Robust Playout Mechanism for Internet Audio Applications

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Abstract

In Internet audio applications, delay and delay jitter affect applications' quality of service mostly. Since packet delays are different and changing over time, the receiver needs to buffer some amount of packets before playout. Therefore, the amount of buffered packets and timing of playout are very important for the performance of the applications. Here we adopt an auto-regressive (AR) model for estimation of packet delay and deploy a robust identification algorithm for adjustment of parameters of AR process. In our preliminary experiments, this robust algorithm leads better performance when the noise is correlated and/or non-stationary, and also it is robust to model uncertainties.

1. Introduction

In Internet audio applications such as real-time voice communication and packet radio service, delay and delay jitter affect applications' quality of service (QoS) mostly. In order to alleviate the problem caused by unpredictability of delay and delay jitter in Internet, the receiver usually needs to buffer some amount of packets before it actually plays them. Little buffering tolerates only small delay jitter, whereas excessive buffering causes large startup delay. Therefore, the amount of buffered packets and timing of playout are very important for the performance of the applications.

Figure 1 shows the sample sequence of operations on a sender and a receiver of an audio application. Two solid lines represent the sequence number of audio packets sent and received respectively, and two dotted lines represent the sequence number of played packets by the receiver. Delay jitter during transmission causes the uneven arrivals of the packets on the receiver. In order to smooth and hide such delay jitter, the receiver delays the initiation of playout of received packets. t_1 and t_2 represent the two different playout time. In the case of t_1, some of packets are delivered to the receiver after their playout time, therefore they are dropped without playing. In the case of t_2, the receiver has much delayed startup time t_2-t_0 than the case of t_1, which is t_1-t_0.

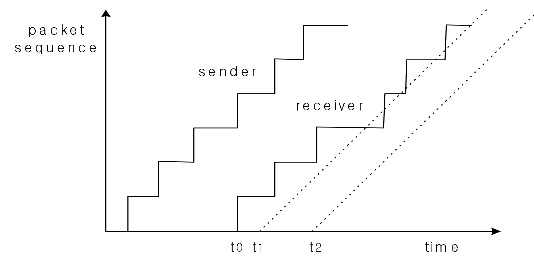

Figure 1. Sequence of operations on audio application

2. Previous work

The main objective of adaptive playout problem is to choose the small startup time (t_1 in the Figure 1) keeping reasonably acceptable packet loss rate.

In the previous work, the basic algorithm [1] of delay estimation used in audio applications such as NeVot [2] and Vat is influenced by TCP round-trip time estimation [3]. In addition, there have been many researches for playout buffering and playout delay adjustment problems [4][5]. However, most AR-based algorithms to estimate a certain parameter use fixed weighting factor, such as α in (3) in [1], the performance of the estimators is highly dependent on how correct the weighing factor is in a given situation, so they are not so adaptive to the situation change. Even though there have been modifications of the previous algorithm, such as [4][6] to make it adaptive and robust, mostly their approaches to adjust the weighting factor are heuristic, therefore, it is hard to justify the adaptability and analyze their performance behavior.

3. Robust playout mechanism

We consider the estimation of packet (flow) delay, d_{n+1} at time $n+1$ as the AR process as follows.

$$d_{n+1} = \sum_{i=1}^{p} \alpha_i d_{n+1-i} + \Phi_n \tag{1}$$

where d_i is the i-th packet (flow) delay, α_i's are the parameters of the AR process to be identified, and Φ_n is an unknown noise sequence. Here, p is called the order of AR process. We use the following vector notation instead of (1).

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$$d_{n+1} = \alpha^T \eta_n + \Phi_n \qquad (2)$$

where $\alpha = (\alpha_p, ..., \alpha_1)$, $\eta_n = (d_{n+1-p}, d_{n+2-p}, ..., d_n)^T$.

3.1. Robust identification approach

For the special case when the driving noise Φ_n is i.i.d. Gaussian, the least squares (LS) approach can be used to obtain an optimal estimator for both its variance and parameter α. However, when the noise is correlated and/or non-stationary, LS is not robust to modeling imprecision. This assumption on the noise is reasonable for delay and jitter in Internet. Therefore, [7] adopted a robust identification approach, which usually shows better performance than any heuristic method and is more robust to modeling imprecision. After having robust identification of α and variance of Φ, v, then similar to [3], the end-to-end delay for playing time can be derived as:

$$d_{e2e,n} = d_n + \beta\sigma_n \qquad (3)$$

Based on [7], the recursive relation of $\hat{\alpha}_n$, estimation of α_n for n-th packet is as follows.

$$\hat{\alpha}_{n+1} = \hat{\alpha}_n + (d_{n+1} - \hat{\alpha}_n^T \eta_n)(\textstyle\sum_{n+1} + \eta_n \eta_n^T)^{-1} \eta_n \qquad (4)$$

$$\hat{\alpha}_0 = \alpha_0 \qquad (5)$$

where $\sum_n$ is a sequence of p*p-dimensional positive-definite matrices. The recursive relation of $\sum_n$ is:

$$\textstyle\sum_{n+1} = \sum_n + \eta_{n-1}\eta_{n-1}{}^T - \frac{1}{\gamma^2} Q_n \qquad (6)$$

$$\textstyle\sum_1 = \overline{Q}_0 - \frac{1}{\gamma^2} Q_0 \qquad (7)$$

Here, if we let $\overline{Q}_0 = Q_0 = I_p$, where I_p is the $p \times p$ identity matrix, and $Q_n = \eta_{n-1}\eta^T{}_{n-1}$, then we get the following update for $\sum_n$.

$$\textstyle\sum_{n+1} = \sum_n + (1-\gamma^{-2})\eta_{n-1}\eta^T{}_{n-1} \qquad (8)$$

$$\textstyle\sum_1 = (1-\gamma^{-2}) I_p \qquad (9)$$

Here γ is a parameter of the algorithm and should be larger than one.

For $\hat{\sigma}_n^2$, estimation of variance σ_n^2, either the method based on spectral estimation introduced in [7] or ordinary method using low-pass filter can be used.

3.2. Robust playout algorithm

Based on the estimation of α and σ_n^2 in [7], the overall robust playout algorithm is shown as follows.

```
1. For each talkspurt,
2.   For each i-th packet,
3.     Calculate α̂_i and σ̂_i^2 .
4.   End
5.   Calculate playout time t_p = d̂_{n+1} + cσ̂_n
     (where d̂_{n+1} = α̂_n^T η_n)
6. End
```

4. Conclusions and future work

This paper focuses on adaptive playout problem for Internet audio applications, and presents our playout algorithm using robust identification approach. In the preliminary simulation on the traces in [4], our algorithm shows better estimation on delay jitter and loss packet loss rate [8]. Currently the estimation approach is applied to other multimedia video streams. In addition, this robust estimation framework can be applied to other applications such as transport retransmission timeout, available bandwidth estimation, etc.

5. Acknowledgement

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6. References

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