Adaptable Technique For Recovering Lost Internet Audio Packets

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Abstract: - With rapid progress in both computers and networks, real-time multimedia applications are now possible on the Internet. Since the Internet was designed to support traditional applications, multimedia applications on the Internet often suffer from unacceptable delay, jitter and data loss. Among these, data loss has the largest impact on quality. When the retransmission technique is used to recover lost audio packets, more obstacles, like the delay time resulted from lost packets retransmission, network traffics overloading, and buffer management, are faced. All above demonstrated problems are due to the real-time communication features. In this paper, we demonstrate a new technique that studies the audio packet's priority and it's effect on the transmission process. Hence, our technique arranges the audio packets depending on there importance. So, before deciding to retransmit a packet, we should determine if this packet could be recovered by other suitable technique or not. Hence, the number of retransmitted packets is decreased. Therefore, all above problems may be calmed. Some additional parameters, like a buffer management and a packet position are taken in the consideration. These parameters help us in selecting an ideal recovery technique for lost audio packets. Finally, NS2 is used to build a simulated environment for testing our technique.

Key-Words: - Multimedia Loss, Internetworking, Multimedia Streaming, Audio packet, TCP/IP.

1 Introduction

Data loss is a common problem in today's Internet. Network congestion and buffer overflow can all result in data loss, which results in a gap in the continuous data stream. Data loss in multimedia streams transmission can impact the continuity in the display. Data loss can occur involuntarily from network congestion or system buffer overflows, or voluntarily in order to avoid congestion at a client, server or network router. Audio conferences on the Mbone have reported data loss rates as high as 40% [1], [2]. Too much data loss can result in unacceptable media quality. To compensate for data loss, much work has been done to find effective data loss recovery techniques. There are two categories of data loss recovery techniques: sender driven and receiver-based. Each of them has its own strength and weakness. These techniques have proven to be fairly effective (but not sufficient) for audio stream data loss, but have yet to be applied to video [3], [4]. In this paper we try to select the ideal technique depending on the state of lost packet. This state can be determined using some new parameters like packet importance and position.

This paper is proceeding as follows: in section 2, our technique is introduced. The detailed simulation is demonstrated in section 3. Finally, the conclusion and the future work are reviewed in sections 4 and 5.

2 Our Technique

In the following subsections, big picture, description, and implementation of our technique are demonstrated.

2.1 Big picture

The basic idea of our technique is to divide the audio packets into groups. Each group is arranged depending on its importance in the audio file. The group importance is determined by its availability of providing clear data to the user (i.e. this provided data should be effective to the user).

2.2 Technique description

When the audio packets are transmitted through the Internet, it is so important to note the bottlenecks and network throughput in the packets path. In the real time communication the retransmission of lost audio packets causes some problems like delay and buffer management. These problems arise due to our need of time during the audio packets presentation and the large audio file size that needs respectively large network bandwidth. Hence; we will try to use more than one recovery technique at the case of audio packet loss. The selected recovery technique depends on importance and position of the audio packet.

2.2.1 Packet importance
We divide the audio file packets into four groups. These groups are arranged ascending each related to its performance. The first group contains the silent periods packets. The second group contains the low voice periods packets. The third group contains the high-speed voice periods packets. The fourth group contains the clear voice periods packets. We use two bits in the RTP (Real-time Transport Protocol) packet header to determine the importance degree. These two bits, called HPF (Header Priority Flag), are added at the RTP padding filed [5]. So, the RTP simplicity is still preserved. HPF is the indicator to packet priority. The HPF of first group is 00. The HPF of second group is 01. The HPF of third group is 10. The HPF of fourth group is 11, see table 1.

Header Priority Flag (HPF)	Description
00	Silent periods (SP)
01	Low Voice Dialogs (LVD)
10	High speed Voice Dialogs (HVD)
11	Distinct Voice Dialogs (DVD)

Table 1: Importance Priority table

Suggested technique is based on selection of the most ideal technique to recover the audio multimedia packets (i.e. more than one technique may be used at one audio session).

2.2.2 Packet position
The packet position is a new important parameter in selection of multimedia lost packets recovering technique. When the retransmission technique is selected, we should consider the delay time resulted by the packet transmission. Hence, we can make a relation between the packet consumption time and the retransmission delay time. By using this relation, an ideal technique can be selected to recover the lost packets in the multimedia system. For simplicity, the new relation can be stated as the following:
If: (the consumption time[1] – the loss detection time[2]) is greater than the retransmission delay time, we can select the retransmission technique as an ideal technique for recovering.
Else: By the packet importance, the ideal technique can be selected.
Note: There are more than one parameter determine the consumption and loss detection times

3 Experiments and Results
After developing our technique, a network simulator called NS2 can be used to evaluate its performance. NS2 is an object oriented discrete-event network simulator written in C++ with an Otcl interpreter as a fronted. It provides support for simulation of TCP/IP, routing and multicast protocols over wired and wireless local and wide area networks [6]. The target of this simulation part is to compare the performance of the audio transportation using our technique and only one recovery technique (old system) as regards the following parameters: late, error, and lost packets, delay jitter, end-to-end delay, **bandwidth utilization**, consecutive lost packets, and frequency distribution. In addition, some experiments to recommend the selected technique for each packet type are demonstrated.

3.1 Simulation Setup

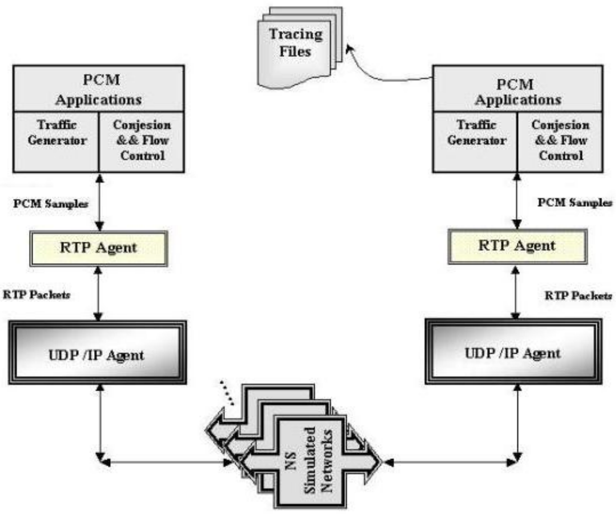

Figure (1): Simulation Model.

Definition	Value
Number of clients	100
Number of servers	1
Session bandwidth	Minimally 265 kb/s
Audio type	PCM
Coding rate	64 Mb/s
Buffer size	8000 byte
Number of sent packets	100000
Packet size	200 bytes
Simulation time	100 seconds

Table 2: Importance Priority table

In our simulation, we represent the old system by selecting different techniques to recover the lost audio packets. But, only one technique is used during our simulation time [4].

1- Consumption Time: The time at which the audio packet is used and presented to the end user.
2- Loss Detection Time: The time at which the packet loss is detected.

3.2 Results and comments

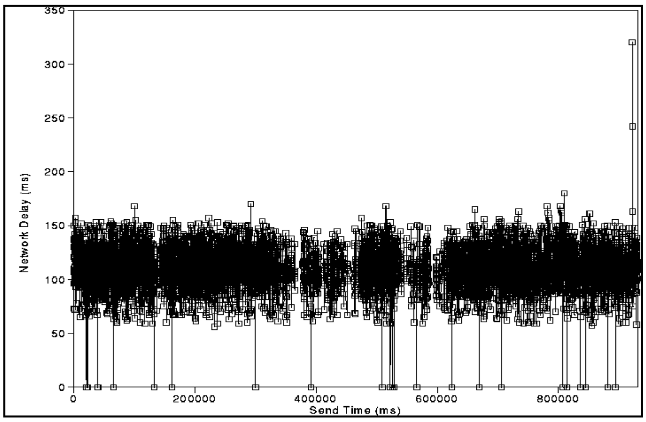

Figure (2): Network delay.

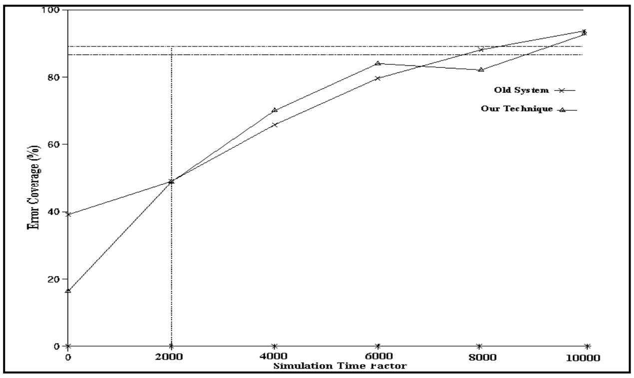

Figure (3): Packet error coverage.

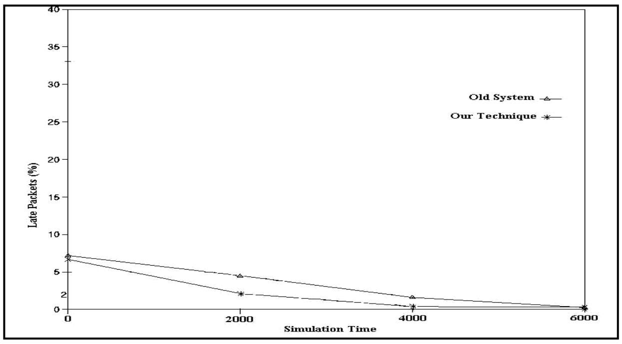

Figure (4): End – to – End delay

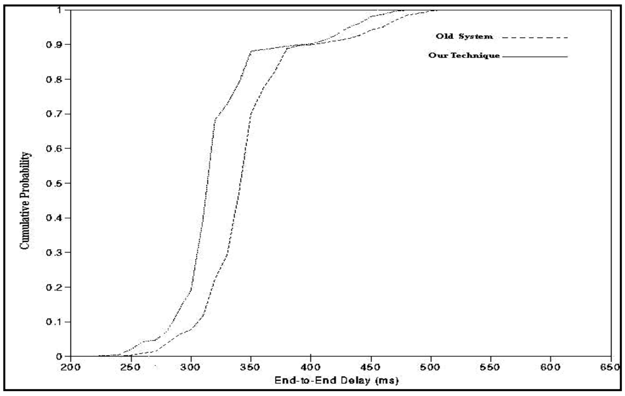

Figure (5): Late packets

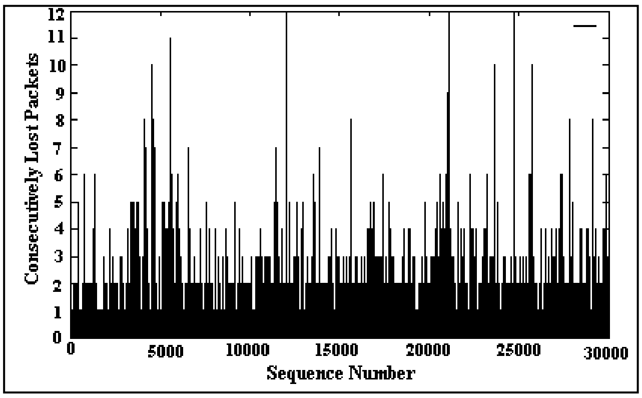

Figure (6): Consecutive lost packets.

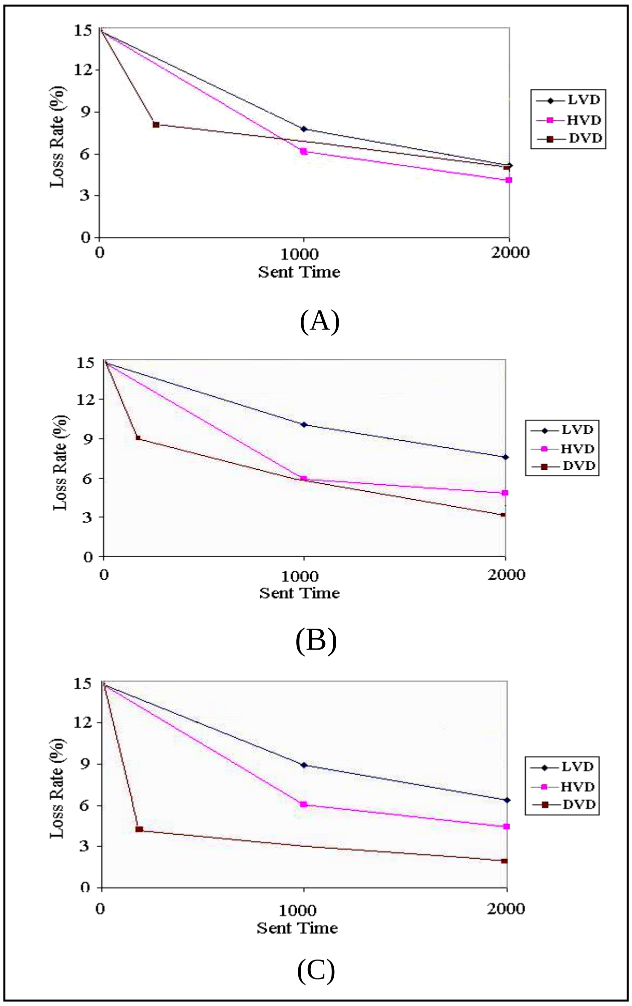

Figure (7): Three trials for loss rate measurement of different packet types

In our simulation, the network delay represents a very important factor. This is because; we should grantee that the additional functions didn't affect the system performance. Fig. 2 represents the delay time of a system supported by our technique. The number of packets that are recovered by silent periods is tested in fig. 3. Fig. 4 measures the end-to-end delay results from the buffer management and system overloading. Fig. 5 demonstrates the number of late packets that are considered lost in the old system. Fig. 6 represents the number of consequent lost packets that will help us to examine our technique as regards the quality of sent audio streams. This parameter is so urgent because if this number is high, the used recovery technique may be changed from the retransmission to the Repetition or the FEC (Forward Error Correction) due to the delay factor consideration [3], [4]. Fig. 7 shows three loss rate tested trials for audio streams contains different proportions of the LVD, the HVD, and the DVD. Table 3 contains a comparison between our proposed technique and other recovery techniques under different audio applications. This comparison shows that our technique has a high efficiency than other techniques. This is due to its sensitivity with audio streams transmission environment.

Application type	**Audio type**	**Recovery Techniques**	**Recovery Percentage**
Non-Interactive Application	Vat-PCM	Media-Independent FEC	89.254 %
		Our Technique	94.632 %
	Vat-GSM	Media-Independent FEC	88.174 %
		Our Technique	95.324 %
Interactive Application	Vat-PCM	Media-Specific FEC	90.045 %
		Our Technique	93.781 %
	Vat-GSM	Media-Specific FEC	91.954 %
		Our Technique	92.654 %

Table 3: Our adaptive technique *vs.* preferable recovery techniques under different audio types and network loads.

4 Conclusion

In our paper, we demonstrated a new adaptive technique to recover the lost audio packets. Our proposed technique arranges the audio packets depending on there importance. It is adaptable as its sensitivity with the ideal recovery technique that should be used to recover the lost audio packets at different audio session times. Finally, we simulate our adaptable recover technique to test it as regards other recovery techniques. We found that our proposed technique efficiency is higher than others. Also, additional computations in our technique didn't affect the system performance.

5 Future Work

We shall try to scale our technique to recover not only audio streams, but also video frames. Also, we shall try to test our scalable technique as regards different encoding types like JPEG and MPEG.

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