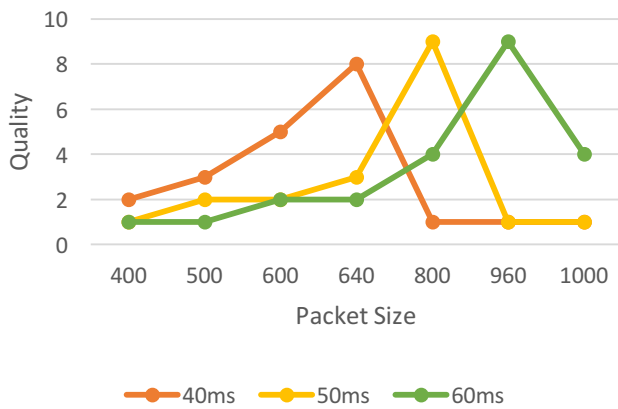
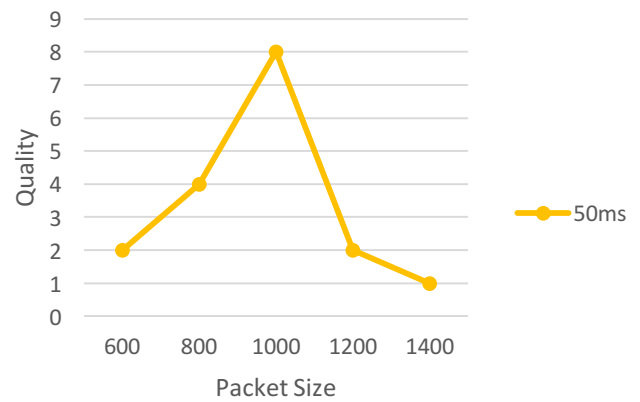


UDP Radio Transmitter Graphs Group M4

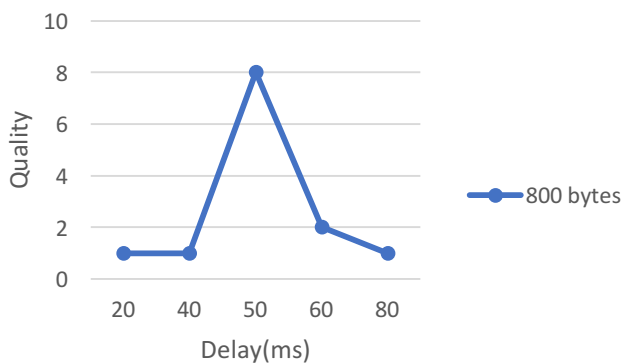
1. Quality of koikahe.mp3



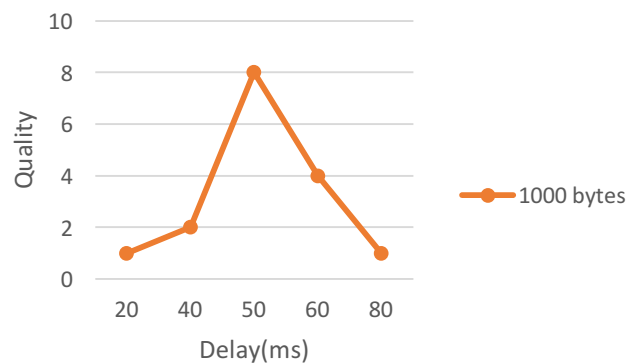
2. quartet.mp3



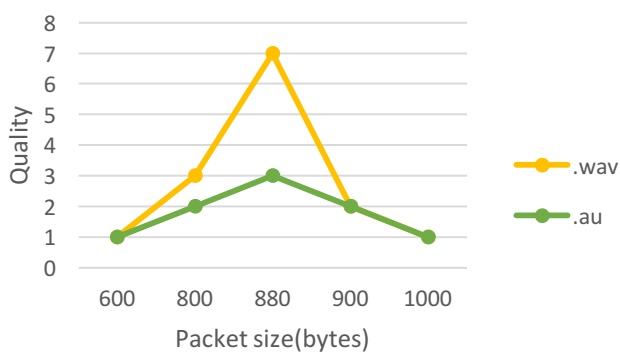
3. Quality of koikahe.mp3 at fixed packet size 800 bytes



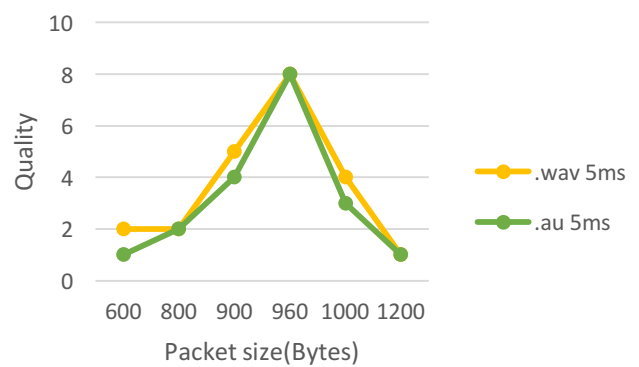
4. Quality of quartet.mp3 at fixed packet size 1000 bytes

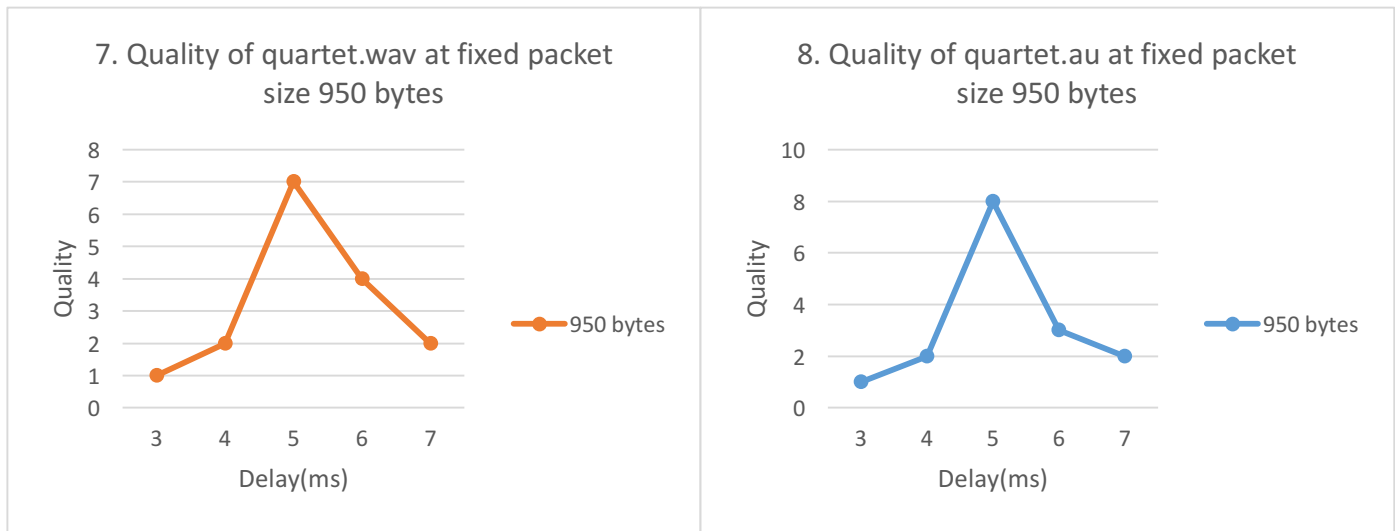


5. Quality of koikahe.wav and koikahe.au at 5ms fixed delay



6. Quality of quartet.wav and quartet.au at 5ms fixed delay





We took two songs koikahe.mp3 and quartet.mp3, and used these to stream via UDP radio and plotted 8 graphs by varying **delay, packet sizes and file format** for each. The program is named M4_udp_radiostream.c, and was compiled using :

```
gcc M4_udp_radiostream.c -o M4_udp_radiostream.
```

This executable was run as:

```
./M4_udp_radiostream <VLC_Host_ID> <Port> <File Name> <Buffer Size> <Packet Size> <Transmission Delay>
```

E.g.: ./M4_udp_radiostream 128.235.25.250 11999 quartet.mp3 1500 1000 50

The .mp3 files were converted to .wav and .au using FFMPEG. The following command and option were used:

```
./ffmpeg -i quartet.mp3 quartet.wav
```

```
./ffmpeg -i quartet.mp3 quartet.au
```

Inferences:

- For both cases, if the packet size is too less, then the audio appears to be broken but there is no loss of packets, and when the packet size is increased gradually, the quality keeps improving up to a point, then the packets start overlapping and quality deteriorates (sudden drop in quality is observed).
- For the koikahe.mp3 file, the best quality is achieved when the ratio of packets sent to the transmission delay is almost equal to 16, while that of quartet.mp3 is 20.
- The size of the .wav and .au files is almost 10x that of the .mp3 files.
- For .wav and .au files the delay is less in the order of 10, i.e. almost 5ms for same packet sizes, instead of 50.
- When checked the bitrate of quartet.mp3 file is 160kbps or 20KBps, and that of the quartet.wav and quartet.au files are 1536kbps or 192KBps.
- On lossless conversion, the .wav and .au files increase in size but have a delay that is less in the order of 10, and have almost 10 times the bitrate as that of .mp3 files.

Conclusions:

- The ideal ratio of packet size to the delay is almost equal to the bitrate of the audio (for quartet.mp3, the ideal ratio is 1000bytes/50ms = 20KBps, which is equal to the bitrate.)
- If ratio is less than the bitrate i.e. small packet sizes, then the audio plays with large gaps in between but there is no loss in packets.
- If ratio is greater than the bitrate then the packets overlap, and the lyrics become undiscernible.