SIPRTPUA: A SIP User Agent with Low Latency Audio Interface.

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Agenda

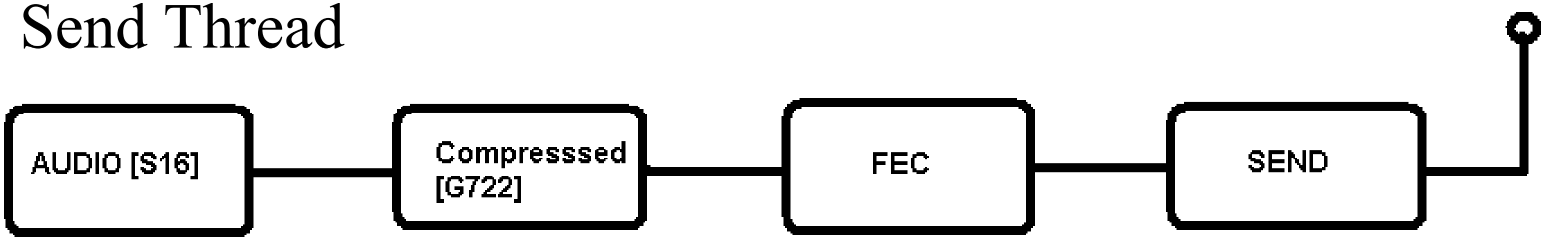
- Project Objective
- Approach
- Design and Initial Setup
- Realization
- Different Modules
- Possible Extensions
- Q & A
- Demo

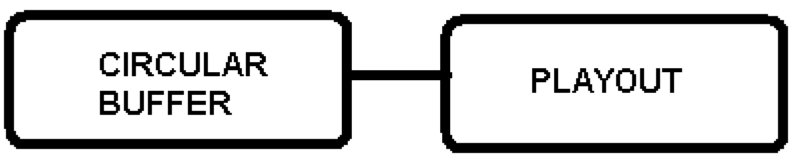
Proposed Approach

- Proposed Plan – *Building blocks*
  - SIPUA with RTP support
  - Audio on Linux
  - G722 Codec
  - Adaptive playout algorithms
  - Implement FEC

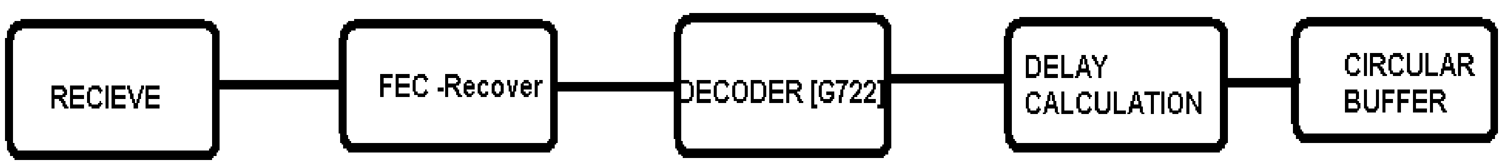
System Architecture

Send Thread





Receive Thread



Module I

- Audio on Linux
  - 16 Khz,S16
- G722
  - 20 ms packetization, 640 bytes to 160 bytes
  - 64 kb/s
- Silence Detection
  - Primitive implementation

Module II

- Adaptive Delay
  - Circular buffer of 1 second
  - Wait for at most 200 ms for late packets
  - Based on
    - *Adaptive Playout Mechanisms for Packetized Audio Applications in WideArea Networks. By: H. Schulzrinne, R. Ramjee, J. Kurose, D. Towsley*

Module III

- FEC as per RFC 2733
  - Reed-Solomon (3,4) implementation
    - XOR of 3 packets
    - Sent out as FEC packets
  - Considered piggyback, RFC 2198

Possible Extensions

- Testing with some packet dropping emulators
- AGC, Automatic Gain Control
- Windows + Solaris platforms
- Configurable codec

References

- RFC 1889: RTP: A Transport Protocol For Real-Time Applications *By: H. Schulzrinne, S.Casner, R. Frederick, V. Jacobson.*
- RFC 2733: An RTP Payload Format for Generic Forward Error Correction.*By : J. Rosenberg and H. Schulzrinne.*
- RFC 2198: RTP Payload for Redundant Audio Data.*By: C. Perkins, I. Kouvelas, O. Hodson, V.Hardman, M. Handley, J.C. Bolot, A. Vega-Garcia, S. Fosse-Parisis.*
- Adaptive Playout Mechanisms for Packetized Audiop Applications in Wide Area Networks. *By: H. Schulzrinne, R. Ramjee, J. Kurose, D. Towsley*
- Integrating Packet FEC into Adaptive Voice Playout Buffer Algorithms on the Internet. *By: J. Rosenberg, L. Qiu, H. Schulzrinne*
- http://www.4front-tech.com/pguide/audio.html: for Audio device on linux.