Project Title: Implementation of Signal Processing for Audio Equalization
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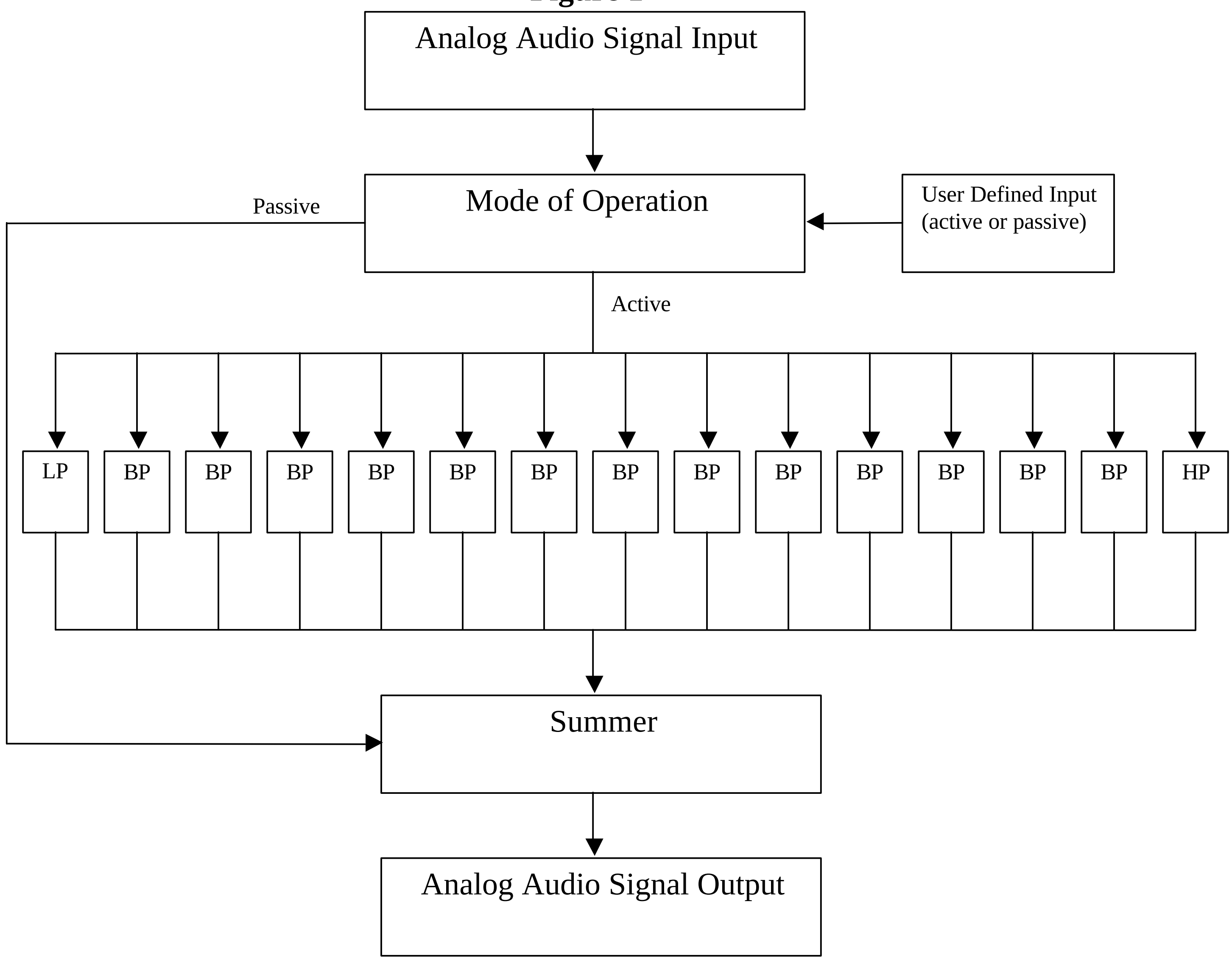
Memo: Complete Block Diagram

Three methods will be utilized in accomplishing audio equalization. The first method's block diagram is shown below in Figure one and consists of a user defined mode of operation and a digital filter bank containing 15 filters. The mode of operation will allow the user to select active mode, where the user can create custom equalization using the variable gain digital filters to amplify or attenuate specified frequency bands, or passive mode where the input signal is sent directly to the output by bypassing all filtering circuitry.

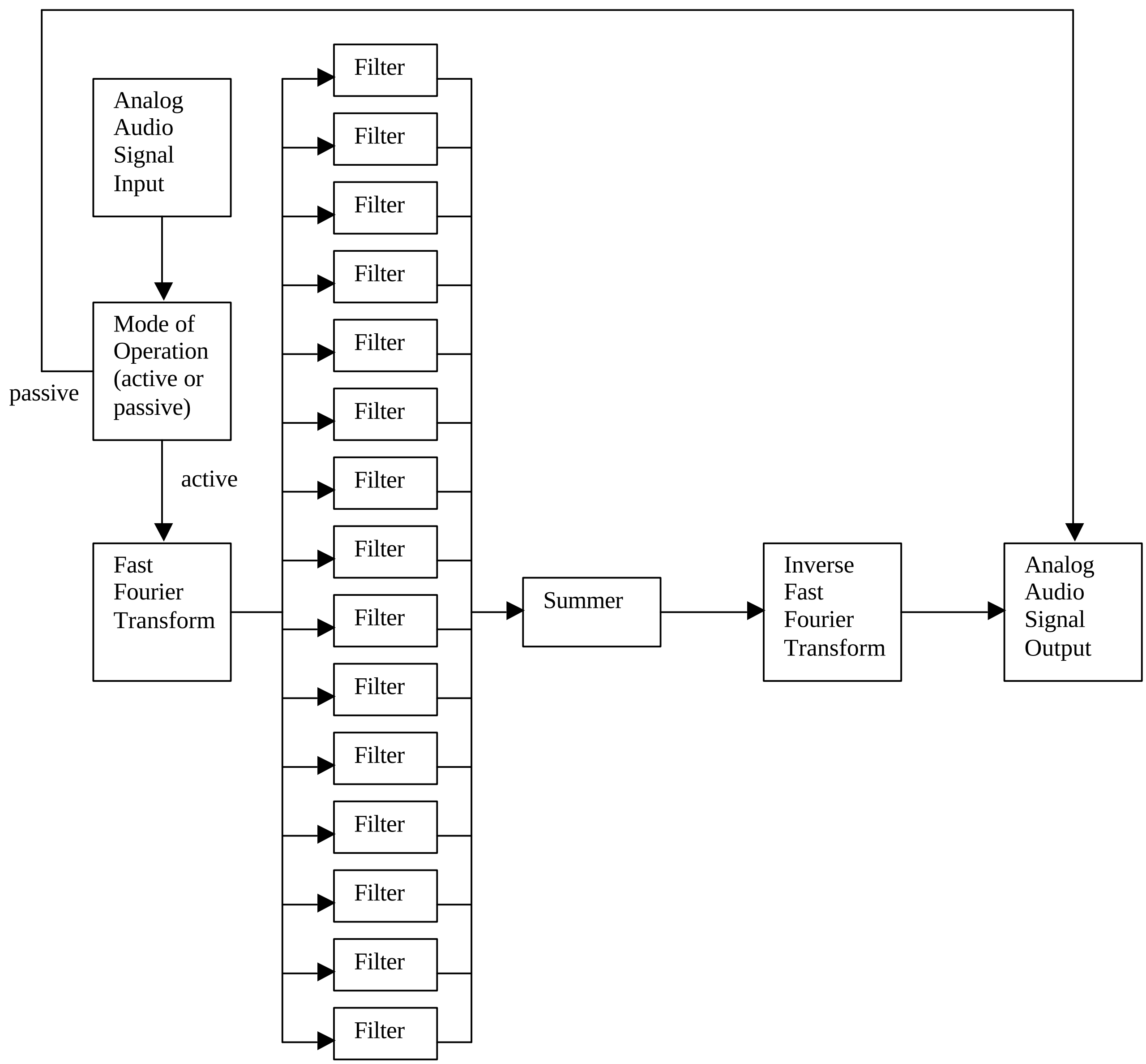
LP = Low Pass Filter
BP = Band Pass Filter
HP = High Pass Filter

Figure 1



The second method to be utilized is very similar functionally to the first method, although the equalization will now take place in the frequency domain. A fast fourier transform of the input audio signal will transform it from the time domain to the frequency domain. Amplification and attenuation will then take place by multiplying the amplitude of specific frequencies to attain the desired sound. The block diagram is shown below in Figure 2. Active and Passive modes will be present.

Figure 2



The final method will utilize the JPEG 2000 data compression technique. The block diagram for this method is shown below in Figure 3. The analog audio input will pass through a high and low pass filter and the frequencies at the output of these filters will be down-counted by a factor of two. This output will then be passed through high and low pass filters identical to the first pair and down-counted. The process continues until the desired amount of filters is reached. For this project 16 filters with variable gain will be used. Recombination of the signal is done using the same method in reverse.

Figure 3