Towards User-Centric Rate Adaptations for VoIP Traffic*

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I. Introduction

Voice traffic by nature is high in data volume and sensitive to network impairments. Bandwidth requirement for voice calls is, therefore, higher. An increasingly serious dilemma, as the amount of VoIP traffic increases, is that it will not be fair to give priority to the multimedia traffic and starve its best-effort counter part; yet the voice data delivered might not be perceptible if each voice call is limited to the rate of an average TCP flow. We propose to approach this problem from a user-centric view and adapt the sending rate of the voice calls based on the user satisfaction. Such a ***user-friendly*** rate adaptation mechanism will also be ***congestion-friendly***, although not strictly TCP-friendly [1].

There has not been a promising rate adaptation mechanism yet for user- and congestion-friendly VoIP services. In [2], the authors propose a TCP-like rate adaptation mechanism. In that, the sending rate is adjusted in an additive-increase-multiplicative-decrease fashion based on the measured loss rate and delay. [3] [4] propose to use speech quality indicator such as E-Model as the metrics to adapt the sending rate. However, the number of parameters required to calculate the E-Model value is high, twenty-one to be exact. Some parameters such as mouth-to-ear delay are not easily accessible. Many of the parameters are over-simplified in the existing works.

Exploiting the User Satisfaction Index (USI) derived from a recent study on Skype traffic [5], we propose a rate adaptation mechanism for Skype voice call which requires only 3 parameters that can be measured and computed in real time. Although the USI is specific to the voice codec used by Skype and might not be generally applicable for all VoIP calls, our work serves as a necessary next step towards user-centric rate adaptation for VoIP traffic on the Internet. The preliminary experimental results show that rate adaptation mechanisms based on USI may result in more satisfying voice communication experience.

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II. User-Centric Rate Adaptation

To conserve network bandwidth, voice data are often encoded or compressed before they are transmitted. Typically, the perceived voice quality will be better when the compression rate is lower. I.e., more details are preserved in the compressed voice stream. Such high resolution voice streams will require more network bandwidth. However, a higher sending rate may incur losses or longer delays in an already congested network. Simply sending voice at a high sending rate may not necessarily result in better user experience. On the other hand, reducing the sending rate whenever packet losses or increasing packet delays are detected reduces also the amount of voice details transmitted. This may not necessarily lead to better user experience either. As observed in real Skype traffic [5], user satisfaction (USI) degrades as the amount of voice data received decreases and the level of network congestion increases:

$$USI = 2.15 \times \log(R) - 1.55 \times \log(J) - 0.36 \times RTT$$

R denotes the receiving voice data rate, J the receiving rate jitter, and RTT the round-trip time. While a higher sending rate may lead to a higher receiving rate, it might result in a higher receiving rate jitter as well. A lower sending rate might help preventing packet losses and stabilizing delays, but it might also result in a lower receiving rate. Gearing for user satisfaction, we adopt USI as the metric to tune up or down the sending rate. Another advantage of using USI is that the three parameters required to calculate the USI value are accessible and easy to compute in real time.

As a proof of concept, we devise a USI-based rate adaptation mechanism following the additive-increase-multiplicative-decrease principle. R, J, and RTT are measured and the corresponding USI is calculated periodically. The source will halve the sending rate when reported USI decreases for H consecutive times. Otherwise, the source will increase the sending rate by a step once every I intervals. Similar to the loss-based mechanism proposed in [2], the source also fine-tunes the sending rate for mildly congested networks. When the receiving rate jitter increases for D intervals, the sending rate will be decreased by a step.

III. Preliminary Results

To demonstrate the potential of the USI-based mechanisms and to highlight the significance of pursuing sophisticated designs, we conducted four sets of experiments. The experimental network consists of: (1) a source node operating a particular rate adaptation mechanism, (2) a destination node receiving data and assist in the measurement of receiving rate, receiving rate jitter, packet loss rate and round-trip delay, and (3) a DummyNet-controlled link between the source and destination nodes to emulate different levels of congestion. The controlled link is configured to run in the bandwidth of 80kbps, 65kbps, 50kbps, 35kbps, 20kbps, 35kbps, 50kbps, 65kbps, and 80kbps for 300 seconds each. Each experiment, therefore, takes 2700 seconds in total. The four mechanisms compared are CONST, LOSS, USI-I, and USI-II. In the CONST experiment, we use a straight-forward constant bit rate source. LOSS is our implementation of the exact algorithm with exact parameter setting proposed in [2]. USI-I and USI-II are our proof-of-concept mechanisms. USI-I and USI-II differ in the parameter settings. They are configured with H=5, I=4, and D=3, and H=5, I=3, and D=3 respectively. Figure 1 plots the resulting USI for the four mechanisms, Figure 2 the packet loss rate, and Figure 3 the round-trip delay respectively.

Potential of the USI-Based Mechanisms. Comparing CONST, LOSS, and USI-I, we can see that when the available bandwidth is abundant, there is no distinct difference in the three mechanisms' performance. When the available bandwidth is mildly sufficient or insufficient, we observe that USI-I outperforms the other two in all three metrics demonstrating the potential of USI-based mechanism for better user satisfaction. LOSS is not far worse and performs significantly better than CONST in the mildly sufficient bandwidth case. In the extreme case, however, the sending rate of LOSS, the state of the art mechanism, fluctuates and results in a lower level of user satisfaction.

Sensitivity of the USI-Based Mechanisms to Parameter Settings. Although USI-I is shown promising providing a higher level of user satisfaction, from the results of USI-I and USI-II, we also observe that the performance of the USI-based mechanisms are sensitive to the parameter settings. A slightly smaller I value in USI-II causes the source to increase the sending rate more aggressively and, as a result, leads to undesirable sending rate fluctuations. In fact, not just the time scale to adapt the sending rate, the granularity of the steps and the rate adaptation criteria may also impact the stability of the rate adaptation mechanism and ultimately the user experience. Systematic exploration of the design space is necessary for a robust rate adaptation mechanism. This is the issue we intend to pursue in the future.

IV. Summary and Outlook

Voice and best-effort data are different in nature. Sending voice data in a TCP-friendly fashion might not lead to perceptible quality when voice and best-effort traffic content for network bandwidth. We think that adapting the VoIP traffic sending rate based on user satisfaction is a promising alternative to explore. Taking Skype, one of the most popular VoIP services on the Internet, as an example, we show with the preliminary results that a user-centric congestion control mechanism may result in a higher level of user satisfaction and Skype, although runs its own rate adaptation mechanism already [6], may potentially be improved by a robust rate adaptation mechanism. A more thorough investigation on whether existing TCP-friendly congestion control mechanisms will result in satisfying VoIP service, whether the existing rate adaptation mechanism used in Skype is adequate, and whether a robust mechanism for consistent user experience can be devised is the subject of our future study.

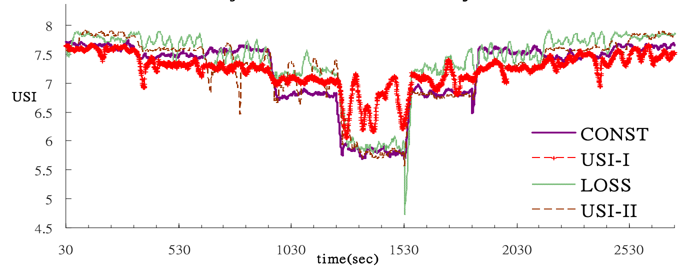

Figure 1 *USI result of the Four mechanisms*

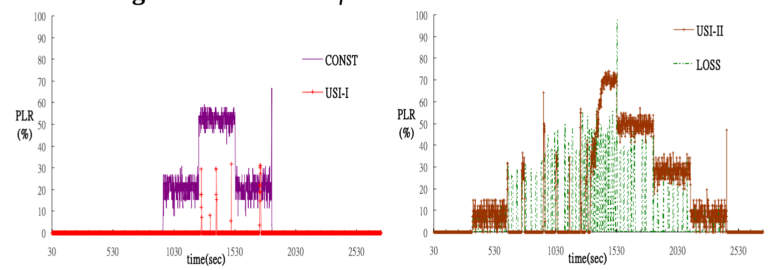

Figure 2 *Packet loss rate of the Four mechanisms*

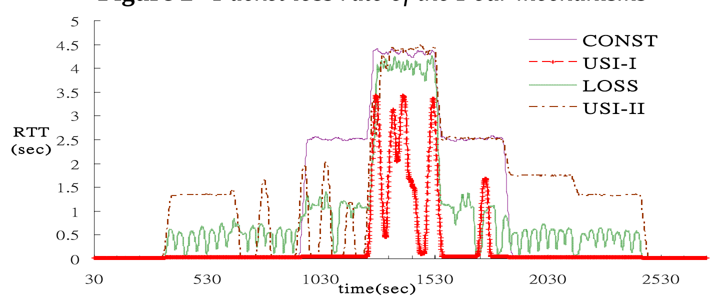

Figure 3 *RTT of the Four mechanisms*

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