# Exponential Quantization: User-Centric Rate Control for Skype Calls

Yu-Chuan Yen†, Cing-Yu Chu†, Chien-Nan Chen*, Su-Ling Yeh†, Hao-Hua Chu†, Polly Huang‡
†‡National Taiwan University
*University of Illinois at Urbana-Champaign
†{b99901079, r99921035, suling, haochu}@ntu.edu.tw
*‡{kcircnc, huang.polly}@gmail.com

## ABSTRACT
As Skype has become popular and a profitable business, the long-standing problem of how to deliver Skype calls deserves a serious revisit from an economic viewpoint. This study proposes a rate control mechanism for Skype calls that satisfies *more users* and satisfies *users more* than the greedy-naïve mechanism, as well as the mechanism implemented in Skype.

## Categories and Subject Descriptors
C.2.2 [**Computer-Communication Networks**]: Network Protocols – *applications.*

## General Terms
Performance, Design, Experimentation, Human Factors.

## Keywords
VoIP, QoE, Skype, Rate Control, Proportional Fairness

## 1. RATIONALE
Recently, many [1][2] have investigated how users perceive the quality of Internet services under different network conditions and found the relationship between QoS metrics and user experience very likely logarithmic. [3] confirmed through subjective experiments and established a logarithmic model between the bitrate and user score for Skype calls. The model is plotted in Figure 1, where the x-axis depicts the bitrate and the y-axis the mean opinion score (MOS). One could observe that as the call bitrates increases, the perceived quality improves. One could also observe that the perceptual *improvement* is relatively lower as the bitrate rises. Given the same amount of extra bandwidth – Δ, a user of low-rate call feels stronger the improvement, as opposed to that of high-rate call. Aiming at cost-effective use of network bandwidth, the rate control mechanism should increase the bitrate aggressively when the bitrate is low and conservatively when the bitrate is high. This gives rise to the idea of Exponential Quantization (EQ), where call bitrates are regulated at a number of levels and the levels are quantized in a non-uniform manner.


*SIGCOMM'13*, August 12–16, 2013, Hong Kong, China.
ACM 978-1-4503-2056-6/13/08.

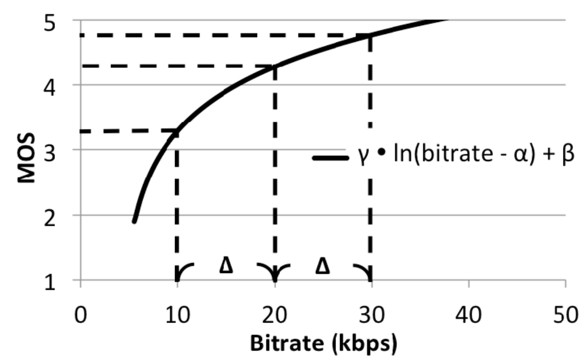


**Figure 1. Logarithm in MOS-bitrate relationship.**

## 2. PROPOSED MECHANISM
In EQ, the MOS range is first evenly dividing into a number of levels. By transposing the parameters of the model in Figure 1, the bitrate can be expressed as below,

$$bitrate = \alpha + e^{\frac{MOS-\beta}{\gamma}}$$

where α = 4.091, β = 1.515 and γ = 1.00. Then, apply the MOS value for each level. The formula returns the corresponding bitrate. The difference between two adjacent levels is greater for higher-rate levels and smaller for lower-rate levels. Under this scheme, when the network bandwidth increases, the bitrate increases to the highest level allowed. Conversely, when the network bandwidth decreases, the bitrate drops by one level.

The proposed mechanism is simple and distributed, and it produces bandwidth allocations that improve upon two important performance metrics: 1) the number of calls served and 2) the overall/accumulated user experience.

## 3. SIMULATION
The behavior of the rate control mechanisms is simulated at the call level, meaning that the call rate is the most basic unit of dynamics to capture. During its lifetime, each call is a periodic process whose bitrate is determined by the mechanism at hand and the bandwidth currently available. The background traffic is a random process modeled from an operational network [4]. The proposed mechanism is compared to two alternatives.

**Naïve Mechanism:** The Naïve mechanism in this study implements an intuitive, fair control method that determines the

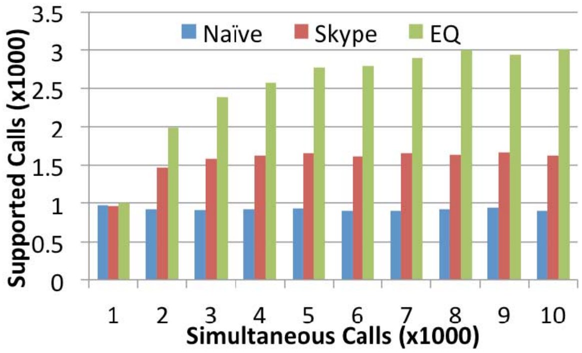


**Figure 2. Number of calls served.**

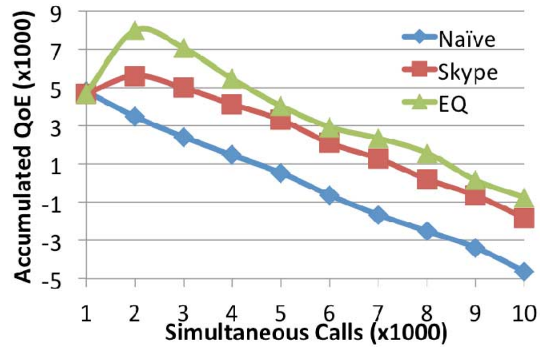


**Figure 3. Accumulated QoE.**

level of call rate increase or decrease based on the amount of increase or decrease in available bandwidth divided by the number of calls.

**EQ Mechanism:** The EQ mechanism implemented in the simulator has five levels. When there is an increase in available bandwidth, the bitrate is set to the level closest to, but not exceeding, the available bandwidth. When there is a decrease, the bitrate is set one level lower.

**Skype Mechanism:** This study includes Skype's rate control mechanism for comparison. Despite various attempts to decipher Skype's mechanism, the exact rate that Skype uses for different available bandwidths in its latest version is unknown. To facilitate this comparison, we adopted a black-box approach and set up a series of dummynet [5] experiments to identify Skype's sending rate at different bandwidths. The measured sending rate per available bandwidth was then implemented in the Skype simulations.

## 3.1 Number of Calls Served

Figure 2 shows the number of calls supported as the number of calls increases. The Naïve mechanism serves the least amount of calls throughout all call populations, whereas the proposed quantization mechanism serves the most. Skype offers new calls a higher chance of joining because of its slower response to spared bandwidth. This characteristic allows Skype to support more calls than the Naïve mechanism. However, it does not outperform the quantization mechanism because it treats calls as more uniform.

## 3.2 Accumulated MOS

The accumulated MOS is derived by summing up the MOS scores of all calls. As a call adapts its rate, an MOS score is derived from the MOS-bitrate relationship. The MOS of a call is calculated by averaging the MOS scores recorded throughout the call. One issue to address is that the MOS score for the dropped calls is not 0. Based on the derived model of SILK [3], a sending rate less than 4.091 kbps takes the corresponding MOS to negative infinity. This means that the quality of dropped calls perceived by users is significantly worse than calls with low quality. In other words, a significantly small negative value is more characteristic of dropped calls. To facilitate quantitative comparison, this section assigns a conservative negative value, -1, to the dropped calls and discusses the effects of using a small negative score later.

Figure 3 shows the accumulated MOS scores. The Naïve mechanism has the lowest accumulated MOS of all cases. The quantization mechanism outperforms Skype despite the over-estimation of the Skype call MOS. Based on the number of calls dropped (Figure 2), the lower the negative MOS score for dropped calls, the further the quantization mechanism out-performs the other two mechanisms.

A noteworthy finding is that the accumulated MOS initially increases with the number of calls, but the network quickly becomes saturated as the number of calls exceeds 2,000. The penalty for dropped calls becomes more pronounced as the call population grows. This leads to another potential use of the MOS-bitrate model: call admission control for local subnets. For a network adopting the Skype or quantization mechanism, allowing 3000 calls does not yield a higher level of overall user experience.

# 4. OUTLOOK

A closer look reveals that the EQ mechanism would converge to a fair bandwidth allocation among all calls. This indicates a potential link to *proportional fairness*, a property suggesting that EQ is not just better than the state of the art, but also the optimal.

# 5. REFERENCES


[1] M. Fiedler, T. Hossfeld, and P. Tran-Gia, "A Generic Quantitative Relationship between Quality of Experience and Quality of Service," IEEE Network, vol. 24(2), pp.36-41, 2010.

[2] P. Reichl, S. Egger, R. Schatz, and A. DAlconzo, "The Logarithmic Nature of QoE and the Role of the Weber-Fechner Law in QoE Assessment," In Proceedings of ICC, 2010.

[3] C.-N. Chen, C.-Y. Chu, S.-L. Yeh, H.-H. Chu and P. Hunag, "Measuring the Perceptual Quality of Skype Sources," in Proceedings of ACM Communications and Computer Networks, SIGCOMM, W-MUST, 2012.

[4] C. Fraleigh, F. Tobagi, and C. Diot, "Provisioning IP Backbone Networks to Support Latency Sensitive Traffic," in Proceedings of IEEE International Conference on Computer Communications, INFOCOM, 2003.

[5] L. Rizzo, "Dummynet and Forward Error Correction," in Proceedings of the USENIX Annual Technical Conference, 1998.