

Successful Multiparty Audio Communication over the Internet

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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

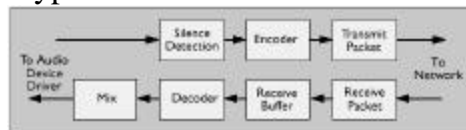
- Many publicly available multicast audio tools

– 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 packet size and quality?)
- Use silence detection
 - Measure of energy
 - Bandwidth savings up to 50%
 - Also, "push-to-talk" button
- (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-recv 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

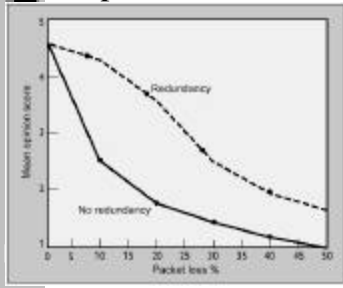
- Has 'hands-free' design
 - Speech detection
- Workstation scheduling features (we'll cover)
- Loss seen as most significant
 - Uses redundancy to repair
 - + Linear Predictive Encoding (LPC) for secondary frame



- Multiple loss repaired through waveform substitution on second packet




Repair in RAT




Packet loss %	Redundancy (Mean Opinion Score)	No redundancy (Mean Opinion Score)
0	5.0	5.0
10	4.5	3.5
20	4.0	2.5
30	3.5	2.0
40	3.0	1.5
50	2.5	1.0

- 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



Conclusions and Future Work

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



Evaluation of Science?

- Category of Paper
- Science Evaluation (1-10)?
- Space devoted to Experiments?

