A Survey of Client Side Transmission Technique for Improve Quality of Service in Multimedia over IP

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Abstract

Multimedia Over IP (MOIP) is one of IP-based real-time communication that has been gaining popularity in recent years. Quality-of-service (QoS) is still one of the major challenges in real-time communication over IP networks. Excessive delay severely impairs communication interactivity, while packet loss results in glitches in audio and poor picture quality and frozen frames in video. In this paper, we perform a survey of the existing client side transmission technique. We list out several features and functionalities for research continuity in this area.

Keywords: Multimedia over IP, VOIP, Packet Loss, Delay, QoS,

1. Introduction

Multimedia over Internet protocol (MOIP) applications, such as IP telephony and video streaming, continue to gain popularity. In MOIP systems, one or several encoded video or audio data are grouped into a packet for the transmission through packet networks. The packet network for most MOIP systems operate based on RTP/UDP/IP, but they do not have any quality of service (QoS) control mechanism. Thus, packet losses could occur due to network congestion [7].

In real-time communications, losses are a result of not only packets dropping over the network, but also late arrival for packets. This paper introduces different loss-resilient techniques for both audio and video in 3 types of techniques: client-side techniques, active techniques and scheduling techniques, depending on whether they require any encoder involvement. One type of loss techniques is passive methods that are implemented at the client side, which do not require any cooperation of the sender or increase the cost of transmission. Client-side techniques impose low overhead for the communication system but can be highly efficient in enhancing the quality of the rendered media. A different type of error-resilience techniques requires the encoder to play a primary role. They are able to provide even higher robustness for media communication over best-effort networks.

The type refers to these techniques as “active” to differentiate them from those only employed at the client side. Active techniques have feedback mechanism from server. The last type of technique is packet scheduling. Packet-scheduling methods have been designed for allocation of a minimum bandwidth to each flow that crosses a link and also provision of throughput and a delay bound. These scheduling mechanisms consider dropping data that misses the deadline and do not consider data loss due to buffer overflow. In this paper, we only describe the client side technique in detail.

The rest of this paper is organized as follows: Section 1 presents the introduction of this project. Following section 2 presents the related work of client side technique. In this section we describe some techniques that have related with the client side technique for research continuity. Detailed performance features of client side transmission techniques presented in section 4. In this section also we present issues of performance quality of service multimedia over IP. Following section 5 presents our approach. Finally, section 6 concludes the paper and discusses some of our future work.

2. Related Work

Edward J. Daniel et al. [3] proposed an inter arrival delay jitter model based on client side technique. The technique was design for solving the problem of delay jitter on audio and video. They using network performance (netperf) for simulation the performance of the technique. Their project were about examines modeling and simulation of network delay jitter for real-time multimedia communications applications. They examine the multi-structure characteristics of network delay and develop a model
for simulation of jitter. The model is confirmed empirically using collected packet network jitter delay statistics. In their research delays are generated from probability distributions that have no correlation between samples, thus not simulating delay spike characteristics. Therefore, an exponential decaying function is used to simulate this phenomenon when the Laplacian or Gaussian distribution produces a delay spike.

Naofumi Aoki et. al. [5] proposed a waveform reconstruction technique that also takes account of the pitch variation between the backward and forward frames of gap frames. From experimental results of objective evaluation, it is indicated that the proposed technique may potentially be useful for improving the speech quality, compared with the conventional technique. This technique was designed to solve the problem of packet loss in quality of service for VoIP.

Figure 1: Jitter Model

However this technique only limited for delay jitter. In QoS of MoIP application, there are several problems of delays like encoding delay, packetization delay, end to end delay and etc. This project also did not mention about packet loss. This technique is very limited for one problem. For MoIP transmission, the two of problems must have been solved for make QoS become good and efficient.

Mi Suk Lee et. al. [4] proposed a voice packet loss concealment algorithm in order to improve voice quality for both multimedia over IP and voice over IP services. The proposed algorithm estimates the coding parameters of lost frames by combining forward and backward prediction from the good frames before and after the lost frames. The algorithm was designed for solving the problem of packet loss and delay in area of multimedia over IP. The algorithm is based on receiver-based algorithm which has advantage compare with sender-based algorithm. Receiver-based algorithms do not need any additional bits, and thus they can use the already existing standard speech encoders without any modification.

Naoven et. al. [6] proposed a TCP-friendly layered video multicast algorithm, which provides a feasible solution to both congestion control and error control in the Internet environment. They first introduce a new receiver oriented multicast congestion control algorithm SPLIT. Secondly, they apply the priority-dropping mechanism (PDM) advocated by to SPLIT using Random Early Detection (RED) queue. Since, the proposed PDM combined with packet classification scheme can distinguish the priority at the packet level within one layer; it can be applied to any type of video transmission over the Internet. The mechanism was designed for solving the problem of
packet loss in area of Multimedia over IP. They use network simulation ns2 for their experiments and simulations.

Figure 3: Multi-layer structure of a typical scalable video coding scheme.

As shown in Fig. 3, the base layer (lowest layer) is the most important one and so it gets the highest priority. Given the priority to each layer or each packet in a layer, a router can selectively drop packets from lower priority layers or packets of lower priority. This will isolate loss region to the enhanced layers of the current layer subscription. This will also minimize error propagation and also reduce redundancy due to proactive error control such as FEC. This mechanism is to minimize the number of losses in the base layer and subsequent enhanced layers. The problem is they did not mention about delay problem. This technique is very limited for one problem. For MoIP transmission, the two of problems must have been solved for make QoS become good and efficient.

3. Comparison Client-Side Transmission Technique

Currently, there are a few numbers of existing active transmission techniques available on the Internet. Some of them have already been commercialized by certain computer communication companies. In this research, there are six techniques will be studied in detail and to be compared with each other by using the comparative study method. The techniques that have been selected in this study are the Error Concealment Technique, Packet Loss Concealment Delay Jitter Model, Forward Concealment Algorithm, Waveform Reconstruction Technique and SPLIT. The techniques were design by year of 1996 until 2007. Some of the techniques have some weakness and need some enhancement to increase the performance and efficiency of quality of service for multimedia over IP application. Table 1.0 will show the comparison of the existing techniques.

<table>
<thead>
<tr>
<th>Multimedia Transmission Technique</th>
<th>Problem Type</th>
<th>Performance Features QoS for MoIP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>√</td>
<td>√</td>
</tr>
<tr>
<td>Packet Loss Concealment [2]</td>
<td></td>
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<tr>
<td>Forward Concealment Algorithm [4]</td>
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<tr>
<td>Waveform Reconstruction Technique [5]</td>
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<tr>
<td>SPLIT [6]</td>
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</table>

| √ = Yes                                           | X = No       | Blank = not mentioned |
| 1 = High/Max/Fast                                 | 2 = Moderate | 3 = Low               |
stable transmission rate, convergence time, stable operation and removal useless packet. The "√" symbol which signifies a "yes" answer, denote that the technique has a particular feature. Meanwhile, the "X" symbol for a "no" value, shows that the technique does not have the particular feature. The “1” symbol is for value high or maximum or fast. Meanwhile, the symbol “2” is value for moderate; symbol “3” is for low and the last one, symbol “4” value for bad.

By analyzing Table 1, it is found that there are six of client-side transmission techniques. Some of these techniques can solve both of the problem packet loss and delay issues. Almost of these techniques are use multicast for the communication. Among all of the features, the highest selected feature is the good quality of services in Multimedia over IP. Hence, this feature needs to be included in the development of the multimedia transmission techniques. The developer must make the techniques or algorithm itself good quality and performance, for example in the real-time live video transmission, the developer must consider the both of problem delay and packet loss in order to provide a feasible and consumer satisfaction.

4.1 Issues in QoS of Multimedia over IP

Real-time IP applications, such as videoconferencing and voice-over-IP are much more sensitive to network quality of service of data applications, such as e-mail and file transfer. Quality of Service (QoS) refers to intelligence in the network to grant appropriate network performance to satisfy an application’s requirements. For multimedia over IP networks, the goal is to preserve both the mission-critical data in the presence of multimedia voice and video and to preserve the voice and video quality in the presence of burst data traffic. Four parameters are generally used to describe quality of service: latency or delay, the amount of time it takes a packet to transverse the network; jitter, the variation in delay from packet to packet; bandwidth, the data rate that can be supported on the network; and packet loss, the per cent of packets that do not make it to their destination for various reasons. [10]

End-to-end latency refers to the total transit time for packets in a data stream to arrive at the remote endpoint. The upper bound for latency for H.323 voice and video packets should not be more than 125-150 milliseconds. The average packet size for video packets is usually large (800-1500 bytes) while audio packet sizes are generally small (480 bytes or less). This means that the average latency for an audio packet may be less than that for a video packet as interveningouters / switches typically prioritize smaller over larger packets when encountering network congestion. [10]

Jitter or variability of delay is refers to the variability of latencies for packets within a given data stream and should not exceed 20 - 50 milliseconds. An example would be a data stream in a 30 FPS H.323 session that has an average transit time of 115 milliseconds. If a single packet encountered a jitter of 145 milliseconds or more (relative to a prior packet), an under run condition may occur at the receiving endpoint, potentially causing either blocky, jerky video or undesirable audio. Too much jitter can cause inter-stream latencies which as discussed next. [10]

Packet loss is refers to the loss or desequecing of data packets in a real-time audio/video data stream. A packet loss rate of 1% produces roughly a loss of one fast video update per second for a video stream producing jerky video. Lost audio packets produce choppy, broken audio. Since audio operates with smaller packets at a lower bandwidth, in general, it is usually less likely to encounter packet loss, but an audio stream is not immune from the effects of packet loss. A 2% packet loss rate starts to render the video stream generally unusable, though audio may be minimally acceptable. Consistent packet loss above 2% is definitely unacceptable for H.323 videoconferencing unless some type of packet loss correction algorithm is used between the endpoints. Packet loss in the 1-2% should still be considered a poor network environment and the cause of this type of consistent, significant packet loss should be resolved. [10]

5. Approach

Our approach is to provide an efficient technique to reducing packet loss and delay for quality of service in MoIP application. The technique is providing a transmission balancing of performance features QoS in MoIP. The word balanced does not refer to equal number of performance features, but based on the technique itself. The technique will also embed the ISO quality of service that has mention in 4.1 in order to fulfill the basic quality performance suggested by ISO. Client-side technique is one category of loss techniques is passive methods that are implemented at the client side, which do not require any cooperation of the sender or increase the cost of transmission. It did not have any feedback mechanism. All of this technique did not have any removal useless packet mechanism. Useless packet is based on the fact that for packetised audio and video,
packet loss rate must be maintained under a given threshold for any meaningful communication. When packet loss rate exceeds this threshold, received audio and video become useless. Our approach technique is to design a technique that have fulfill the performance quality of service for MoIP such as TCP Throughput and TCP Friendliness. Our approach also will design one mechanism to remove the useless packet through the network congestion.

Based on the evaluation, toolkit like Network Simulation NS-2 is chosen for the implementation of data modeling, collision or interaction response and data attenuation to transmit on network for multi-users. Module SIP will be added in Network Simulation NS-2 in order to make exact simulation for the proposed technique.

6. Conclusions and Future Work

In this paper, we have conducted a survey of existing client-side transmission techniques. We have found that the transmission technique must have solved the both of problem delay and packet loss in order to fulfill the standard quality performance. We also found, there are eight performance features of quality of service for MoIP application. In this paper, our approach is to provide a technique that will be able to satisfy the user’s needs and requirements. We strongly believe that, to achieve such condition the performance features must be in balanced state. We will have to develop an efficient technique for quality of service Multimedia over IP (MoIP) which combining the best features and functionalities form MPEG-4 different from other techniques. For future work, we will increase the level of TCP Friendliness and focus with other features; that are Stable Operation and Convergence Time.

Acknowledgements

This work was supported in part by research contract between MOSTI and University Technology of Malaysia and in part by the Malaysia Government through the Ministry for University and Research (MOSTI).

References


