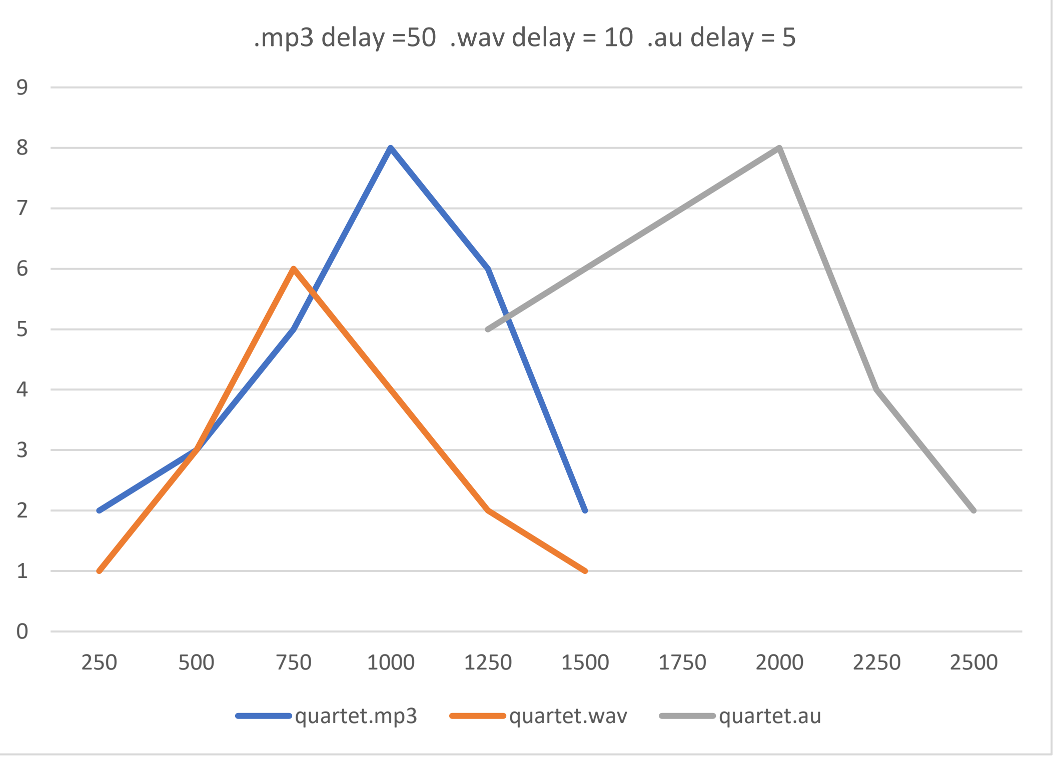
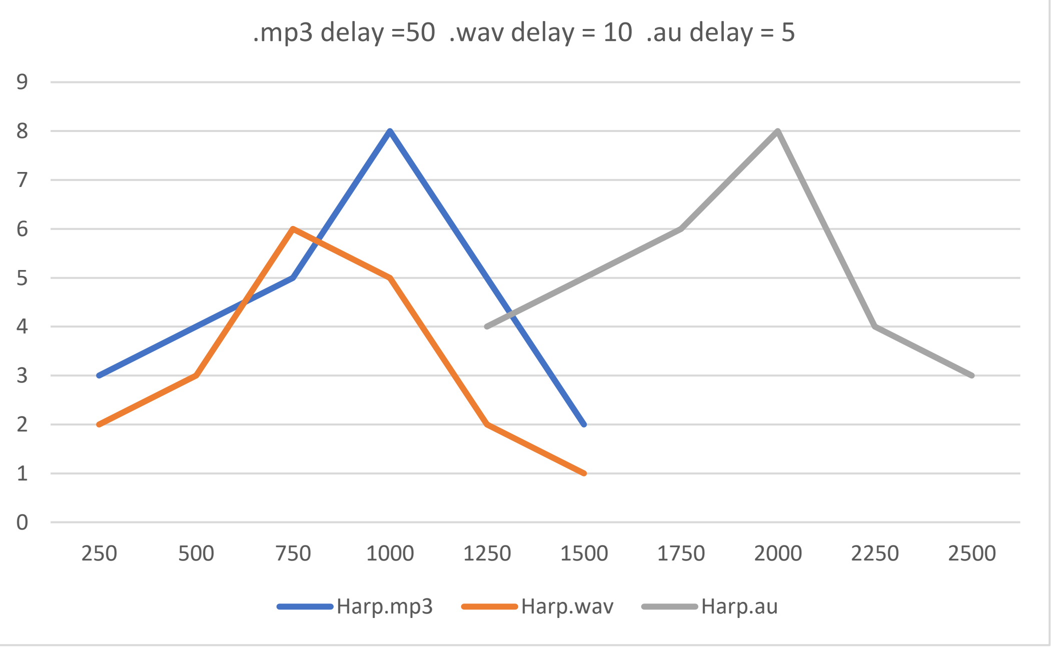
PROJECT REPORT CS 656 - 003

"Internet Radio"





Observation:

For both the audio clips (Quartet and Harp), in **'.MP3'** format, for fixed transmission delay (50 ms), quality increases with increase in packet size until some certain threshold (1000 bytes). After that, the quality starts to degrade as we increase the packet size further. This is expected behavior because as packet size increases, there will be less no of packets which will decrease overall processing delay. However, after some limit, large packets will fill up destination queue and there will be packet loss at the destination. This can be resolved by increasing transmission delay when packet size is very large and vice versa.

For **'.WAV' and '.AU'**, files are very large as compared to '.MP3' format (For ex. a 597 KB MP3 file turns to around 1.27MB and larger than 5 MB in case of '.WAV' and '.AU' respectively). Thus, minimum packet size needed to play audio is very large hence they have quite higher threshold value than '.MP3' file. But otherwise they show similar behavior.

Conclusion:

Thus, we believe that, to maintain audio quality, packet size should be varied according to the transmission delay of the channel. If the transmission delay is very less, packet size should be small and for channels with more transmission delay, packet size should be larger.

To run the code:

- ➔ java Udp 128.235.24.24 (destination IP) 23456 (port number where VLC will listen) 3000 (Packet size in bytes) 150 (Transmission delay in ms)

Command: `java Udp 128.235.24.24 23456 3000 150`

To convert files:

from **'.MP3'** to **'.WAV'**: `ffmpeg -i harp.mp3 harp.wav`

from **'.MP3'** to **'.AU'**: `ffmpeg -i harp.mp3 harp.au`