Literature review – Web Conferencing Systems

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ABSTRACT

Web-conferencing or online meeting tools allow remote meeting and collaborative work. Poor Internet service however makes most Web conferencing solutions unreliable for South Africa and developing countries in general. This paper reviews the literature on improving the user experience with low bandwidth and unstable Internet conditions for Web-conferencing. A special focus is put on audio/video stream optimization, which is the most affected feature of a web conferencing system. The ongoing research in this area can be grouped into three main domains. Firstly, is research on rate adaptation schemes that aims to provide the best quality multimedia stream to several receivers with optimal use of the available bandwidth. Secondly, is research on compression which attempts to reduce the bandwidth requirement with acceptable content quality. The last research domain studies how to weaken the influence of transmission errors and problems over the content provided.

1. INTRODUCTION

Web conferencing systems are Internet-based tool offering a virtual environment for remote meeting and collaborative work among geographically dispersed participants. Remote conferencing can avoid travel expenses and time for face-to-face meetings (Gurhan, et al, n d). This explains the worldwide spreading of web based conferencing tools.

South Africa, and other developing countries in general, often rely on low bandwidth and unstable Internet connections. This badly affects the service quality offered by Web conferencing solutions, often developed with the assumption of a good Internet connection. That is why most current conferencing tools cannot reliably be used in the African context (Egido, 1998).

The most affected feature of online conferencing systems is the real-time audio/video communication, which requires a high speed and steady Internet link (Gurhan, et al., n d). This paper is then a review of the literature on improving the quality of user experience for web based conferencing using an extreme Internet service. This review focuses on multimedia streaming with low bandwidth and unstable networking conditions.

Section 2 is a general presentation of Web meeting software. In section 3, the influence of poor Internet transmission on user experience is discussed. Section 4 reviews the literature. Finally, section 5 compares and discusses the current state of the art and how the findings can positively affect the Web conferencing user experience.

2. ON LINE CONFERENCING SYSTEM

On-line Conferencing Systems, sometimes referred to as Electronic Meeting Systems (EMS), are Internet-based services offering a virtual environment for real-time remote meetings between geographically dispersed participants (Ramesh, et al, 2003). EMS are part of the broader field of Collaborative Internet Systems that encompasses the use of computers and Web technologies to support coordination and cooperation of two or more people attempting to perform a task or solve a problem together(Bafoutsou, 2002).

EMS environment offers tools, methods and features to support remote meetings. Some of the common features include slide show presentation sharing, audio/video communication, shared white board for annotation and text chatting (Ramesh, et al, 2003). Fig.1 illustrates a common EMS user interface.

Fig.1 - Example of user interface for EMS

EMS and Collaborative tools in general perform an important role into organizations. They can offer a functional meeting platform where regular face-to-face meeting is infeasible (due to travel constraints, for example) or unaffordable (Reushle, 2008).

Current EMS solutions range from sophisticated executive meeting suites for immersive experience to simple computer to computer solutions. A server hosting the meeting is often involved; it can be hosted on the Internet or into a local network (for local meeting sessions) (Aguilar, et al, 1986). The client can be a regular computer (or a mobile handset) with audio and video capture devices for audio/video conferencing. The communication relies on Web technologies and Internet protocols (Aguilar, et all, 1986). Table 1 lists some of the current Web meeting tools[1].

Table 1- Existing Web-conferencing software

Software	License	capacity
Adobe Acrobat Connect	Commercial	500 users
BuddyMeeting	GPL + Free	10 users
Cisco Unified Meeting Place	Commercial	500 users
Dimdim	Commercial	-
Zoho	Commercial	-

3. EFFECTS OF POOR INTERNET TRANSMISSION ON THE QUALITY OF USER EXPERIENCE

When using Electronic Meeting System, users not only interact with the computing environment, but with other peers through various communication channels (text, audio, video, etc.). The table below summarizes how some important Quality of Service (QoS) degradations can affect the user experience quality (Scholl, et al, 2005) (Watson & Sasse, 1996) (Hargreaves & McCown, 2008).

Table 2- QoS degradation effects on user experience

QoS degradation	Consequence on user experience
Long and random delays and packet loss	Directly affect fluency of multimedia stream, leading to poor user concentration
Frequent interruptions	Badly alter the sense of "presence", can be very irritating for the user
Low bandwidth	Delivery of poor quality content which cuts off the user enjoyment and immersive experience

As per today, it is impossible to ensure a perfect QoS over the Internet. So, lots of research is being conducted to enhance the user experience under poor QoS. The next section discusses some of the current state of the art in this challenging research area.

[1] Wikipedia, Comparison of web conferencing software, "http://en.wikipedia.org/wiki/Comparison_of_web_conferencing_software", received at April 25th, 2011

4. HOW TO IMPROVE THE USER EXPERIENCE UNDER POOR INTERNET CONNECTION

Several research avenues are exploited to enhance the user experience under limited Internet transmission, which can be summarized in the following three main domains:

Transmission rate adaptation: research in this area studies how to efficiently use the available bandwidth to broadcast a multimedia stream to several recipients.

Error tolerance: the aim here is to develop correction schemes to make transmission errors less noticeable by the user.

Compression: a good compression scheme reduces the network load while keeping a reasonable content quality. Findings in this area help to reduce the effect of low bandwidth on the user experience.

4.1 Transmission rate adaptation

Research in this domain aims to optimally adapt the multimedia stream rate to user bandwidth capacity in order to provide the best QoS possible (McCanne, et al, 1996). The main challenge is to accommodate heterogeneous environments were several users have different bandwidths (Gill, et al, 2008).

4.1.1 Source based rate adaptation

This approach is the simplest solution to the problem. An uniform representation of the signal is sent to all interested receivers using IP multicast (Deering & Cheriton, 1990). So the sender or source broadcasts at a fixed rate without regard to changing network conditions. The source based rate adaptation is very simple to implement and supposes that receivers have almost the same bandwidth, which do not change a lot over the time. In realistic condition though, the bandwidth is not stable and receivers have different amount of bandwidth. So, the current method performs poorly in heterogeneous environments since low capacity regions of the network suffer congestion while high capacity regions are underutilized (McCanne, et al, 1996).

4.1.2 Receiver based rate adaptation

The Internet’s heterogeneity makes multi-point communication design a difficult problem. The receiver based rate adaptation struggles to solve this problem. The objective is to broadcast a live signal from any particular sender to an arbitrarily large set of receivers along paths with potentially high variability in bandwidth (McCanne, et al, 1996). To achieve this, the transmission rate is adjusted to match the available capacity in the networks.

The general principle is to broadcast several flows and each receiver, depending on its actual available bandwidth, will subscribe to one or more streams. Hence, it is up to the receiver to adapt the flow rate in this approach (Amir, et al, 1997).

There are several implementations for the receiver based rate adaptation, which can be grouped into two main models: simulcast and multilayer.

4.1.2.1 The simulcast model

In this model, the sender transmits multiples copies of the same signal simultaneously at different rates (resulting in different qualities). Depending on its available bandwidth, the receiver subscribes to only one flow, which is the optimal quality for the actual capacity. Should the network condition change, the receiver can adapt by subscribing to another “flow channel” to avoid either congestion or underutilization of the bandwidth. This approach requires a good bandwidth at the sender side to supply parallel streams (Gill, et al, 2008).

4.1.2.2 The multilayer model

The multilayer model encodes the stream into a number of layers that can be incrementally combined to provide progressive refinement (McCanne, et al, 1996).

The receivers can then connect to more than one layer at the same time. Each upper layer to which the user connects provides further content refinement. Fig. 2 illustrates a situation where three receivers with different amounts of bandwidth (R1: 512 Kbs/s, R2:

256 Kbs/s and R3: 128 Kbs/s) connect to a sender S with 512 Kbs/s bandwidth.

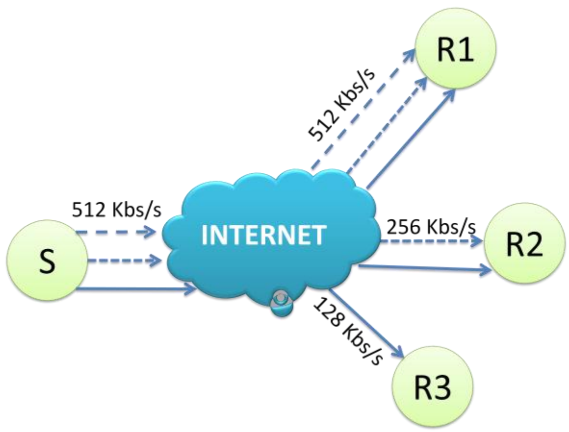

Fig.2 – Rate adaptation based on multilayer model

In the implementation of this model, the receiver connects initially to the first or base layer. If reception is good (no congestion, small error rate, etc.), it adds the second upper layer, which adds a certain refinement to the stream and the reception quality is analyzed again. This process is repeated to eventually reach the best tradeoff between bandwidth and quality (McCanne, et al, 1996). If on the opposite, the bandwidth decreases while receiving, the user can simply successively drop the upper layers to adapt the stream to the actual capacity.

Another important advantage of this model is the ability to associate a priority with each layer. The basic layer can get the highest priority and upper layers lower priorities. So when the bandwidth decreases, the routers can drop the upper layer packets which carry refinement information first. This technique leads to a graceful degradation of the stream over low and unstable bandwidth, substantially improving the quality of the user experience.

4.2 Error tolerance

The Internet often suffers from packet loss and random delay (jitter), degrading the user's perceived quality of multimedia stream (audio and video) (Claypool & Tanner, 1999). The following section discusses some approaches enhancing the quality of user experience under usual Internet transmission problems.

4.2.1 Stream buffering

In this approach, transmitted frames are buffered in memory by the receiver, allowing each frame to be played out with a constant latency, achieving a steadier stream (Claypool & Tanner, 1999). The added latency can badly alter interactive communication. So the buffer size should be chosen to provide the best perceptual quality, taking into account the tradeoff between decreased jitter and increased latency.

4.2.2 Error tolerance and correction

This approach privileges real time traffic by minimizing the delay as much as possible. Some error correction methods insert redundant information into the packet for later recovery. But this approach augments the bandwidth usage. A second approach is to retransmit erroneous packets. Under high error rate conditions, this approach worsens the jitter problem and generates a network overhead. A third approach is to generate small packets, so when an error or loss occurs, the packet is merely ignored (replacement with a silent packet for audio stream, for example). Under low error rates, the small ignored packets will not be noticed by the user. But above a certain error threshold (depending on the nature of the stream), this approach is almost useless (Claypool & Tanner, 1999).

4.3 Compression schemes

In the electronic meeting context, the video stream consumes most of the bandwidth (around 10 times more than the audio) (Chen, 2002). The compression aims then to reduce the bandwidth needed when keeping an acceptable quality for the user experience.

4.3.1 The Discrete Cosine Transform (DCT)

DCT is used in most of videoconferencing systems. A modern DCT compressor requires roughly 100 Kbps for a 320*240*15 fps video of a person's upper body (Chen, 2002). Below this rate, the video image motion may appear jerky (emergence of blocking artifacts).

To cope with very low bandwidth, two alternative approaches have been developed. In the first approach, only the outlines of images are encoded. Experimentation has shown that people can actually recognize the identity and facial expression of a person by the outlines of facial features (Chen, 2002). An implementation of this idea can delivers usable video at less than 10 Kbps. On the second approach, some key facial features are encoded in order to animate a 3D model of the person's head. The Moving Picture Experts Group (MPEG) is standardizing this approach and a good implementation can deliver a usable video at less than 1 Kbps! (Chen, 2002).

4.3.2 Frame rate minimization

The bandwidth requirement can be lowered by decreasing the video frame rate (which is the number of images displayed per second). Experiment demonstrated that a frame rate as low as one update every five minutes is enough to provide environment awareness. But studies on user behavior suggest that 5 frame per second (fps) is the acceptable lower bound for a direct human to human interaction (Chen, 2002).

An enhancement of this approach is to use a dynamic frame rate. The idea is to detect user gestures and increase the frame rate only when movements are spotted. Otherwise keep a very low frame rate(Chen, 2002).

5. COMPARISON AND DISCUSSION

Lots of current research results (in multiple domains) can contribute to improve the quality of user experience when using poor Internet connection for Electronic Meeting Systems. This section compares and discusses some of main findings and their possible application to the Web meeting context.

Research on rate adaptation aims to provide the best streaming broadcast quality adapted to the available bandwidth. The source based approach poorly uses the available bandwidth. On the other hand, the multilayer model approach offers the best rate adaptation since it allows the definition of different QoS on layers (leading to graceful signal degradation).

Error correction techniques based on retransmission and redundancy are simply not applicable in Web meeting contexts, as they increase the network overhead. The buffering approach is a better solution. Since meeting exchanges are not as interactive as a phone call, for example, the small delay introduced by the buffering will almost not be perceived by other participants, as long as the delivered stream is smooth.

There is good progress on video compression too, but the bandwidth required (100 Kbps) to keep an acceptable quality can still be too much (for African contexts especially). So, it's better to combine a compression scheme with frame rate minimization to deliver a usable video stream over very low bandwidth conditions.

CONCLUSION

This paper is a review of the literature on improving the Web conferencing user experience under low bandwidth and unstable Internet connection. A good understanding of how specific degradation of Internet Quality of Service (QoS) can alter the Quality of Experience (QoE) helps to develop mechanisms to deliver acceptable content on top of poor networking conditions.

Interestingly, lots of reviewed approaches can complementarily be used to improve the service quality of Web-conferencing systems. For example, a good compression scheme can be combined with dynamic framing methods to provide an acceptable video stream under very low bandwidth conditions. The same way, buffering and error tolerance algorithms can be mixed to supply smooth audio streaming on unstable Internet links (presenting interruptions, random delays and packet loss).

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