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MPEG 1 – Layer III Audio Codec

System Block Diagram

From the high-level system diagram presented in the previous assignment, we dissected our project into the twelve subsystems shown in figure 1. Each subsystem generally has one input and one output, making this system relatively linear. Since the project is comprised of a simulation phase and a hardware build phase, relevance to each phase will be included with the description of the blocks.

Input & Encoding Subsystems (Blocks 1-8)

The input (block 1) consists of a digital audio signal read into MATLAB as a PCM-coded WAV file. This file can be any kind of audio signal sampled at 44.1Khz with either 16-bit or 8-bit resolution. (The 44.1Khz WAV format was chosen because it can be easily manipulated with common audio software, such as Cool Edit.)

Blocks 2 through 8 decompose the digital data from the WAV file (block 1) into specific frequency subbands and quantize each of these according to the psychoacoustic model. The digital audio output is fed simultaneously into two paths, both of which will compress the data. The first of these paths is the bandpass filter bank (block 2). These digital bandpass filters will dissect the signal into 32 subbands, each with a bandwidth of 750Hz. An 18 channel modified discrete cosine transform (MDCT) filter bank (block 3) is applied to break each of the 32 subbands into 18 separate frequency bands, for a total of 576 bands. Next, each subband is downsampled (Block 4) to 1.5KHz by retaining every 32nd data point and discarding all other data points (48Khz / 32 = 1.5KHz).

The second data compression path includes two subsystems – a 1024 point Fast Fourier Transform (FFT) and the psychoacoustic model (blocks 6 and 7, respectively). The FFT supplies the psychoacoustic model with the frequency components of the input signal. The model uses this information to identify components of the input signal that are imperceptible to the human ear. The quantizer (block 7) uses the spurious signal components detected by the psychoacoustic model to determine how each of the 576 subband signals should be encoded. Subbands containing spurious data below a certain threshold will be discarded or encoded with less accuracy than subbands with more perceptually important signals. The bitstream produced by the quantizer is formatted into standard MP3 frames in block 9.

Decoder & Output Subsystems (Blocks 9-12)

Blocks 9 through 12 make up the decoder subsystem and will be used for both simulation and hardware phases of the project. The MPEG data frames are fed into the bitstream extraction logic (block 9), which recovers the audio bitstream from the data frames. The inverse modified discrete cosine transform (IMDCT, block 10) receives this data and reconstructs the signal into the 32 subbands. All of the subbands are summed (block 11) and delivered as PCM digital audio (block 12).

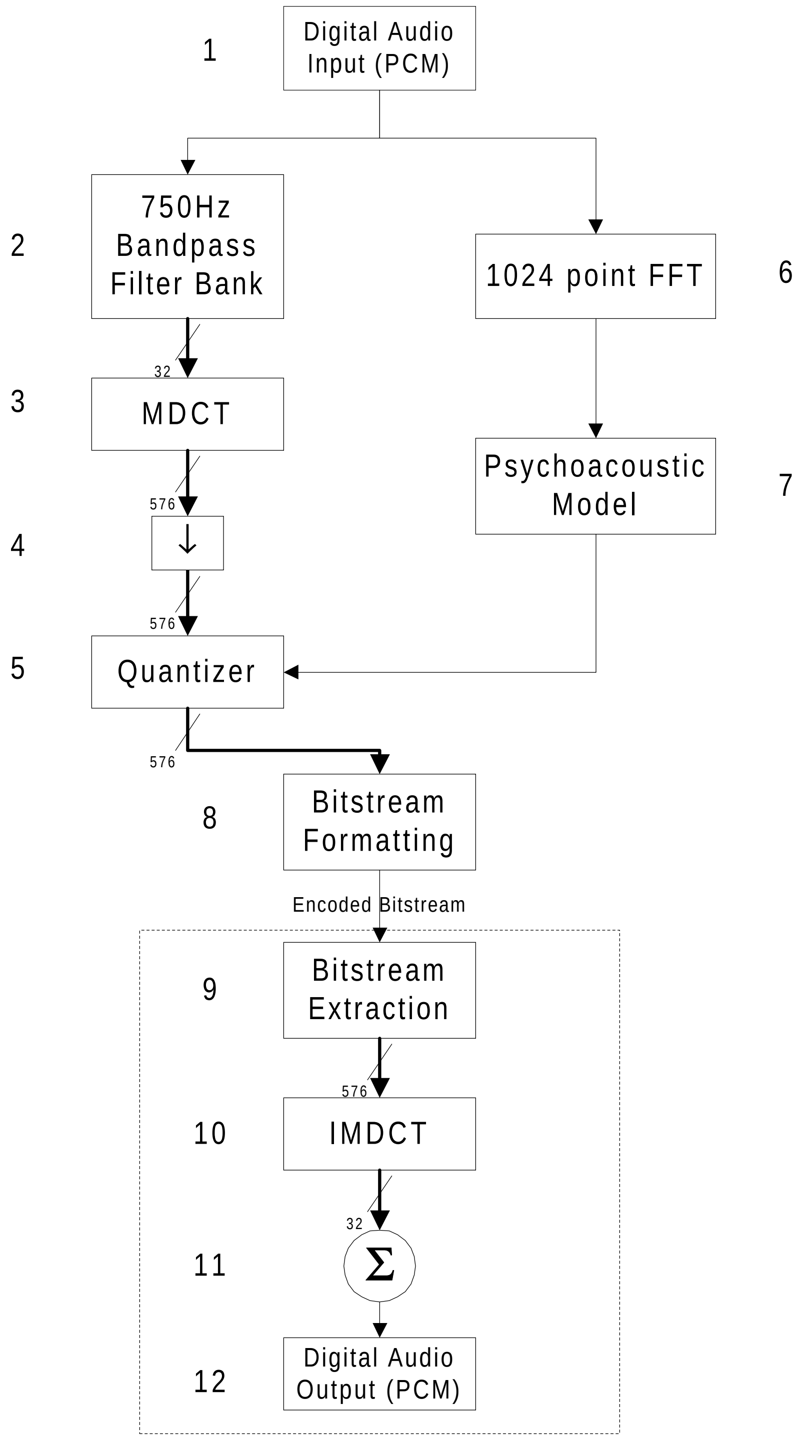

Figure 1 – System Block Diagram
(dashed line indicates the hardware phase of the project)