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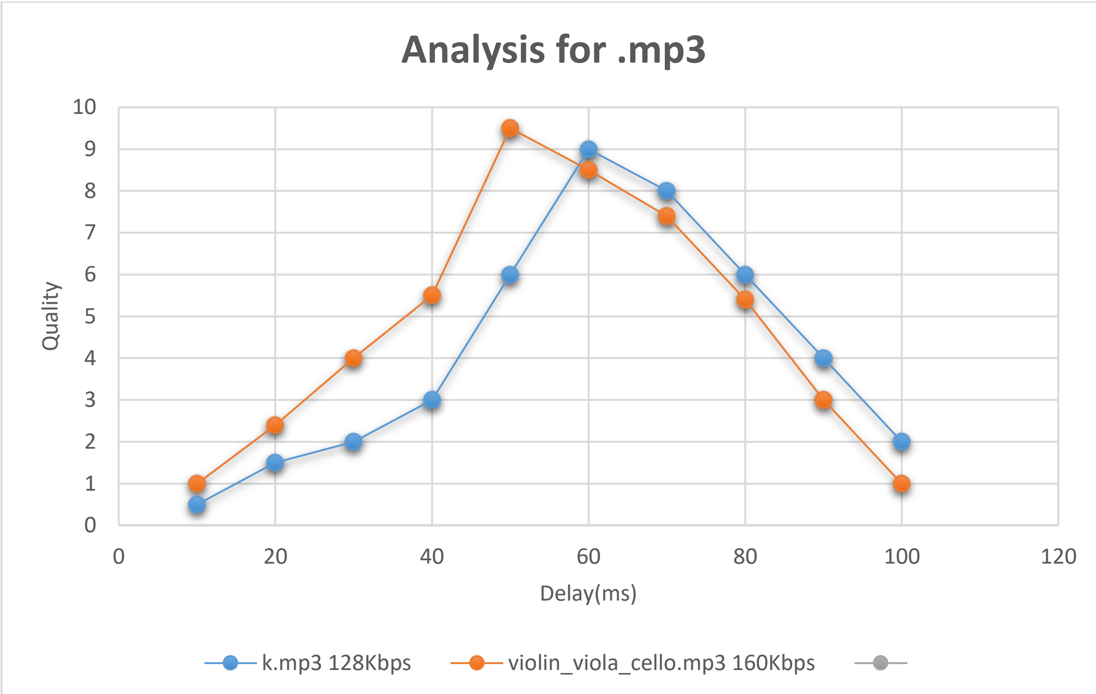
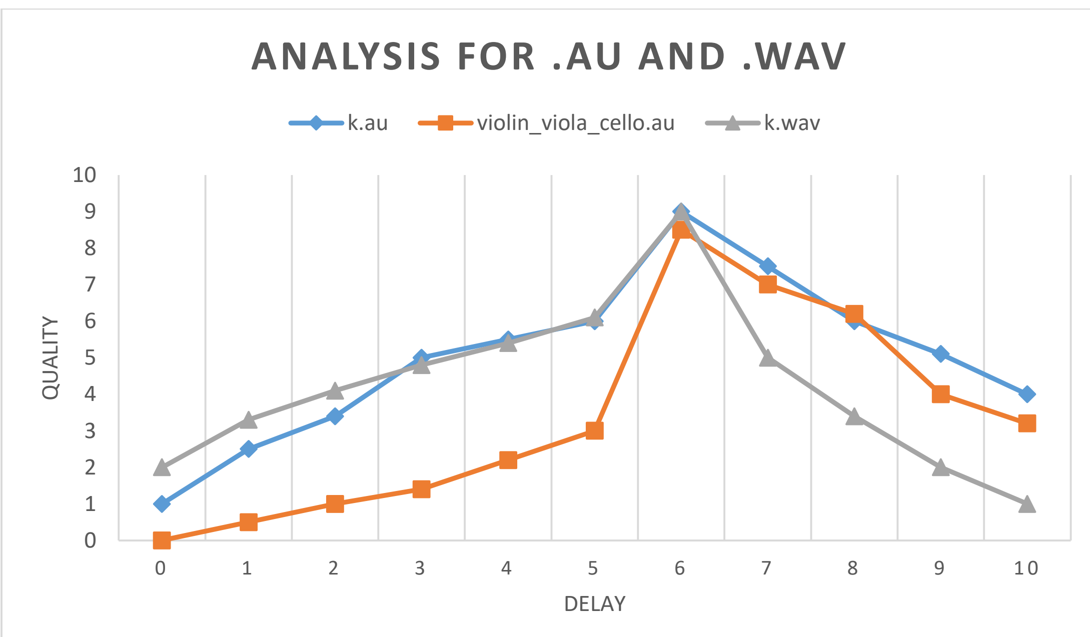
UDP RADIO ANALYSIS





We have used three file types (.mp3, .WAV, and .au). Each of them have a different bitrate. Two mp3 files (k.mp3 and violin_viola_cello.mp3) have a bit rate of 128Kbps and 160Kbps respectively. For k.mp3 the audio is almost perfectly heard at 62ms and for violin_viola_cello.mp3 its 51ms. For the .WAV and

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.au files (k.wav, violin_viola_cello.wav, k.au, and violin_viola_cello.au) the bitrate 1411kbps. The audio is almost perfectly heard at a delay of 6ms.

The ideal delay can be found by dividing the packet size by the bitrate. In this case, each packet is of size 1024bytes.

At delays below the ideal delay, there are packets arriving at a higher than what is required. There will be negative jitter in this case and the packets would overlap. If the delay is above the ideal delay, the packets arrive at a rate less than what is required. So, there will be loss of packets. At delays above 9 ms for .au and .wav files, there would be lots of loss. This is because the size of data sent in each packet is high since it has a bit rate of 1411 Kbps. So, even a small increment in delay above ideal delay would lead to a substantial increase in packet loss. In case of mp3 files since the bit rate is lower, the data sent over each packet would be much small. So, there will be more overlapping at delays below ideal delay.

So, the streaming depends on the bitrate of the file being sent, the type of encoding, and the delay.

ARCHITECTURE OVERVIEW DIAGRAM