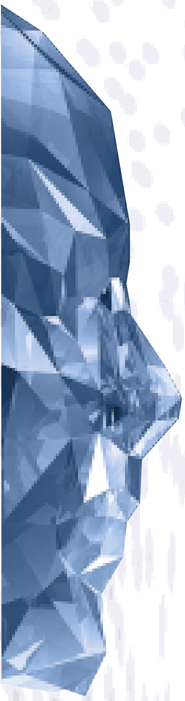


# update on audio CODING

New audio coding systems (such as AAC) can halve the bit-rate

Quality assessment is not a simple measurement

Discrete multi-channel sound is possible at 320 kbit/sec

Don't squeeze the bit-rate

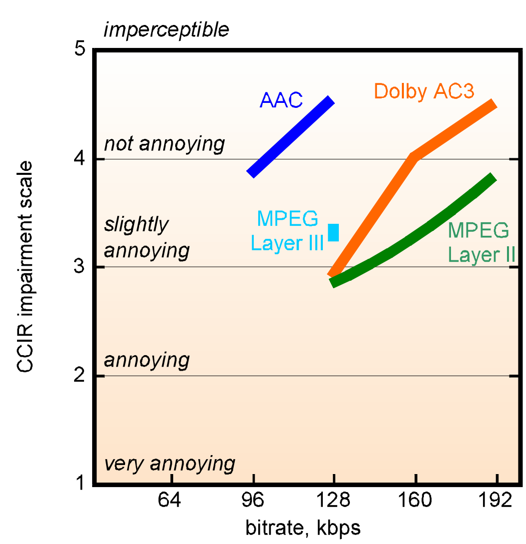
Low bit-rate coding schemes make transmission of digital audio possible and storage of audio cheaper. Newer systems are arriving which can reduce the bit-rate even further. These systems will expand the possiblitities for audio content publishing and production into areas such as the internet and true multi-channel surround sound. The implications of using such low bit-rate systems need to be borne in mind in all parts of the broadcast chain.

Modern audio coding systems in use today are lossy – they attempt to remove redundancy and irrelevancy. Models of the human hearing system are used to predict whether we will hear impairments introduced by the coding system. For example, the model may determine that only the louder of two tones may be audible – so only the louder tone is coded for transmission.

## Newer Systems

Newer systems require a lower bit-rate for the same quality, but at the cost of increased complexity and increased coding/decoding delay. The improvements made by MPEG AAC (Advanced Audio Coding) can be seen from the graph of mean results from one recent stereo listening test. The graph shows that AAC at 96 kbits/sec gives a similar quality to MPEG Layer II at 192 kbits/sec.

Lower bit-rates also mean that 5-channel surround sound becomes feasible. Listening tests have shown that AAC can provide excellent quality at 320 kbits/sec and quite good quality at 256 kbits/sec.




**Results of one recent stereo listening test of the best available software based codecs.**

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## Quality Assessment

Measures such as signal-to-noise, frequency response or distortion cannot fully describe a bit-rate reduction system. Listening is currently the only way to assess such systems, but unfortunately some material will sound fine to some people and terrible to others. Such systems should not be assessed by individuals – subjective tests with critical material and many listeners are needed to give a reliable indication of performance. There are newer objective perceptual measurement methods which use psychoacoustic models and subjective test data, but they are still not as reliable as subjective tests.

## The Whole System

Each part of the broadcast chain has an impact on the quality of the delivered programme. When bit-rate reduced systems are used one after the other (cascaded), the quality deteriorates. If several parts of the chain are only "just good enough" in isolation, the final result may not be "good enough" to transmit.

Cascading of coding systems should be avoided wherever possible. When cascading is unavoidable, some coding "headroom" should be built in to each part of the system, e.g. by using a bit-rate that is more than "just good enough". For example, MPEG Layer II stereo coding at 384 kbits/sec used in production provides very good quality and can be cascaded many times without significant loss in quality.

## Glossary

**MPEG** Moving Pictures Experts Group, the name of a family of standards.

**MPEG-1** Consists of three layers of complexity and delay, where Layer I is the simplest and has the lowest delay.

- Layer I: High bit-rates (~400 kbits/sec), used for Digital Compact Cassette (DCC).
- Layer II: Medium bit-rates (~256 kbits/sec), used for Digital Radio and TV, ISDN and desktop editors.
- Layer III: Low bit-rates (~128 kbits/sec), used for ISDN and Internet music delivery.

**MPEG-2** Adds low sampling frequency and discrete multi-channel sound to MPEG-1.

**MPEG-4** Uses a tool-based approach to create audio at very low bit rates (2 kbits/sec – 64 kbits/sec). The tools include: text-to-speech, music synthesis, speech codecs, surround sound, audio coding and the ability to mix and change the received audio.

- MPEG AAC: Advanced Audio Coding is the audio coding tool. For a stereo signal at 96 kbits/sec it provides a similar quality to MPEG Layer II at 192 kbits/sec. It can provide excellent sound quality for 5-channel surround sound at 320 kbits/sec.

**Dolby AC-3** Capable of stereo and multi-channel sound, it is similar to Dolby Digital used in the cinema. Adopted for the USA digital TV system (ATSC), also used in DVD and optional for DVB multi-channel.

**Dolby-E** A low-delay "mezzanine" compression format designed to carry multi-channel signals around the studio in a single AES/EBU stream, and able to survive a number of cascades.

**Sony Minidisc** Portable recording medium using a proprietary coding scheme known as ATRAC (~300 kbits/sec).

**RealAudio & NetShow** Internet based real-time streaming systems. They use a variety of codecs which can be chosen for the material or bit-rate being used (including speech codecs).

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