PERFORMANCE ANALYSIS AND PLAYOUT TIME ESTIMATION FOR MULTIMEDIA OVER INTERNET PROTOCOL (MOIP)

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ABSTRAK

This thesis presents the algorithms to estimate minimum buffering delay and playout delay. The possibility to use the buffering delay estimation in Multimedia Application at the receiver site will reduce the effect of jitter and will also optimize the packet loss. The increasing of Traffic of loaded generator will influence the network behaviour and affect to transmission data using video conferencing netmeeting application over the network connection.

***Keyword** : Playout Algorithm, buffering delay, delay, jitter, packet loss*

1. INTRODUCTION

The real time transmission for video and audio through internet is a considerably difficult to playout the received data at the receiver perfectly because of delay, variation delay (jitter) and loss of packet[4]. A typical delay and loss can not be estimated due to some parameters of the network and the distance between sender and receiver sites. But the values considered acceptable for the packet loss and latency are 0-20% and 5-500 ms respectively.

To reduce the effect of variable network delays, buffering delay at the receiver will smooth the jitter to the appropriate times. The use of a buffer time to generate a queuing time of packet as they received at the receiver will compensate the fluctuating end to end delays and variable network delays. Commonly the larger the jitter, the bigger the buffer delay time inserted to the playout time at receiver will be. Unfortunately this additional delay will impair the human understanding and the QoS, whereas the adjustment to the low playout times will cause some packets to arrive too late and also it affect the perceived QoS. Therefore the buffering delay has to tradeoff between the loss of information and the determining of the waiting time to playout.

There has been some investigation regarding the buffering delay in few decades to present the determination of the buffer delay time to produce the scheduled time to playout arrived packet at the receiver site[4][5][6][7]. Some investigations have applied the fixed buffering delay time which used the fixed method of determination to fix buffer size of delay time. It is easy to implement because it is just to determine the fixed buffer for each session and using the fixed buffer for each packet arrived. Unfortunately, it is unsatisfactory of an audio or video quality. On the other hand, some investigations have presented the adaptive method. This method performs continuous estimation of the network delays and dynamically arranges the playout delay at the beginning of each talkspurt. The arrangement is applied on the first packet of the talkspurt where all packets in the same talkspurt are scheduled to play out at fixed intervals following the playout of the first packet.

The main purpose of this investigation is to continue the presented investigation by Sakuray regarding the VoIP communication to Multimedia packet data transmission using video and audio data over internet protocol. In addition, the author more presents mathematical algorithms to optimize buffer delay used in application that involve the transmission of an audio and video packet using the interactive communication network application like netmeeting and video streaming file. The main objectives to this investigation is to show the presented algorithm can perform better application for the determination of buffering delay and playout delay time than the previous and to ensure the acceptable satisfactory user for communication regarding multimedia communication over internet protocol.

2. BUFFERING DELAY

2.1. THEORETICAL DETERMINATION OF BUFFERING DELAY

In Video and audio applications generate data packet with intervals time $\Delta \mathbf{t}$ in an active periods or talkspurt. A packet is transmitted at instant t_i and is received at instant a_i and executed at instant p_i, as shown in figure 1, the i-th packet of talkspurt k is sent at time t_k^i, it arrives at the receiver at time a_k^i, and is held in the smoothing receiver's playout buffer until time p_k^i, when it is played out. Inside a talkspurt, packets are equally spaced at the sender by time intervals of length Δt in seconds.

In Figure 1, a dropped packet due to a late arrival is viewed by a dashed line. A packet is artificially dropped if it arrives after its scheduled deadline p_k^i . This loss can be reduced by increasing the amount of time that packets stay in the playout buffer. An efficient playout algorithm must take into account the trade-off between loss and delay in order to keep both parameters as low as possible.

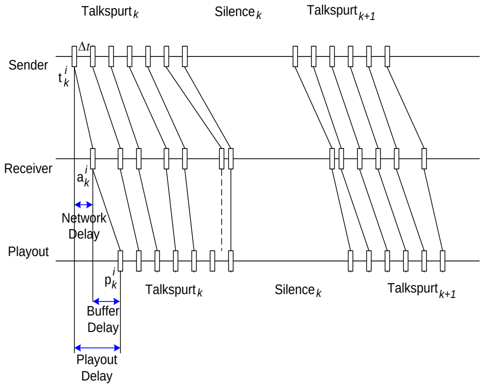

Figure 1. Transmission of packets of a talkspurt

2.1 PROBLEM

Network latency $|\ell|$ is necessary to be determined because it will represent the time difference existing between the clock transmitter and receiver. The time t_i will be reference at the receiver site, and will be replaced by $(t_i- \ell)$ in such a way that if $\ell<0$ ($\ell>0$) the sender client will be advanced (delayed) compared to the receiver client and in case $\ell=0$. The time clock of both sender and receiver site will be synchronized by using NTP (Network time protocol) or using GPS (Global Position System).

The execution time period *pi* of the packets must respect the periodicity Δt applied by the sender site to transmit them, i.e., $p_i - p_i\text{-}1=\Delta t$ for $i=2,...,n$, or even $p_i = p_1+(i\text{-}1)\Delta t$

The execution time is scheduled based on the buffering delay time according to the playout algorithm which is calculated each buffering delay of p_i in one talkspurt.

Notice that packets that are loss with index i in a talkspurt is mathematically characterized when $p_i < a_i$ or $p_i - (t_i\text{-}\ \ell) > L$, in order to avoid the packet loss the algorithm has to meet the equation $p_i - a_i \geq 0$ and $p_i - (t_i - \ell) \leq L$, for every $i \in N$. The first equation is known as playout restriction and the other is as latency restriction. This way, the playout delay (Pd) of the packet with index i is given by

$Pd_i = p_i - (t_i - \ell)$.

3. DETERMINATION OF BUFFERING DELAY AND PLAYOUT DELAY

In a talkspurt, if a packet with index i does not disturb the playout restriction, and the difference between the reception instant and sending instant overcomes the latency L, then the latency restriction is influenced by the packet with index i, no matter the buffer delay T used in talkspurt. In this session, consider a talkspurt with n number packets, having packet indexes given by $N=\{1,2,\ldots,n\}$. the first result we present is a property referred to the latency restriction

The equation of Latency restriction $p_i - (t_i - \ell) \leq L$, where $p_i = a_1 + T + (i-1)\Delta t$ and T is an arbitrary buffer delay. Notice that $p_i - (t_i - \ell) = p_i - a_i + a_i - (t_i - \ell)$, besides $p_i - a_i \geq 0$ and $a_i - (t_i - \ell) > 0$, so $p_i - (t_i - \ell) > L$.

The network conditions impact the determination of the buffer delay. The previous property figures this fact, viewing under which conditions the network can determine unavoidable losses, independently from the choice of buffer delay. In this investigation, the conditions that deal to dimensioning the buffer delay have been tried to control by the algorithm. Without loss of generality of the results in order to consider that $a_i - (t_i - \ell) > 0$, for every $i \in N$

In a talkspurt where a buffer delay T is inserted, no packet will theoretically be lost, if and only if $\max_{i \in N}\{\delta_i - (i-1)\Delta t\} \leq T \leq \min_{i \in N}\{\gamma_i - (i-1)\Delta t\}$ where $\gamma_i = (t_i - \ell) - a_1 + L$ for every $i \in N$ and $\delta_i = a_i - a_1$.

To prove the above equation, we can consider that if there is no packet loss at the talkspurt, this is equivalent to say that : $p_i - a_i = 0$ and $p_i - (t_i - \ell) > L$ for every $i \in N \Leftrightarrow p_i \geq a_i$ and $p_i = (t_i - \ell) + L$ for every $i \in N \Leftrightarrow a_i \leq p_i \leq (t_i - \ell) + L$ for every $i \in N \Leftrightarrow a_i \leq a_1 + T + (i-1)\Delta t \leq (t_i - \ell) + L$, for every $i \in N \Leftrightarrow \max_{i \in N}\{\delta_i - (i-1)\Delta t\} \leq T \leq \min_{i \in N}\{\gamma_i - (i-1)\Delta t\}$, where $\gamma_i = (t_i - \ell) - a_1 + L$, for every $i \in N$.

From the latter equation, we can note that $T_{\min} = \max_{i \in N}\{\delta_i - (i-1)\Delta t\}$ and $T_{\max} = \min_{i \in N}\{\gamma_i - (i-1)\Delta t\}$, respectively, the minimum buffer delay and the maximum buffer delay, to which the packets loss is not verified.

The insertion of the maximum and minimum buffering delay will prove the playout algorithm to meet the trade-off minimum loss and the maximum latency. With $T_{\min} = \max_{i \in N}\{\delta_i - (i-1)\Delta t\} = \delta_r - (r-1)\Delta t$, then:

$$p_i = a_1 + T_{\min} + (i-1)\Delta t, \forall i \in N \Leftrightarrow$$
$$p_i = a_1 + \delta_r - (r-1)\Delta t + (i-1)\Delta t, \forall i \in N \Leftrightarrow$$
$$p_i = a_r - (r-1)\Delta t, \forall i \in N$$

By considering that $i=r$, we have that $p_r = a_r$, the packet with index r, which defines T_{min}, is applied at the instant time of its reception. On the other hand, the part of equation with the maximum buffering delay will be applied to playout delay to get the bound of maximum latency, by considering $T_{\max} = \min_{i \in N}\{\gamma_i - (i-1)\Delta t\}$, then:

$$p_i = a_1 + T_{\max} + (i-1)\Delta t, \forall i \in N \Leftrightarrow$$
$$p_i = a_1 + \gamma_s - (s-1)\Delta t + (i-1)\Delta t, \forall i \in N \Leftrightarrow$$
$$p_i = (t_s - \ell) + L - (s-1)\Delta t, \forall i \in N \,.$$

Assuming $i=s$, we have that $p_s = (t_s - \ell) + L$, the packet with index s, which defines T_{max}, is executed with maximum latency.

The insertion of the maximum and minimum buffering delay will prove the playout algorithm to meet the trade-off minimum loss and the maximum latency.

4. RESULT

The data sets are analyzed using the playout algorithm based on the formula and some equations to determine the buffering delay time and playout delay time. All result buffering delay and playout delay time have been underestimated for each talkspurt in one traces session. Over the result of the buffering delay estimation and playout algorithm will consider as packet loss

The next figurre show the first talkspurt of netmeeting application and graphic of Playout algorithm result for one session from the data collection over VLAN network by using Load Utilization 5 %, 10%, 15%, respectively.

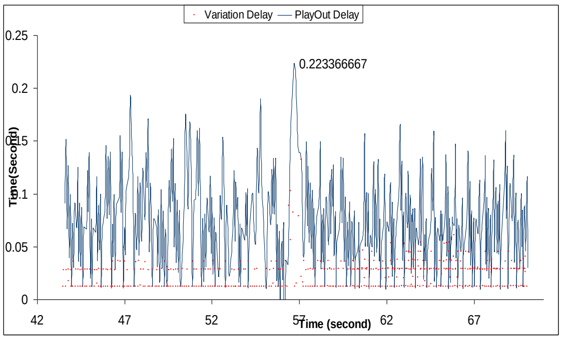

Figure 2. PlayOut Algorithm in one session trace using netmeeting application with Load utilization 5 %

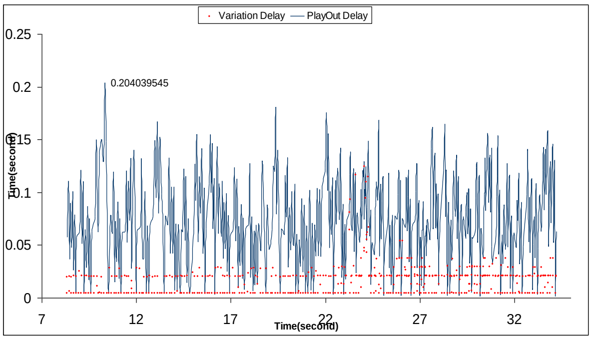

Figure 3. PlayOut Algorithm in one session trace using netmeeting application with Load utilization 10 %

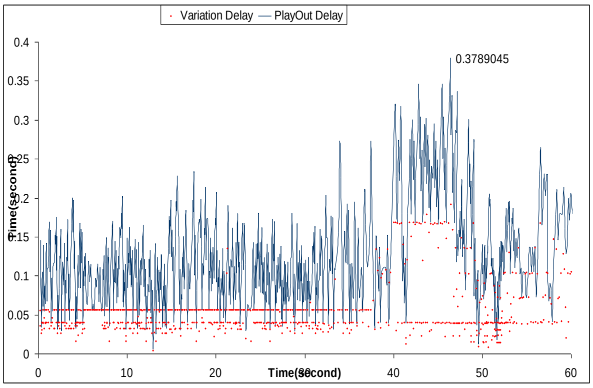

Figure 4. PlayOut Algorithm in one session trace using netmeeting application with Load utilization 15 %

5. CONCLUSION

In multimedia streaming file, the increasing traffic load generation by using the network analyzer will influence to the determination of the buffering delay time and playout delay time and packet loss. When the packet loss increases in the previous session, the buffering delay will be generated longer in the next session to reduce the packet loss.

The increasing buffering delay time and playout delay time will influence the packet loss of receiving packet in one session and our investigated playout algorithm also can keep lower tradeoff for packet loss and buffering delay time than the previous algorithm

6. REFERENCES

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