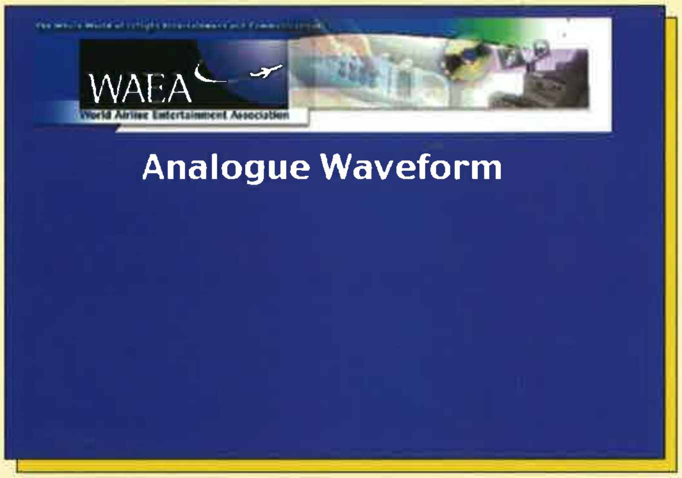
Audio Compression

The Potential Impact of Low Bitrate Encoding

By John Salzman

As more content is delivered on aircraft via file servers, audio compression becomes a necessity to reduce the file sizes loaded. This article, highlighted by a number of diagrams that should prove helpful, is intended to convey a perspective on the impact that lower bit rates and lower sample rates have on the music we listen to onboard.

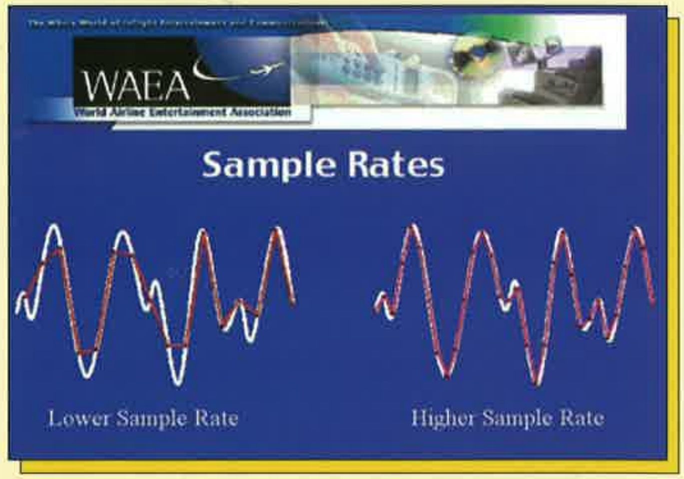
Instruments, whether they are strings, woodwinds, brass, or percussion, produce analogue waveforms.



This is true of our vocal cords as well. How we handle or how we process these analogue waveforms in the digital domain will have an impact on the quality of the music.

When a musical group enters a studio for recording, they will probably be mastered using one of several software applications such as ProTools, Sadie, or Sonic Solutions. Their original analogue waveforms are broken down into individual digital samples.

The sample rate tells us how often a series of these successive binary approximations are used to represent the original waveform. Each sample has 24 bit resolution, or three bytes. Most often in mastering, the sample rate is 48 thousand per second. The higher the sample rate, the closer the digital waveform will represent the analogue waveform.



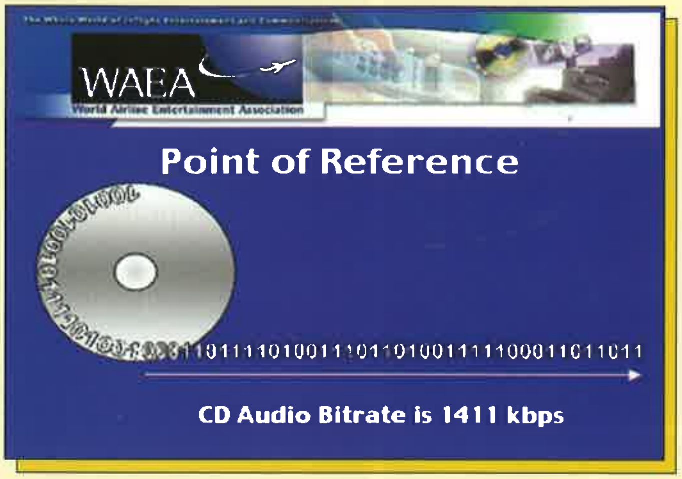
So often we read ads about products that claim to reproduce CD quality audio. The inference is as though CD quality is the best there is, and while CD audio is very good, we are able to discern a difference from the original digital master. The Red Book standard for CD audio specifies 16-bit resolution at 44,100 samples per second.

From the original 24/48 master, the audio stream must now undergo sample rate conversion and resolution truncation.

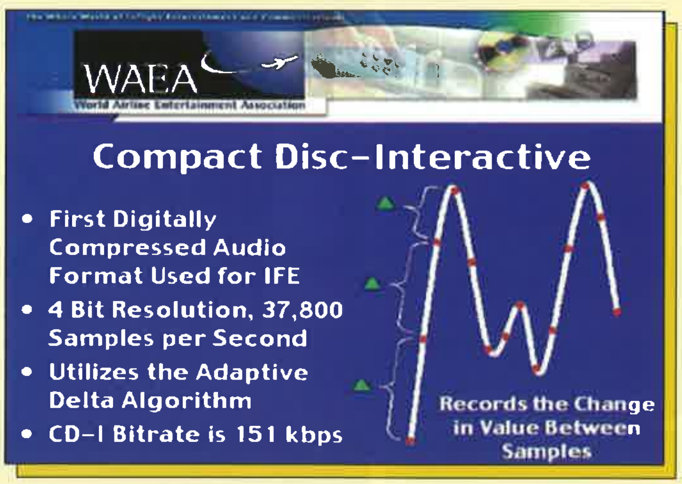
From 24 bits to 16, the last eight bits of resolution are cut off. In CD authoring, there is a process called dither which assigns random values to those eight bits before they are cutoff. Dithering will reduce the digital noise level and help to compensate for the inevitable quantization error before truncation. Bit depth determines dynamic range. The greater the dynamic range, the less digital noise in the recording. Yes, I did say digital noise. We think of digital audio having no noise, but each

Editor's Note: Previously, AVION featured an article, authored by Rory Briski, that explained an aircraft multiplex system ("Mux -What it is"). That article has been reproduced and circulated, highlighted on the WAEA's internet library, and quoted in many presentations. It was a superb example of a complicated topic reduced to simplified terms by an expert. The following article is similar. It explains a complicated and technical topic in easy to understand terms. No doubt it will be used as a primer for years to come.

resolution bit represents 6 db of noise reduction. I mention these things because Compact Disc is the most widely used source for airline audio programming. Just for a point of reference, the bitrate for CD audio is just over 1400 kilo-bits per second (kbps).



The first compressed audio format used for IFE was the airline CDI format. It is a subset of the Green Book Standard, Level B. This format has 4-bit resolution at 37,800 samples per second. It utilizes the Adaptive Delta encoding method. This algorithm records the difference between two successive samples.

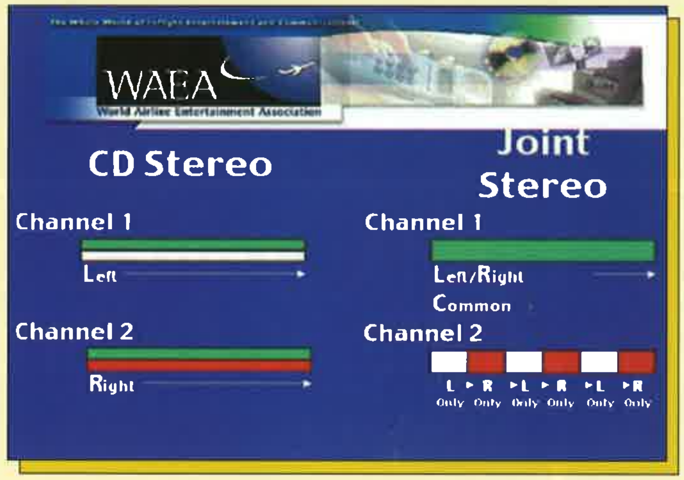


For the purpose of comparison, the bitrate for airline CDI is 151 kbps.

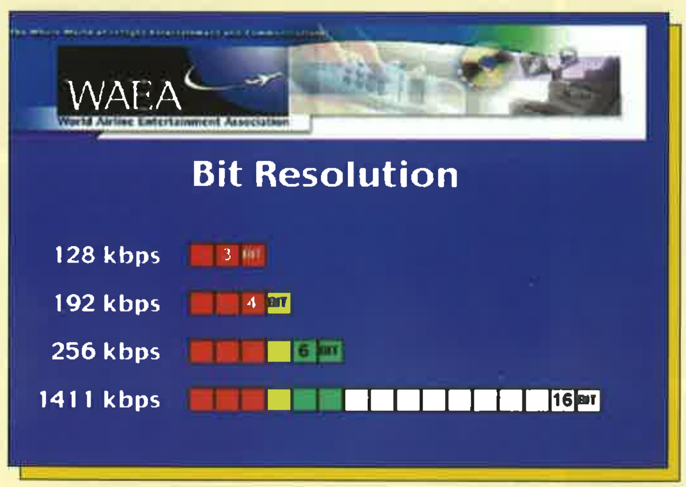
Since then, several other encoders have been designed for audio compression. The most commonly used in our industry at this time is MPEG 1, Layer 2. Layer 3, more commonly referred to as MP3, is already a ten-year old technology. Both are perceptual-encoding algorithms, designed to exploit the properties of human hearing. The concept, in basic form, is to eliminate redundant frequencies and lower level audio that would not normally be heard. By not digitizing audio that cannot be heard anyway, we reduce the amount of data required to encode the music passage.

A couple of the by-products of using lossy compression algorithms, and this will vary from song to song, are the too rapid a decay of a diminishing tone, and the loss of the harmonic ambience that adds such a fullness to the sound. Overly compressed audio can have an almost clinical, lifeless sound. It's just too sanitary.

One effective method of reducing the amount of data for a stereo channel is compressing in joint stereo.



In a normal stereo recording, much of the audio is common between the left and right tracks. In joint stereo mode, the common audio data is stored on one track, and the difference audio, that is, left only, and right only, is stored on the other track. This process occurs within the compression algorithm. This helps a great deal in resolving low bitrate audio signals.



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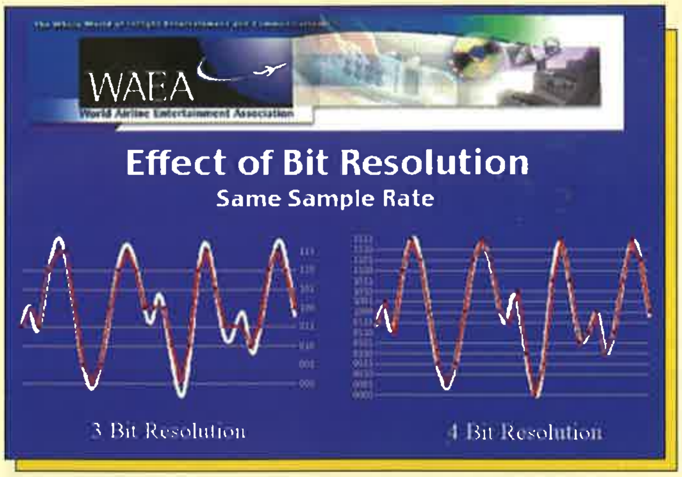
Using MPEG 1, Layer 2 or MP3, the frequently used bitrates are 128 kbps, 192 kbps, and 256 kbps. At sample rates of 44,100 per second, the respective bit resolutions are 3 bits for 128k, 4 bits for 192k, and 6 bits for 256k. Now think back for just a moment. CD audio is 16-bit resolution at a bitrate of 1411 kbps.

Lower bitrates will have higher quantization error.



With three-bit resolution, each sample has only eight chances to accurately represent the analogue waveform. Each bit can be a '1' or a '0'. With three bits, there are eight possible combinations. In binary arithmetic, 23 = 8.

With four-bit resolution, by adding just one bit, we double the chances for each sample to more accurately represent the waveform, 24 = 16.

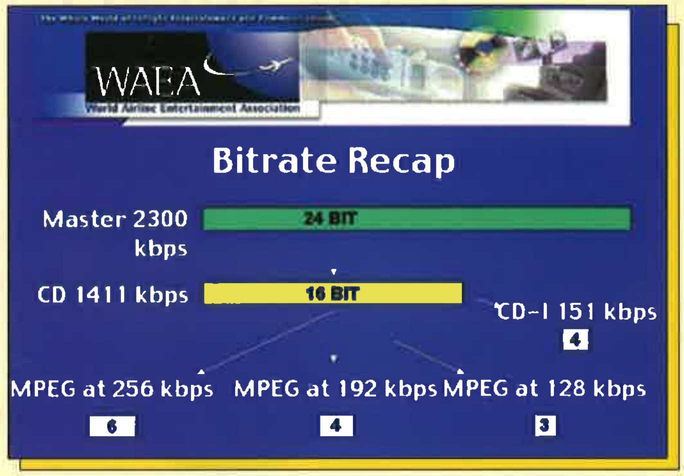
The pattern continues in binary function. Five-bit resolution equals 32, and six-bit resolution equals 64. Think back once again to the original 24-bit digital master, 224 = 16,777,216. This bitrate will very accurately represent the original waveform, and that is why it sounds so good!

The reason three and four bit resolutions work at all is attributed to the perceptual encoding and decoding process of the algorithms.

MP3 Pro is the next generation of the MPEG Layer 3 codecs. Codec is an acronym for encoder/decoder. It, too, is a perceptual encoding algorithm, claiming to offer MP3, 128 kbps quality at a bitrate of 64 kbps. MP3 and MP3 Pro codecs are backward compatible. Thus, an MP3 Pro decoder will process Layer 1, Layer 2, and Layer 3 bitstreams.

About a year ago, Dolby announced the availability of licenses for a new compression codec, AAC, or Advanced Audio Coding. It was developed within the MPEG standard, however, it is not backward compatible with the previous Layers 1, 2, or 3. Although I have not yet listened to an AAC bitstream, the reports I have read claim the audio quality is better than MP3 at approximately 30% lower data rate.

Just to recap,



we start with a digital audio master with 24-bit resolution at 48,000 samples per second. Then to record compact discs, that bitstream is sample rate converted to 44,100 samples per second, dithered, and truncated to 16-bit resolution.

For the CDI format, we apply the Adaptive Delta algorithm at Level B, yielding a bitrate of 151 kbps.

For file server formats, we compress and perceptually encode using MPEG algorithms to reduce file sizes. Our final resolutions, based on bitrate are:

3 bits 128 kbps;
4 bits for 192 kbps; and
6 bits for 256 kbps

Audio quality improves as the bitrate increases. If we require lower bitrates, we see the importance of using the best algorithms available.

It is not the intent of the Audio Technologies Working Group, at least not at this time, to standardize or specify a certain algorithm or bitrate. We do encourage, however, the use of the highest bitrate possible dependent upon the storage capacity available, the amount of content offered, and the download characteristics, processor speeds, and the size of the pipeline of the systems on board.

John Salzman began his Inflight career at Trans-Com in 1976. In 1984 John went to work at AEI Music, currently DMX Inflight, as Manager of Audio Production. He later became Director of Technical Services. John joined the WAEA's Technical Committee in the mid '80s and continues to be an active and valued participant.