Low Bit Rate MP3 Audio Preference In the Modern Music Streaming Environment

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Abstract

The present study assessed the effect of signal-to-noise ratio and level on the perception of quality between lossless, CD-quality musical audio and lossy, mp3-encoded audio of various bit rates. As expected, the results show strong listener preferences for bit rates greater than 32kbps for all conditions. Generally, listeners showed no significant preference for bit rates above 96 kbps, regardless of presentation level or the presence of background noise. These results suggest that an MP3 encoding bit rate of at least 96 kbps will be audibly indiscernible from the original, CD-quality audio by the vast majority of listeners utilizing streaming music services, even in quiet environments using studio-quality headphones. Even though improvements have been made to perceptual coding since 1991, our results show the original MPEG-1 Layer 3 (MP3) algorithm to still be a dramatically efficient, robust, and transparent lossy audio codec, especially in the modern music streaming environment

Background

Popular audio compression codecs:

- MP3:

The Moving Pictures Expert Group (MPEG) was established by the International Organization for Standardization and the International Electrotechnical Commission (referred to collectively as the ISO/IEC) in 1988 to develop encoding and compression standards for digital video and audio. In 1991, this group developed the MPEG-1 Layer 3 (MP3) audio codec, which successfully became a standard in digital audio due to its widespread use during the growth of the internet's early digital music broadcasting and transmission services [7].

The MP3 compression algorithm is a "lossy" compression algorithm, meaning that it removes information in the original, CD-quality file (commonly referred to as a "lossless" file) to create a file that sounds nearly identical to the original file, but is much smaller in size, and thus more convenient to transmit and store digitally. This compression algorithm uses what is known as perceptual coding to reduce the size of the audio file. The primary goal of perceptual coding is to reduce the size of an audio file while retaining a listening experience identical to that of the original file [9]. By taking advantage of fundamental limitations of the human hearing system, a well-designed perceptual coding algorithm can have a range of encoding bit rates for which the reconstructed audio is perceptually indiscernible from the original lossless audio file. The MP3 algorithm, for example, was designed from its inception to work well at bit-rates as low as 128 kilobits per second (kbps) [7], a more than ten-fold reduction from typical lossless bit rates of 1411 kbps (the common standard for lossless CD-quality stereo audio, which has a sample rate of 44.1kHz and a bit depth of 16 bits).

However, even twenty-five years after the invention of the MP3, many audio enthusiasts insist that bit rates as low as 128 kbps - a loss of more than nine-tenths of the original lossless audio information - must deteriorate the quality of the listening experience. To understand how the MP3 compression algorithm can reduce the bit rate of an audio file so drastically while retaining a perceptually identical listening experience, it is important to understand the basic perceptual coding fundamentals used in the algorithm itself.

The MP3 algorithm, and most perceptual coding algorithms developed since, are founded upon a basic structure of encoding digital audio, as outlined by the block diagram in Figure 1 as taken directly from [7] and [9].

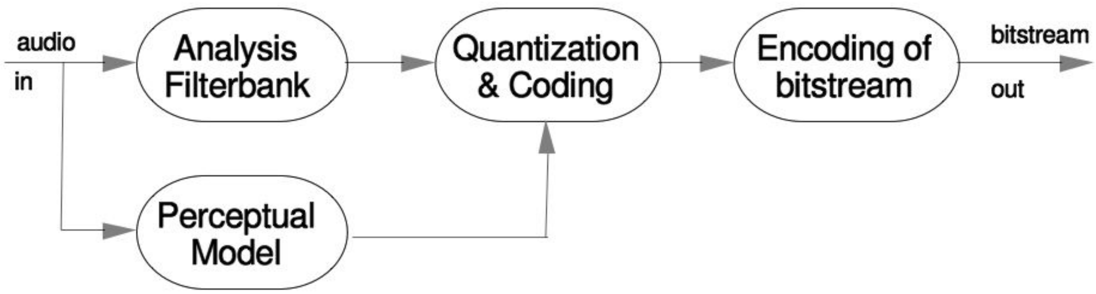

Figure 1: Simplified Block Diagram of the MP3's Perceptual Coding Algorithm

In essence, the MP3 algorithm hides its own distorting of the audio signal underneath the auditory masking threshold for a given audio signal. In other words, the musical audio signal in the resulting MP3 encoded audio will be stronger than the noise added by the MP3 algorithm to a degree that it will be imperceptible in as many frequency bands as possible. The algorithm's perceptual model continually calculates masking thresholds within frequency subbands critical to human hearing and makes sure a majority of its quantization noise lies below these thresholds [21].

Lossy audio created by this type of compression scheme (as well as the ones outlined below) are especially prone to a few different types of undesired artifacts. One of these is the complete removal of extremely high or low frequency information. This occurs when spectral data is so low for a given frame and subband that the quantization noise cannot possibly fall below the desired masking threshold, and the subband is completely zeroed out. Another possible artifact, called a pre-echo, occurs when quantization noise is synthesized back into a given frame such that it precedes the transient from which it was created. Other artifacts occur only at very low bit rates, and are negligible to the present study [7].

Both of these artifacts are, however, still more likely to occur at lower bit rates. It is also important to keep in mind that different pieces of music will be more or less prone to each of these artifacts, e.g. an audio file with lots of very low-level high frequency content will be more prone to have that content deleted than an audio file with lots of high-level high frequency content [7]. Adding yet more variability to quality between different MP3 files, the MP3 scheme does not dictate exact encoder specifications, though it does make detailed recommendations [8]. Many encoders of different quality levels exist, though most widely used encoders today are of high quality (one example being the open source LAME encoding software [10]). It is fair to assume that most streaming music services use high quality encoders when encoding their audio files.

Additional technical information on the MP3 algorithm can be found in [7] [8], [18] and [19].

- AAC:

In 1994, the ISO/IEC developed the MPEG-2 Advance Audio Codec (AAC), an improvement upon the MP3 compression scheme which largely served to increase the efficiency of the coding algorithm. AAC was seen as the "second generation" of MP3, and, at an algorithmic level, the two compression schemes are very similar, with the AAC adhering to the same basic compression process as the MP3. As outlined in Table 1, AAC is one of the most popular compression formats utilized by music streaming services today.

In terms of coding efficiency - that is, quality versus bit rate - the AAC codec boasts an impressive improvement of, on average, 70% lower bit rate for the same audio quality when compared to MP3 [7]. Thus, an AAC file at 50% the bit of an MP3 file will almost certainly contain *more* spectral information than its MP3 counterpart.

AAC improved upon the MP3 codec in a number of ways, most being subtle improvements on select blocks within the MP3 compression scheme. Two of the most notable technical improvements are (1) the shortening of the filterbank's impulse response to reduce pre-echo artifacts - and (2) increasing of frequency subbands to 1024 from the 576 used in MP3 - giving finer resolution in the frequency domain [7].

- Other popular modern compression schemes (AAC+, Ogg Vorbis, Opus):

AAC+ is a popular extension of AAC optimized for low bit rates between 24 kbps and 64 kbps [22]. More information on AAC+ and other AAC extensions can be found in [23].

Ogg Vorbis and Opus are both free and open source audio codecs, and, being free of licensing worries, are used by a few notable streaming services today (see Table 1). Though these algorithms are both founded in the same basic perceptual coding theory outlined for the MP3 codec, it is important to note that both of these formats support variable bit rate encoding, which is not offered by the MP3 algorithm. Ogg Vorbis and Opus both have the option for either constant or variable bit rate [13] [14].

Reliable subjective quality comparison studies between these two codecs and the AAC/MP3 codecs are lacking in today's literature, though objective quality comparisons, where they exist, show a close level of similarity. Some objective quality evaluations of Ogg Vorbis and Opus can be found in [2], [5], and [15].

The present music streaming environment:

Demand for high-fidelity music has grown as music-streaming services such as Spotify, YouTube, and Pandora have risen to prominence as industry standards. According to a study done by Next Big Sound, in 2015 there were just over one trillion songs streamed by consumers across the globe. The rise of streaming audio has introduced a great deal of scrutiny on lossy audio compression. Specifically, audio quality enthusiasts and professionals (referred to collectively henceforth as 'audiophiles'), have created a demand for streaming services to offer lossless music streaming. Some streaming services, such as Tidal have recently begun to offering lossless audio streaming to consumers. Other popular services, to the dismay of audiophiles, offer streaming bit rates as low as 64 kbps.

The following chart outlines the minimum and maximum bit rates, as well as the compression methods, of the most popular music streaming services in use today[1]:

Streaming Service	Minimum Bit Rate [kbps]	Maximum Bit Rate [kbps]	Compression Method
Spotify	96	320	Ogg Vorbis[11]
YouTube[2]	128	198	Opus / AAC[3]
Pandora	64	192	AAC+
Apple Music		256 (only available bit rate)	AAC
Tidal	96	1411	96 kbps - AAC + 320 kbps - AAC 1411 - none [12]

Table 1: Bit rates and compression methods used by popular streaming services

It is important to note the ubiquity of the AAC and AAC+ algorithms in the music streaming market, as well as the wide range of bitrates utilized by music streaming services.

Because higher bit rate files are more expensive for music services to host and serve, there is a clear need to determine a minimum bit rate below which audio becomes subjectively unsatisfactory to the listener using the streaming service. For an audiophile in a high end studio environment, the definition of "unsatisfactory" will of course be more stringent than that of the average casual listener using a mobile streaming app during their commute.

Subjective quality tests of popular compression formats:

Numerous studies have shown that the average listener has difficulty discerning lossless and lossy audio above a certain encoding bit rate. This cutoff has previously been shown to lie between 320 kbps and 192 kbps for most genres of music under ideal listening conditions [1] [2] [4] [20]. The most critical audio quality enthusiasts claim to perceive lossless music as sounding distinctively better than mp3-encoded music of any bit rate, though this perception has yet to be proven reliably in a blind study. In fact, some studies (including ours) have demonstrated that

[1] This chart excludes the popular streaming service Soundcloud, which hosts songs of various codecs, both lossless and lossy, with the quality of each file being determined entirely by the file uploader.

[2] It is important to note that much of YouTube's musical content is user uploaded, and is often encoded at bit rates lower than those provided, using encoders of varying quality.

[3] Depending on user's browser and operating system

average listeners may prefer lossy audio in some circumstances [2], seemingly in cases of limited dynamic range of the original CD-quality audio.

Other experimental factors identified in the literature as possibly significant to listener preference include listener familiarity with the songs, quality of the sound source, and the transducer used to deliver sound [1] [3]. However, despite their importance in most music streaming environments, no studies have characterized the effects of sound level and signal-to-noise ratio (SNR) on listener preference of lossy audio.

Introduction

The present study:

The present study incorporated real-world factors to best represent the listening environments in which music streaming most commonly occurs. Given the average consumer's preponderance for use of entry-level headphones, such as those included with mobile phones, this study used studio-quality headphones to provide a 'best-case-scenario.' To simulate realistic environments with background noise, this study included spectrally shaped noise resembling ambient sounds present on a subway car. Background noise is likely to have an adverse impact on the perception of music quality due to the poor acoustical isolation of most portable transducers. Based on this assumption, conclusions drawn from studies done on the perception of music quality without competing background noise may be ecologically irrelevant to real-world listening conditions.

The present study examines subjective versus objective degradation of digital audio quality in both silent and noisy listening environments, as well as at both high and low listening levels. Through the use of a forced-choice paradigm between two audio samples of varying bit rate, listeners chose their preferred audio under four different environments (lower level and noiseless, higher level and noiseless, lower level and noisy, and higher level and noisy) and with three song samples differing in genre and dynamic range.

Methods

Listeners

Listener recruitment was completed primarily through Northeastern University and included 14 listeners ages 18-30; 10 females and 4 males. All listeners reported that they had normal hearing and had never been diagnosed with a hearing loss. Several of the listeners reported some degree of musical training and possessed knowledge of audio fidelity and digital audio compression.

Materials

This experiment was run using MATLAB (2015a) on a Windows XP computer using a LynxTWO sound-card and Sony MDR-V6 headphones. Sony MDR-V6 headphones are commonly used in music studios for mixing due to their relatively flat frequency-response curves and affordability. They are objectively superior in fidelity to most popular portable transducers.

The three musical samples used in this study were:

Figure 1

Beethoven - 9th Symphony: (16:05 - 16:35)	High dynamic range
Daft Punk - Around the World: (5:00 - 5:30)	Medium dynamic range
Green Day - American Idiot: (1:40-2:10)	Low dynamic range

As noted above, thirty seconds from a representative section of each song was used. Those thirty seconds looped repeatedly until the listener had made a preference decision. The lossless tracks were encoded to MP3 via the digital audio workstation Logic X (which uses the original MPEG-2, Audio Layer 3 encoding algorithm [16]) at three additional bit rates. The bitrates used are included in Figure 2.

The MP3 algorithm was chosen over the AAC algorithm because of the importance of the MP3 algorithm as the first widely used perceptually coded audio compression algorithm, and its continued use to this day. The AAC algorithm currently utilized by most streaming services is largely an improvement of the MP3 algorithm, rather than a replacement, and uses the same basic structure with more efficient subtleties, retaining more spectral content for the same given bit rate. Thus, our results can be reasonably extrapolated to services using AAC compression.

Using these four bitrates, four principal conditions were then created. Figure 3 lists the level and background noise conditions that were used.

Figure 2

Compression bit rate	Real World Approximate Correlate
1411 kbps	Lossless, CD Quality
160 kbps	Spotify Desktop Standard / Mobile High Quality[4]
96 kbps	Spotify Mobile Normal Quality
32 kbps	Low Quality

Figure 3

	Music Level	Noise Level	SNR
High Level, No Background Noise	80 dB SPL	0 dB SPL	--------

[4] As noted previously, Spotify uses the lossy Ogg Vorbis format rather than the MP3 format. A direct comparison to Spotify's services, therefore, must be based on the assumption that Ogg Vorbis and MP3 formats of our given bit rates are similar enough to be subjectively indistinguishable.

High Level, Background Noise	80 dB SPL	74 dB SPL	+6 dB
Low Level, No Background Noise	60 dB SPL	0 dB SPL	--------
Low Level, Background Noise	60 dB SPL	54 dB SPL	+6 dB

Song levels were adjusted to have an average RMS (root-mean-square) value which corresponded to an output of 80 dB SPL and 60 dB SPL for the "low" and "high" listening levels. To emulate the average acoustic environment of a transit commuter, background noise was created using normally distributed Gaussian white noise filtered by an averaged spectral fit of a recording taken inside of a Boston Green Line MBTA subway car in motion.. The subway recording was taken using a Zoom H6 portable recorder through its built in XYH-6 X/Y capsule held at an ear level typical of standing commuters. The recording was forty-four seconds in length. It captured the sound of the subway car in motion and included other ambient noises, such as passenger's speech. For the noisy conditions in the present study, the derived background noise was set to 6 dB below the average RMS level of each song sample.

There were three repetitions for each pair of conditions tested (level and noise). This resulted in 120 trials for each of the three song, or a total of 360 trials. Given the attentional challenges associated with lengthier test durations, this study was limited to 360 total trials. It was estimated that this number of trials would take our listeners approximately 90 minutes to complete.

Procedures

Listeners were seated in a double-walled, sound-attenuating listening booth in the Forsyth building at Northeastern University. A monitor displaying the Matlab GUI was positioned so listeners could comfortably view it through the booth's window. Listeners were instructed to select which of the two samples (Labeled 'A' and 'B'), they preferred after listening to each as many times as desired. Listeners were forced by the GUI to listen to each sample at least once before they could proceed to the next trial. The two samples differed only in bit rate to provide a comparison between bitrate preferences for the genre, level, and background noise conditions. It is important to note that listeners were specifically told to select the sample they preferred, rather than attempt to identify the higher bit rate option. Due to the large number of trials, listeners were allowed short breaks throughout the experiment. The time spent on each trial was recorded, but was not displayed to the listeners.

Results and Discussion

Using SPSS (v.23), a repeated-measures ANOVA ($\alpha = 0.05$) with Huyhn-Feldt correction was performed. Bit rate comparison ($df = 3.05$, $F = 139.764$, $p < 0.001$) and bit rate comparison X noise ($df = 1.799$, $F = 3.495$ $p < 0.02$) were found to be statistically significant. Statistically significant differences were not observed for the noise or level conditions.

Each panel in Figure 4 corresponds to one of the songs used in the experiment. The x-axis of the graph represents the different forced-choice bit rate comparisons (a different bit rate "matchup"), while the color of the particular bar within a matchup represents the SNR and level combination, in accordance with the legend in the upper right corner. The y-axis of the graph represents the relative frequency that the higher bit rate file was chosen, averaged from all trials and all participants. Each bar has three horizontal lines corresponding to, ascendingly, (1) the lower boundary of the 95% confidence interval, (2) the mean of the data, and (3) the upper boundary of the 95% confidence interval. Where there is only one line (as in some 32 kbps cases), all decisions were in agreement across the entire listener and trial pool.

A 95% confidence interval near a probability of 1 on the y-axis means that the higher bit rate was chosen consistently as better, while an interval near 0 means that the lower bit rate was chosen consistently as better. An mean of 0.5 infers that listeners were equally likely to select either sample as the better one, presumably because the difference between the two was imperceptible. Intervals that did not cross 0.5 then, exemplified a slight preference for either the higher or lower bit rate versions.

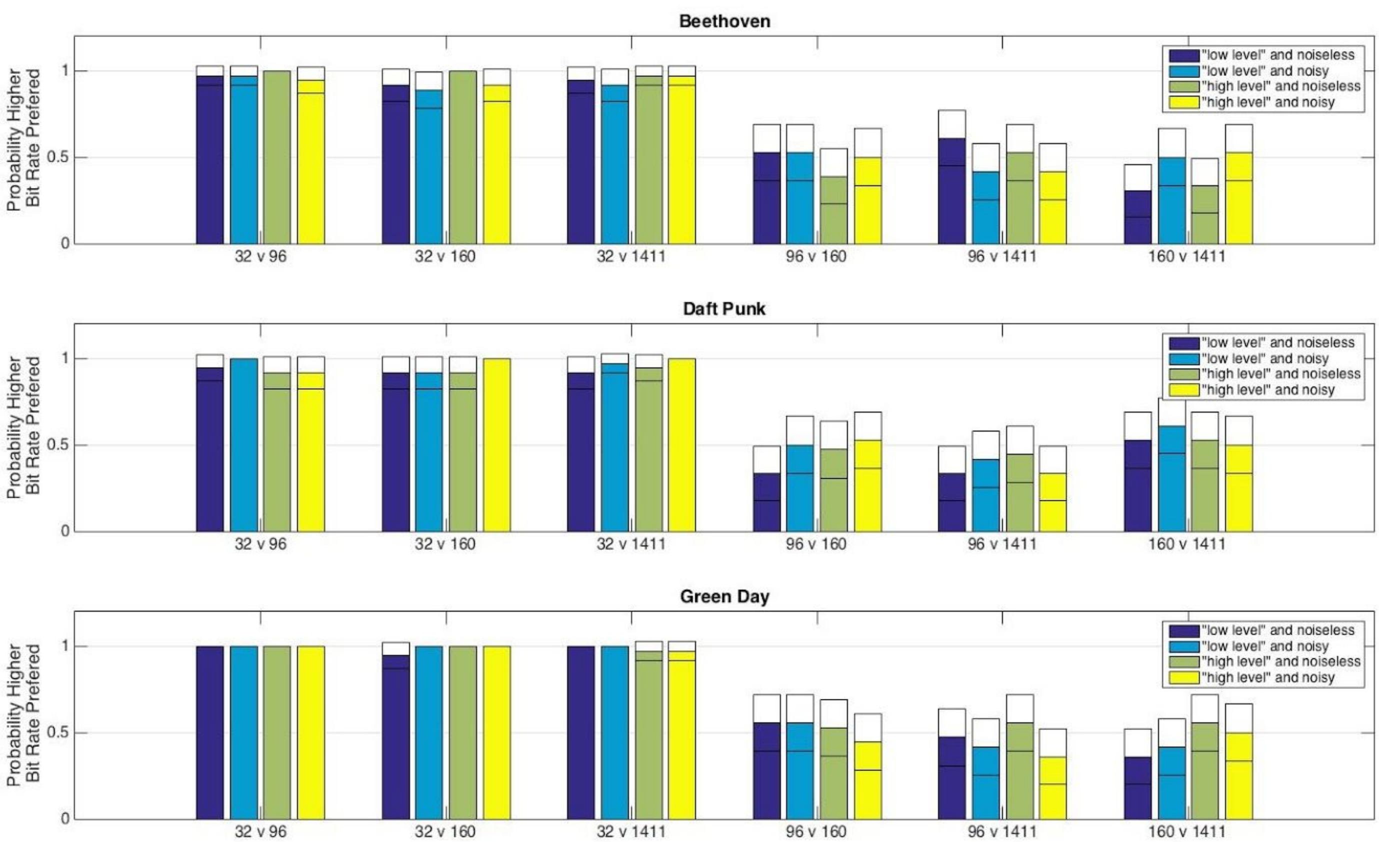

Figure 4. Probability that higher bit rate file was chosen by listeners across trial and listener pools for each condition and bit rate "matchup"

As predicted, listeners preferred higher bit rates over the 32 kbps samples nearly 100% of the time. Objectively, 32 kbps encodings represent a CD-quality music signals very poorly. The 32 kbps condition served as a control to ensure that listeners were not making their selections randomly.

A more interesting pattern emerges for bit rate comparisons above 32 kbps. Intuitively, one would expect the higher bit rate to be preferred in most case; however, for comparisons of 96 vs. 160, 96 vs. 1411, and 160 vs. 1411 kbps, the frequency with which a higher bit rate was preferred by the listener fell to around 50% for most conditions, suggesting that listeners did not prefer higher bit rates in these cases.

There is one notable exception to this trend observed under a few conditions in the 96 kbps comparisons for Daft Punk's "Around the World," where the 95% confidence interval fell significantly below 50%, indicating that listeners prefered the lower bit rate of 96 kbps over both 1411 kbps and 160 kbps. While it is unknown why listeners preferred the lower bit rate in this instance, we posit it results from the Sony MDR-V6's inability to accurately reproduce very low frequencies. To investigate, we took the Welch's power spectral density estimate of both the 1411 kbps file and the 96 kbps file and subtracted the latter from the former, giving the spectrum in Figure 5. At 20 Hz, the lossless sample was 20 dB higher in sound intensity level than the lossy sample. A loss of low frequency energy may have improved the perceived sound quality in two ways. First, it may have limited low-frequency distortion induced by the transducer. Secondly, reduction of a large quantity of low-frequency energy may have reduced the spread of masking. [6]

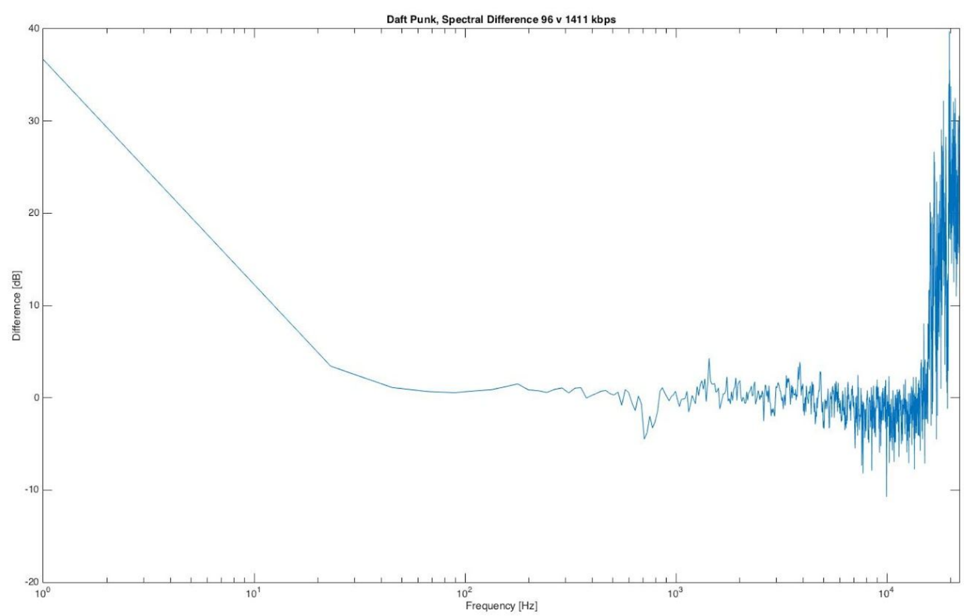

Figure 5. The energy lost as a function of frequency during the compression to 96 kbps for Daft Punk's "Around the World".

Another notable exception lies in the noiseless conditions of the 160 v 1411 kbps comparisons of Beethoven's 9th Symphony, where listeners once again again demonstrated a significant preference for the lower bit rate. The spectral subtraction of the 160 kbps file from the 1411 kbps file is provided in FIgure 6. The cause of this preference remains unclear, even after investigation.

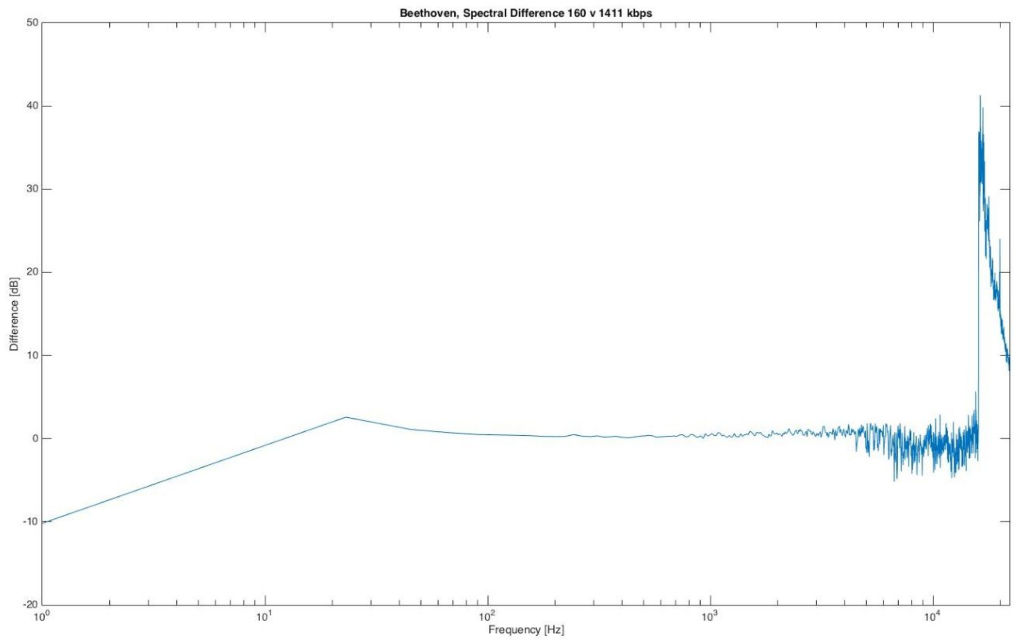

Figure 6. The energy lost as a function of frequency during the compression to 96 kbps for Beethoven's 9th Symphony.

Bit rate comparison X noise was a significant cross-factor as well. Analyzing the results, it is difficult to tell exactly how noise affected bit rate choice systematically. However, it is likely that listeners had a more difficult time comparing sound samples with competing background noise. This difficulty is likely due to subtle acoustic cues, such as high-frequency percussion 'hits,' being masked by the noise. It is possible that listeners relied upon these cues when determining a bit rate preference. However, in most cases, listeners did not make a consistent preference decision, even in the absence of noise. We had hypothesized that Beethoven's 9th would be easier to distinguish due to its larger dynamic range, but the results do not demonstrate this.

There are some potentially observed differences in bit rate comparisons among songs, though none were found to be statistically significant. The p-value for this cross-factor was not significant ($p = 0.06$), but is approaching significance. Previous studies have noted similar trends between bit rate preference and dynamic range [1], and of higher dynamic range being related to increased subjective quality in general [16].

No other factors or cross factors were found to be statistically significant. These results do not support the common belief that most listeners can easily tell the difference between lower bit rate compression and uncompressed audio. Neither level increase nor the presence or absence of background noise impacted listeners' frequency of choosing the higher bit rate. This indicates that discerning differences between bit rates at or above 96 kbps is challenging, even with relatively high fidelity listening equipment, and even using the MP3 algorithm (as opposed to the more efficient AAC algorithm).

If listeners found discerning between samples challenging, it is likely that they might spend more time attempting to make their decision. Figure 7 shows the mean decision time for each comparison across listeners and trials.

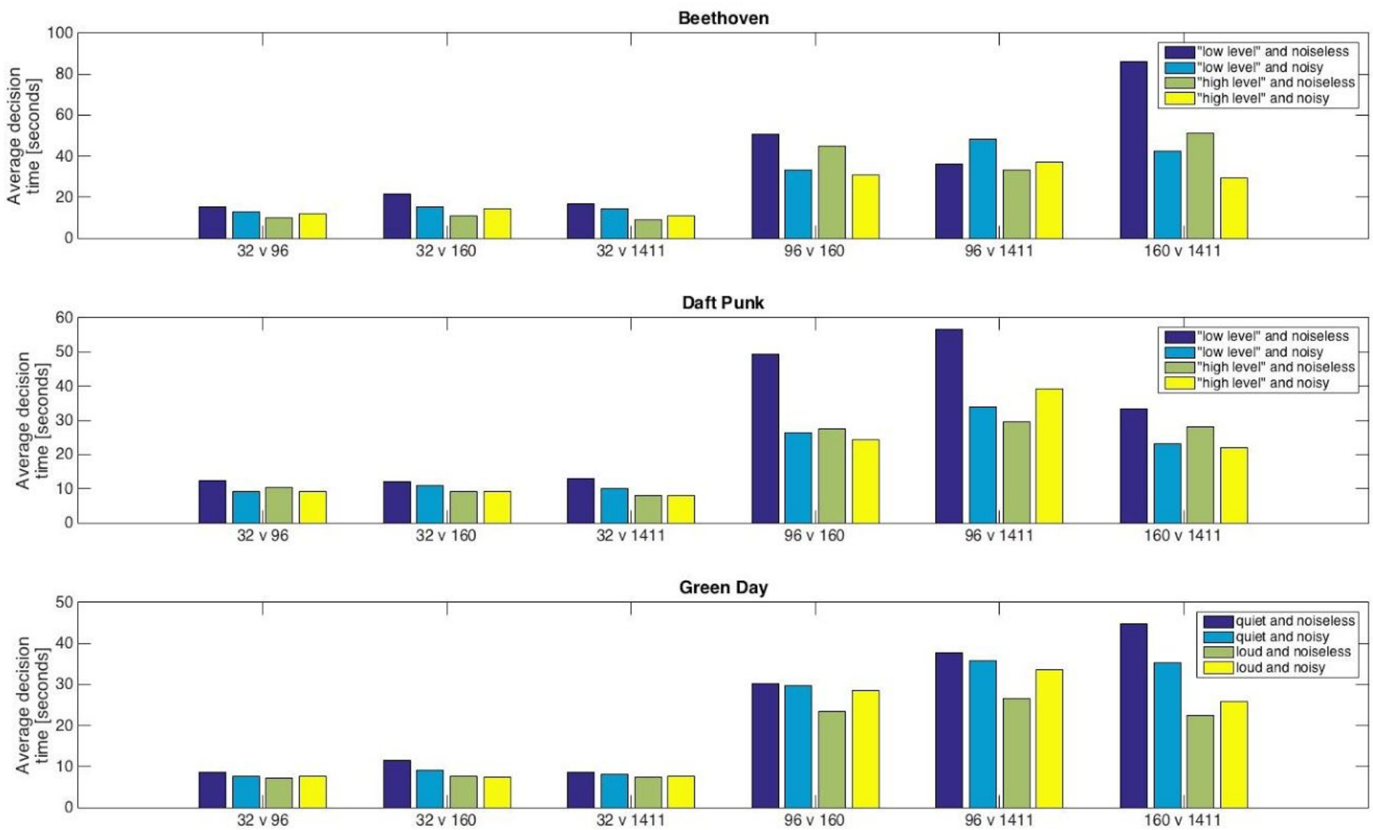

Figure 7. Mean time to determine preference for each condition

Figure 4 shows that listeners quickly made their selections on trials in which one of the sound files was 32 kbps. This is in stark contrast to the higher bit rate trials, which took three to four times as long as the 32 kbps trials. This demonstrates that listeners required more time to decide on their preference with bitrates above 96 kbps, likely because they could not hear a significant difference between the samples. These timing data support the assertion that it is not easy to distinguish between sound files with bit rates above 96 kbps.

It should be noted that the within-listeners results appeared very similar to the averaged results. No individual listeners' data varied significantly from the average data. Everyone in the study exhibited similar preferences, or lacks thereof. Even the self-reported musicians and 'audiophiles' of the group displayed little preference for higher bit rates over lower bit rates outside of the 32 kbps cases.

Conclusion

In agreement with existing literature examining less ecological conditions, this study's results suggest that scrutiny surrounding digital audio compression is largely unwarranted, particularly for mobile, casual listening in realistic environments. Listeners could not consistently identify a significant preference for bit rates higher than 96 kbps in all but a few conditions tested, in which lower bit rates were actually preferred. It is certainly possible that more trials or a larger sample size may have revealed a significant trend favoring higher bit rates under the more favorable listening conditions. However, even the results of trained listeners in this study did not hint at the existence of such a trend.

These results do not support the idea that consumers need high bit rate audio to improve their listening experience. Thus, streaming platforms may not benefit from devoting resources towards the delivery of higher bit rate audio. Standard streaming bit rates as low as 96 kbps (and likely lower for modern compression schemes) will offer the average listener an audibly indistinguishable listening experience from higher bit rates in most circumstances. Further experiments will be required to determine whether trained and experienced audio experts listening on high-end equipment in ideal environments would be able to distinguish between bit rates, but the trends observed in the present work do not indicate that concern about encoding rate would be applicable even to the most discerning music listeners.

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