Lossless audio compression standards

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Outline

1. Compression

2. Basic principles of lossless compression
   - Framing
   - Intra-channel decorrelation
   - Entropy coding

3. Performance evaluation

Compression

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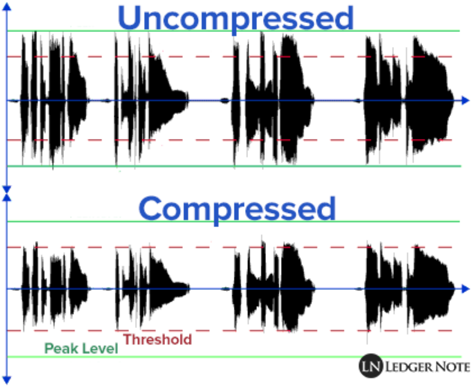
- Lossy audio compression algorithms provide higher compression at the cost of fidelity and are used in numerous audio applications. These algorithms almost all rely on psychoacoustics to eliminate or reduce fidelity of less audible sounds, thereby reducing the space required to store or transmit them.
- In both lossy and lossless compression, information redundancy is reduced, using methods such as coding, pattern recognition, and linear prediction to reduce the amount of information used to represent the uncompressed data.

Compression

- The acceptable trade-off between loss of audio quality and transmission or storage size depends upon the application.
- Lossless audio compression produces a representation of digital data that decompress to an exact digital duplicate of the original audio stream, unlike playback from lossy compression techniques.



Compression

- The figure is a block diagram representation of the operations involved in compressing a single channel.
- Generally, stereo or multiple channels are compressed separately.

Figure 1 The basic operations in most lossless compression algorithms.

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- The most commonly used algorithms use a 13-26 ms time frame durations, translating into 576-1152 samples at 44.1 kHz.

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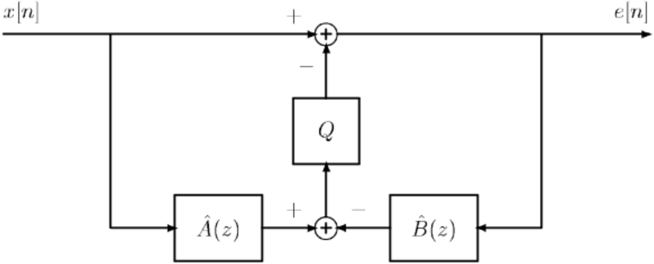
- 2 approaches:
    1. Modified linear predictive modelling of the signal.
    2. Obtaining a low bit-rate quantized or lossy representation of the signal, compute the difference between the signals and losslessly compress the computed signal.

Intra-channel decorrelation

Linear predictive model



$$e[n] = x[n] - Q\left\{\sum_{k=1}^{M} \hat{a}_k x[n-k] - \sum_{k=1}^{N} \hat{b}_k e[n-k]\right\},$$

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- Three widely used methos: Huffman coding, run length coding, Rice conding.
- We will discuss the less known of the three, Rice coding.

Entropy coding

Rice coding

- Rice coding is characterized by one parameter (m).
- In this type of coding, a number is divided into 3 parts: a sign bit, the m low-order bits, and the remaining higher-order bits.
- The bit sign is self-explanatory, the second part consists of the m least significant bits of the binary representation of the number's absolute value, and the third part is made of N consecutive zeros, where N has the same binary representation of the yet unused most significant bits; a 1 bit is inserted to terminate the sequence.
- The table below show the Rice code representation for parameter m=3.

Number	Sign bit	m Lower bits	Number of 0's	Full code
0	0	000	0	00001
18	0	010	2	0010001
-12	1	100	1	110001

Performance evaluation

Test suite

- All of the tracks are in CD format (44.1 kHz, stereo, 16 bits).
  - Track 4 of [17] (3 min 11 s): Madonna's "Like a Virgin" song.
  - Track 20 of [7] (39 s): Saxophone, low frequency musical signal.
  - Track 27 of [7] (20 s): Castanets, high frequency musical signal.
  - Track 48 of [7] (28 s): Voice quartet.
  - Track 66 of [7] (18 s): Wind ensemble.
  - Track L=R (18 s): this track was constructed using the left channel of Track 66 of [7]. The right channel is an exact copy of the left channel. Therefore, the difference between the left and right channel is zero. This track serves as a probe to allow us to determine if a given compression scheme takes advantage of inter-channel correlation.

Performance evaluation

Results

Track	File size (bytes)	H_0 Left channel (bits per sample)	H_0 Right channel (bits per sample)
Track 04	33,715,920	14.33	14.28
Track 20	6,879,600	10.86	11.03
Track 27	3,528,000	9.27	9.26
Track 48	4,939,200	11.59	11.48
Track 66	3,175,200	10.81	10.92
Track L=R	3,175,200	10.83	10.83

Table 3 File size and first-order entropy for left and right channels ($x[n]$).

Performance evaluation

Results

Track	Integer arithmetic AudioPaK	Fixed-point arithmetic MUSICompress	Fixed-point arithmetic Sonarc
Track 04	1.39	1.37	1.42
Track 20	3.12	2.93	3.13
Track 27	2.47	2.46	2.71
Track 48	2.56	2.41	2.65
Track 66	2.46	2.33	2.55
Track L—R	4.93	4.58	2.56

Table 4 Compression ratios for integer and fixed-point arithmetic lossless audio coders (we set the frame size to 1152 samples for the integer coders, and we used the default arguments for the fixed-point coders).

Performance evaluation

Results

Track	Floating-point arithmetic					
	Shorten -p0	Shorten -p10	OggSquish	LTAC	Sonarc -x	WA -c5
Track 04	1.38	1.43	1.43	1.41	1.42	1.46
Track 20	3.11	2.72	3.01	3.11	3.16	3.28
Track 27	2.46	2.69	2.67	2.70	2.72	2.83
Track 48	2.54	2.32	2.53	2.65	2.69	2.77
Track 66	2.46	2.42	2.51	2.54	2.57	2.64
Track L=R	2.47	2.44	5.01	2.70	2.58	5.28

Table 5 Compression ratios for floating-point arithmetic lossless audio coders (we set the frame size to 1152 samples for Shorten and we used the default arguments for OggSquish, Sonarc, LTAC, and WA).

Performance evaluation

The data

- The data in tables 4 and 5 is very revealing. Firstly, we note that the compression ratios obtained by various state-of-the-art algorithms are quite similar.

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- The data for Track R=L shows which of the algorithms attempts to take advantage of intra-channel dependencies in a way that works best when the two channels are tightly correlated (in this case they are the same).
- Finally, the tests show that WA with the -c5 consistently gives the best compression throughout the 6 tracks.

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- Furthermore, in spite of the wide range of approaches and complexities of the differenet compressors, they all perform similarly on the same audio signals, generally with less than one bit per sample in difference.
- It is noteworthy that HP's AudioPaK has similar performance to other modern compressors while keeping the complexity and number of arithmetic operations down. It is likely that upcoming compressor designs will focus on reducing algorithm complexity.