Tuning Skype™ Redundancy Control Algorithm for User Satisfaction

Te-Yuan Huang, Kuan-Ta Chen, Polly Huang

Network and Systems Laboratory
National Taiwan University

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
Outline

- Motivation
- Related Work
- How does Skype adapt its redundant data?
- Is Skype’s mechanism good enough?
- How can we do better?
- Conclusion
Experiment Setup

PC-PSTN(G.729)

PC-PC(iSAC)
Observation

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

• Definition
  • The percentage of packets that carry redundant voice data

Redundancy Ratio = 2/4 = 0.5
Estimate Redundancy Ratio of G.729

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

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

- Redundancy Ratio of iSAC (PC-PC Calls)
Outline

- Motivation
- Related Work
- How does Skype adapt its redundancy ratio?
- Is Skype’s mechanism good enough?
- How can we do better?
- Conclusion
Skype’s Redundancy Control Algorithm

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

![Graph showing the comparison between G.729 and iSAC codecs. The x-axis represents the loss rate, ranging from 0 to 0.1, while the y-axis represents the redundancy ratio, ranging from 0 to 1.0. The graph illustrates that iSAC generally outperforms G.729 in terms of redundancy ratio for a given loss rate.]
Effect of Network Loss Burstiness

- G.729 (PC-PSTN)

![Graph showing the effect of network loss burstiness with different burst ratios (BR=1, BR=1.5, BR=2) on the effect of loss rate on redundancy ratio. The graph illustrates how increasing burst ratios affect the redundancy ratio at various loss rates.]
Outline

• Motivation
• Related Work
• How does Skype adapt its redundancy ratio?
• Is Skype’s mechanism good enough?
• How can we do better?
• Conclusion
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

Encoder → Simulation of frame dropping and piggyback → Decoder

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?