Real Time Chat

Audio chat between sender and receiver over network is facilitated by UDP and the voice is recorded and played using pulse audio. Uncompressed audio is sent over UDP. Frequency used 44100hz and 16 bit linear signed PCM is used so that we can transfer audio.

Procedure Employed.

1. Audio is recorded and send to receiver using Send Thread.
2. Audio is received and played using receive thread.

Observations of Experiment.

Problems found in the implementation of voip :

1. Problem:
   Initially TCP protocol is used to send the audio packets from the sender to the receiver ,Audio packets were transmitted and received without any loss but the delay was huge and hence real time experience was not possible with the college network.
   Debugging:
   Practically listening to the audio at the receiver side showed the delay introduced by TCP.
   Solution:
   To overcome this UDP was used, by using UDP delay was far reduced
2. Problem:
   UDP had packet loss and this loss depends on network congestion if network is congested packet loss was very high.
   Debugging:
   Practically listening to the audio at the receiver side showed distortion in audio and some of the words were not heard .
   Solution:
   If network congestion is low then audio quality was better.
3. Problem:
   UDP Packets arrived out of order at the receiver side.
   Debugging:
   Practically listening to the audio at the receiver side showed some of the words were out of order.
   Solution:
   Implementing RTP header solved the issue of out of order arrival. Sequence number was used to eliminate this.
4. Problem:
   Voice clarity was not good
   Debugging:
   Practically listening to the audio at the receiver side showed this problem.
   Solution:
   Buffer size was increased which resulted in increase call clarity.
5. Problem:
   There was lot of background noise heard in between the packets.
   Debugging:
   Practically listening to the audio at the receiver side showed this problem.
   Solution:
   Opening and closing the stream for each packet removed this noise.
6. Initially only voice was heard when 8000hz was used later that rate was changed to 44000hz to incorporate all the frequency in audio range.

Improvement Suggestion : The real time chat which is now implemented by using UDP alone has to incorporate RTP header inside so that audio corruption is avoided.

Release Time:

The release time refers to how long the compressor will take to relax the compression once the signal has fallen below the threshold.
The longer the release time, the longer the compressor holds on to the signal, and the smoother the sound. A faster release means the compressor lets go quickly, and the signal retains more of the original dynamic.

Working of code on raspberry Pi:

Code is working fine on Raspberry and working in the same way as laptop.