RTP based Real time voice phone

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1 Objective

The objective of experiment is to build a RTP based Real time voice phone.

2 Backgroud

2.1 RTP

Real time transport protocol is a network protocol for delivering audio over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media such as telephony, video teleconference applications etc.

It is used over UDP rather than TCP because TCP prefers Reliability over timeliness.RTP is fairly insensitive to packet loss so it doesnt require the reliability of TCP. UDP has less overhead for headers so that one packet can carry more data, so the network bandwidth is utilized more efficiently.

Port number is even port number between 16384 to 32767. linear PCM 16 bit audio is used to transmit all the frequencies in the audio range uncompressed and at a rate of 44100hz.

RTP header size is minimum of 12 bytes and can even support extra optional headers. Fields of RTP header are as follows
1. **Version:** This field caries information about version of protocol and 2 bits in size.
2. **Padding:** This field is used to indicate padding bytes at the end of packet and this of size 1 bit.
3. **Extension**: This field gives information of any optional header after the main header in RTP and is of size 1 bit.
4. **Contributing source Count (CSRC):** Number of other contributing sources to the packet other than audio and is of size 4 bits.
5. **Marker:** Provides boundary in the data stream and is of size 1 bit.
6. **Payload type:** Is the type of data format used in payload for example L16 for 44100hz

single channel and is of 7 bits size.
7. **Sequence Number:** A unique sequence number is given for each packet and incremented for the next packets in line. The size of this field is 2 Bytes.
8. **Time Stamp:** Each packet gets its own sampling time stamp and is used at receiver side to play back the samples. This field is of size 4 Bytes.
9. **Synchronization source identifier(SSRC):** This field is used to uniquely identify the source. Size of this field is 4 Bytes.

RTP Header fields for the Experiment:
1. **Version:** Value of this field is 2 as the latest version of protocol is 2.
2. **Padding:** Value of this field is 0.
3. **Extension:** Value of this field is 0 as there is no optional header.
4. **CSRC Count:** Value of this filed is 0 as there are no other contributing sources other than audio.
5. **Marker:** Value of this field is 0.
6. **Payload type:** Value of this field is 11 frequency 44100hz.

2.2 Pulse Audio

Pulse Audio acts as a sound server, where a background process accepting sound input from one or more sources is created.
Pulse audio Api is as follows:
1. pa_simple_new creates a new connection to the server.
2. pa_sample_spec specifies channels ,rate and format.
3. pa_simple_drain wait until all the data has been played.
4. pa_simple_read records the data.
5. Pa_simple_write plays the audio.

3 Procedure

1. Audio is recorded from client side using Send Thread which uses pulse audio api pa_simple_read to record and the thread sends the rtp packet over UDP.
2. As soon as the recording buffer is full it is sent.
3. Audio is sent uncompressed, Pulse audio uses PA_SIMPLE_S16LE which uses linear PCM of 16 bit signed.
4. Audio is played after receiving the data packet Using Receive Thread which internally uses pulse audio api pa_simple_write to play.
5. Sequence number of the received packet is used to ensure in order playing of the packets. Which is a big advantage and it solves the problem which UDP faced.
6. Pa_simple_free is used to free the connection to the server.

4 Observations

1. Able to send voice over the IP with decent quality.
2. Able to even send Songs which have frequency upto 20Khz with decent quality.
3. Buffer Size used is 20000 bytes so as to increase the sound clarity.
4. Noise between packets was observed which was lowered by opening the playback stream and closing the same for each packet.
5. Due to RTPs sequence number audio corruption is avoided.

5 Improvements

1. Various compression techniques can be used to send compressed data over network.
2. Network slowness has some effect on real time feel of the chat. So a better network in terms of speed is needed.

6 Result

RTP based Real time Voice phone is implemented.

7 References

1 http://0pointer.de/lennart/projects/pulseaudio/doxygen/
2 https://wiki.multimedia.cx/index.php/PCM
3 https://en.wikipedia.org/wiki/Real-time_Transport_Protocol