Successful Multiparty Audio Communication over the Internet

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Communications of the ACM

May 1998

Outline

- Introduction
- Problems with Current Tools
- The Robust Audio Tool
- Conclusions

Introduction

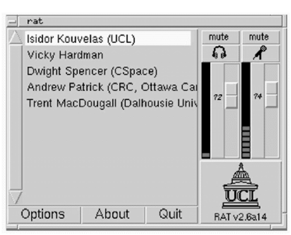
- Internet has point-to-point (unicast)
- Web allows download and playout of MM
- Multicast conferencing requires
  - Real-time interactive audio and video
  - Many participants at once
- Of audio and video, audio most important
  - Get it right, first

Introduction

- Mbone provides test bed for diverse apps
  - Teleconferencing
  - Telepresence
  - Distance education
- Multicast audio quality hurt by
  - Packet loss
  - Lack of real-time operating system support

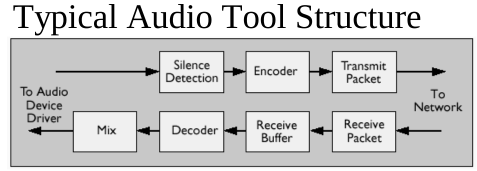
Introduction

- Many publicly available multicast audio tools
  - Visual Audio Tool



- Power meters to indicate send-receive volumes (why?)
- Highlight speaker (why?)

Typical Audio Tool Structure



- Sample 20, 40 or 80 ms
  - (Tradeoff between sample size and quality?)
  - (Tradeoff between sample size and network load?)
- Use silence detection
  - Measure of energy
  - Bandwidth savings up to 50% (2-way)
  - Instead, use “push-to-talk” button (why?)
- (What is “Mix”?)
- Buffer to remove some jitter

Problems with Existing Tools

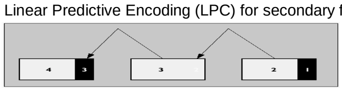
- Delay not usually a big problem
  - Typically under 400-600 ms
- Loss can be problematic
  - 10% is max tolerable with silence substitution
  - International links typically have 20-25% loss
- No real-time OS support
  - Processes to send-receive may not be scheduled in time, leading to gaps in playout

Acoustic Problems with Existing Tools

- Echoes and feedback
  - Push-to-talk helps, but not silence detection
- Silence detection
  - Clips words, may pick up background noise
- Lack of distance cues
  - Causes mismatch of volume
- Restricted intelligibility
  - Caused by mere toll quality audio (?)
- Monaural sound
  - Hard to identify speakers

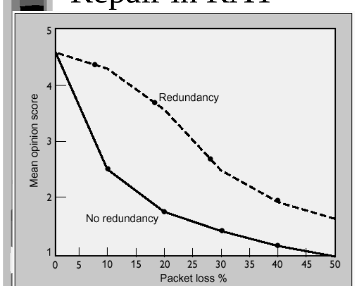
The Robust-Audio Tool

- Has ‘hands-free’ design
  - Speech detection
- Workstation scheduling features (we’ll cover)
- Loss seen as most significant
  - Uses redundancy to repair
    - + Linear Predictive Encoding (LPC) for secondary frame



  - Multiple loss repaired through waveform substitution on second packet
    - + Brain can smooth over

Repair in RAT

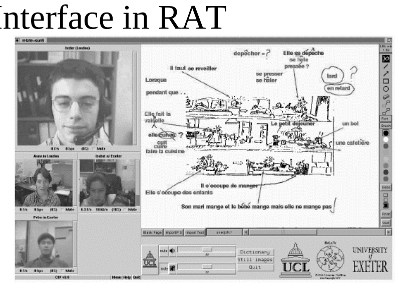


- Up to 20% loss ok
  - 30% with waveform (not shown)

Silence Detection in RAT

- Uses simple energy measure
  - Supplement with ‘rule-based’ approach(?)
- Adaptive silence threshold
  - For changing background noise
- Works so well, do not need ‘push-to-talk’ feature

Interface in RAT



- Video and shared whiteboard
- Audio and video separate, so need synch
- Power-meter for volume

Future Work?

Conclusions and Future Work

- Multicast has potential, but problems to fix
- RAT fixes loss, has improvements to interface
- Loss now ok, acoustic problems next
  - (A previous paper demonstrated this)
- Need to have responsive protocols
  - (Our previous topic)
  - Use alternate encoding to reduce rate
- Network may use priority queuing to help
  - (Upcoming topic)