**Stereo and Mono Sound Equalizer**

This Software code involves the creation of a 7-channel equalizer primarily by using the Butterworth filter approximation method. The mathematics modeling program MATLAB was used to model the equalizer. A stereo sound file was input into the program, where 1 low-pass filter, 1 high-pass filter, and 5 band-pass filters divided the stereo sound signal into 7 different frequency channels. Each filtered channel was then adjustable including a dB gain variable to change the gain for each respective channel, and then the results were recombined into two stereo channels. The stereo channels were then saved to a new stereo sound file. The results either will sound will work as on a mono filter or as a stereo filter.

The 7-channel equalizer filter for this MATLAB code can be modeled using basic principles of low-pass, band-pass, and high-pass filters. The Butterworth approximation method is used to filter these 7 individuals’ channels and was accomplished using the mathematics modeling program MATLAB.

The first channel is obtained by a 3rd order LPF with a cutoff frequency of 64Hz giving the first channel range from 0Hz to 64Hz. The second channel is obtained from the 1st bandpass filter with the order of 2, the range of this bandpass filter is from 64Hz to 500Hz. The third channel is obtained from the 2nd bandpass filter with the order of 5, the range of this bandpass filter is from 500Hz to 1000Hz. The fourth channel is obtained from the 3rd bandpass filter with the order of 6. The range of this band is from 1kHz to 4kHz. The fifth channel is obtained from the 4th bandpass filter with the order of 14, the range of this bandpass filter is from 4kHz to 8kHz. The sixth channel is obtained from the 5th bandpass filter with the order of 25, the range of this bandpass filter is from 8kHz to 16kHz. The 7th channel (last one) is obtained from the high pass filter with the order of 20 and a cutoff frequency of 16kHz. The gain for each channel was then calculated by applying the fast Fourier transform (FFT) on all 7 filtered channels individually and then converting the FFTs’ amplitudes to dB gain values.

Then the left and right channel of the stereo sound is separated in the code. This allows separate control of the amplitude of the left channels and the right channels signals for each filtered channel. for example, the code splits the left and right channels of the 32 Hz to 64 Hz filtered channel. This process was repeated for every other filtered channel. The gain control was done by setting a variable equal to the desired gain adjustment, where the default is 0 dB for no gain change. Then, the dB value is converted back into the original amplitude unit for the stereo sounds signal so it can be scaled correctly.

After the gain control of the left and right signals separately for each channel of the seven filtered channels, the signal was restored by adding all 7 of the filtered channels together after adjusting the gain to the desired level. The variable calculated by converting the dB value back into the unit used by the stereo sound signal was used to scale each filtered channel by multiplying each respective one together.

Finally, the output of the code can be chosen to be a stereo type or monotype. By default, the code was set to have stereo output, yet it can output a mono sound by simply uncommenting the last two lines in the code. The conversion from stereo sound to mono sound was done by calculating the mean of the left and right channel of the stereo signal, then output this means to be played for both the left and right channels. Figure 15 shows the plot of the 7-channel equalizer designed using MATLAB when the gain of all the channels was set to 0 dB.

Chart, line chart

Description automatically generated

Figure 15 the plot of the 7-channel equalizer at 0 dB

These filters were designed using basic principles of low-pass, band-pass, and high-pass filters. The Butterworth approximation method was used to find a transfer function that would appropriately filter 7 individual channels of the input stereo sound signal, and MATLAB was used to execute this function. The dB gain was adjusted after separating each channel into separate stereo components, and then the channels were recombined per left and right stereo components and the new filtered stereo sound signal was written. Finally, the output was determined to be either stereo sound or mono sound.