (Pentium with 75 MHz clock speed) and faster modem (28.8 Kbps). However the video were still found to be slow and jerky although image quality was good. Few Web sites were also surfed to find the reported quality of audio and video at various connection speeds. Based on these researches, the quality of the audio/video on different connection speeds is shown in Table 1.

The analysis of the data reveals that at present, for modem speeds, technology is acceptable for audio broadcasting but barely useable for video broadcasting. Another contributing factor for both audio and video broadcast, is the type of programme: recorded or live. Because of the high demands of the compression on-the-fly, live audio/video delivery results in more deterioration in quality.

V. FUTURE SCENARIO

Rabiner [11] states that the vision of multimedia communi-

Table-1

Connection Speed and Quality

Connection type	Speed (Kbps)	Audio Quality	Video Quality
Dial-up modem	9.6-14.4	8Khz. AM radio[1] (breaks)	
	14.4	AM radio[1]	Pixelated & blocky frames (every 10 or 12 seconds)[1]
	28.8	16/22Khz.mono [1] Near FM[2]	Good image quality displayed (every 2 or 3 seconds)[1]
Fram relay/ISDN	56-64	44 Khz Stereo, Near CD[2]	Good low motion video, High motion clips distorted.
ISDN, two B channel	128	44 Khz Stereohi-fi	30 fps full motion video at quarter screen resolution.[3]
T1	384+	VHS	MPEG-1 quality

Source: 1. Listening / Viewing Observations at Delhi (India), 2. http://www.firstadio.com/listen.html,

3. http://www.xingtech.com/sw.winclient/info/tsg-win.html

cation in the year 2001 contains simple-to-use ways for people to manipulate their complex and rich communication environment comprising of speech, music, still and motion video. Some of the technology that is in offing for video/audio on the net and which shall play a key role for broadcasting through cyberspace are:

A. *Very Low BIT Rate Coding using Model Based Approaches*

To avoid blocking and mosquito artifacts at a very low bit rate, Object based and Knowledge based codings are being researched. In Video applications, set of parameters defining the motion, shape and texture of the moving object, wire frame model and the changes in the structure from frame to frame etc. can be used to synthesize the next frame at the receiver side. [12]

B. *Web Voice Browser*

Web-On-Call Voice Browser uses text-to-speech technology to read back information on a Web Server to a user calling into the Web site. This has special application in broadcasting since the news of specific importance can be received on an ordinary phone or a cell phone [13].

C. *One-Way Virtual Broadband Network*

The technology is based on the fact that the Internet user does not need high speed two-way transfer but needs high-speed one-way transfers from host computers to home site. This allows user to dial their Internet provider using low-speed telephone lines but receive the audio/video file over user's own satellite dish using Ku-band. The data is routed through an intermediate server using high speed lines (upto 400 kbps). The technology is able to provide "real time audio, video and multimedia file transfers" "to an unlimited number of locations in continuous feeds, scheduled regular transmissions or occasional, as needed, broadcasts". [14]

D. *Multicast IP*

IETF (Internet Engineering Task Force) is finalising standards for multicast IP traffic over ATM [15].

VI. CONCLUSION

Broadcasting in the cyberspace has opened new challenges for broadcasters where a Broadcast station can be operated by a single individual [16]. A number of operators viz. ABC (American Broadcasting Corporation), CBS (Canadian Broadcasting Corporation), Hong Kong Radio, RTM (Radio Television Malaysia), MTV, CNN etc. have already started the service. With the popularity of the Internet increasing and the technology maturing, the Short Wave transmission may be on its way out.

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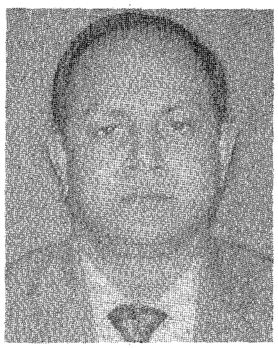
BIOGRAPHY



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