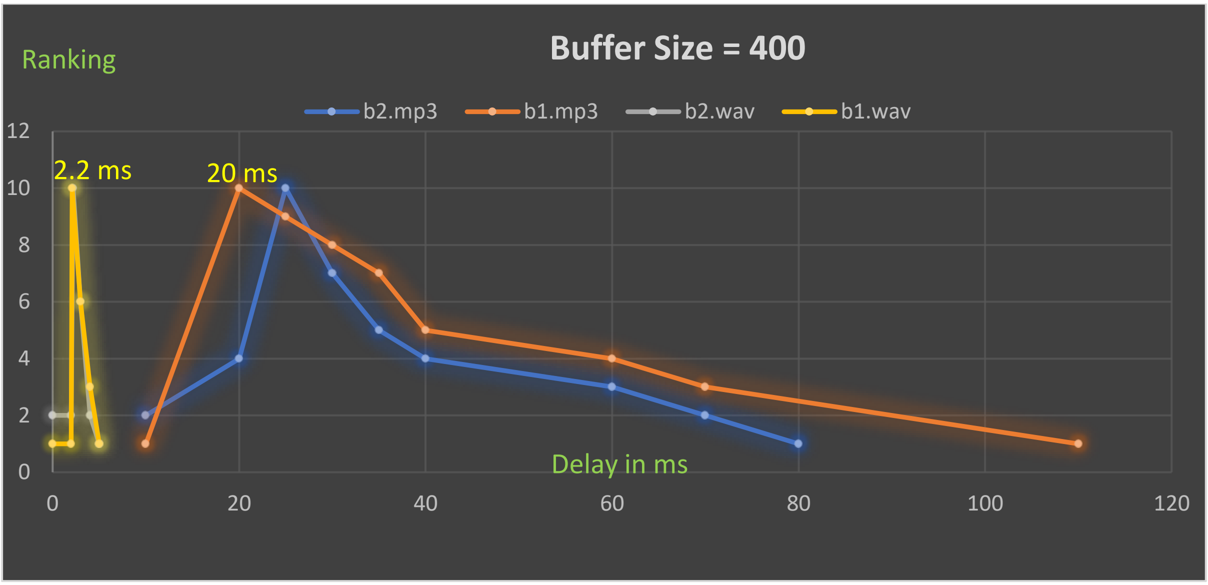
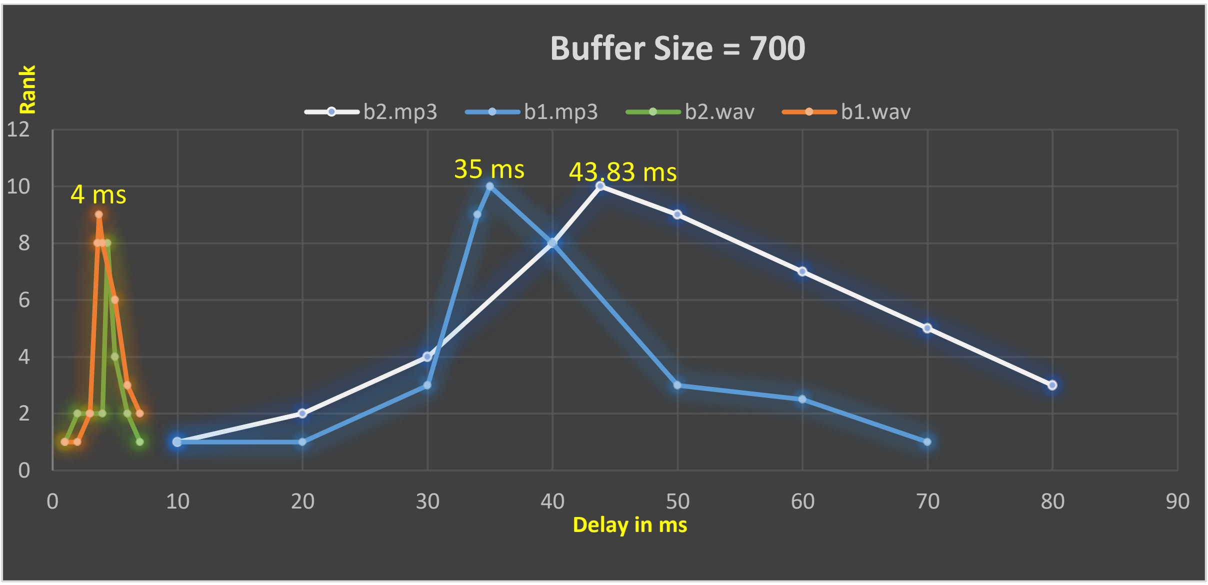
To convert mp3 to wav files used command:./ffmpeg -i b2.mp3 b2.wav and ./ffmpeg -i b1.mp3 b1.wav

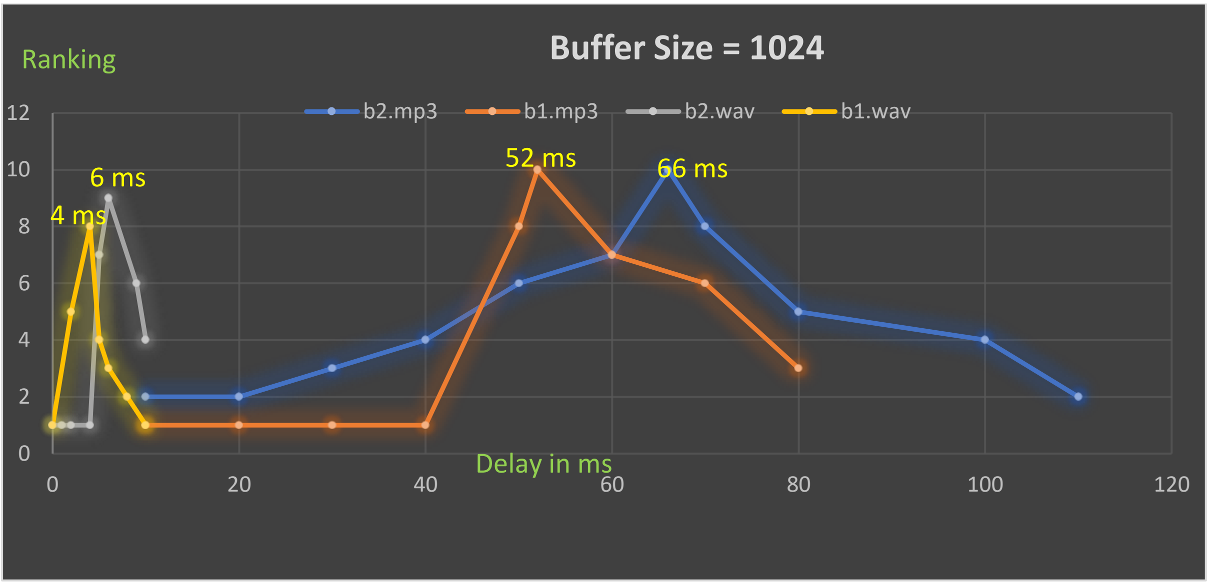
Graph:



Results for b1.wav and b2.wav were almost same. For b2.mp3 best quality was observed at 25 ms.



Combination buffer size 700 and delay 43.83 gave the best results among all combinations of buffer and delays.



Could not here anything till 40 ms for b1.mp3, At 0 ms b2.wav was the fastest among all the buffer and delay combinations.

Analysis:

- For mp3 files on delay 10ms it seemed like packets were playing too fast and most packets were skipped. As moved forward towards the best quality, quality was becoming better than previous one. For wav files delay at 1ms or 2ms, in starting could barely hear anything and when it played, played too fast.
- After best quality delay, as increased delay timing, pauses in audio were increased.
- Jitter won't affect the transmission as traceroute shows hops which are consistent and not showing any significant latency:

  1 32 ms 32 ms 31 ms afsaccess1.njit.edu [128.235.208.201]
  2 34 ms 36 ms 31 ms afsaccess1.njit.edu [128.235.208.201]
  3 31 ms 31 ms 37 ms afsaccess1.njit.edu [128.235.208.201]

Factors impacting quality:

- Command ./**ffprobe -v error -show_format -show_streams file_name** was used to get the bit_rate.
  1. When input bitrate (It can also be seen in vlc's statistics) matches to file's bitrate best quality was observed. Bit rate for b2.mp3 is 128Kbps. For buffer size 700 at 43.83 ms (43.83*23=1 sec) delay, sending 700*8*23 = approx 128 bits per second, that's why getting best quality. As we take lesser delay, for i.e 10, we are sending more bits per second (560 bits per second) that's why playing too fast, and for larger delays sending lesser bits that's why getting delays/pauses while playing.
  2. Similarly, for other combinations where file's bitrate and input bitrate are same we get best quality. As we can see for uncompressed file format file size is larger, they have higher bit rates and thus their best quality satisfies at lower delays as compare to compressed files.
  3. Finally, we conclude that its combination of byte size and delay which decides input bit rate, if matching to file's bit rate then it gives the best quality.