IMPROVING THE INTERACTIVE REAL-TIME VIDEO COMMUNICATION WITH NETWORK PROVIDED CONGESTION NOTIFICATION

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Abstract

A typical interactive real-time video communication scenario imposes high demands on the transport channels and sometime the network does not match the demand from the application which results in congestion. In those situations, a fast and efficient end to end adaptation scheme can play a vital role to secure service performance. In this paper the benefits of using network assisted congestion control for RTP/UDP are discussed with the help of some simulation results.

Introduction

In today's Internet, a larger portion of the traffic is video related and a fair amount of that portion is coming from interactive real-time video communication. With the advance of new technologies the amount of traffic related to real-time video communication is expected to increase. A typical interactive real-time video communication scenario imposes high demands on the transport channels. It requires low delay, error free packet transmission and high throughput from the transport network. Current network infrastructures and deployments already try to handle this pressure. However, sometime the network does not match the demand from the application which results in congestion. The effects of congestion include packet loss and rapidly increasing delay (Figure 1 and Figure 2) which results in choppy video, lip synchronization problems and more. In those situations, a fast and efficient end to end adaptation scheme can play a vital role to secure service performance by ensuring high quality throughout the session, even in extreme conditions, and can keep the ongoing services by avoiding session termination. However, this all will depend on how well the application predicts or detect the network congestion in time and adapt accordingly. In this paper, the benefits of using network assisted congestion control for RTP/UDP are discussed with the help of some simulation results.

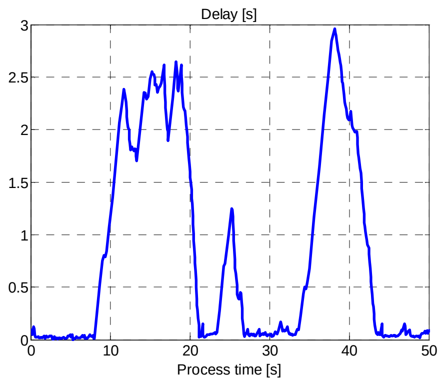

Figure 1. Field test of congested network, no AQM

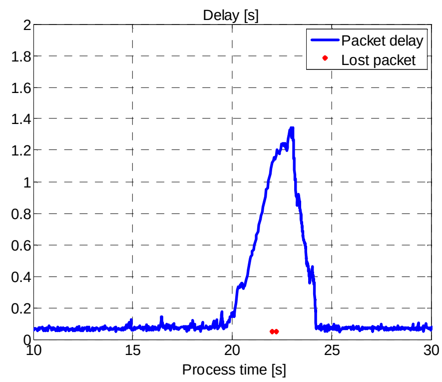

Figure 2. Simulated user in a congested network

Network provided congestion notification

Perhaps the most popular way to deliver interactive real-time video over the Internet is using RTP [1] over UDP/IP. Since none of these protocols have built-in congestion control, the media adaptation is usually done at the application layer. This media adaptation can be made more efficient by providing congestion related information to the application from the network. This is because the network has better knowledge about current network load and resource scarcity and can predict the congestion quite accurately. One of the ways to convey the congestion information to the application is using Explicit Congestion Notification (ECN) [2].

Recently ECN usage for RTP over UDP/IP [3] has been approved as standard in IETF which gives an unambiguous way to convey the network congestion indication for the RTP receivers. Using RTCP, this information can be shared with the sender of the flow. The application can then use this information to adapt the media to handle the upcoming congested situation.

There are other implicit mechanisms to predict or detect the network congestion. Such mechanisms use packet loss, delay or jitter observed in the communication path. These metrics do not really provide aggregated information about the network congestion (we can only assume that the metrics are proportional to the bitrate used) and are heavily dependent on how the application perceives the information. On the contrary, ECN gives explicit information about the network congestion. The congested network node can combine various local as well as cell wide information (in case of cellular network) to set the ECN bits in the IP header to indicate that congestion will occur in communication path if no preventive measures are taken by the end points. Figure 3 shows how a base station (eNodeB) in a LTE network can detect upcoming congestion and start setting ECN bits in the IP header before the congestion becomes severe.

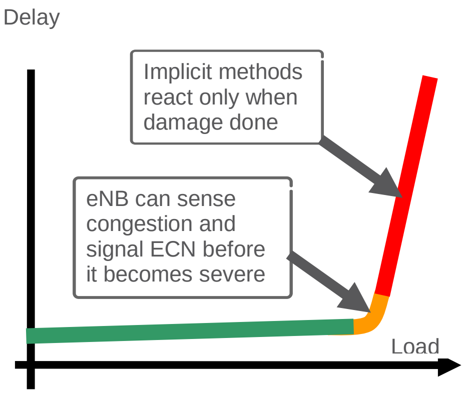

Figure 3 congestion prediction

Simulation results

In this study, a typical interactive video communication scenario is simulated in a LTE simulator which uses RTP/UDP for the media transport. The RTP AVFP profile is used both for fast feedback and to enable ECN feedback signaling. AQM (Active Queue Management) is disabled in the network to simulate a delay based implicit congestion control scheme. The delay based congestion control scheme has 200ms jitter as delay threshold However, it should be noted that with AQM end points will observe more packet losses which may not be good for interactive real-time video communication. LTE network is chosen because mobile broadband access covers a big portion of the current Internet and generates huge amount of mobile data traffic. The assumption here is that if we can use ECN to solve congestion related problems in the dynamically changing network properties of a LTE system then the same solution should also work in other less complex network systems.

Figure 4 shows the average bitrate of the users in the simulation system using ECN based rate adaptation which is lower than adaptation using an implicit mechanism based on delay. It shows that with ECN marking the congestion is spread over the current flows which involves more users in the adaptation and helps better sharing the cost of congestion. By involving more users we can see that the ECN based adaptation is able to improve the performance for all users in the system (Figure 5). This show how an ECN based adaptation and careful ECN configuration in the network node can improve overall system performance.

If we look at the troubled user of Figure 2 in the simulated network, we can see the delay based rate adaptation (Figure 6) improves the situation slightly compared to no adaptation with fixed bitrate. The problem is the late reaction (see the lower portion in Figure 6). It can also be mentioned that packet loss based rate adaptation is not very helpful here as packet drops are quite rare even though the delay is quite high. Delay based adaptation can possibly be made more sensitive but in reality this is tricky to achieve as ”normal” delay and jitter varies greatly depending on the network one is using. On the other hand, ECN based rate adaptation (Figure 7) has the main benefit of initiating the rate adaptation in advance of the actual congestion. In this particular case in the simulation, the congestion was predicted almost 6 seconds earlier than the implicit mechanism. This is mainly because ECN can exploit knowledge about local network conditions.

It is a general assumption that the use of AQM with packet drops will reduce the delay. However, this will result in higher packet loss rate which is especially bad for video. In contrast, ECN can in theory provide zero congestion related packet loss.

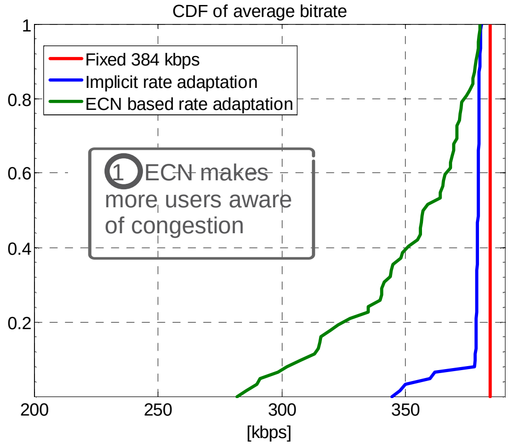

Figure 4 CDF of Avg. bitrate

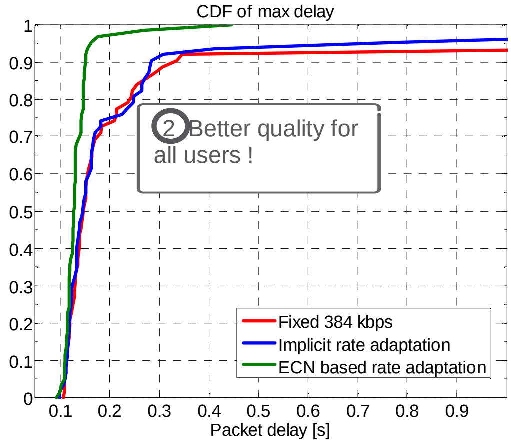

Figure 5 CDF of max delay

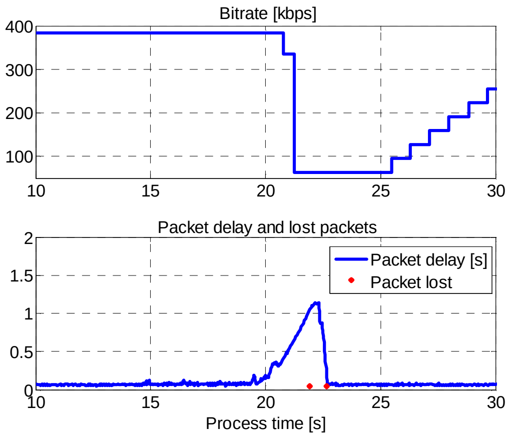

Figure 6 Delay based implicit adaptation

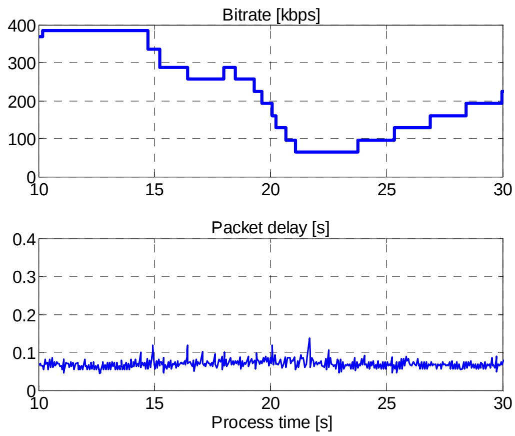

Figure 7 ECN based explicit adaptation

Conclusion

In this paper we have discussed the benefit of using ECN based congestion control mechanism in interactive real-time video communication. The simulation results in a LTE simulator shows that ECN based adaptation can certainly improve the media quality in a real-time communication by means of delay and packet loss. It can capitalize on available network information in the network and predict the congestion earlier than implicit mechanisms. This results in higher user satisfaction through out the whole system improving the overall user experience of the provided service.

Reference

[1] Schulzrinne, H., Casner, S., Frederick, R., and V.Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[2] Ramakrishnan, K., Floyd, S., and D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", RFC 3168, September 2001.

[3] Magnus Westerlund et.al., "Explicit Congestion Notification (ECN) for RTP over UDP", http://tools.ietf.org/html/draft-ietf-avtcore-ecn-for-rtp-08.