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A Scheme for Jitter Elimination in IP Real-time Service

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***Abstract*—With the development of wireless communication technology, more and more conversation is based on IP protocol, the time-delay jitter is unavoidable in IP conversation due to it provide a best-effort service. In this paper, a scheme uses jitter buffer to eliminate jitter is proposed, the adaptive jitter buffer is set to solve the jitter problem and control the packet loss rate. In the meantime, LMS(Least Mean Square) algorithm is used to adjust the length of buffer by predicting the average jitter in next period. To prove the performance of the scheme, time-delay and packet loss rate are simulated on OPENT. Compared with the existing method, the scheme is proved that it can provide lower packet loss rate and smaller jitter.**

Keywords- time-delay jitter; adaptive buffer; LMS algorithm

I. INTRODUCTION

Recently, more and more services based on the IP protocol appear in our life, like TVIP (Television over Internet protocol) and VOIP (Voice over Internet Protocol), etc. Due to IP protocol provides a best-effort service and the processing time for packets transmission in the network is different, the time-delay jitter has been produced and will cause the packet loss. Packet loss is a severe problem in IP Real-time services, such as some words cannot be heard clearly or the flashing in video communication. Therefore, the elimination of jitter is the guarantee of real-time services.

In order to eliminate jitter, Kang Zhou proposed a method sets fixed buffer to cache the packets and receiving data in steady rate to eliminate time-delay jitter[1]. But the problem in this method is the fixed length of buffer. The improper length for network condition will cause unnecessary time-delay or packet loss.

To solve above problems, Kevin M. McNeill proposed the adaptive buffer instead of the fixed buffer, the length of buffer can be adjusted to suit different conditions in network, but the deficiency is that no algorithm had been put forward to judge how long the length should be changed so that to precisely control the packet loss rate, these defects will lead the scheme not to meet the requirements of real-time service.

In real-time services, packet loss rate and time-delay are both important parameters which are contradictory: packet loss is caused by jitter, so increasing of buffer length will reduce the packet loss and amplify the time-delay, vice versa. Therefore, both parameters should be controlled in a reasonable range

In this paper, LMS (Least Mean Square) algorithm is used in receiving procession to predict the jitter in network and the length changed are based on the prediction value. Through this scheme, the performances will meet the request of real-time services.

II. THE DESIGN FOR ADAPTIVE JITTER BUFFER

A. The Overall Structure of Adaptive Jitter Buffer

Fig.1 shows the overall structure of the adaptive buffer, when packets received, they are processed in the buffer to restructure the sequence they were send before, so the buffer can solve the jitter eliminate problem in IP real-time services. In once sampling period, jitter buffer samples and judges the jitter in network, then LMS algorithm is used to predict the jitter in next period, thus adjust the buffer length to adapt the network condition.

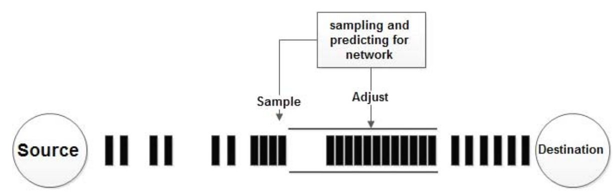

Figure1. The overall structure of the adaptive jitter buffer

B. Introduction of LMS Algorithm

LMS (Least Mean Square) is an adaptive algorithm which only needs to adjust internal parameter through the observing value instead of getting the statistics of signal and noise. LMS algorithm optimizes the specific objective function, thereby, the performance of the system will be improved .

As Fig.2 shows, the LMS system consists of several adaptive transversal filters, the value of $x(n+1)$ can be obtained from $x(n)$. Filter AF_1 is converged to the optimal weight vector. The output of AF_1 is $y(n) = \hat{x}(n)$,which is the k steps prediction value for $x(n-k)$. The filter AF_2 has the same weight vector as AF_1 , thus $\hat{x}(n+k)$ is the k steps prediction value for $x(n)$.

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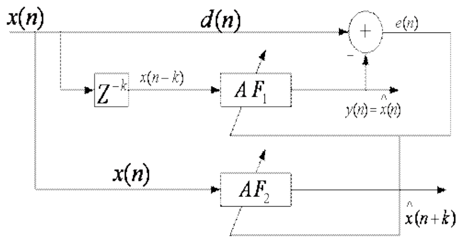

Figure2. LMS adaptive prediction system

The LMS algorithm is put forward for predicting the average jitter in next sampling period in network, which is used to judge the length change of buffer. In this paper, the average jitter $\overline{J}$ for m periods as the input signal in LMS system is set to predict the jitter in next period, so that the change of length can be calculated accurately according to the prediction.

The expression and calculation for $\overline{J}$ as follows: Assuming that L packets are received in each period. The local system time $T_{receive}$ is regarded as the packet arrive time when the packet is received, in the meantime, the packet send time $T_{transmit}$ is gained from upper-layer protocol (RTP or GTP protocol). For the ith packet, the jitter of which can be expressed as:

$$J(\mathrm{i}) = T_{receive} - T_{transmit} \qquad \mathrm{i}=0,1,2...\mathrm{L} \tag{1}$$

The average jitter in mth period is:

$$\overline{J_m} = \frac{J(0)+\mathrm{J}(1)+\mathrm{J}(2).....+\mathrm{J}(\mathrm{L}-2)+\mathrm{J}(\mathrm{L}-1)}{L} \tag{2}$$

After m periods, the average jitter is:

$$\overrightarrow{J(\mathrm{m})} = [\overline{\mathrm{J}_0}, \overline{\mathrm{J}_1}, \overline{\mathrm{J}_2}, ...\overline{\mathrm{J}_{m-2}}, \overline{\mathrm{J}_{m-1}}]^T \tag{3}$$

The average jitter for m-1 periods is:

$$\overrightarrow{J(\mathrm{m}-1)} = [\overline{\mathrm{J}_0}, \overline{\mathrm{J}_1}, \overline{\mathrm{J}_2}, ...\overline{\mathrm{J}_{m-2}}]^T \tag{4}$$

We assume that the average jitter in the past period $\overline{J_{m-1}}$ is marked as V_1. V_1 and $\overrightarrow{J(\mathrm{m}-1)}$ are put into the LMS system to predict the average jitter in next period, the predicted value is defined as V_2 . V_1 and V_2 are used to judge the length change of buffer

C.The Design of Receiving Procession for Jitter Buffer

Combined with the time-delay requirements of real-time services set by 3GPP[3], the length of buffer dynamically adjusted within 20 ~ 400 ms is deemed to reasonable.

A design of receiving procession is proposed in this paper, which is the key to eliminate jitter. Before introduction of the procession, assumptions have been put forward:

S_{send} : The latest packet sequence number in sending procession

S_{new} : The latest packet sequence number in receiving procession.

P : The packet loss rate for N packets in statistics.

$P_{threshold}$: The threshold of packet loss rate set before receive procession, like 6% 7% 9%, etc.

T_{buffer} : The sampling period in receiving procession.

The parameters J , L_{buffer} , V_1 and V_2 have been described in Chapter B. The receiving procession Fig.3 shows can be indicated by the structure:

(1)Initialization: Before starting real-time services, a jitter buffer is created for receiving packets and T_{buffer} is initialized, Jitter buffer counts P and judges whether P is smaller than $P_{threshold}$: if $P < P_{threshold}$, the length of buffer cannot be adjusted, else the length should be initialized to an appropriate value.

(2)**Update value of** J : at the end of each T_{buffer} , calculate $\overline{J_m}$ in mth T_{buffer} and $\overrightarrow{J(\mathrm{m})}$ in m T_{buffer} (use formula (3)), V_1 and $\overrightarrow{J(\mathrm{m}-1)}$ are put into the LMS system to predict the average value of jitter V_2 in next T_{buffer} .

(3)**Procession for packets:** when a new packet is received, if the buffer is full, drop the packet, else comparing S_{send} and S_{new} . If $S_{new} \le S_{send}$ or S_{new} has been existing in the buffer, drop the packet. Else save this packet in buffer and wait for next packet.

4)Increase the length of buffer: When packet loss or end of each T_{buffer} , the length of buffer is adjusted by comparing $\overline{J_m}$ with L_{buffer} , if $\overline{J_m} > L_{buffer}$, the length of buffer increases 0.02s then start a new T_{buffer} .

(5)Decrease the length of buffer: If $\overline{J_m} < L_{buffer}$ and $P \le P_{threshold}$, compare the value of V_1 with V_2 . When $|V_1| > |V_2|$, length of buffer is decreased by 0.04s ,else 0.02s and then start a new T_{buffer} .

The length of buffer increases or decreases per time is defined in [4] which mentions the proper buffer length in real-time services. Through the process of the receiving, the jitter has been eliminated. The performances will be shown in next chapter.

III. SIMULATION

A. The Environment Of Simulation

In OPENT, the software we used for simulation, the scene has been established by the gateway(GateWay), base station (Femto) and user (UE) is shown in Fig.4.

The gateway sends a packet in every 0.02s plus a random time-delay to simulate a congestion condition in network, the base station receives these packets and adjusts the length of buffer according to schemes which are proposed respectively by Kevin M. McNeill[1] and this paper.

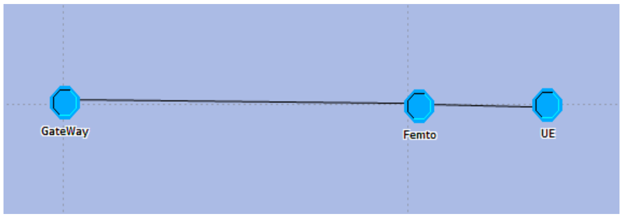

Figure4. The scene for packets transmission

B. The Results of Simulation for Existing Scheme

The packet loss rate in the existing method by Kevin M. McNeill [1] as Fig.5 shows, the horizontal axis indicates simulation time(minutes) and the vertical axis indicates the percentage of packet loss rate. For there is no set threshold of packet loss rate, packet loss rate is floating between 9%-10%, but cannot be smaller. This defect will lead packet loss not meeting the requirement of some real-time service, and also an inadequate result in jitter elimination.

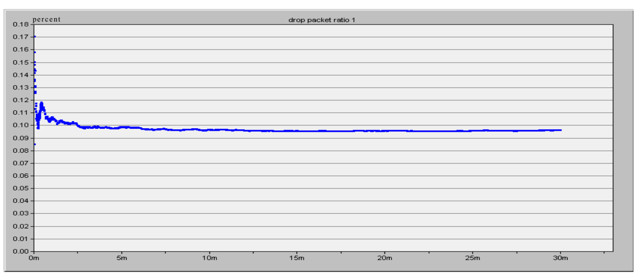

Figure5.the packet loss rate for existing method

C. The Results of Simulation for Proposed Scheme

In the proposed scheme ,the threshold of packet loss rate is set to precisely controlling the rate and further eliminating the jitter. In this part, we set three threshold of packet loss rate which are $P_{threshold} = 6\%, 5\%, 4\%$. In this scene, the packet loss rate and time-delay are shown in Fig.6 and Fig.7 respectively.

In Fig.6, the horizontal axis indicates the simulation time(minute) and the vertical axis indicates the percentage of packet loss rate. The packet loss rate can be precisely controlled on each threshold value, so that can meet any jitter request for the packet loss rate in real-time services

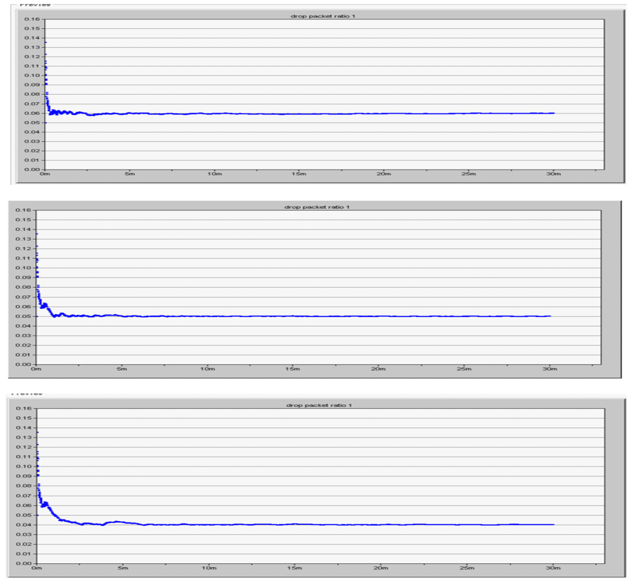

Figure.6 the packet loss rate in simulation ($P_{threshold} = 6\%\ 5\%\ 4\%$)

The time-delay is the same as length of buffer empirically, in the Fig7, the horizontal axis indicates the simulation time(minutes), and the vertical axis indicates the length of buffer(seconds).The results show that the changes of $P_{threshold}$ has obvious effect on the time-delay. With the decreasing of $P_{threshold}$, the time-delay increases. But as the result shows, time-delay in the scheme can be accepted in real-time services, and it floats in a small range which means the jitter is small.

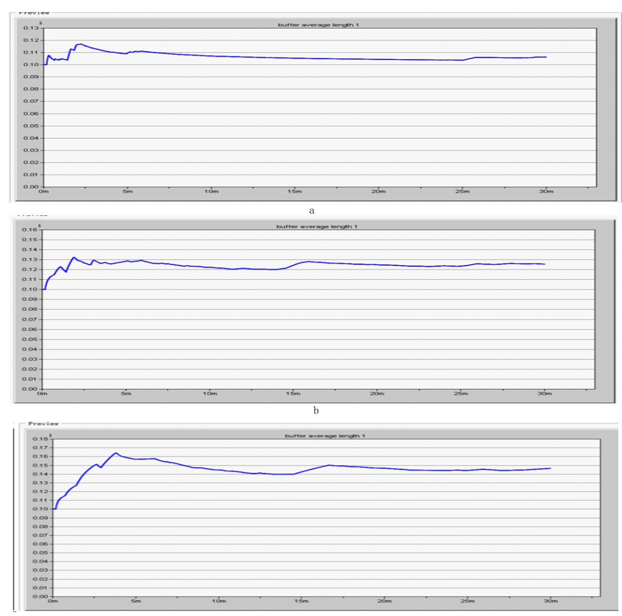

Figure7.the buffer length in simulation ($P_{threshold} = 6\%\ 5\%\ 4\%$)

In summary, the proposed scheme is able to eliminate jitter further than existing method, it can control packet loss rate more accurate and provide better performance in real-time services.

IV. CONCLUSION

In this paper, a scheme is proposed for jitter elimination in real-time services. It can provide an accurately control of packet loss rate and small jitter in a reasonable time-delay range. The adaptive jitter buffer structure is introduced and the LMS algorithm is used to predict the jitter which can be regarded as the judgment of length changing .The design of receiving procession is put forward and build a simulation on OPENT. Two schemes are compared in the simulation, ,and the results proves that the proposed scheme can provide good QOS for real-time services.

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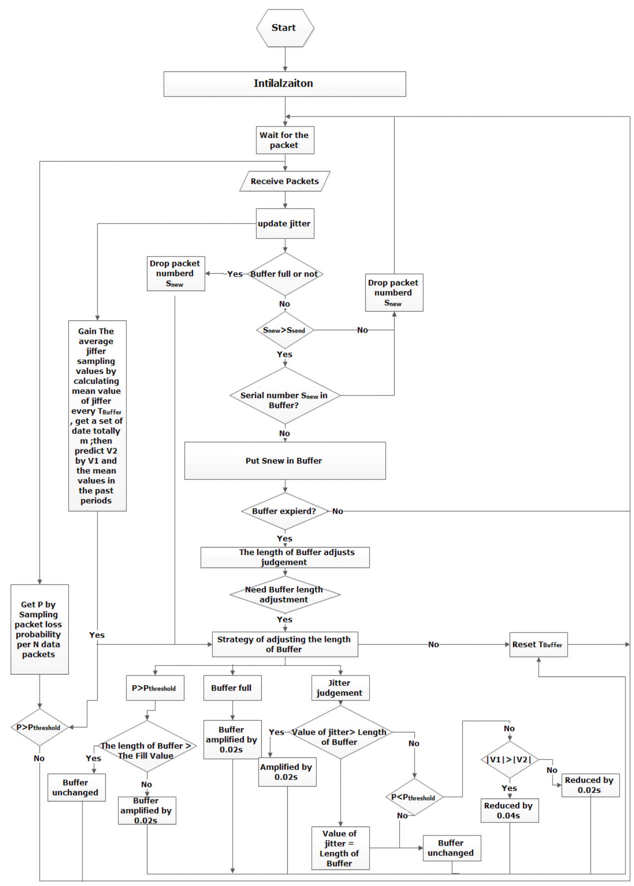

Figure2. The reception of the adaptive jitter buffer