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Functional Description

Our senior project involves two major phases. The goal of the first phase is to simulate the standard MPEG 1 – Layer III (MP3) audio encoding and decoding algorithms using MATLAB and a DSP simulator. The goal of the second phase is to design a DSP-based portable MP3 decoder suitable for production. Figure 1 below depicts the functionality of both these goals.

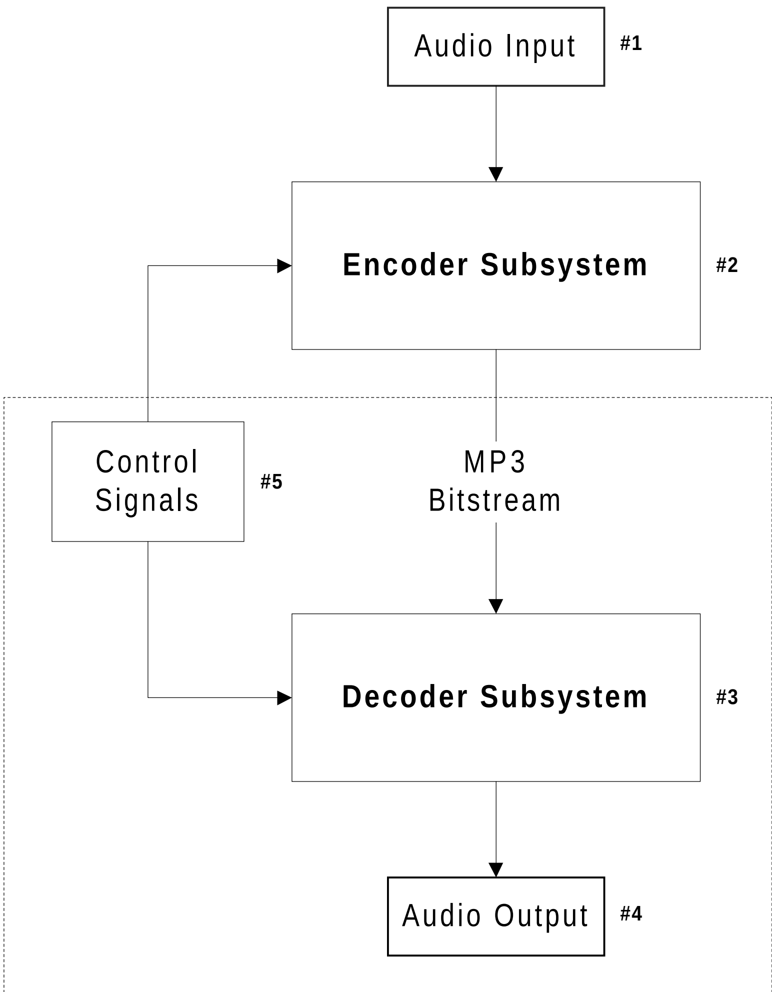

Figure 1 – High-Level Functional Block Diagram
(Dashed box indicates phase 2 of the project)

#1) The system will accept simulated analog audio inputs ranging from roughly 20Hz to 20kHz.

#2) This analog signal will be converted to a digital bitstream according to the standard MPEG-1 layer III encoding algorithm. This algorithm achieves a data compression ratio of approximately 10:1 by using a psychoacoustic model that exploits the limitations of the human auditory system. In other words, the model codes only the components of the input signal essential to reproduce a perceptually equivalent output signal and ignores any other signal components that cannot be heard.

#3) The decoder subsystem then reassembles the analog audio signal by deciphering the bitstream created by the encoder subsystem. The standalone decoder box will receive MP3 encoded data from a host PC and store them in non-volatile RAM. The algorithm used to reassemble the analog signals is also part of the standard MPEG-1 layer III specification.

#4) A speaker will convert the analog signal into audible sounds. The sounds generated by the decoder should be perceptually equivalent to the audio input (**#1**).

#5) The operation of the encoder and decoder subsystems is governed by a set of undetermined control signals. These are combination of both external hardware controls and software functions.