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Broadcasting to Netcasting

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Audio and video broadcasting using Internet have become a reality today with many sites already offering audio / video-on-demand. The present paper reports the result of listening/viewing observations, in India, of several Internet broadcasts. It is observed that for modem speeds, technology, as of now, is acceptable for audio broadcasting but barely usable for video broadcasting. The paper also describes the technology behind the netcasting and the emerging scenario.

THE world's most powerful new broadcast station reach listeners/viewers from one part of the globe to other, yet they dont't have transmitters or antennas. Welcome to Internet radio, which brings news, talks, interviews, concerts, documentaries and music to a PC, wherever it may be. Cyberspace is opening new possibilities for administering broadcast services. Several radio and TV stations now broadcast live on the web although marriage of on-line hypertext with packet audio and video presents some tough challenges to software architects, network operators and content providers. For starters, more bandwidth and better compression schemes are needed. The internet must also undergo redevelopment at the protocol level if it is to become a medium for broadcasting and with the internet poised for widespread home use, the network must also contend with content filtering for the first time.

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. The use of stream (continuous-delivery) technology permits playing of a single audio/video packet on receipt[1,2]. The transfer communication being bi-directional, the player can request the server to send a specific packet.

Compression Techniques

Streaming 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.

Broadcasters bringing video to the internet face trade-offs between the quality of video they can provide and the bandwidth available to the average end user. Image size, frame rate and color depth are all traded off to dovetail with the capacity of today's modems, ISDN lines and T 1 lines (1.536 Mbits/s). 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. To support full-motion, full-screen video in the future, Broadcasters are looking to advanced compression technology and the emergence of cable modems and digital-subscriber-loop technology. Higher-speed ISDN Internet access promises to improve video quality, potentially yielding quarter-screen images at 15 to 30 frames/second.

Several application specific compression techniques are used for authoring audio and video on web. Metavoice compression algorithm is best suited to low bandwidth (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 techniques used for authoring video on the web are Wavelet[3] and derivative of MPEG algorithm.

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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 User Datagram Protocol (UDP), Transmission Control Protocol (TCP) and IP multicasting. Making the narrowband connections work, however, requires a clever blending of two Internet protocols: the Transmission Control Protocol and Interent Protocol (TCP/IP). Audio/video is sent back in one continuous stream via the User Datagram Protocol (UDP), a connectionless transport-layer protocol. Data moves faster over UDP since this lightweight protocol doesn't have the retry mechanisms that TCP/IP uses to prevent packet loss[4]. Multicast Transport Protocol (MTP) with adaptive time-out exponential back off and packet sequence numbering (RFC-1123) are being experimented for scheduled programming or live audio.

Broadcast Delivery System

The core of the system is the broadcast (media) server that interfaces with an http (web) server[5]. There can be remote audio/video databases with multiple servers. The processes involved are media file digitization, media file encoding, embedding multimedia into a web page and integration of multimedia files into the server environment.

Receipt System

The receipt system consists of multimedia personal computer (MPC) with a sound card, a 28.8 kbps modem or better connectivity, direct SLIP or PPP connection to Internet along with 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.

Quality

The listening observation of audio received from a number of web sites was made at Delhi (India), using a 19.4 kbps modem. With line connection 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. However, when a line speed of 14.4 kbps was achieved using new lines provided by VSNL, the quality of audio was AM and continuous. FM quality of audio has been claimed by manufacturers at 28.8 kbits/s. 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 300 MHz clock speed) and faster modem (28.8 kbps). However the video was still found to be slow and jerky although image quality was good.

The analysis of the data reveals that at present, for modem speeds, technology is acceptable for audio broadcasting but barely usable 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.

FUTURE SCENARIO

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:

* 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[6].

* 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 will gain popularity for broadcast of sports and other news[7].

* 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 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[8].

* Multicast IP

  Broadcasters may see improvements in the capabilities underlying the network when the industry steps up to the next-generation internet protocol (IPng) standard. IPng has a scalable multicast address format to help web developers build bandwidth-thrifty servers that implement rate matching, which will allow users connecting at different speeds to tap into the same broadcast.

CONCLUSION

Broadcasting in the cyberspace has opened new challenges for broadcasters where a broadcast station can be operated by a single individual[9]. A number of operators viz ABC (American Broadcasting Corporation), CBS (Candian Broadcasting Corporation), Hong Kong Radio, RTM (Radio Television Malaysia), MTV, CNN, All India Radio 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. Broadcasters will do well in thinking twice before sinking national resources in to shortwave expansion.

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