

Network and Systems Laboratory
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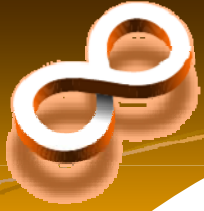
Tuning Redundancy Control Algorithm for User Satisfaction

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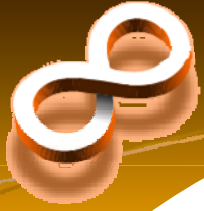
Institute of Information Science
Academia Sinica, Taiwan

INFOCOM, 2009



Motivation

- Voice traffic is **sensitive** to network impairment
- Why VoIP sending rate is important?
 - Most important factors on user satisfaction
 - **Sending Rate** and its **Variation**
 - **High** and **Stable** voice quality
- Why adapting sending rate is difficult?
 - Aggressively?
 - Conservatively?



Motivation – Cont.

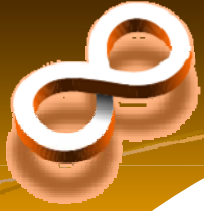
- Skype – one of the most popular VoIP software

Q1: How Skype adapts its voice traffic?





Q2: Is their mechanism good enough?

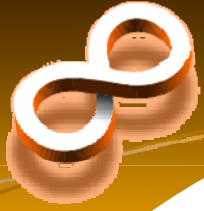
Q3: How can Skype's policy be improved?



Related Work

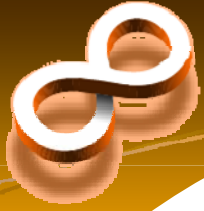
- Skype's voice traffic is governed by: [Bonfiglio et al.]
 - Bit Rate
 - Framing Time
 - Redundant Data
- PC-PSTN calls 
 - G.729
- PC-PC calls 
 - iSAC

Only **Redundant Data** is controlled by Skype



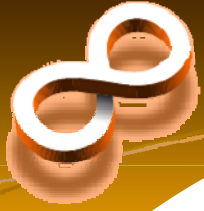
Outline

- Motivation
- Related Work
- How does Skype adapt the redundancy ratio?
- Is Skype's mechanism good enough?
- How can we do better?
- Conclusion

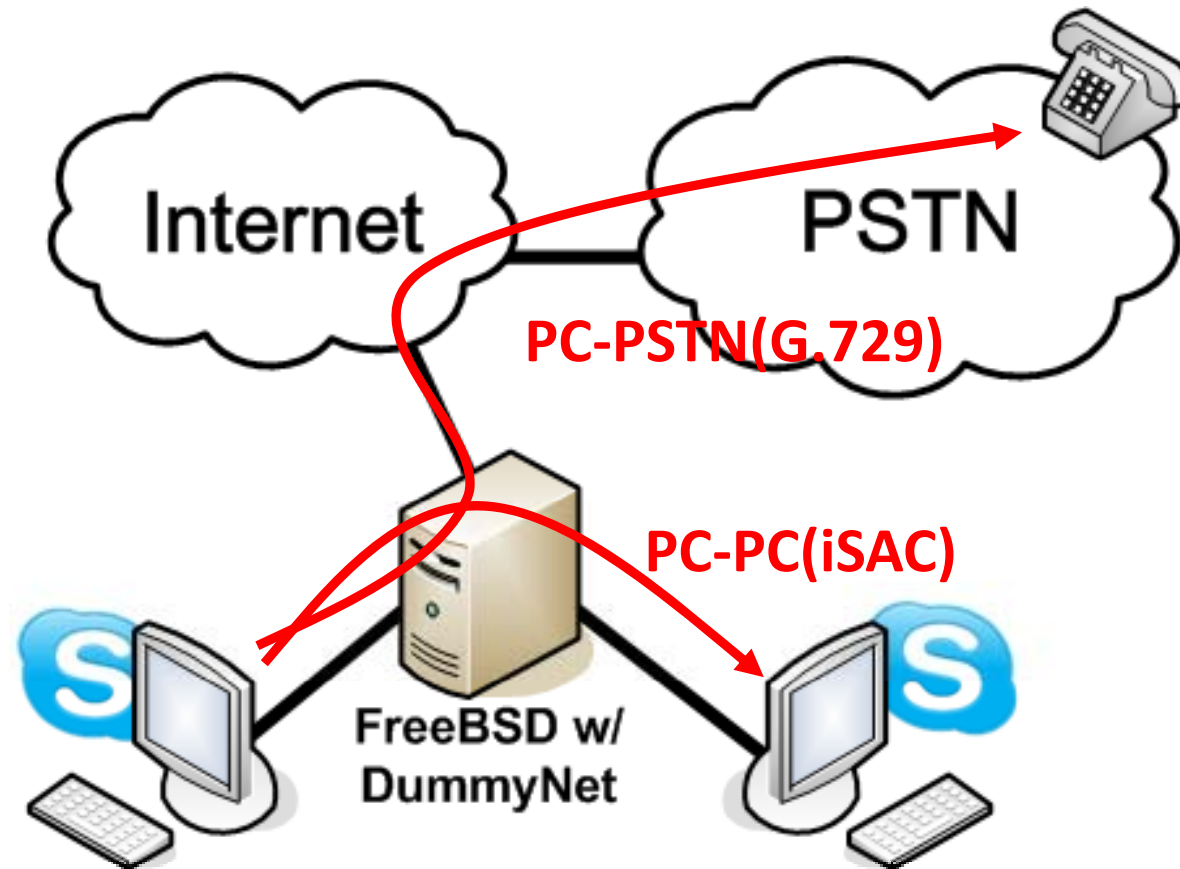


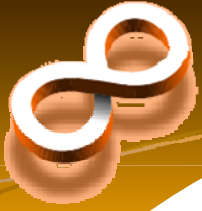
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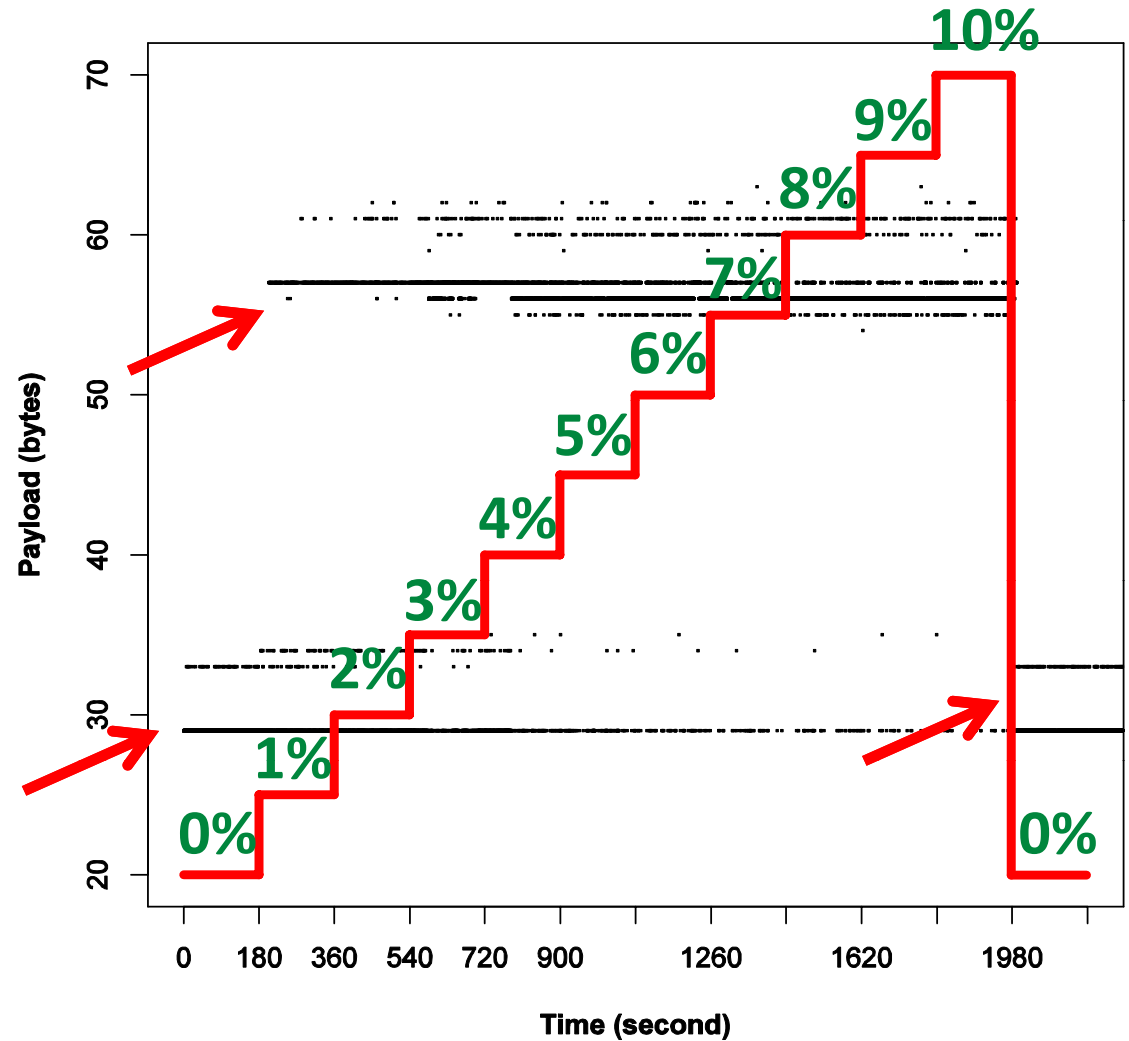
Experiment Setup

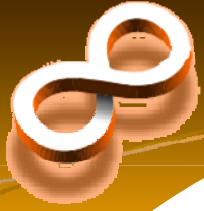




Observation

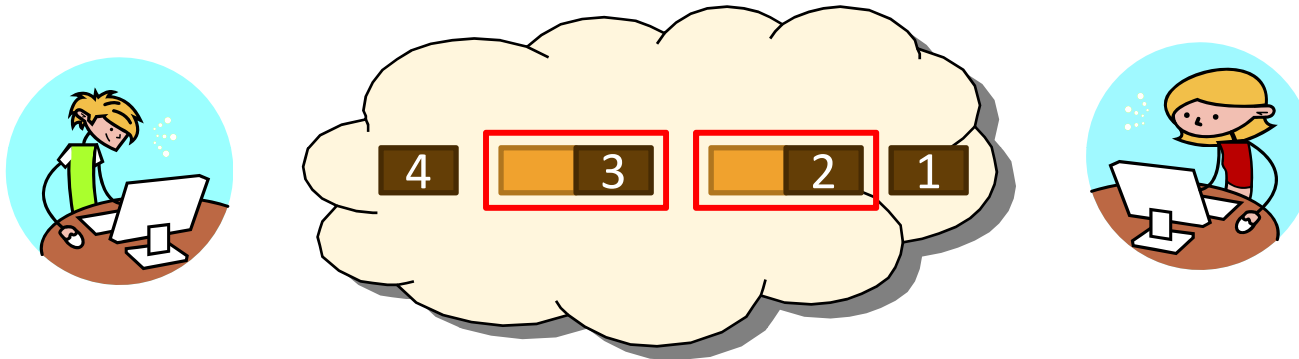
- G.729 (PC-PSTN)
 - Constant bit rate
 - Constant framing time



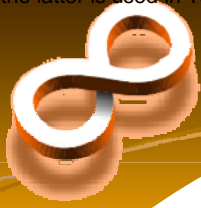


Redundancy Ratio

- Definition
 - The **percentage** of packets that carry redundant voice data

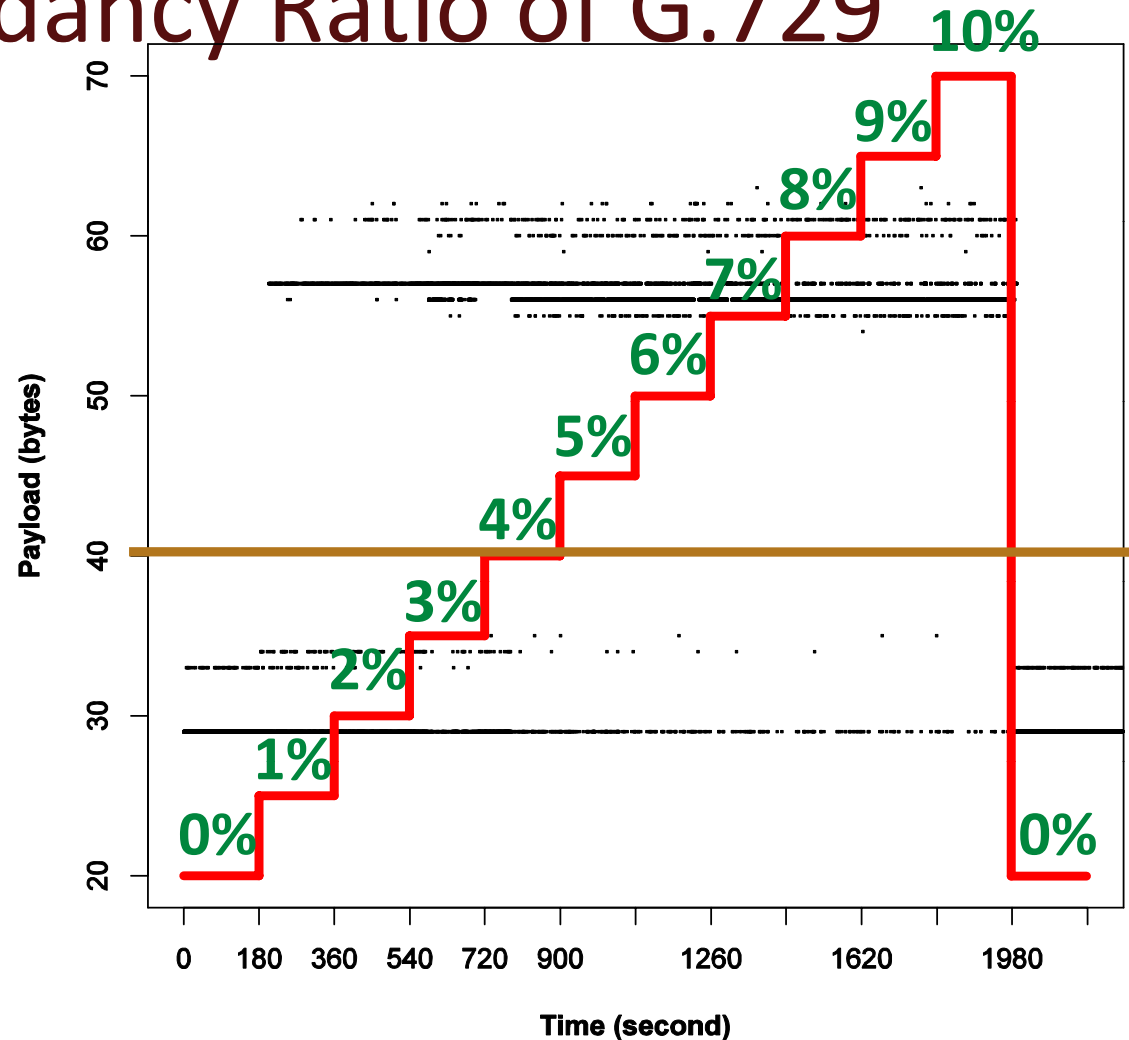


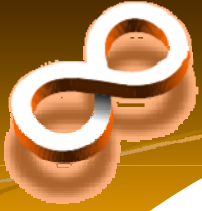
$$\text{Redundancy Ratio} = 2/4 = 0.5$$



Estimate Redundancy Ratio of G.729

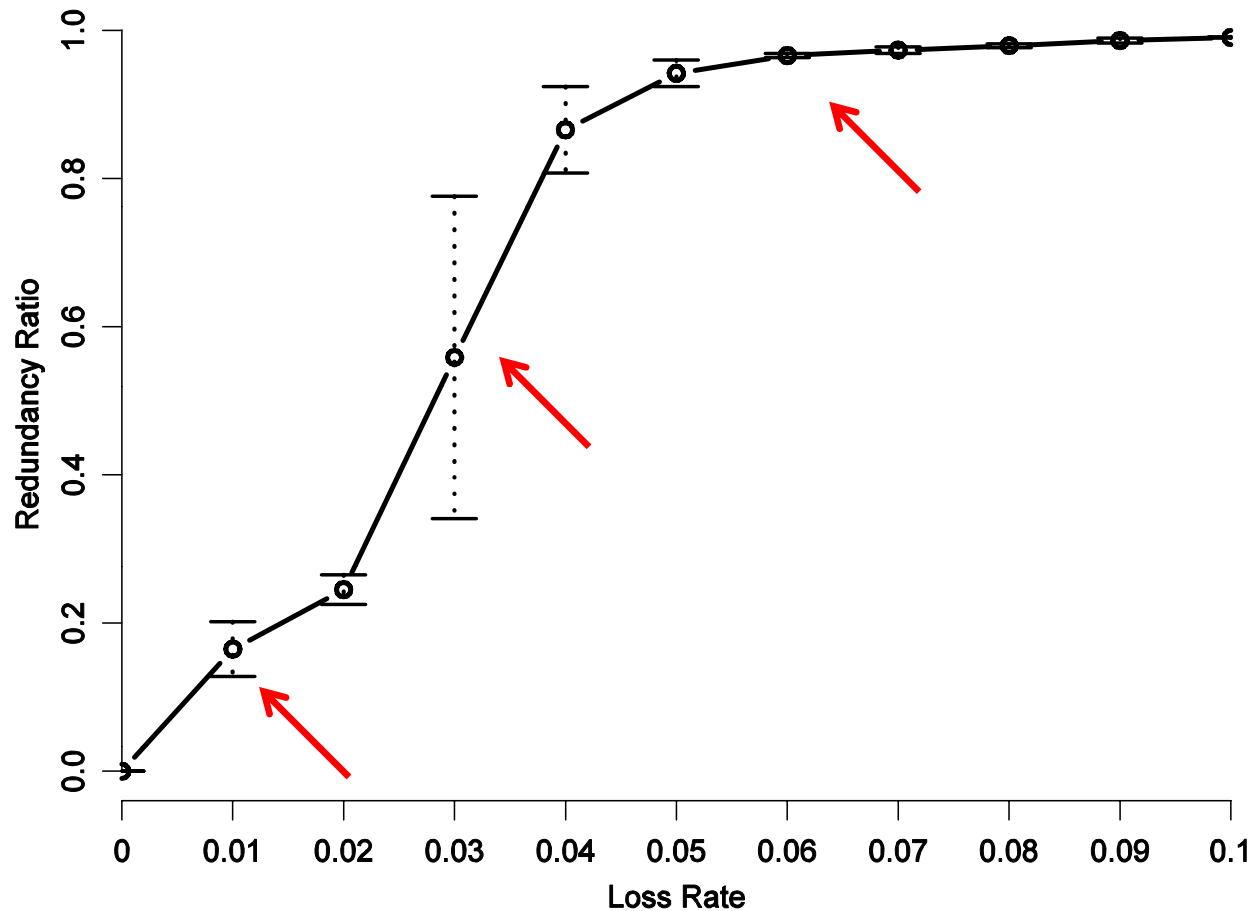
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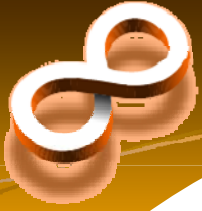




Identify Redundancy Ratio

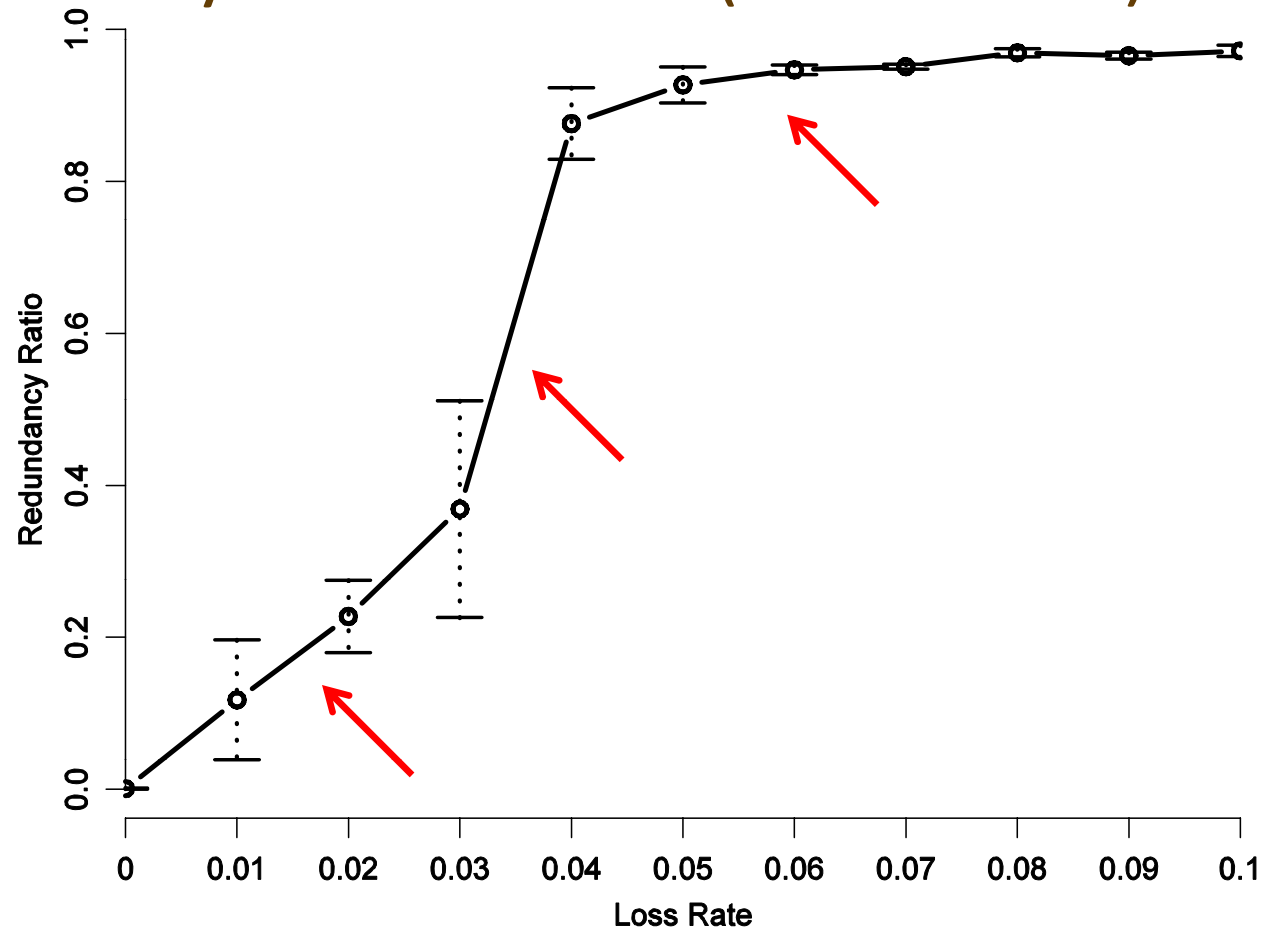
- Redundancy Ratio of G.729 (PC-PSTN Calls)

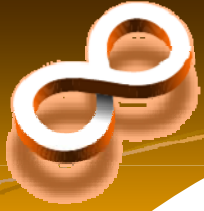




Identify Redundancy Ratio

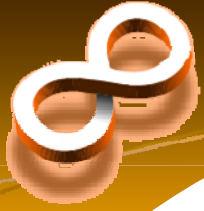
- Redundancy Ratio of iSAC (PC-PC Calls)





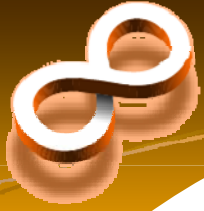
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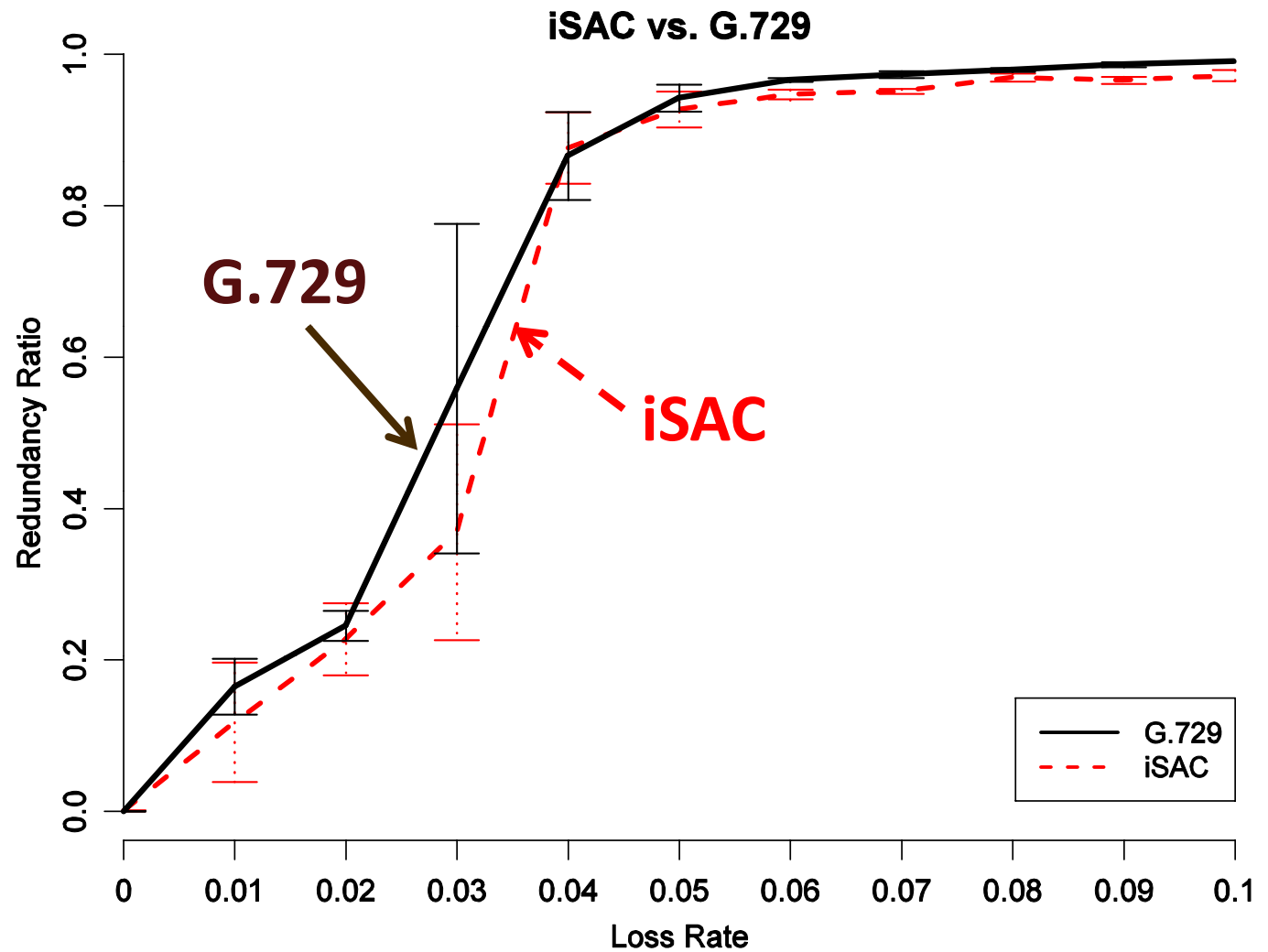


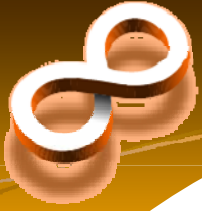
Skype's Redundancy Control Algorithm

- Adapt to network loss rate
- Adapt to other factors?
 - Codec
 - Network Loss Burstiness



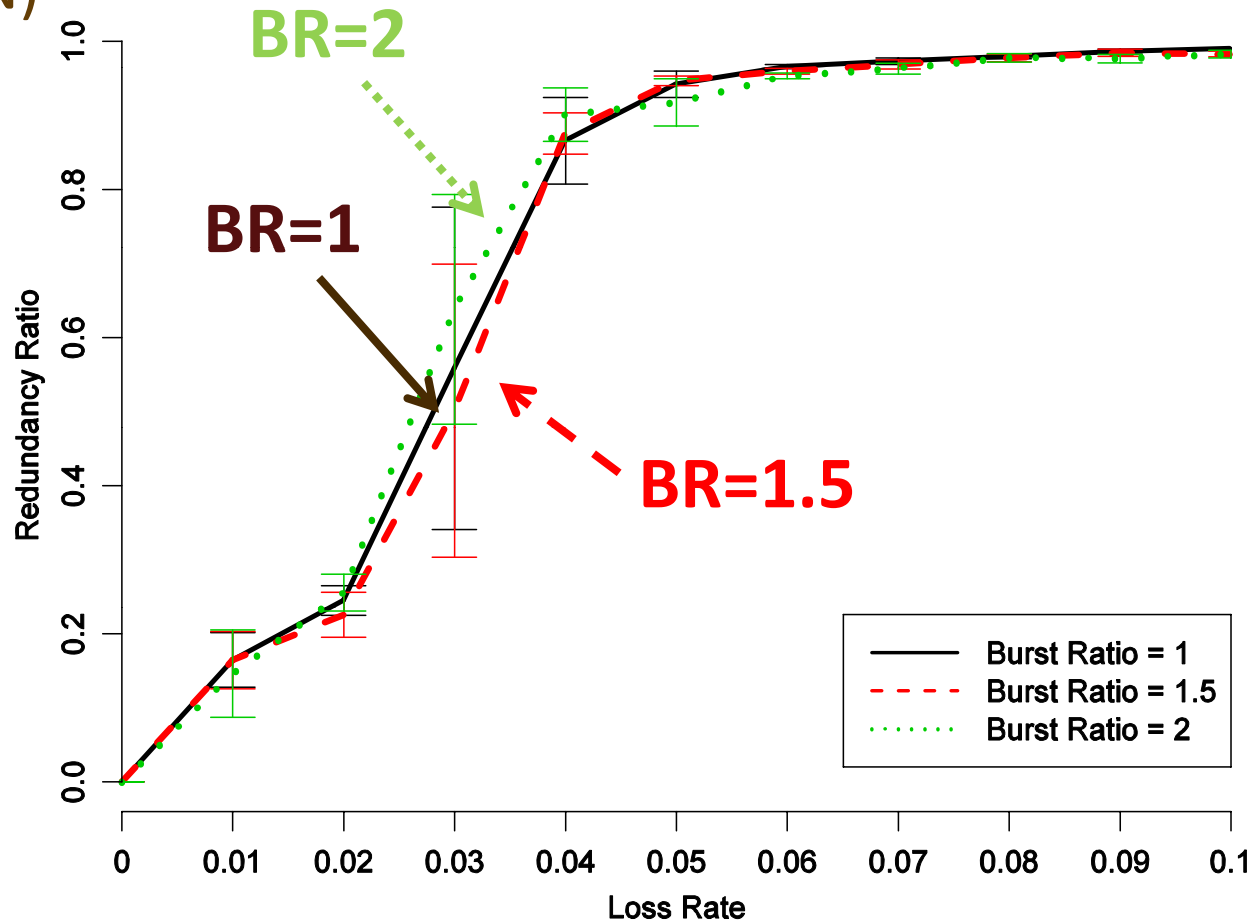
Effect of Codec

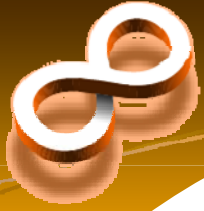




Effect of Network Loss Burstiness

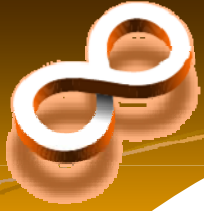
- G.729 (PC-PSTN)





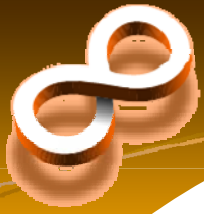
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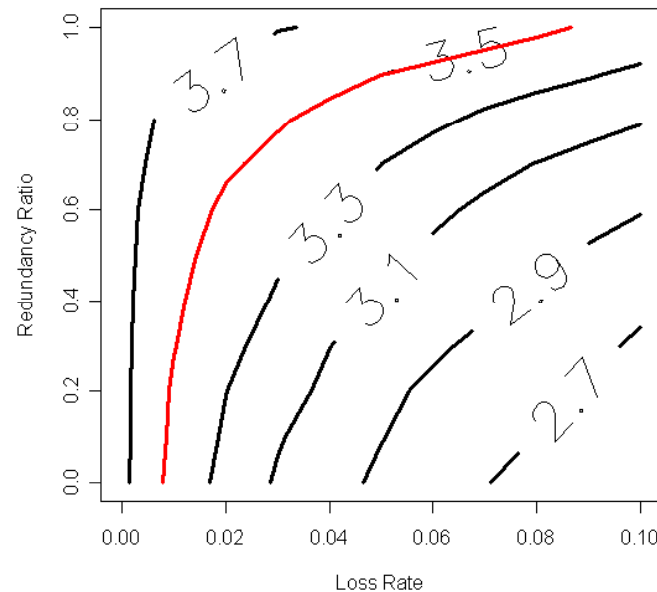


Optimal Redundancy Control Policy

- What's the Optimal Policy?
 - **Minimum** amount of redundancy data we need to sustain **the same audio quality** under different network conditions

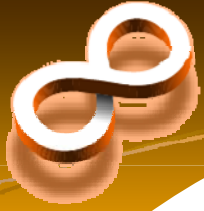


Emulation Flow

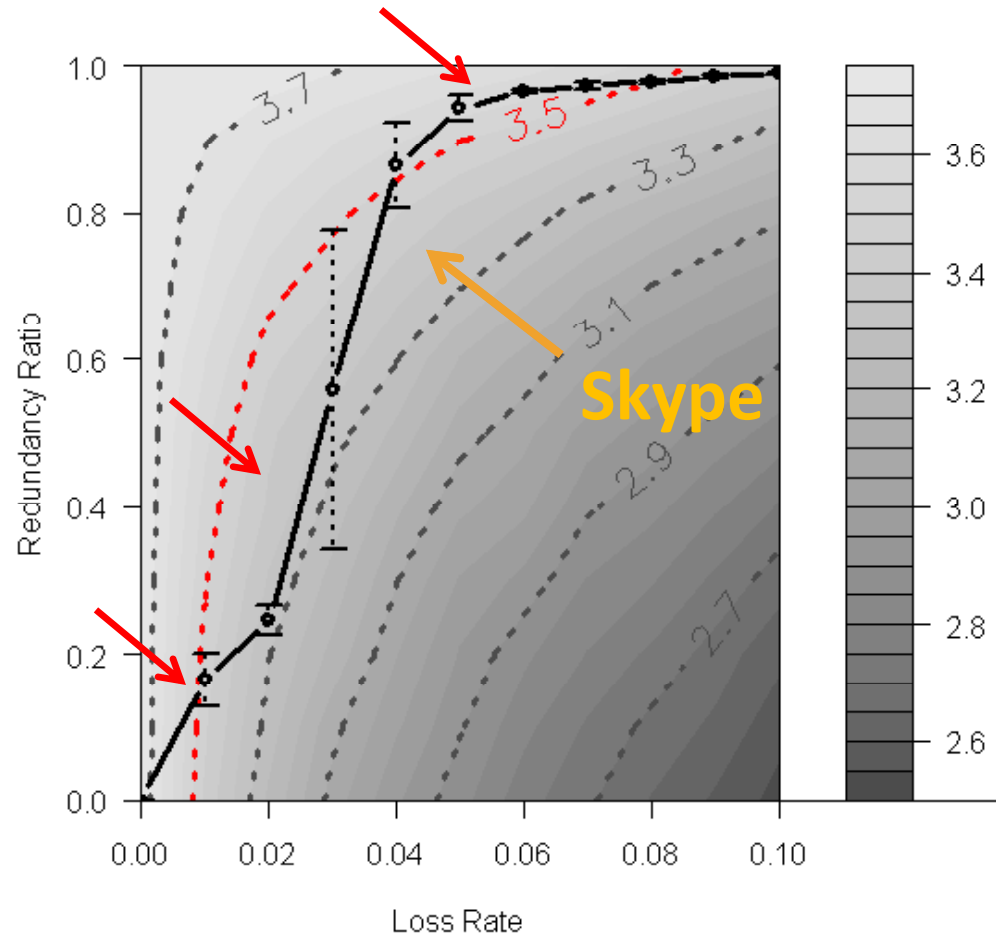


**Optimal
Redundancy
Ratio**

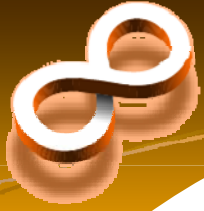
G.729



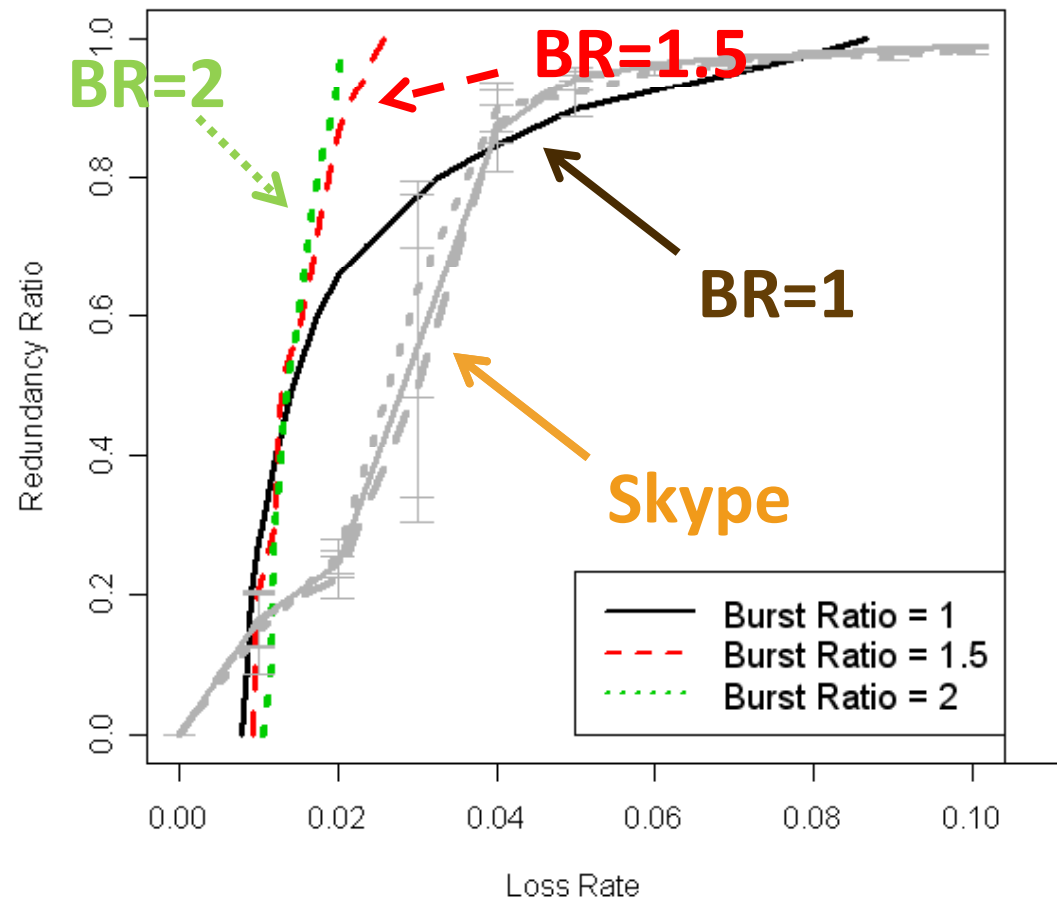
Skype vs. Optimal – G.729



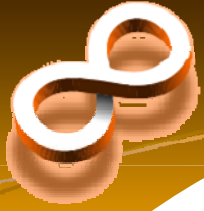
Burst Ratio = 1



Optimal Redundancy for the Burst Ratio



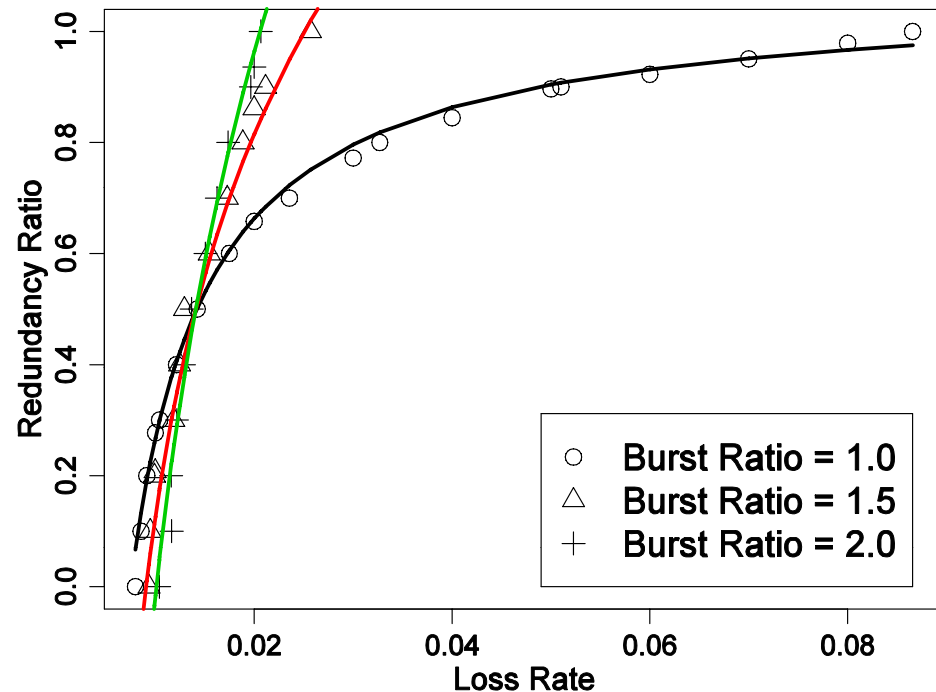
G.729, MOS=3.5

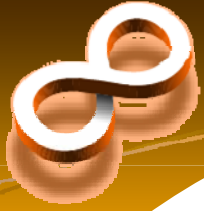


Modeling Optimal Redundancy Ratio

- Based on the targeted voice quality
- Take **codec** and **burstiness** into consideration

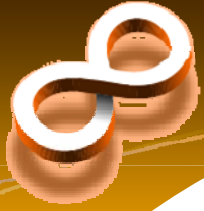
**Optimal Policy for
G.729, targeted
MOS=3.5**





Conclusion

- Explore how Skype adapts its voice traffic
 - Redundancy Ratio
- Skype's policy does not factor in the **individual codec** and **loss patterns** in to consideration
- Propose a general model for optimal policy
 - Consistent user satisfaction
 - Extensible to general VoIP software



Thanks You
Question?