Tuning Skype's Redundancy Control Algorithm for User Satisfaction

Te-Yuan Huang†, Kuan-Ta Chen‡, and Polly Huang†

†Department of Electrical Engineering, National Taiwan University

‡Institute of Information Science, Academia Sinica

***Abstract*—Determining how to transport delay-sensitive voice data has long been a problem in multimedia networking. The difficulty arises because voice and best-effort data are different by nature. It would not be fair to give priority to voice traffic and starve its best-effort counterpart; however, the voice data delivered might not be perceptible if each voice call is limited to the rate of an average TCP flow. To address the problem, we approach it from a user-centric perspective by tuning the voice data rate based on user satisfaction.**

Our contribution in this work is threefold. First, we investigate how Skype, the largest and fastest growing VoIP service on the Internet, adapts its voice data rate (i.e., the redundancy ratio) to network conditions. Second, by exploiting implementations of public domain codecs, we discover that Skype's mechanism is not really geared to user satisfaction. Third, based on a set of systematic experiments that quantify user satisfaction under different levels of packet loss and burstiness, we derive a concise model that allows user-centric redundancy control. The model can be easily incorporated into general VoIP services (not only Skype) to ensure consistent user satisfaction.

***Index Terms*—MOS, PESQ, Piggyback, QoE (Quality of Experience), QoS (Quality of Service), VoIP**

I. Introduction

Effective end-to-end transport of delay-sensitive voice data has long been a problem in multimedia networking. Voice traffic, by nature, is high in data rate and it is sensitive to network impairments. With the increase in multimedia traffic on the Internet, a growing dilemma is that it would not be fair to give priority to voice traffic and starve its best-effort counterpart; however, the voice data delivered might not be perceptible if each voice call is limited to the rate of an average TCP flow. To address this problem, we approach it from a user-centric viewpoint by adapting the sending rate of voice calls based on user satisfaction. Such a user-friendly rate adaptation mechanism would also be congestion-friendly, although it is not strictly TCP-friendly [7].

Adapting the voice sending rate is a subtle issue because users prefer calls with a higher bit rate [5]. However, sending voice data with an unnecessarily high bit rate could be a waste of network resources or result in congestion, and that in turn could compromise the user's experience. In July 2008, eBay announced that Skype had 338.2 million registered users and earned US$136 million in revenue, representing 51% year-over-year growth[1]. Skype, as one of the largest and fastest growing VoIP services on the Internet, seems to note the subtlety and does not indulge its voice data with unlimited network bandwidth. Recently, Skype launched a very ambitious monthly plan worldwide, which is expected to attract even more users and voice transmissions from the traditional telephone services to the Internet. The surge in demand raises an important question: *How should Skype or competing VoIP services adapt their voice sending rates to meet customers' QoS expectations*. To address this question, we investigate three issues: (1) how Skype adapts its voice rate, (2) whether Skype's rate adaptation mechanism is geared to user satisfaction, and (3) how Skype and any other VoIP services should adapt their voice data rates to ensure consistent user satisfaction.

Bonfiglio et al. [4] observed that Skype's voice data rate is governed by three factors: the bit rate, the framing time, and redundancy. Among them, the bit rate and framing time are determined by the codec. Skype uses G.729 as the audio codec for SkypeOut (PC-to-PSTN) calls; while for PC-to-PC calls, iSAC, an audio codec developed by Global IP Solutions [9], is used in most of the calls. In particular, G.729 provides a constant bit rate (CBR) for voice data. Thus, the rate variation in SkypeOut calls is the result of adapting redundancy to network conditions. Furthermore, we found that the bit rate and framing time adaptation in calls using iSAC, the popular variable bit rate (VBR) codec, is very likely implemented by the codec developer [9], instead of Skype. Therefore, the only parameter tuned by Skype is the *redundancy factor*.

Focusing on the rate adaptation issue at the redundancy control level, we present our methodology for automatically identifying the redundancy ratio, i.e., *the percentage of packets piggyback a previous packet*, in general Skype calls, and derive the relationship between the redundancy ratio and the network loss rate. The major findings are (1) Skype increases the redundancy ratio as the network loss rate increases; however, (2) Skype's control algorithm does not take the individual codec and packet loss patterns (burstiness) into consideration. These findings indicate that, although Skype's rate adaptation mechanism somehow addresses the subtle relationship between sending rate and user satisfaction, there are yet discrepancies towards consistent user satisfaction.

To address the problem, we adopt implementations of public domain codecs and quantify user satisfaction, i.e., the mean opinion scores (MOS), for calls under different levels of packet loss and burstiness. Our results suggest that the adaptation policy should be codec-specific. To sustain the voice quality at MOS value 3.3, more redundancy should be added to G.711 voice calls than to those in G.729 calls. More redundancy

[1] http://www.tgdaily.com/content/view/38431/122/

should also be added when the network loss is bursty. Therefore, given the desired MOS level, we develop a model to tune the redundancy ratio based on the measured loss rate and loss burstiness. This model can be easily implemented and used generally by any VoIP service to provide consistent user satisfaction.

The remainder of this paper is organized as follows. Section II contains a review of related works on Skype. In Section III, we describe our experiment setup and methodology for quantifying redundancy . In Section IV, we discuss Skype's redundancy control algorithm. In Section V, we describe the simulation setup and discuss the optimal policy for controlling the amount of redundancy. Section VI details the simulation results. Then, in Section VII we provide some conclusion remarks.

II. Related Work

As the popularity of Skype has increased, a great deal of research has been devoted to understanding the phenomenon. Some works have focused on the design of Skype and the protocol used. For example, Baset et al. [1] analyzed the operation of the peer-to-peer infrastructure of Skype, while [2] performed detailed reverse engineering of the protocol and packet format of Skype. Other works have focused on identifying Skype traffic. For example, [10, 19] focused on identifying relayed Skype traffic, [4] tried to identify direct Skype sessions, and [21] proposed to detect Skype flows based on the signaling traffic between a node and its supernodes. In addition, [10] studied the behavior of Skype users, such as their usage patterns, and the characteristics of Skype's supernodes, e.g., their bandwidth consumption.

The present study is closely related to two previous works. In [5], the authors quantified the effect of network factors on the level of user satisfaction in Skype VoIP calls. They analyzed the relationship between network factors and the length of VoIP sessions, and found that the bit rate, bit rate variation, and round trip time have the most impact on user satisfaction. The same authors later proposed *OneClick* [6], a lightweight framework for measuring network applications' quality of experience from users' perspective, in the hope to verify the passive measurement results by user experiments. Bonfiglio et al. [3] analyzed how Skype adapts its traffic to different network conditions. They found that when available bandwidth is decreased, the bit rate and payload size of Skype traffic are also decreased. On the other hand, when packet loss is detected, Skype mitigates its impact by sending voice packets with redundancy. More specifically, Skype adopts a piggyback technique, which appends previously sent voice blocks to to-be-sent packets. The authors conducted a series of experiments to evaluate Skype's redundancy control algorithm, and demonstrated that the payload of some packets is doubled when artificial packet losses are introduced. Moreover, as the loss rate increases, the percentage of packets with a double payload size also increases. The authors proposed a source traffic model of Skype. Based on the model, Skype's traffic is decided by three parameters: 1) the bit rate used by the codec; 2) $\triangle T$, the framing time of human speeches; and 3) the redundancy factor, which is the percentage of previous voice frames piggybacked by the current frame.

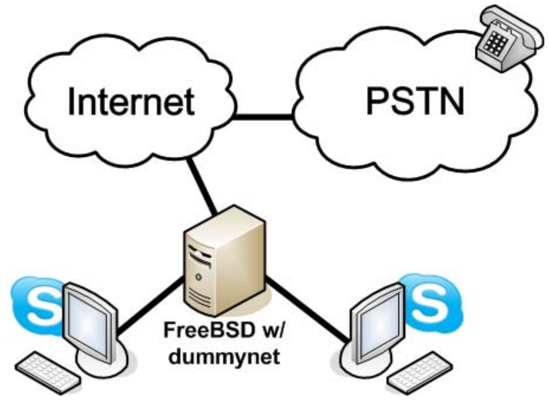


Fig. 1. The network setup for collecting Skype traffic.

Inspired by Bonfiglio et al.'s work, we tried to determine whether Skype adjusts the three parameters properly so that an optimal level of user satisfaction can be achieved. To this end, we conducted some experiments with the iSAC codec, an audio codec used by Skype, and found that the first two parameters, i.e., the encoding bit rate and the speech framing time, are controlled by the codec, and only the redundancy factor is controlled by the Skype program. For this reason, we believe that Skype's redundancy control mechanism might be the key to its good voice quality. Thus, in this work, we address the following questions: 1) *Is Skype's redundancy control optimal?*; 2) *if it is not optimal, how should a VoIP application like Skype adjust the redundancy ratio to achieve a balance between bandwidth consumption and user satisfaction*. We consider these two issues in the following sections.

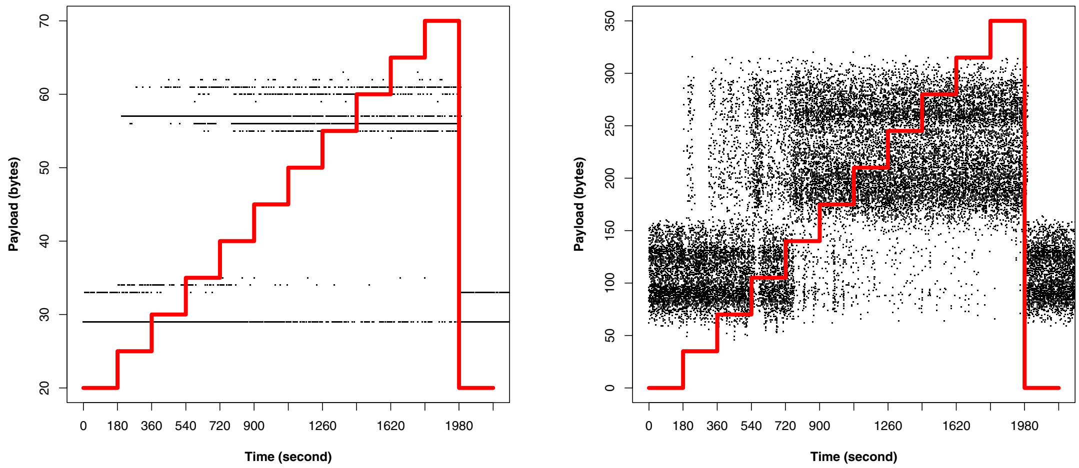
III. Estimating Skype's Redundancy Ratio

In this section, we describe our methodology for quantifying the amount of redundancy Skype adds into its voice traffic.

A. Experiment Setup

To collect Skype traces, we make Skype calls in a controlled network environment, as shown in Fig. 1. A FreeBSD box, which acts as a layer-2 bridge, is used to control the traffic passing through by `dummynet` [17]. Skype is installed on two Windows XP machines, which are connected to the Internet through the FreeBSD box. To simulate human conversation, audio files downloaded from the Open Speech Repository [20] are played during the Skype VoIP calls.

Skype can transmit its voice packets by either UDP or TCP. Since TCP guarantees in-order and reliable transmission, there is no need for Skype to add redundancy to TCP flows. Therefore, only UDP flows are the subjects of the present study. To increase the probability that Skype transmits voice traffic using UDP, each Windows XP machine is assigned a public IP address [1]. Since version 3.2, Skype adopts an in-house developed audio codec, SVOPC [16]; thus, we use different versions of Skype for experiments on different codecs. We use Skype version 3.1 for experiments on iSAC and SkypeOut, and use version 3.8 for experiments on SVOPC.



(a) G.729 (b) iSAC

Fig. 2. The impact of the network loss rate on the payload size of Skype packets.

For both versions of Skype, the codec G.729 is used for SkypeOut sessions.

We collect Skype traces on the FreeBSD box with the program `tcpdump` [15]. In addition, to avoid the interference caused by initial setup traffic, we only record the traffic after the call has been established for 60 seconds. In each experiment, we increase the network loss rate from 0% to 10% in 1% increments every 180 seconds. For the iSAC traces, we filter out the control and signaling packets by inspecting the "Start of Message" (SoM) field, which contains the message ID and the function of the packet [2].

B. Observations

Fig. 2(a) shows the scatter plot of the payload sizes of the packets in the SkypeOut (G.729) trace. From the graph, we observe that when the loss rate is 0%, i.e., between 0 and 180 seconds, the payload size remains around 30 bytes. However, as the loss rate increases, we find there are more packets with a payload size around 60 bytes. When the loss rate reaches 10%, i.e., between 1800 and 1980 seconds, the majority of the packets have a payload of around 60 bytes. However, when the loss rate returns to 0% after 1980 seconds, the payload size of most packets drops to around 30 bytes. This phenomenon indicates that Skype changes the proportion of packets with redundancy information based on the network loss rate. Note that Skype may also piggyback signaling data in voice packets. This explains why we can still observe various payload sizes when no redundant voice information is introduced, even though G.729 is a constant-bit-rate codec.

The iSAC trace exhibits the same behavior, as shown in Fig. 2(b). When the loss rate is 0%, the payload size remains within the range $(0, 160)$ bytes approximately. As the loss rate increases, we find more packets with a payload size in the range $(160, 320)$ bytes; and when the loss rate reaches 10%, the majority of the packets have a payload size in the range (160, 320) bytes. Note that although iSAC is a codec with several framing time options, the framing time of the observing iSAC traffic stays at 30 ms during the whole call; thus, the variance in the payload sizes is not a consequence of changes in the speech framing time.

C. Redundancy Ratio Identification

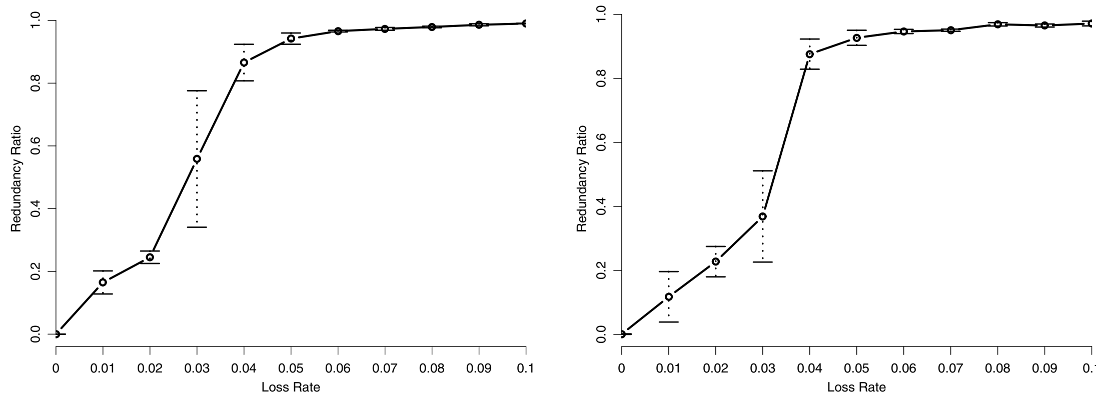
In order to understand the redundancy control algorithm used by Skype, we attempt to quantify the amount of redundancy added to Skype voice traffic. We define the *redundancy ratio as the percentage of packets that carry redundant voice data*. If all packets carry redundant information, the redundancy ratio is equal to 1. Conversely, if none of the packets carry redundant information, the redundancy ratio will be 0.

In the following, we present our method for inferring the redundancy ratio used by Skype based on the traces collected in the above experiments. We take G.729 and iSAC as examples, though the method can be extended to other codecs supported by Skype.

1) G.729: It is easier to deal with the SkypeOut traces, as the G.729 codec uses a constant bit rate of 8 Kbps and a constant framing time of 10 ms. For this reason, the codec's payload size is more stable than that of iSAC, as shown in Fig. 2(a). We use a simple threshold method, with the threshold set at 40 bytes, to determine whether a packet contains a piggybacked frame. In other words, we assume that a packet contains redundant information if its payload size is larger than 40 bytes. I.e., if there are 30% of packets with payload size larger than 40 bytes, then the redundancy ratio will be 0.3.

2) iSAC: It is more difficult to deal with the iSAC traces because iSAC supports variable bit rate and variable framing time. We use the the following steps to infer the redundancy ratio in each of the iSAC traces:

i. First, we determine the framing time of a packet, as it will affect the payload size of iSAC packets. When the framing time is longer, the payload size will be larger, since each packet would carry more information.



(a) G.729

(b) iSAC

Fig. 3. The relationship between the network loss rate and the redundancy ratio that Skype uses for Skypeout (G.729) and iSAC.

The framing time can be estimated from the inter-packet time, i.e., the time difference between successive packets. Because the framing time may be changed during a call due to network conditions, we estimate the framing time on a window basis. For a window of n packets, we calculate the average inter-packet time based on $(n-1)$ inter-packet times. Assuming that inter-packet time is normally-distributed and centered at the actual framing time, we compute the likelihoods of the averaged inter-packet time on the distribution of each possible framing time. Then, we consider the framing time that yields the maximum likelihood as the actual setting. According to [4], the possible framing times of iSAC are 30 ms or 60 ms.

ii. Second, by assuming a canonical framing time, we normalize the packets' payload sizes based on the estimated framing time. For example, we assume that the canonical framing time of iSAC is 30 ms. Thus, for a packet with a framing time of 60 ms, we normalize its payload size by a factor of 2, i.e., its payload size is divided by 2 in the normalization step.

iii. Third, we determine whether a packet carries redundant information based on the normalized payload size. Similar to the method we used for G.729, we set the threshold at 160 bytes to identify packets that containing redundancy. We choose this threshold because it is the maximum observed payload size when there is no packet loss.

D. Identification Results

We repeat the experiment five times and estimate the redundancy ratio for each trace. First, we analyze the G.729 traces. Fig. 3(a) shows the average redundancy ratios and their 95% confidence intervals with each network loss setting. We observe that the redundancy ratio increases gradually when the loss rate is between 1% and 2%, and increases dramatically when the loss rate is between 3% and 4%. The redundancy ratio stays higher than 0.9 when the loss rate is higher than 4%.

Next, we analyze the iSAC traces and plot the relationship between the average redundancy ratio and the network loss rate, as shown in Fig. 3(b). From the figure, we find that Skype adjusts the redundancy ratio for iSAC traffic in a similar way to that used to adjust G.729, which suggests that Skype adjusts the redundancy ratio regardless of the codec used.

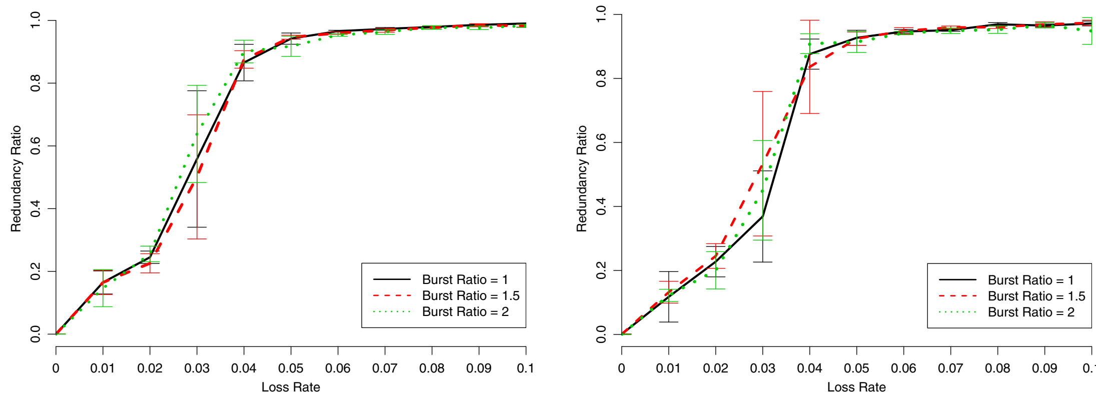
IV. Understanding Skype's Redundancy Control Algorithm

The experiments in Section III show that Skype adjusts its redundancy ratio based on the current network loss rate. In this section, we investigate whether Skype considers other factors when it adjusts the redundancy ratio.

We consider three factors that may affect VoIP quality. The first factor is available bandwidth, as reduced bandwidth may cause some packets to be dropped and force the codec to use a lower encoding bit rate whenever possible. The second factor is the audio codec used, as different codecs may interpret frame losses in different ways. Moreover, some codecs may be robust to frame loss, while others may not. Thus, it may be appropriate to adjust the redundancy ratio with different methods for different audio codecs. The third factor is the burstiness of network loss, which characterizes the degree of successiveness on packet drops, since different patterns of packet loss might cause different levels of voice quality impairment. In the following sub-sections, we discuss Skype's redundancy control policy in response to these three factors.

A. Effect of Available Bandwidth

How Skype adapts to changing bandwidth has been discussed in [3]. The authors have found that when they reduced the bandwidth, the payload size decreased, which suggests the codec switches to a lower bit rate. Although additional packet loss may occur with a reduced bandwidth, the authors did not observe any packets with a double payload size, i.e., did not observe any packets carrying redundant information. This is probably because the codec successfully switches to a lower bit rate before the packet loss rate is high enough to trigger the redundancy control algorithm. The authors concluded that the bandwidth setting does not affect Skype's redundancy control decisions.



(a) G.729 (b) iSAC

Fig. 5. Comparison of Skype's redundancy control algorithms for different levels of network loss burstiness for G.729 and iSAC.

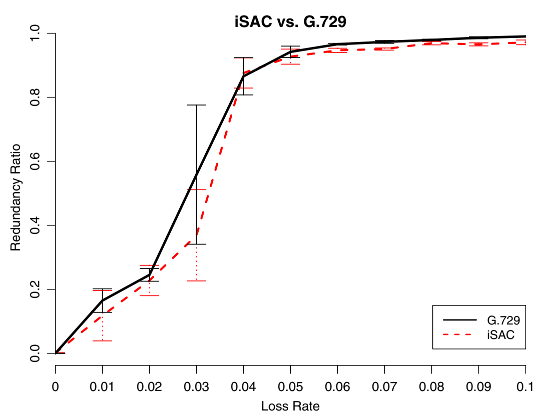


Fig. 4. Comparison of Skype's redundancy control algorithms over different network loss rates for G.729 and iSAC.

B. Effect of the Codec

Fig. 4 shows the redundancy ratios for various network loss rate under G.729 and iSAC. Our objective is to determine whether Skype uses different redundancy control algorithms for different codecs. In the figure, the 95% confidence interval of two curves collide with each other. This observation strongly suggests that Skype applies the same redundancy control algorithm for different codecs, even though it leads to different levels of user satisfaction, as we will show in the next section.

C. Effect of Network Loss Burstiness

To quantify the burstiness of network loss, we adopt the metric *burst ratio* defined in ITU-T G.107 [12]:

$$\frac{\text{The average length of observed consecutive losses}}{\text{The average length of consecutive losses under random losses}}.$$

In this definition, the burst ratio is equal to 1 when packet loss is purely random, and it is larger than 1 when packet loss is bursty. Specifically, a burst ratio equal to 2 indicates that the average length of consecutive losses is twice longer than that of purely random losses.

The experiment are similar to those in Section III, except that the packet loss is now bursty rather than uniformly distributed. We implemented the Gilbert model [8] to determine whether a packet should be dropped in `dummynet` [17] in order to simulate different levels of loss burstiness. The Gilbert model comprises two states, the *received state* and the *loss state*; and two transition probabilities, p and q, where p is the probability of a transition moving from "received" to "loss" and q is the probability of a transition moving from "loss" to "received." In this model, the packet loss rate is formulated as $\frac{p}{p+q}$ and the *burst ratio* is formulated as $\frac{1}{p+q}$. Thus, by setting the values of p and q, we are able to control both the network loss rate as well as the burst ratio.

Fig. 5(a) shows the observed redundancy ratio of G.729 when its traffic experiences packet losses with different burst ratios. As shown in the figure, each curve is corresponding to a burst ratio setting and their 95% confidence intervals overlap with each other. Similarly, in Fig. 5(b), each curve represents the redundancy ratios observed from iSAC calls with packet losses under different burst ratios. Again, we found that the 95% confidence intervals of the curves are also overlapped.

In summary, our experiment results strongly suggest that Skype adjusts redundancy ratios only based on the network loss rate; i.e., it does not consider the codec or network loss busrtiness.

V. Deriving an Optimal Redundancy Control Policy

In this section, we present a methodology that can derive the optimal redundancy control policy for a desired VoIP quality under a certain network condition. We then compare the inferred Skype redundancy control policy with the optimal policy to determine whether Skype's policy is optimal.

A. Methodology

We develop a simulator that can grade the voice quality of audio clips transmitted using a specific codec with a given network condition. To evaluate the quality of an audio clip, we use PESQ [14], which compares a degraded audio clip with its original version and output a Mean Opinion Score (MOS) [13].

Fig. 6. The information flow of our methodology for computing audio quality under given network conditions.

The steps of our methodology for deriving the optimal redundancy control policy are as follows:

i. Encode an audio clip into voice frames by using one of the encoders provided by the Intel IPP (Integrated Performance Primitives) library [11].
ii. Simulate network loss with the Gilbert model; that is, drop a frame if the model is currently in the "loss" state and retain the frame otherwise.
iii. Determine whether a frame is piggybacked by the desired redundancy ratio. If the redundancy ratio is set to p, then each frame has a probability p of being transmitted twice. Thus, even if a frame was dropped in step 2, it will be restored if the subsequent frame was not dropped and was selected to carry redundant voice data (with probability p).
iv. Use the corresponding decoder to decode the resulting stream of voice frames into a degraded audio clip.
v. Use PESQ to quantify the quality of the degraded audio clip by comparing it to the original clip.
vi. Repeat the above steps for a range of redundancy ratios, and consider the ratio with the desired PESQ score as the *optimal redundancy ratio*. For example, if the desired PESQ score is 3.3, then under each network loss settings, the redundancy ratio achieves exactly 3.3 is considered as the optimal redundancy ratio.

The information flow of the methodology is illustrated in Fig. 6. In our simulations, we use an audio clip concatenated from several speech recordings in American English provided by the Open Speech Repository [20]. The recordings are concatenated using the Sound eXchange (SoX) Library [18]; the length of the concatenated audio clip is 3 minutes and 27 seconds.

B. Optimal Redundancy Ratio for the Codecs

First, we consider whether the optimal redundancy ratios are the same for different audio codecs. To address this issue, we conduct the simulations described in the previous sub-section for G.711, the most common codec used in digital speech systems, and for G.729, the codec used by SkypeOut[2].

The optimal redundancy ratios inferred by our methodology for G.711 and G.729 are shown in Fig. 7. On each graph, the contour curve labeled with a number, say 3.3, represents the combinations of loss rates and redundancy ratios that yields the same MOS score, 3.3. We can see that, for a certain loss rate, higher redundancy ratios yield to higher MOS scores. On the other hand, for a certain redundancy ratio, higher loss rates lead to lower MOS scores. If we compare the contour plots of both codecs, we find that the redundancy ratios required to maintain a certain MOS score for G.711 and G.729 are different. Generally, redundancy should be added more aggressively for G.711 in order to achieve the same quality as G.729. For example, assuming the network loss rate is 2% and the desired MOS score is 3.3, the redundancy ratio should be set to 0.5 for G.711. In contrast, we can achieve the same sound quality by setting the redundancy ratio to 0.2 if G.729 is used.

[2]We did not evaluate iSAC because it is a proprietary codec of Global IP Solutions [9]; hence, we do not have access to the source codes of its encoder and decoder, and therefore not be able to include the iSAC into the simulation.

C. Optimal Redundancy Ratio for the Burst Ratios

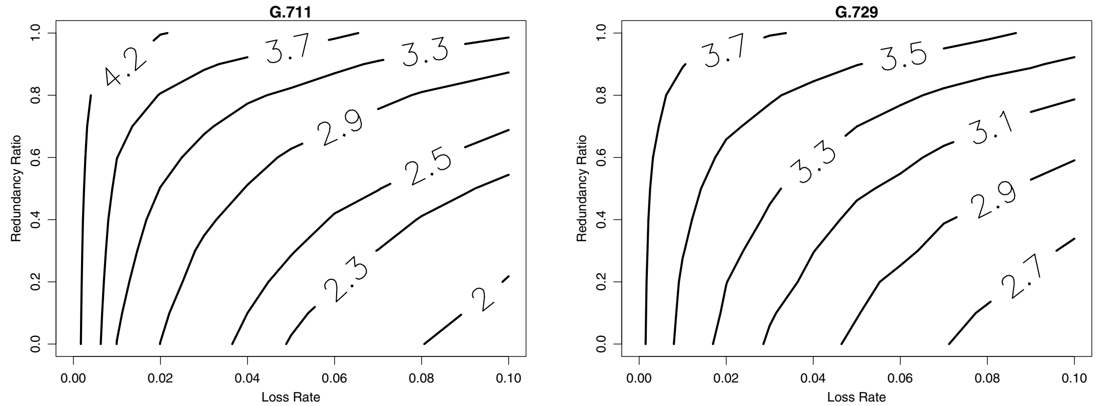
We repeat the above simulations, except that we now infer the optimal redundancy ratios for different burst ratios. Fig. 8 shows the contour lines for G.711 and G.729 that corresponding to the MOS score 3.5 for the burst ratios 1, 1.5, and 2. From the graphs, we observe that the redundancy ratio should be increased more aggressively if we wish to maintain the same audio quality under higher burst ratios. For example, to maintain a consistent level of user satisfaction with G.711 under network loss rate of 2%, the redundancy ratio should be set to 0.7 when the burst ratio is 1; however, it should be set to 0.8 and 1 when the burst ratios are 1.5 and 2, respectively.

D. Is Skype's Policy Optimal?

Now that we have determined Skype's redundancy control algorithm and derived the optimal redundancy control policy, we can now assess whether Skype's redundancy control policy is optimal by comparing it with the optimal policy.

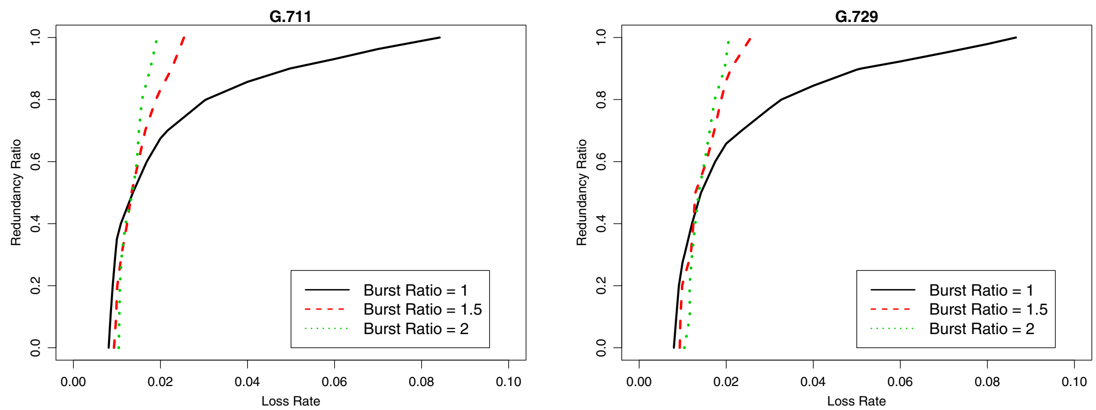
In Fig. 9, we overlap Skype's redundancy control decisions and the optimal redundancy policy for G.729; each graphs corresponds to a certain burst ratio. By comparing the contour curves and Skype's redundancy ratio curve, we find that *Skype fails to maintain a consistent voice quality at a certain level*.

For example, in Fig. 9(a), Skype achieves an audio quality better than that of a MOS score of 3.5 when the loss rate is higher than 4% or lower than 1%. At the same time, its quality level is much lower than 3.5 when the loss rate is between 2% and 4%. The inconsistency in voice quality would be frustrating for users. On the other hand, assuming that the desired MOS score is 3.3, this phenomenon indicates that Skype may inject more than enough traffic into the network by adjusting the redundancy ratio too aggressively and obtains an unnecessarily high MOS score as a consequence. In contrast, by adjusting the redundancy ratio to the optimal redundancy



(a) G.711 (b) G.729

Fig. 7. The contour plots of audio quality scores for different combinations of redundancy ratios and network loss rates.



(a) G.711 (b) G.729

Fig. 8. The redundancy ratios needed to sustain voice quality for a MOS score of 3.5 with different loss rates and burst ratios.

ratio derived by our methodology, we can ensure a balance between bandwidth utilization and voice quality.

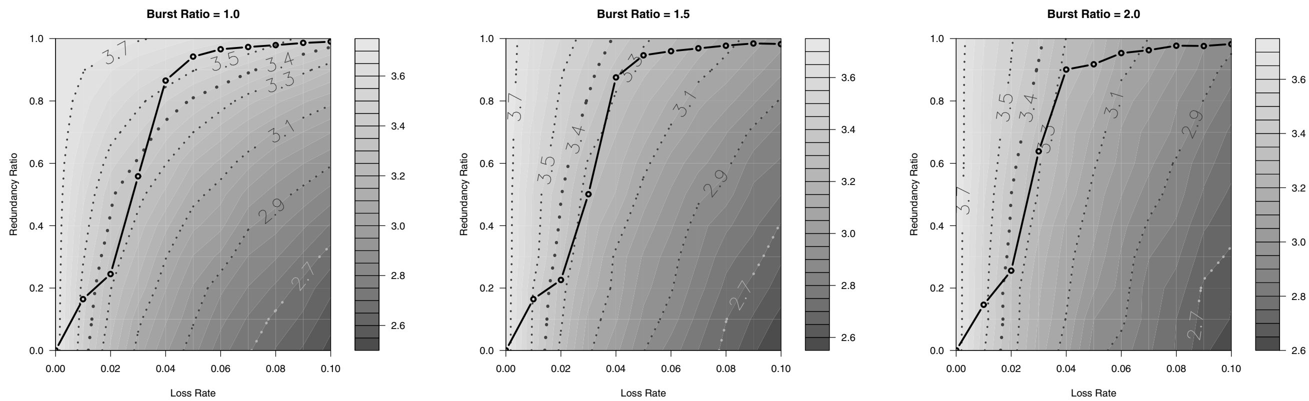
We use Fig. 9 to quantify the degree that Skype's redundancy control algorithm deviates from the policy that achieves a consistent audio quality under various network conditions. The graphs are computed based on the assumption that the desired MOS score is 3.4, as Skype's audio quality is mostly around this level in our simulation scenarios. For each network setting, we plot the bandwidth Skype uses and the MOS score Skype provides on the respective normalized scales. The desired MOS score and the bandwidth required to achieve the desired audio quality are both set to 100. On the graphs, the left-hand side of the y-axis marks the normalized bandwidth utilization, and the right-hand side marks the normalized MOS score[3]. We observe that the bandwidth utilization and audio quality of Skype fluctuate under different network settings. Sometimes, Skype uses too little bandwidth and results in worse quality scores than the desired score; for example, the scenario with 2% loss rate in Fig. 9(d), and those with 2% and 3% loss rates in Fig. 9(e) and 9(f) respectively. At the same time, Skype sometimes injects too much redundant information and thus achieves a quality level better than the desired level, e.g., the scenarios with a loss rate higher than 4% and a burst ratio equal to 1, as shown in Fig. 9(d).

Our results show that Skype's audio quality is not consistent as *it adjusts the redundancy ratio independently of the codec used and the network loss burstiness*. The inconsistency in voice quality may result in frustration for users or over-utilization of bandwidth. To balance the needs of users and ensure network efficiency, a more sophisticated redundancy control algorithm that considers all the necessary factors is required.

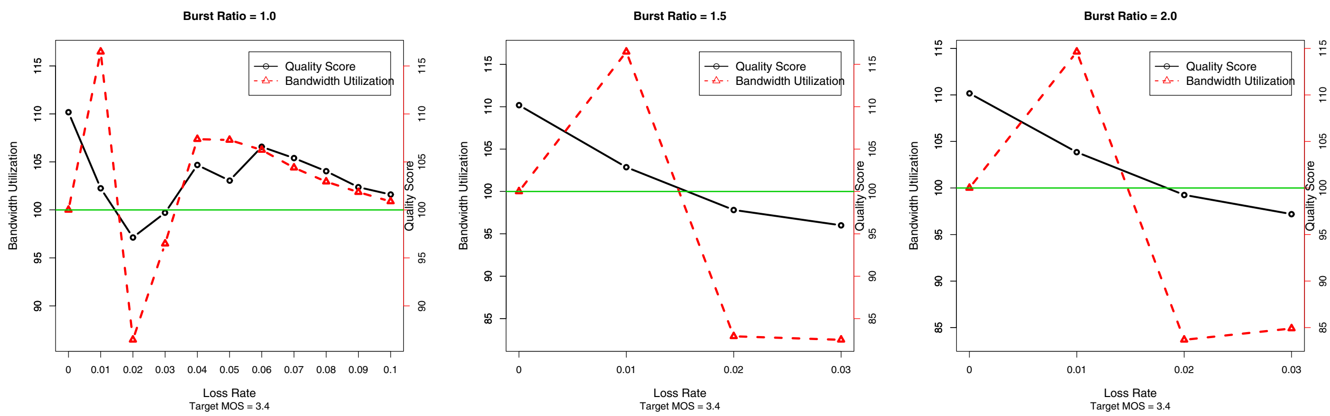
VI. Modeling Optimal Redundancy Ratios

We have shown that the redundancy control algorithm used by Skype is suboptimal. Moreover, we have proposed computing the optimal redundancy control policy based on PESQ quality estimation. The policy can be adopted by real-time audio streaming tools to provide consistent user experiences no matter how network conditions change. However, the procedures for deriving the optimal redundancy ratios are time consuming and therefore need to be performed beforehand. Note that it is not possible to compute an optimal redundancy ratio for any combination of network factors, as many of the factors, including the network loss rate, are real-valued.

[3]Note that the x-axes of Fig. 9(e) and Fig. 9(f) end at 0.03. The reason is that it is impossible to achieve a MOS score of 3.4 when the burst ratio is 1.5 or 2 with a loss rate higher than 3%.



(a) Burst Ratio = 1 (b) Burst Ratio = 1.5 (c) Burst Ratio = 2



(d) Burst Ratio = 1 (e) Burst Ratio = 1.5 (f) Burst Ratio = 2

Fig. 9. Top row: Comparisons of Skype's redundancy control policy vs. the redundancy ratios required to achieve certain audio quality levels. Bottom row: Quantifying how Skype's redundancy control policy deviates from the optimal algorithm assuming that a MOS score of 3.4 is desired.

TABLE I
COEFFICIENTS IN THE MODEL

Variable	Coef	Std. Err.	t	$Pr > \|t\|$
$(constant)$	1.06×10^{0}	4.47×10^{-2}	23.68	$< 2 \times 10^{-16}$
plr	-1.47×10^{1}	8.2×10^{-1}	-17.84	$< 2 \times 10^{-16}$
plr^{-1}	-5.03×10^{-3}	5.53×10^{-4}	-9.10	3.45×10^{-11}
$plr : br$	1.48×10^{1}	0.18	16.25	$< 2 \times 10^{-16}$
$plr^{-1} : br$	-2.89×10^{-3}	1.90×10^{-4}	-15.18	$< 2 \times 10^{-16}$

Therefore, we believe it is necessary to develop a model that can determine the optimal redundancy ratio for any network condition.

Take G.729 as an example. Using an ordinal polynomial regression approach, we develop a model that can predict the optimal redundancy ratio based on a given network loss rate and burst ratio. The model computes the optimal redundancy ratio by

$$\begin{aligned}&(constant) + \\ &coef_{plr} \cdot plr + coef_{plr^{-1}} \cdot plr^{-1} + \\ &coef_{plr:br} \cdot plr \cdot br + coef_{plr^{-1}:br} \cdot plr^{-1} \cdot br,\end{aligned}$$

where plr denotes the packet loss rate, and br denotes the burst ratio. The coefficients are listed in Table I. To evaluate the model's adequacy, we show both the computed and the predicted optimal redundancy ratios for G.729 under various network conditions in Fig. 10. The prediction curves on the graph describe the computed optimal redundancy ratios very well based on the two network factors. The R^2 value of the regression model is as high as 0.986, which indicates that the model fits the data very well.

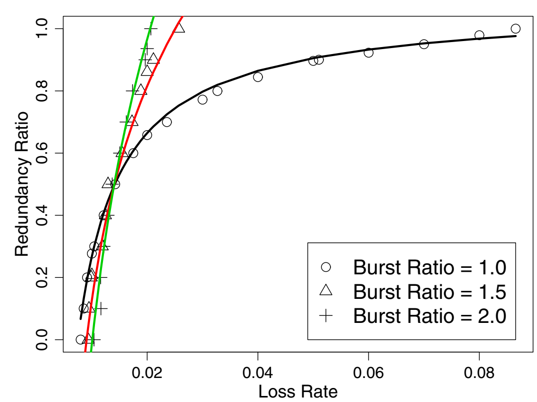


Fig. 10. Comparison of computed and predicted optimal redundancy ratios for G.729 with different combinations of network loss rates and burst ratios.

This approach for predicting optimal redundancy ratios can be extended to other audio codec and generalized by

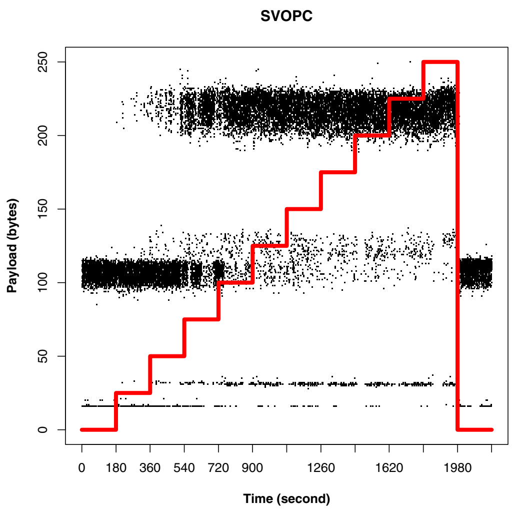


Fig. 11. The impact of network loss rate on the payload size of SVOPC packets

incorporating additional network factors. The advantage of our model is that it is easy to compute, as only simple arithmetic is needed to calculate the optimal redundancy ratio given the network factors. Therefore, any VoIP application that adopts the model can provide consistent service quality with minimum computation and network overhead.

VII. Conclusion

In this paper, we have determined how Skype adapts its redundancy levels to network loss rate and burstiness; shown that Skype's rate adaptation mechanism is not really geared for user satisfaction; and proposed a general model for various codecs to tune the redundancy for consistent user satisfaction. The methodology used to derive the general model can be extended by Skype developers to facilitate tuning of different proprietary codecs, such as iSAC and SVOPC.

During our research, Skype has changed to SVOPC for most PC-to-PC calls. The results of our preliminary experiments (Fig. 11) show that Skype's current redundancy control mechanism is much the same as that used in previous releases. This is consistent with our findings on G.729 and iSAC, and confirms that Skype's redundancy control mechanism is probably not codec specific.

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