Review of audio compression formats for post-processing

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Abstract – In this paper we investigate the impact of audio compression on the analysis of noise metrics using sound level meters. Various audio compression formats, including MP3, AAC, Ogg Vorbis, Opus, FLAC, ALAC, WavPack, and Speex, are applied to a series of test signals. Key noise metrics such as Leq, Lmax, Lmin, SEL,, statistical percentiles and 1/3 octave band analysis were compared.

1 INTRODUCTION

In the field of acoustics, accurate measurement and analysis of sound are critical for applications ranging from environmental noise monitoring to audio signal processing. Sound level meters (SLMs) are widely used to quantify various noise metrics, including equivalent continuous sound pressure levels (Leq), peak levels, and statistical percentiles. These measurements are fundamental to assessing noise exposure, regulatory compliance, and overall sound quality. However, in many practical scenarios, audio data is often compressed for storage and transmission, potentially altering its acoustic characteristics.

In this paper we investigate the impact of audio compression on recorded signals when analysed in a method comparable to a sound level meter, and some high level analysis of the required computational effort for the various compression methods. A variety of audio compression formats, including MP3, AAC, Ogg Vorbis, Ogg Opus, FLAC, ALAC, WavPack, and Speex, are applied to a series of generated signals. Compression levels are varied to examine the extent to which different compression methods affect key sound metrics such as Leq, Lmax, Lmin, and sound exposure level (SEL), as well as percentiles of 1, 5, 10, 50, 90, 95, and 99. A comparison of 1/3 octave band Leq results across different compression methods was also performed. By comparing these metrics with the original uncompressed wave files, this research aims to quantify the extent of distortion or loss of information introduced by various compression techniques, particularly when signals are later analysed for noise metrics.

2 ANALYSIS METHOD

The following steps outline the methodology used for signal creation, compression, and subsequent noise metric analysis.

2.1 Signal Generation

A set of test signals was generated to serve as the baseline for this analysis. These signals include a variety of waveforms commonly used in acoustic testing, such as sine waves, white noise, and pink noise, as well as complex synthetic signals designed to mimic real-world environmental noise. Each signal was sampled at 44.1 kHz, ensuring high fidelity for subsequent analysis.

The signals chosen for analysis were:

- Noises: White, Pink, Blue, MLS
- Log Sweeps (chirps): 20Hz – 20kHz in sine, square and triangle
- Sine wave (1/3 octave centres 20Hz – 20kHz)
- Square wave (1/3 octave centres 20Hz – 20kHz)
- Triangle wave (1/3 octave centres 20Hz – 20kHz)

A wave file of 1 min in length was created for each signal source. These files were used as the reference files against which all the compressed files were assessed.

2.2 Audio Compression

The generated signals were compressed using eight different audio codecs: MP3, AAC, Ogg Vorbis, Ogg Opus, FLAC, ALAC, WavPack, and Speex. Each compression method was applied across a range of predefined compression levels, as follows:

- Ogg Vorbis: [-1, 0, 1, 3, 4, 5, 6, 8, 9, 10]
- Ogg Opus: [6, 34, 62, 89, 117, 145, 173, 200, 228, 256 kbps]
- MP3: [8, 43, 77, 112, 147, 181, 216, 251, 285, 320 kbps]
- FLAC: [0, 1, 2, 3, 4, 5, 6, 7, 8]
- AAC: [8, 64, 120, 176, 232, 288, 344, 400, 456, 512 kbps]
- WavPack: [0, 1, 2, 3, 3, 4, 5, 6]
- Speex: [0, 1, 2, 3, 4, 6, 7, 8, 9, 10]
- ALAC

These compression levels were selected to represent the full range of possible data reductions, from lossless (FLAC) to heavily compressed formats, which are commonly used in both environmental and multimedia applications.

2.3 Noise Metric Calculation

After compression, each audio file was analyzed using software simulating the functionality of a sound level meter (SLM). This analysis was performed for both the original uncompressed signals and the compressed versions, with the following metrics calculated:

- Equivalent Continuous Sound Level (Leq): A-weighted (A) and unweighted (Z) Leq values were calculated to assess the overall energy of the signal.
- Lmax and Lmin: Maximum and minimum sound levels to identify the dynamic range of the signals.
- Peak Sound Level: The highest instantaneous sound pressure level recorded.
- Sound Exposure Level (SEL): The total energy of the signal over its duration.
- Statistical Percentiles: The levels exceeded for 1%, 5%, 10%, 50%, 90%, 95%, and 99% of the time.

Each of these metrics were computed using fast time-weighting, a standard method for capturing sound level variations that simulates real-time acoustic environments.

2.4 1/3 Octave Band Analysis

In addition to the overall noise metrics, a 1/3 octave band analysis was performed on each signal. The frequency spectrum was divided into 1/3 octave bands, and the Leq for each band was computed. This analysis allowed for detailed examination of how each compression method affected the frequency content of the signal across the audible range (20 Hz to 20 kHz).

2.5 Comparison with Uncompressed Signals

To assess the impact of compression, the metrics and 1/3 octave Leq results of each compressed signal were compared against the original uncompressed wave file. Percentage differences were calculated for each metric, highlighting how compression alters the signal properties. Particular attention was paid to changes in the frequency spectrum, as compression algorithms are known to affect different frequency bands to varying extents.

2.6 Tools and Software

All analysis was conducted using InstraLabs proprietary software (yet to be released) which processed the audio files and calculated the relevant metrics. The scripts utilized standard libraries for signal processing, including naudio and FFMPEG various compressions and decompression tasks This software has been calibrated against Type 1 verified noise meters and is known to have an accuracy of better than 0.1dB across all metrics.

2.7 CPU Usage and Power Analysis

An additional step in the analysis involved measuring the CPU usage required to compress and process each audio file. This was done to estimate the computational power required for each compression format and level. CPU usage was monitored during the file creation process, and the total processing time for each file was recorded.

By comparing the CPU load across the various compression methods, this analysis provides insights into the energy efficiency of different codecs. Compression formats that require more CPU power may be less suitable for energy-constrained environments, such as portable devices or large-scale processing systems. This aspect of the study is particularly relevant for applications where processing power or battery life is a limiting factor.

The overall analysis method is shown in Figure 2-1.

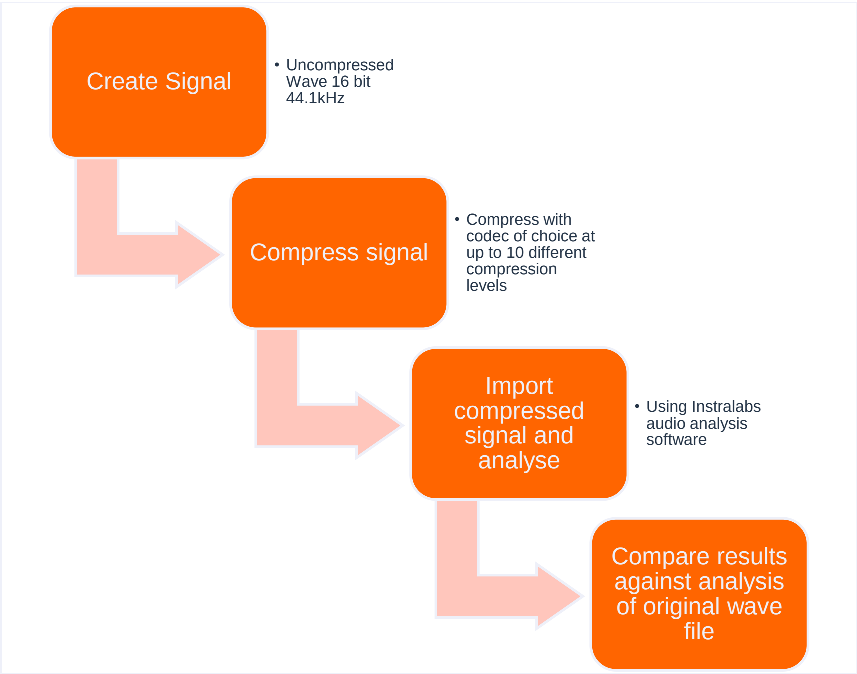


Figure 2-1 Audio and analysis process

3 COMPRESSION TYPES ASSESSED

3.1 Wave

Waveform Audio File Format (WAV) was developed by Microsoft and IBM in 1991 as a standard for storing audio bitstreams on PCs. WAV files store uncompressed audio data in Pulse Code Modulation (PCM) format, ensuring no loss of audio information, which makes them ideal for professional audio recording and editing. Due to this, WAV files have large file sizes. For instance, CD-quality audio at 44.1 kHz, 16-bit, stereo results in a bitrate of 1,411 kbps (roughly 5 meg per minute in mono).

3.2 MP3

Developed in the late 1980s and standardized in 1993, MP3 (MPEG-1 Layer III) was created by the Fraunhofer Institute to enable significant compression of audio files. It achieves data compression ratios of about 10:1 to 12:1 without greatly compromising perceived sound quality. For instance, a CD-quality audio file can be reduced from 50 MB to approximately 4-5 MB at bitrates ranging from 128 kbps to 192 kbps, facilitating the widespread distribution and storage of digital music.

3.3 Ogg Vorbis

Ogg Vorbis, developed by the Xiph.Org Foundation and released in 2000, is a free, open-source audio compression format designed to provide high-quality sound at lower bitrates than MP3. It achieves similar or better fidelity at bitrates ranging from 96 kbps to 160 kbps, compressing CD-quality audio (1,411 kbps) by a factor of about 12:1 to 16:1. Its efficient compression and open-source nature make it a popular alternative to proprietary formats.

3.4 Ogg Opus

Standardized in 2012, Opus is an open, royalty-free codec designed for high-quality, low-latency audio streaming. Developed collaboratively by Xiph.Org, Mozilla, Skype, and others, Opus efficiently handles a range of bitrates from 6 kbps to 510 kbps. For stereo music, it provides good audio quality at 96-128 kbps, achieving compression ratios of approximately 11:1 to 15:1. Opus is well-suited for applications requiring low latency, such as VoIP and live audio streaming.

3.5 FLAC

FLAC (Free Lossless Audio Codec), first released in 2001 by Josh Coalson and later part of Xiph.Org, offers lossless compression, reducing file sizes by 30% to 50% while preserving the original audio quality. A CD-quality audio file can be compressed to 700-1,000 kbps without losing any data, making FLAC ideal for archival storage, high-quality playback, and professional audio editing.

3.6 AAC

Advanced Audio Coding (AAC), developed in the late 1990s as a successor to MP3, was created by a consortium of companies including Fraunhofer and Dolby. AAC is known for better efficiency at lower bitrates, delivering similar or superior sound quality to MP3 at bitrates between 96 kbps and 128 kbps. With compression ratios of about 11:1 to 15:1, AAC has become widely adopted in streaming services, digital radio, and Apple devices.

3.7 WavPack

WavPack, developed by David Bryant in 1998, offers versatile compression options including lossless, lossy, and hybrid modes. In lossless mode, WavPack reduces file sizes by 30% to 70%, compressing CD-quality audio to 400-1,000 kbps. The hybrid mode allows users to store a smaller lossy file along with a correction file for full restoration, making WavPack flexible for both efficient storage and high-fidelity audio.

3.8 Speex

Speex is an open-source codec designed specifically for speech encoding, released in 2003 by Jean-Marc Valin under the Xiph.Org Foundation. It compresses speech efficiently at bitrates from 2.15 kbps to 44 kbps, achieving compression ratios as high as 176:1 at 8 kbps. Speex is optimized for applications like VoIP and audio streaming, offering features such as variable bitrate and voice activity detection.

3.9 ALAC

Apple Lossless Audio Codec (ALAC), introduced by Apple in 2004 and open-sourced in 2011, provides lossless compression, reducing file sizes by 40% to 60%. A CD-quality audio file can be compressed to 700-900 kbps without any loss of quality, making ALAC popular among audiophiles and professionals who need high-quality audio with reduced storage demands.

Table 3-1 Codec compression comparison

Codec	Min Data Rate (kbps)	Max Data Rate (kbps)	Typical Data Rates (kbps)	Compression Ratio	Compression Type	Notes
WAV	64	4,608	1,411	1:1	Uncompressed	Data rate depends on sample rate, bit depth, and channels
MP3	8	320	128 – 320	~11:1 to 4.4:1	Lossy	Widely used; quality varies with bitrate
AAC	8	512	96 – 320	~14.7:1 to 4.4:1	Lossy	More efficient than MP3 at lower bitrates
Ogg Vorbis	45	500	96 – 192	~14.7:1 to 7.3:1	Lossy	Open-source alternative to MP3 and AAC
Opus	6	510	96 – 128 (music)	~14.7:1 to 11:1	Lossy	Versatile; excels in both music and speech
FLAC	500	1,411	700 – 1,000	~2:1 to 1.4:1	Lossless	Compresses without any loss of quality
ALAC	500	1,411	700 – 900	~2:1 to 1.6:1	Lossless	Apple's lossless codec; similar to FLAC
WavPack	500	1,411	700 – 1,000 (lossless mode)	~2:1 to 1.4:1	Lossless	Supports lossless and hybrid modes
Speex	2.15	44	2.15 – 24.6	~656:1 to 57:1	Lossy (Speech)	Optimized for speech; not suitable for music

4 RESULTS

4.1 Compression results

Although these tests are preliminary, the analysis of CPU usage and file compression performance across different codecs reveals significant differences in both efficiency and compression effectiveness. The tests were conducted on a Zenbook Pro with a 12th generation Intel i7 14-core processor and 16GB of RAM. The original wave files were all 5,169kB in size.

Each compression process was performed five times and the results for the five runs were averaged. Table 4-1 presents the average results of all signals for each compression method and quality level.

Table 4-1 Compression processing results

Data rate/ quality level	Compression Ratio	CPU Use %	Calculation Duration (s)	Compressed size (KB)
		AAC		
64	10.7	3.2	0.9	484.0
96	7.2	3.3	1.0	718.8
128	5.4	3.4	1.0	954.2
160	4.6	3.8	1.2	1125.8
192	4.3	4.2	2.7	1233.6
256	4.2	4.0	3.0	1292.0
320	4.2	3.9	3.1	1291.6
384	4.2	3.9	3.1	1291.6
448	4.2	3.9	3.1	1291.6

Data rate/ quality level	Compression Ratio	CPU Use %	Calculation Duration (s)	Compressed size (KB)
512	4.2	3.8	3.0	1291.6
		ALAC		
6	7.9	3.1	0.1	2457.2
		FLAC		
0	2.3	2.8	0.1	3432.7
1	2.4	2.7	0.1	3282.1
2	2.6	2.8	0.1	3197.7
3	2.6	2.8	0.1	2916.1
4	2.7	2.6	0.1	2848.6
5	2.7	2.7	0.1	2815.7
6	2.9	2.9	0.1	2769.4
7	3.1	2.9	0.1	2748.1
8	3.0	2.9	0.1	2649.1
		MP3		
32	22.0	3.0	0.3	234.7
40	17.6	3.0	0.4	293.3
48	14.7	3.1	0.4	351.9
64	11.0	3.0	0.4	469.2
80	8.8	3.1	0.4	586.5
96	7.3	3.0	0.4	703.8
128	5.5	3.0	0.4	938.3
192	3.7	3.1	0.4	1407.5
256	2.8	3.2	0.4	1876.6
320	2.2	3.0	0.4	2345.7
		Ogg OPUS		
6	100.9	2.9	0.4	53.4
32	12.5	2.9	0.3	432.2
48	8.8	3.0	0.3	612.7
64	6.9	2.9	0.3	779.1
96	5.0	3.0	0.4	1060.4
128	4.0	3.0	0.4	1335.3
160	3.3	3.1	0.4	1610.1
192	2.9	2.9	0.4	1884.7
224	2.5	2.9	0.4	2157.9
256	2.2	3.0	0.4	2390.7
		Ogg Vorbis		
-1	24.7	3.6	0.5	377.0
0	33.7	3.7	0.5	233.3
1	29.0	3.9	0.5	291.2
2	26.7	3.6	0.5	335.5
3	24.7	3.3	0.5	377.0
4	25.6	3.1	0.4	410.6
5	23.1	2.7	0.4	471.5
6	20.8	2.5	0.4	578.0
8	18.4	2.2	0.5	743.8
10	9.9	2.0	0.5	1063.5

Data rate/ quality level	Compression Ratio	CPU Use %	Calculation Duration (s)	Compressed size (KB)
		Speex		
0	24.6	3.0	0.3	209.8
1	24.6	3.1	0.3	209.8
2	24.6	3.4	0.3	209.8
3	24.6	3.4	0.5	209.8
4	24.6	3.4	0.5	209.8
5	24.6	3.3	0.6	209.8
6	24.6	3.2	0.8	209.8
7	24.6	3.3	0.8	209.8
8	24.6	3.7	1.0	209.8
10	24.6	4.4	1.1	209.8
		WavPack		
0	5.0	3.4	0.1	2818.7
1	6.3	3.4	0.2	2240.8
2	7.9	3.9	0.4	1972.5
3	6.7	4.4	1.4	2119.2
4	12.6	4.1	2.7	1466.5
5	12.7	3.9	4.3	1459.2
6	13.4	3.8	4.2	1318.0

The data reveals several key trends regarding the performance of audio file formats in terms of compression ratio, CPU usage, processing time, and file size. Ogg Vorbis and Speex demonstrate the highest average compression ratios, achieving the most significant reductions in file size. In contrast, FLAC, being a lossless format, maintains a much lower compression ratio, as it prioritizes file integrity over size reduction. Formats like AAC, MP3, and WavPack offer a middle-ground, balancing compression efficiency with quality.

CPU usage varies significantly across formats, with WavPack requiring the highest average CPU power, while Speex, Ogg Vorbis, and AAC also show higher demands. In comparison, Ogg Opus and MP3 are more CPU-efficient, making them better suited for real-time processing or use in devices with limited resources. Processing times also differ, with Speex taking the longest, while FLAC, Ogg Vorbis, and Ogg Opus demonstrate faster processing, indicating their efficiency.

In terms of file size, ALAC and FLAC generate the largest files, which is expected from lossless formats. Speex and Ogg Vorbis create the smallest files, reflecting their highly efficient compression. MP3, AAC, and WavPack fall in between, offering moderately sized files that strike a balance between storage and quality. Overall, the data shows that Ogg Opus and Ogg Vorbis are among the most efficient formats, offering a strong balance between file size, compression efficiency, and processing demands.

Over 350,000 results were generated in the analysis, and this paper will not provide a comprehensive review of all results. Rather it will focus on the results of most interest – those being the highest compression ratios, with the lowest error rates.

4.2 Conversion Accuracy

The results of the analysed compressed audio files were compared directly with the results of the analysed wave files. The difference between the compressed results and the wave results will show any errors that are present in the analysis post-compression. All metrics and compression data rates were analysed, but for brevity, only a few are presented here.

Table 4-2 shows the dB error of the compressed formats of the L_{Aeq} for the log sweep sine wave, pink noise and white noise signals.

Table 4-2 dB error of LAeq, LA10, and LA90 (compressed – wave)

Data rate/level	LAeqT			LA10			LA90		
	Log Sweep	Pink noise	White noise	Log Sweep	Pink noise	White noise	Log Sweep	Pink noise	White noise
aac									
128	-0.5	-0.2	-0.3	-0.4	-0.3	-0.2	-1.1	-0.3	-0.3
160	-0.5	-0.2	-0.3	-0.4	-0.2	-0.3	-1.1	-0.2	-0.3
192	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
256	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
320	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
384	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
448	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
512	-0.4	-0.1	-0.2	-0.4	-0.1	-0.2	-1	-0.2	-0.2
64	-0.6	0.5	0.1	-0.5	0.5	0.5	-2.4	0.3	-0.1
96	-0.5	-0.2	-0.3	-0.4	-0.1	-0.1	-1.1	-0.4	-0.4
alac									
fixed	0	0	0	0	0	0	0	0	0
flac									
0	0	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	0	0
2	0	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0	0
8	0	0	0	0	0	0	0	0	0
mp3									
128	-0.5	-0.4	-0.5	-0.4	-0.4	-0.5	-1.1	-0.5	-0.5
192	-0.3	-0.2	-0.3	-0.2	-0.3	-0.3	-0.9	-0.3	-0.3
256	0	0	-0.1	0	0	-0.1	-0.7	-0.1	-0.1
32	-0.9	-0.4	-2	-0.4	-0.4	-1.9	-26.1	-0.5	-2.2
320	0	0	-0.1	0	0	-0.1	-0.7	-0.1	-0.1
40	-0.7	-0.2	-1.3	-0.4	-0.3	-1.2	-12.6	-0.3	-1.4
48	-0.7	-0.3	-0.9	-0.4	-0.3	-0.8	-11.1	-0.4	-0.9
64	-0.5	-0.2	-0.1	-0.4	-0.2	-0.1	-3	-0.3	-0.1
80	-0.5	-0.2	0	-0.4	-0.3	0	-1.1	-0.3	0
96	-0.5	-0.3	-0.3	-0.4	-0.4	-0.3	-1.1	-0.4	-0.3
opus									
128	0	0	-0.1	0	0	-0.1	-0.1	0	-0.1
160	0	0	-0.1	0	0	-0.1	-0.1	-0.1	-0.1
192	0	0	-0.1	0	0	-0.1	-0.1	-0.1	-0.1
224	0	0	-0.1	0	0	-0.1	-0.1	-0.1	-0.1
256	0	0	-0.1	0	0	-0.1	-0.1	-0.1	-0.1
32	0	-2.2	-0.5	0	-2.2	0	0.6	-2.3	-1.9
48	0	-1.6	-0.4	0	-0.2	-0.1	0	-2	-1.8

Data rate/level	LAeqT			LA10			LA90		
	Log Sweep	Pink noise	White noise	Log Sweep	Pink noise	White noise	Log Sweep	Pink noise	White noise
6	-2.7	-2.8	-5.4	-1.6	-2.1	-4.6	-27.7	-3.9	-6.5
64	0	0	-0.1	0	0	-0.1	-0.1	0	-0.1
96	0	0	-0.1	0	0	-0.1	-0.1	0	-0.1
Vorbis*									
0	-0.1	0.3	-	0	0.2	-	-3.3	0.2	-
1	-0.1	0	-	0	0	-	-2.5	0	-
-1	0	-0.2	-	0.1	-0.2	-	-1.1	-0.3	-
10	0	0	-	0.1	0	-	0.1	0	-
2	-0.1	-0.1	-	0	-0.1	-	-1.8	-0.2	-
3	0	-0.2	-	0.1	-0.2	-	-1.1	-0.3	-
4	0	-0.2	-	0.1	-0.2	-	0	-0.3	-
5	0	-0.2	-	0.1	-0.2	-	0	-0.3	-
6	0	-0.2	-	0.1	-0.2	-	0	-0.3	-
8	0	-0.2	-	0.1	-0.2	-	0	-0.2	-
wavpack									
0	0	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	0	0
2	0	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0	0

**vorbis white noise files produced an error during decoding which has not yet been identified.*

The analysis of various audio codecs shows that lossless formats like ALAC, FLAC, and WavPack maintain perfect fidelity across all test signals, introducing zero error (0 dB). These codecs are ideal for situations where preserving the exact audio quality is critical. In contrast, lossy codecs such as AAC, MP3, Ogg Vorbis, and Opus exhibit varying levels of error depending on the bitrate and signal complexity. Higher bitrates (128 kbps and above) across all lossy codecs show minimal errors, typically between 0 dB and -0.2 dB, indicating good performance for preserving audio fidelity. However, lower bitrates lead to more noticeable errors, particularly for complex signals like white noise.

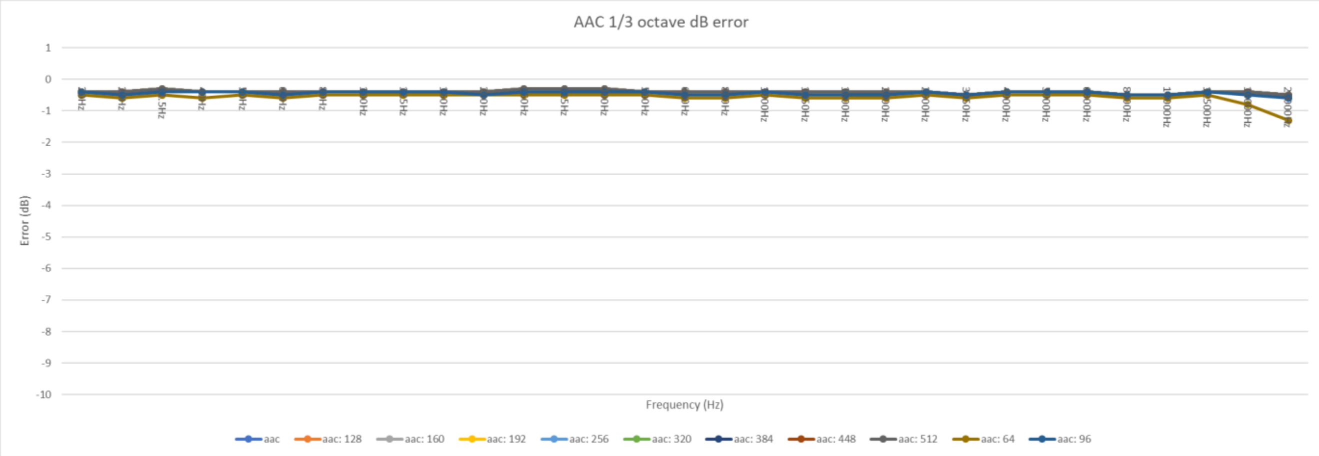
Among the lossy formats, Opus and AAC generally outperform MP3 and Ogg Vorbis at maintaining fidelity at lower bitrates. Opus, in particular, shows strong performance at bitrates as low as 64 kbps, with errors as small as -0.1 dB. However, at extremely low bitrates (such as 6 kbps), Opus exhibits significant degradation, especially with complex signals like white noise. Overall, while lossy codecs can reduce file size significantly, higher bitrates are necessary to minimize error and maintain audio quality close to the original.

The Speex method consistently produced result that were 10-20 dB lower than the wave file. It is not known whether this was the result of the compression method, or the decoding of the file or some other method.

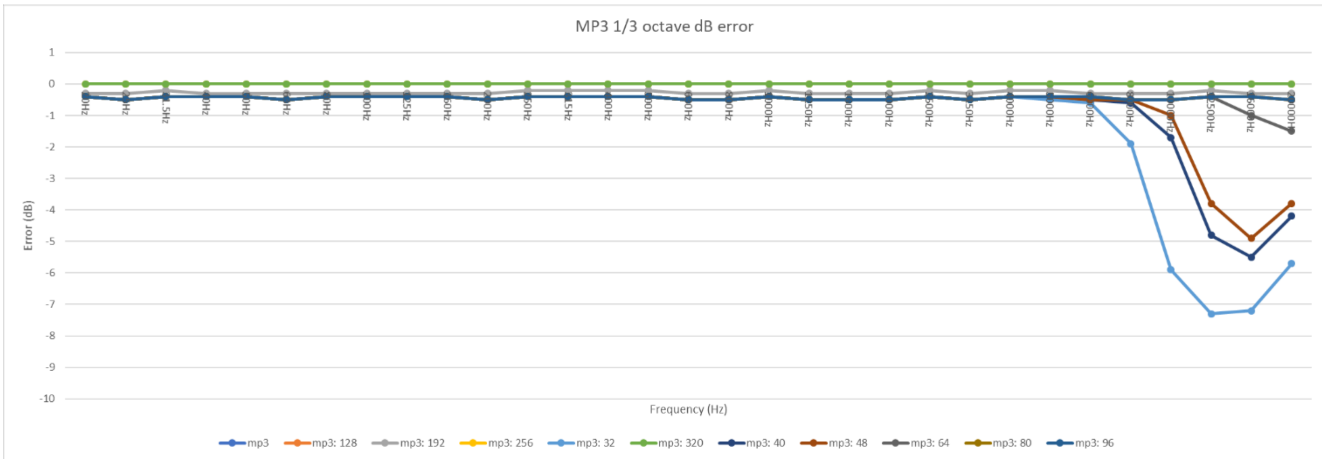
4.3 Third octave results

Results of the 1/3 octave analysis revealed that all lossless compression methods had 0dB deviation from the 1/3 octave results from the wave file. This in itself is a significant result.

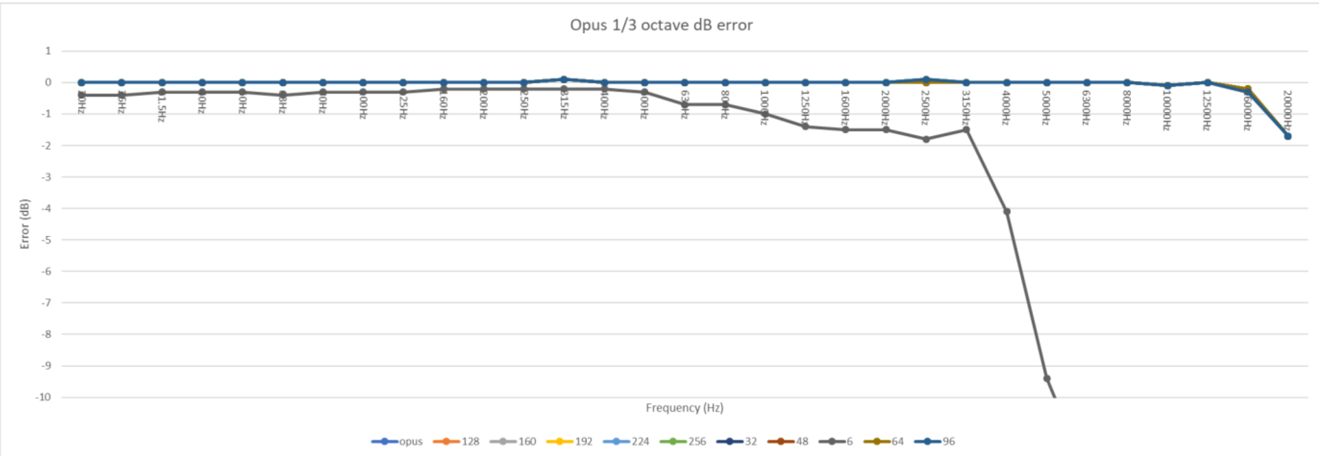
The lossy compressed 1/3 octave results showed some difference from the original wave file results. The results of the difference between the AAC, MP3 and Opus, at their varying compression rates and the original wave file results (original minus compressed in 1/3 octave bands) are show in Graphs 4-1 to 4-3.



Graph 4-1 AAC accuracy compared with original wave



Graph 4-2 MP3 accuracy compared with original wave



Graph 4-3 Opus accuracy compared with original wave

All three lossy compression methods reproduce 1/3rd Leq results within 1dB of the original wave analysis with the following notes:

- AAC is under predicting by roughly 0.5dB across the frequency range
- MP3 is highly accurate at high bit rates with a significant loss in high frequency accuracy at bit rates less than 64kbps
- Opus is highly accurate at high bit rates with a notable loss of accuracy when bit rates fall below 32kbps.
- Opus appears to have some inbuilt high frequency roll off at 20kHz.

5 CONCLUSIONS

The study demonstrates that lossless compression formats effectively preserve audio fidelity, while lossy codecs, despite offering significant file size reductions, introduce errors in noise metrics analysis. High bitrates for lossy formats, particularly Opus, maintain accuracy close to the original signals. However, lower bitrates lead to increased errors, particularly in high-frequency content. These findings show that compressed audio signals can be reanalysed and produce results that are highly accurate compared with the uncompressed format provided that lossless compression or lossy compression with high bit rates are used.

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