Lossless Audio Coding: Not Losing Any Bits Here

Matthew Roney
Bradley Department of Electrical and Computer Engineering, Virginia Tech

Lossless audio coding is the process of compressing speech or music into a smaller size without losing any information. This is the opposite of lossy coding techniques, like MP3, that achieve smaller sizes by throwing information away. In lossy compression, performance is measured by how closely the compressed audio signal approximates the original, uncompressed audio signal. Subjective performance measures include human perception of audio quality and objective performance measures usually rely on mean-squared-error. However in lossless compression, there is no performance degradation since the original, uncompressed audio signal can be exactly reconstructed from the compressed audio signal. Not surprisingly, lossless compression results in larger sizes than lossy compression. Performance measures for lossless audio coding include: compression ratio (i.e. size of the compressed audio signal), encoding time (i.e. time to generate the compressed signal), and decoding time (time to reconstruct the original signal).

Lossless audio compression plays an important role in many real-world applications such as archiving original music and speech recordings. Like the JPEG standard for image compression, lossless audio compression is part of the international MPEG-4 standard entitled audio lossless system (ALS). ALS is still an active research area—the algorithms that will comprise the final, published standard are still being determined. State-of-the-art lossless audio compression techniques typically achieve compression ratios on average of 2:1: in other words, the compressed signal is ½ the size of the original signal. How can we discard 50% of an original signal and not lose any information? Audio signals are redundant—they contain multiple copies of the same information; the idea behind lossless audio compression is to identify and remove this redundancy.

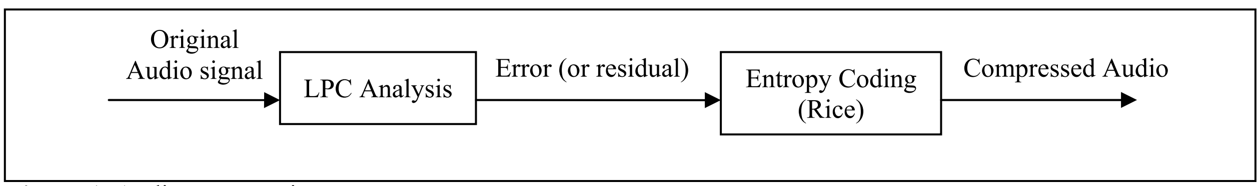

Figure 1. Audio compression process

Matlab code was written that utilized the most advanced techniques for lossless audio compression and follows the block diagram implementation in Figure 1. The two stages of compression, modeling and entropy coding, were implemented using linear prediction (LPC) and Rice coding, respectively. Various commercial lossless compression programs were compared in terms of their compression performance. The results indicate that the various approaches to lossless audio compression all deliver roughly the same performance in terms of the compression ratio. The primary difference between them lies in the speed required to perform the compression. Our ongoing work explores hardware architecture design for fast hardware implementations of a state-of-the-art lossless audio codec.