ABSTRACT

The compression of any data (audio, video, image) must be done in such a way that the decoding process can reproduce the original data exactly. However, where human perception is involved, a lot of data can be removed from the original signal whilst retaining the accuracy required to be acceptable - this can be seen in image compression algorithms such as JPEG (Joint Photographic Experts Group) dan some of audio compression algorithm, like MPEG (Moving Picture Experts Group).

Digital audio coding, which - in this context - is synonymously called digital audio compression as well, is the art of minimizing storage space (or channel bandwidth) requirements for audio data. Uncompress data needs a large of storage space and channel bandwidth.

ATRAC is an audio coding system based on psychoacoustic principles. The input signal is divided into three subbands which are then transformed into the frequency domain using a variable block length. Transform coefficients are grouped into nonuniform bands to reflect the human auditory system, and then quantized on the basis of dynamic sensitivity and masking characteristics.

Block diagram of ATRAC will be implemented in MATLAB v6.1 using .m file. So, the sound quality of a compressed audio will be known objectively as well as subjectively.