Comprehensive Study of Transmission Techniques for Reducing Packet Loss and Delay in Multimedia over IP

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Abstract
Quality-of-service (QoS) is still one of the major challenges in real-time communication over IP networks. That needs one solution mechanism to overcome these problems. In this paper, we perform a comprehensive study of the existing transmission technique. We categorize these techniques into three groups; client side technique, active technique and packet scheduling technique. We list out several features and functionalities for research continuity in this area. Some parameter will be selected to make the approach technique more effective for reducing packet loss and delay in MoIP application.

***Keywords**: Multimedia over IP, VOIP, Packet Loss, Delay Jitter, 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 [13].

Today, the underlying infrastructure of the Internet does not sufficiently support quality of service (QoS) guarantees. As a result, in the future users may have the capability to request specific end-to-end QoS even over the Internet, but this is not feasible today. In addition, many researchers take the stand that the cost for providing end-to-end QoS is too big, and it is better to invest on careful network design and careful network monitoring, in order to identify and upgrade the congested network links.

Unicast and multicast transmission of real-time multimedia data is an important component of many current and future emerging Internet applications, like videoconference, distance learning and video distribution. The heterogeneous nature of the Internet makes the unicast and multicast transmission of real-time multimedia data a challenge. The proposed mechanism uses real-time transmission protocol/real-time control transmission protocol (RTP/RTCP) [13] for the transmission of the multimedia data. The RTP protocol seems to be the standard protocol for the transmission of multimedia data over IP.

The rest of this paper is organized as follows: Section 2 presents the background and motivation.. Following section 3 presents the related work of proposed technique. In this section we describe some techniques that have related with the proposed technique for research continuity. Detailed performance features of transmission techniques presented in section 4. In this section also we present ISO of performance quality of service multimedia over IP. Following section 5 presents our approach for proposed technique. Finally, section 6 concludes the paper and discusses some of our future work.

2. Background and Motivation

Consequently, in recent years, the area of multimedia communication and networking has emerged not only as a very active and challenging integrative research topic across the borders of signal processing and communication, but also as a core curriculum that requires its own set of fundamental concepts and algorithms that differ from those taught in conventional signal processing and communication courses. Now, research has been going on from around the world to make quality of service in MoIP become more efficient.

In real-time communications, losses are a result of not only packets dropping over the network, but also late arrival for packets. This literature 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

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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 [1] [2] [3] [4].

The type refers to these techniques as "active" to differentiate them from those only employed at the client side [5] [6] [7] [8] [9] [10]. 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 [11] [12].

Because network conditions are dynamic and there is no Quality of Service (QoS) guarantee in the current MoIP, this means bandwidth adaptation becomes very essential for video distribution over the Internet. Generally, a no adaptive stream suffers from two deficiencies. First, it can under-utilize the available bandwidth or cause congestion collapse, which eventually degrades receiver video quality. Secondly, it can potentially lead to unfair bandwidth allocation, as the TCP traffic (which is the prominent traffic in current Internet) is adaptive. To solve such problem, one possible solution is to provide necessary QoS support via resource reservation or service differentiation.

3. Transmission Technique Classifications

3.1 Client-side Technique

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.

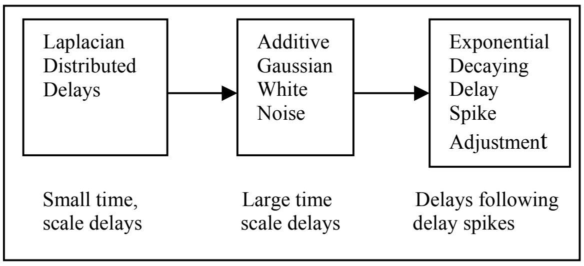

Figure 1: Jitter Model

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. [5] 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 design 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 algorithm do not need any additional bits, and thus they can use the already existing standard speech encoders without any modification.

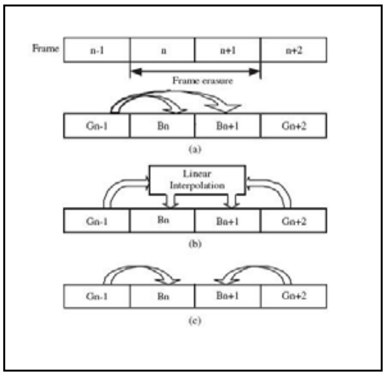

Figure 2: Procedure of each PLC algorithm; (a) forward PLC, (b) interpolative PLC, and (c) forward-backward PLC algorithms, where B and G mean a bad (lost) frame and a good frame, respectively.

Fig. 2 demonstrates the basic idea of three PLC algorithms such as forward PLC (F-PLC), interpolative PLC (I-PLC), and FB-PLC. In the figure, the n-th and (n+1)-th frames are lost. This algorithm is good for quality of services for MoIP but it only focus for voice and not for video. Some research and alternative mechanism additional need in order to this algorithm can be use for both of media, audio and video.

3.2 Active Technique

Ch Bauras and A.Gkamas [7] proposed a unicast mechanism for adaptive transmission of multimedia data, which is based on real time protocols. The proposed mechanism can be used for unicast transmission of multimedia data over heterogeneous networks, like the Internet, and has the capability to adapt the transmission of the multimedia data to network changes. The mechanism was design to solve the problem of packet loss and delay jitter. They evaluate the adaptive multicast transmission mechanism and compare it with a number of similar mechanisms. They use JMF (Java Media Framework) for the testing and built the unicast mechanism. Actually their focus is in the feedback analysis module which is responsible to analyze the feedback information that the client sends to the server, concerning the transmission quality of the multimedia data.

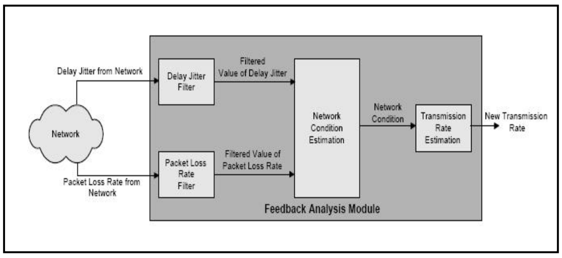

Figure 3: Feedback Analysis Module

Figure 3 displays the components of the feedback analysis module. The feedback analysis module extracts the packet loss rate and the delay jitter of the RTCP receiver report sent by the client and passes them through the appropriate filters (packet loss rate filter and delay jitter filter respectively).[14]

However this mechanism also has some weakness and need some enhancement to make this mechanism become more suitable for unicast transmission. The weakness is the mechanism did not have ability to avoidance useless packet through the network transmission. 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. This mechanism also did not mention about TCP Throughput. Our proposed technique will solve the problem of unicast technique.

Another technique is from Jim Wu and Mahbub Hassan [4] that proposed some algorithms to avoid Useless Packet Transmission (UPT) over single and multiple congested links have been analyzed previously in the context of a single multimedia flow sharing network links with TCP flows. The algorithm was design for solving the problem of delay and packet loss in multimedia over ip. Some simulation of MPEG-2 video had been used to test the effectiveness of UPT under various network scenarios. The main philosophy behind these algorithms is to drop all packets from a multimedia flow whenever the flow experiences a packet loss rate beyond the tolerable threshold. In this paper, we investigate UPT in the context of multiple multimedia flows. They find out that previously proposed UPT avoidance algorithms lead to poor link utilization, as they take action on each and every multimedia flow.

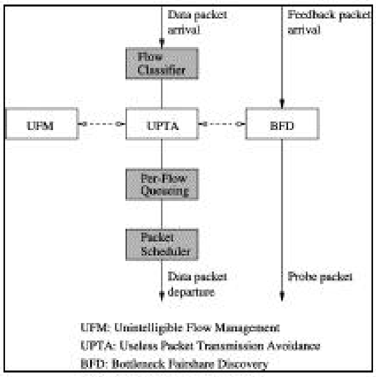

Figure 4: Algorithm of UPTA

Unintelligible flow management (UFM) is proposed to manage U flows efficiently in networks with multiple multimedia flows. It arbitrates which U flow should be converted into I flow based on available bandwidth. Fig. 4 illustrates the high-level architecture of UPTA enhanced with UFM.

These algorithms are useful to make the multimedia transmission become quality and effective. The TCP Throughput has become improve compare with various algorithms. However this algorithm was design for multicast of multimedia communication. So some alternative must be design for make the algorithm suitable for both of multimedia communication, unicast and multicast. In this paper, we proposed this algorithm for part in unicast transmission technique.

3.3 Packet Scheduling Technique

Other techniques for Multimedia over IP problems are packet scheduling technique. 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. The technique is new and was design in year of 2006. Y. Bai and M.R Ito [10] were proposed a new technique for minimizing network loss from users' perspective. The technique is designing for solving the problem of packet loss. Loss control is an essential issue for video transportation over IP networks**.** They proposed a new innovative use of packet scheduling to meet the loss constraints of video communication**.** A packet-scheduling scheme at a router that assures that intra and inter-stream loss requests for video transportation are met was developed and evaluated using real video data. The results demonstrate the advantages of the proposed scheme, in terms of loss performance and fairness.

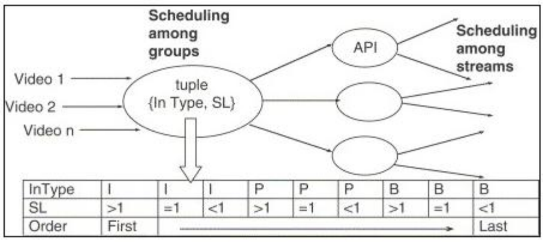

Figure 5: Multilevel priority adaptive packet scheduling (MPAPS) Scheme.

The MPAPS scheme, depicted in Fig. 5, uses two-level scheduling: Group Level and Stream Level. The group-level scheduling maps the video streams into the groups with different transmission priorities according to tuple {In Type, SL}. In Type represents the type of incoming packets and SL is a measure of the service level that a video receives from the network, and is defined as the ratio of the actual packet loss and the maximum allowable packet loss.

This scheme actually benefits for video application through manipulating video data streams such as high-level data format, which is the key distinctive feature of Active Networks. However, this feature is not available in a traditional network. Current routers have been designed to forward packets, not to process them**.** This scheme also cannot solve the problem of delay in multimedia transmission over IP network.

Another packet scheduling technique was design by Kyriakos Vlachos et al. [11]. They proposed a new burst assembly algorithm based on the average delay of the packets comprising a burst. This method fixes the average delay of the packets belonging to an assembled burst to a desired value T_{AVE} that may be different for each forwarding equivalence class (FEC). **T**he proposed method significantly improves the *delay jitter* experienced by the packets during the burst assembly process, when compared to that of timer-based and burst length-based assembly policies. Minimizing packet delay jitter is important in a number of applications, such as real-audio and streaming-video applications. They also find that the improvement in the packet delay jitter yields a corresponding significant improvement in the performance of TCP, whose operation depends critically on the ability to obtain accurate estimates of the round-trip times (RTT).

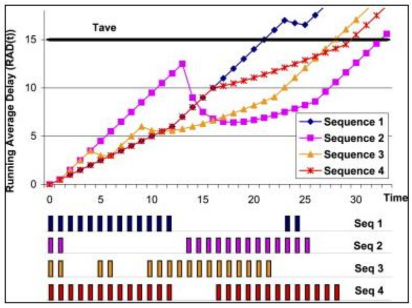

Figure 6: Running average delay of packets in the queue Running Average Delay ($RAD(t)$) for different packet sequences.

Since $RAD(t)$ increases proportionally to time with slope 1 when no packets arrive at the queue, the assembly process can be performed with the following timer-based algorithm that does not require continuous-real time monitoring of $RAD(t)$, but updates $RAD(t)$ only at discrete time instances, whenever new packets arrive at the assembly unit.

The algorithm is suitable for solving the problem of delay jitter. However, this algorithm cannot solve the problem of packet loss. The good of algorithm is that can solve both of problems, delay and packet loss.

4. Performance Features in Transmission Technique

Currently, there are a few numbers of existing transmission techniques available on the Internet. Some of them have already been commercialized by certain computer communication companies. In this research, there are twelve 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 Bottleneck Fairshare Discovery, Traffic Control Algorithm, Delay Jitter Model, Useless Packet Transmission Avoidance, Forward Concealment Algorithm, Waveform Reconstruction Technique, Unicast Technique, DMIF, SPLIT, Packet Scheduling, New burst assembly algorithms, and Forward error correction. The techniques were design by year of 2002 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 1.0: Comparison of Techniques

Multimedia Transmission Technique	**Techniques**			**Problem**		**Type**		**Performance Features QoS for MoIP**							
	Client Side	Active	Packet Schedule	Packet Loss	Delay	Unicast	Multicast	TCP Throughput	Feedback Mechanism	Complexity	TCP Friendliness	Stable Transmission Rate	Convergence Time	Stable Operation	Removal Useless Packet
Bottleneck Fairshare Discovery [1]		√		√	√		√	1	√	1	X	X			√
Traffic Control Algorithm [2]		√			√		√		√	3					X
Delay Jitter Model [3]	√				√		√						1	2	X
Useless Packet Transmission Avoidance [4]		√		√			√	1	√	1	X	X			√
Forward Concealment Algorithm [5]	√			√	√	√	√		X						X
Waveform Reconstruction Technique [6]	√			√					X				2		X
Unicast Technique [7]		√		√	√	√		X	√		2	X	1	2	X
DMIF. [8]		√		√	√	√			√	1	2				X
SPLIT.[9]	√			√			√	1		2	2				X
Packet Scheduling [10]			√	√			√		X	1			2		X
New burst assembly algorithms. [11]			√		√		√	1	X	2					X
Forward error correction [12]		√			√		√		√	1				2	X
Proposed Technique		√		√	√	√		1	√	2	2	1			√

√ = Yes X = No Blank = not mentioned
1 = High/Max/Fast 2 = Moderate 3 = Low

There are various characteristics that have been studied in order to find the similarities and the differences of performance features among each of the techniques. As depicted in Table 1, the table shows us the different of techniques, problem and type of multimedia communication. The tables also show that there are eight main performance features that have been used on the existing transmission techniques. The features are TCP Throughput, feedback mechanism, complexity, TCP friendliness, 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 four of the existing techniques come from the client-side technique; six of the existing techniques are from active technique, while the remaining comes from the packet scheduling technique. 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. The proposed technique is using unicast communication. Actually the main advantage of unicast is the fact that each receiver receives a separate stream from the sender as result the transmission of each stream is optimized for a specific receiver. In multicast one stream is used in order to server many receivers and there are limitation if the group of the receiver that receiver the same stream have difference capabilities (in terms of available bandwidth, processing capabilities etc).

The table shows us the comparison of 12 techniques of multimedia transmission. 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 ISO 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. [16]

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 intervening outers / switches typically prioritize smaller over larger packets when encountering network congestion. [16]

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. [16]

Packet loss is refers to the loss or desequencing 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. [16]

5. Approach

Our approach is to provide a hybrid technique to reducing packet loss and delay for quality of service in MoIP application. We will combine Unicast Technique [7] and Useless Packet transmission Avoidance (UPTA) Technique [4] as the new technique that has ability to increase the performance multimedia transmission over internet. 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.

We have found some issues in the unicast technique that must be improved in order to make this technique have capability of QoS in MoIP. Our proposed technique is enhancement from the unicast technique and has some improvement in the technique. The improvement is the unicast mechanism has the capability to remove the useless packet in the transmission over internet. 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. We formally defined the Useless Packet Transmission problem, and use the avoidance algorithm that called Useless Packet Transmission Avoidance (UPTA) [13]. Our new hybrid techniques called Unicast UPTA Technique. 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 comprehensive study of existing multimedia transmission techniques. We classified the current transmission techniques into three categories; client-side technique, active technique and packet scheduling technique. 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.

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