Journal of Theoretical and Applied Information Technology
20th July 2013. Vol. 53 No.2

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

A SURVEY ON VIDEO STREAMING OVER MULTIMEDIA NETWORKS USING TCP

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ABSTRACT

Video Streaming has been used recently for various interactive services. Video Streaming is playing the content of the video in real time. Today's research is focused on video streaming through TCP. Regardless of the retransmission procedures, researchers focus on TCP Streaming due to its reliable service. This paper focuses more towards TCP Video Streaming and various congestion control algorithms used for video streaming.

Keywords: *TCP, Video Streaming, Congestion Control*

1. INTRODUCTION

Figure 1. describes an architecture of TCP Video Streaming. The Video server splits the video into streams. The Video Streams are then transmitted through TCP to the clients. There are 'n' clients connected to the internet. The clients are connected with varying bandwidth limits. The challenge is towards efficient video streaming with minimal loss and delay.

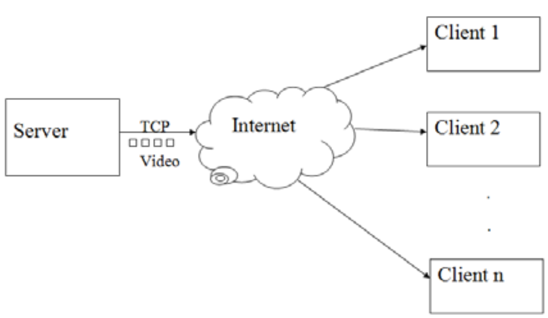

Figure 1. TCP Video Streaming Architecture

2. FEATURES OF TCP AND UDP IN VIDEO STREAMING IN WIRELESS NETWORKS

Chi-Fai Wong et al [1] discussed the comparisons of TCP and UDP video streaming over wireless networks. The Advantages and Disadvantages of TCP/UDP for streaming videos over wireless networks are given below.

2.1 Disadvantages of Using UDP in Wireless Video Streaming

a. UDP is an unreliable and non congestion control protocol. Packet loss occur during video streaming in UDP because of its unreliable service and UDP are in need of the error correction and retransmission mechanisms to avoid packet loss. However, the above mechanisms have certain draw backs. It is very difficult to implement efficient retransmission mechanisms and it increases overhead at the client side .Forward Error Correction schemes increases delay in the encoder part at the server end and error concealment schemes are not suitable for burst error in the case of wireless channel.

b. High packet loss occurs with varying bandwidth in the case of wireless networks .

c. In video streaming, some frames like I frames and synchronization bits have to be protected. Due to wireless errors, there is a chance for data loss in UDP. Such data loss may reduce the quality of the video.

d. Some protocols such as RTP use UDP, but many applications prefer the usage of TCP because of the firewall penetration problem with UDP.

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2.2 Advantages of Using TCP in Wireless Video Streaming

a. TCP is a reliable congestion control protocol. Error recovery and error concealment mechanisms are not required .

b. TCP provides Selective frame transmission and the proxy can be designed in such a way that it provides flexibility in selecting the frames to be transmitted.

c. Frame overhead may occur in RTCP and RTP protocols and TCP avoids frame overhead .

d. TCP is friendly and bandwidth adaptable in nature. Even if congestion occurs TCP utilizes the resources using that bandwidth

e. TCP can be implemented in applications because it penetrates the firewall with the use of HTTP.

To our knowledge we have inferred, though there are some drawbacks in TCP such as retransmission which can be avoided by buffering mechanisms, TCP would give better Quality of service due to its fairness. Since UDP is a non congestion control protocol, they are in need of adaptive control mechanisms. The disadvantage of UDP and the Advantages of TCP in wireless video streaming have been given in Table 1.

Table 1. Summary of Features of TCP and UDP in Wireless Video Streaming

Features	TCP	UDP
Reliability	Reliable. Uses Sequence number and Ack. Avoids Packet loss	Unreliable. No sequence number and Ack so Packet loss occurs
Error Recovery and Error Concealment	TCP is a Congestion control protocol. No need of error recovery and error concealment mechanisms	UDP is a non congestion control protocol. FEC, Retransmission mechanisms are needed
Firewall penetration	TCP penetrates with the firewall by means of HTTP	UDP does not penetrate firewall
Selective Frame transmission	TCP provides selective frame transmission	UDP do not provide selective frame transmission

3. ADVANCED TCP FOR WIRELESS ENVIRONMENT

Danny De Vleeschauwer and David Robinson [2] have proposed the HTTP adaptive video streaming mechanisms and the authors have suggested the researchers to follow the following features to make TCP more robust. H. Inamura et al [3] states the additional features of TCP.

1. Client Side Buffer: The size of the buffer at the client end has to be chosen such that no delay occurs while streaming video

2. Early Congestion Notification (ECN): ECN signals are used to notify whether packet loss occur due to error or congestion

3. Selective Acknowledgement (SACK): Instead of sending the entire packets for retransmission, the lost packets alone are retransmitted.

Multimedia streaming via TCP-analytical performance study have been discussed in Bing Wang et al [4] .The best results are obtained in this paper, if its throughput is two times the bit rate. Kim and Ammar [5] worked on stored media streaming using TCP based on receiver buffer size. TCP streaming model and quality of experience has been assessed by Jinyao Yan et al [6].

4. CONGESTION CONTROL ALGORITHMS IN THE INTERNET

To achieve TCP Fairness and friendliness, the congestion control methods should have the following capabilities

i. It should maintain Network Stability
ii. Effective Bandwidth utilization is needed
iii. Smooth playback is to be followed

5. CLASSIFICATION OF TCP FRIENDLY CONGESTION CONTROL ALGORITHMS

The classification of congestion control algorithms have been depicted in Figure 2. Qian Wang et al [7] has classified the Congestion Control algorithms in the internet.

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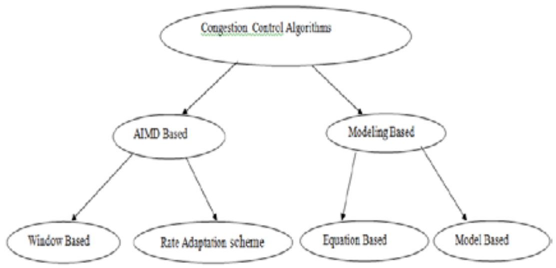

Figure 2. Classification of TCP Congestion Control Algorithms

5.1 AIMD Congestion Control Scheme

According to Qian Wang et al [7], AIMD Congestion Control Algorithm is given by AIMD(A,B)

(or)

$$E: W_{t+s} \leftarrow W_t + A,\ A>0 \qquad (1)$$

$$F: W_{t+\delta t} \leftarrow (1-B)\, W_t,\ 0<B<1,$$

Where
E-Window size increase
F-Window size decrease
W_t –Window Size at time t
S-Round Trip
A=1 packet
B=1/2

Disadvantage of TCP AIMD

- Oscillations in the sending rate
- Sending rate is halved for a single packet drop

5.2 Window Based Approach

D. Bansal, H. Balakrishnan [8] presents a TCP Friendly congestion control schemes by modifying the Equation1.The parameters m and n are used in equation 2.

$$E: W_{t+s} \leftarrow W_t + AnW_t^{m};\ A>0 \qquad (2)$$
$$F: W_{t+\delta s} \leftarrow W_t - B\, W_t^{n}\ ;\ 0< B<1$$

where,
m and n are the parameter space.

The Equation 2 becomes TCP Friendly if it satisfies the following

m+n=1 and n<=1

Disadvantage

- It is difficult to configure the parameters (m,n) space according to the required application.

5.3. Rate Based Approach

Reza Rejaie et al [9] has discussed the Rate Adaptation protocol for real time streams. The rate based scheme are not acknowledgement based when compared with window based scheme. Packet sending is not based on acknowledgements but it is based on the sending rate timer. Rate Adaptation Protocol(RAP) and LDA are the examples of this scheme. RAP is used in unicast playback of real time systems. RAP uses Inter Packet Gap(IPG).IPG doubles multiplicatively if congestion occurs or it decrease additively if congestion does not occur.

Advantage

- Adaptation of the sending rate reduces oscillations
- Reduces traffic in real time application

Disadvantage

- Sending rate is reduced due to single packet loss, so performance is reduced in the case of real time applications

5.4. Modeling Based Approach

J. Padhye et al [10] proposed the Modeling schemes to solve the above drawbacks. The Response Function of this scheme is given in Equation 3.

$$TR=Q/(S\sqrt{2/3p}+t_n(3\sqrt{3p/8}).p(1+32p^2)) \qquad (3)$$

Where,
TR - Maximum sending rate
Q - Size of the packet
S - Round Trip
t_n - Time out value

5.5 Equation Based schemes

These schemes are developed for Unicast traffic .The transmission rate depends upon rate of loss and round trip time. These parameters are adjusted to achieve a reasonable transmission rate. According to this technique, the client has to send the feedback . TFRC is an example for this scheme

Advantage

- Sending rate is reduced to half for successive loss events

5.5.1 Design goals of TFRC

S. Floyd et al [11] proposed the TFRC for Unicast traffic. The authors have discussed several

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

design goals. Transmission rate is reduced to half on successive loss events and if the feedback is not received from the client end, again the transmission rate is reduced to half. So, the transmission rate purely depends upon the successive loss rate and the feedback. The parameters are calculated based on the Response Function as stated in Equation 3.

Advantages

- TFRC reduces its sending rate more smoothly thereby reducing the oscillations in the sending rate .

5.6 Model Based

This scheme is based on the the formulation

Throughput=f(Wmax ,R,p,B),

Where,
Wmax-Receiver Declared window size
R-Transmission Rate
p-Loss Rate Probability
B-Round trip time

TFRCP is an example for this scheme

Advantage

- Reduces oscillations in the sending rate
- Suitable for real time multimedia applications

6. DYNAMIC TCP FRIENDLY AIMD ALGORITHM (DTAIMD)

Lin Cai et al [12] proposed the DTAIMD algorithm which is based on AIMD(α,β)

Step 1 If(cwnd≥1/1- β) //congestion window is large
Step 2 α=3(1- β)/1+ β
Step 3 else if(cwnd= =1) //congestion window is minimal
Step 4 α=1
Step 5 else //if 1<cwnd≤1/1- β
Step 6 α=3/2cwnd-1

The results in this paper showed that the above algorithm is suitable for multi rate multimedia applications. The scenario has been tested with different (α,β) pairs and the proposed DTAIMD algorithm yield better throughput when compared with TCP.

7. DESIGN OF MEDIA TCP FRIENDLY CONGESTION CONTROL (MTCC) APPROACH

Hsien-Po Shiang and Mihaela van der Schaar [13] proposed a new Congestion Control approach for Wired IP Networks. The authors have proposed an algorithm for independent and interdependent packets. Window size is modified without altering the design of the receiver side. The design of this approach are given below.

1. The RTP packets are classified into M classes

2. MTCC uses the retransmission mechanism of TCP but the expired packet in the buffer is not retransmitted

3. MTCC adjust the congestion window based on the Transmission Scheduler and Network Estimator

4. Transmission Scheduler selects the number of packets to be sent in k time slots

5. The Network Estimator updates the packet Loss rate

8. CONCLUSION AND FUTURE WORK

The paper has discussed about the TCP Video Streaming and the congestion control algorithms used in the internet. Though TCP has certain limitations with video streaming, the buffer management schemes and congestion control methods involved in it improves the video streaming which is still a challenging problem to be solved. Past researchers have focused on the window size and retransmission procedures and our future direction of research will be towards improving quality of service in video streaming through TCP.

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