ISSN 2229-5518

Auto-Equalizer For Music Applications

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Abstract- When a song is recorded, it undergoes a lot of processing before it is produced. One of the most important and difficult parts of the mixing process is equalizing the tracks. Equalization is nothing but changing the gains at the various frequency components present in the song. An auto-equalizer is an easy-to-use tool which will help any lay man to apply basic equalization to his music track and produce a good quality song.

Keywords— Attenuate, Boost, Filter Bank, Fundamental Frequency, Gain, Pitch, Timbre, Vocal Range.

1 Introduction

In music industry today, while recording a song, every instrument is recorded on a separate track. The vocals are later recorded on another track. One of the major tasks during the recording process is mixing of these tracks together. Certain effects such as compression, equalization, delay, reverb etc need to be applied to the vocal track for a perfect mix. This job is done generally by a sound engineer. One important effect that needs to be applied is equalization. The sound engineer listens to the background music, sometimes analyzes the frequency spectrum of the music as well as the vocals and then decides which frequencies to boost or cut from the voice signal. This project aims at automatically equalizing the voice signal by taking into consideration a few factors such as frequency spectrum of the signal, frequency response of the microphone, and range of the vocals. This tool is especially helpful to people who are not professionals and do not understand the technicalities of mixing. By using this tool they will be able to produce a good quality song.

The equalization will be carried out in two main steps as follows:

1. For the 1st step, the user will have to tell the computer which microphone was used and the range in which the vocals lie. There will be a list of a few commonly used microphones. There will also be an option 'other', which when selected the frequency response of the microphone will be assumed as flat. There will also be an option to select the lowest frequency note present in the vocals and an option to select the highest frequency note. Depending on the frequency response of these microphones and the range of the vocals, the tracks will be equalized
2. The second and the more important step will be analyzing the frequency spectrum of the instruments tracks and compare it with the frequency spectrum of the vocals track. Depending on these two spectrums the instruments track will undergo some cuts at a few frequencies in order to make space for the vocals. Also at places the vocals track may undergo a few boosts. The user however will need to tell the computer which track is the instruments track and which track is the vocals track.

2 Important Concepts

2.1 Equalization

It is necessary to have a basic idea about equalization before we can talk about auto-equalizer. Equalization, as the name suggests means trying to equalize or in other words it is the process of trying to maintain the balance within the frequency components of any electronic signal.

Fourier proposed that any signal can be represented by a sum of many sinusoidal signals of different frequencies, amplitudes and phases[1]. In audio signals, the amplitude of each frequency component in the frequency response plays a major role on the quality of the sound. Even the pitch of the sound is determined by the frequency. Pitch is nothing but that frequency component which is the most dominant. The timbre of the sound signal is determined by the amplitudes and the number of harmonics of the fundamental frequencies present in the frequency response of the signal.[2]

When we talk about equalization in music, we talk about slightly boosting or cutting the gains of some of the frequency components present in the music. This process helps enhance the quality of the music. The changes in the gains are very minute and subtle usually, because a big change might affect the quality of the sound completely, unless that is required. Cutting frequencies is the more subtle way of equalizing. It is preferable to cut the frequencies rather than boosting. Boosting can lead to clipping of the sound signal and other undesirable effects like harmonic distortion. Hence our auto-equalizer will also focus more on attenuating rather than boosting.

2.2 Filter Bank

Equalization involves changing the gain at various frequencies. This job is done by using filters. We will need to

ISSN 2229-5518

boost the gains at certain frequencies and cut the gain at certain frequencies, so for this purpose we can make use of filter banks. Filter Bank is a series of band pass filters. It contains a number of triangular filters with adjustable cutoff frequencies, adjustable number of filters and adjustable length of filters.[3] The gains of each filter can be changed and then the frequency spectrum of the track to be equalized is multiplied by the filter bank, thus the equalization of the entire signal is done with a single multiplication. Figure 1 shows an example of a filter bank.

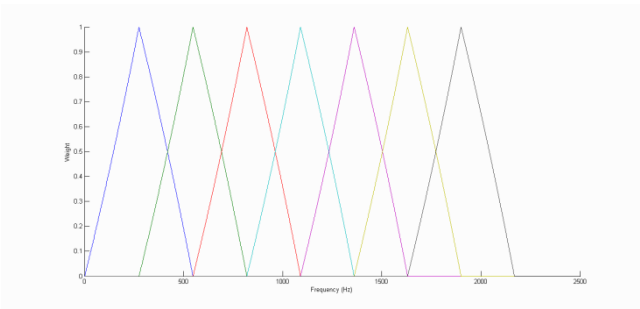


Figure 1:Filter Bank

3 Equalization Steps

3.1 Equalization Based On Microphones

Microphone is a transducer which converts sound pressure variations into changes in electrical voltage. When we speak of a sound signal, we are referring to the sum of various frequencies. While recording a song with the help of a microphone, the different amplitudes at different frequencies of our voice signal gets either boosted or attenuated because of the frequency response of the microphone, changing the sound in general. The graph of the frequency response indicates the level change suffered by that one sound of each frequency. The ideal graph would be a flat response, which would mean that it would not affect the original sound, and you would get an exact sound into the PC as it sounds in real. A drop or rise of up to 3db is considered as acceptable. The frequencies where the gain is more than 3 dB below the ideal response need to be boosted whereas the frequencies where the gain is more than 3 dB above the ideal response need to be attenuated in order to get a flat frequency response of the microphone.

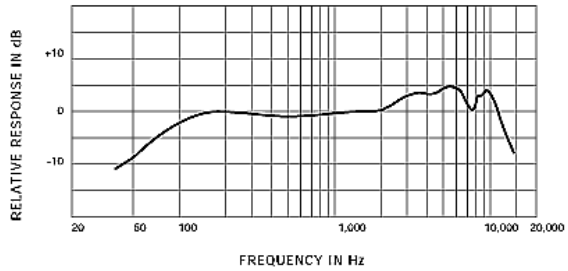


Figure 2.a: Shure SM58.[4]

Figure 2.a shows the frequency response of the microphone Shure SM58. In this amplitude versus frequency plot, the gain at 60 Hz is -10dB, so the auto-equalizer will boost the gain at these frequencies. The gain between 2000 to 8000 Hz will be attenuated in order to bring the response closer to ideal.

We will be providing the presets of a few commonly used microphones. The user will have an option to select the preset of the microphone he is using. Depending on the selection the gains at the required frequencies will be boosted or attenuated in an attempt to achieve a flat frequency response. It is important to note that the boosting will always be more subtle than the cutting. Following are the frequency responses of the microphones that are commonly used in the industry.

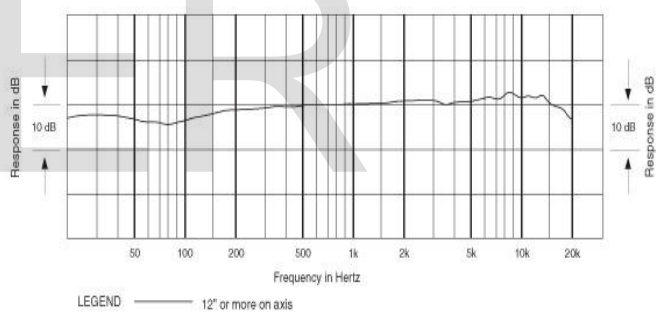


Figure 2.b: AT2020 Frequency Response.[5]

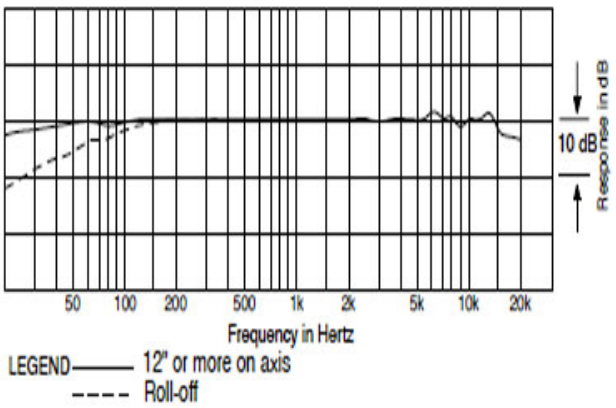


Figure 2.c: AT2035 Frequency Response.[6]

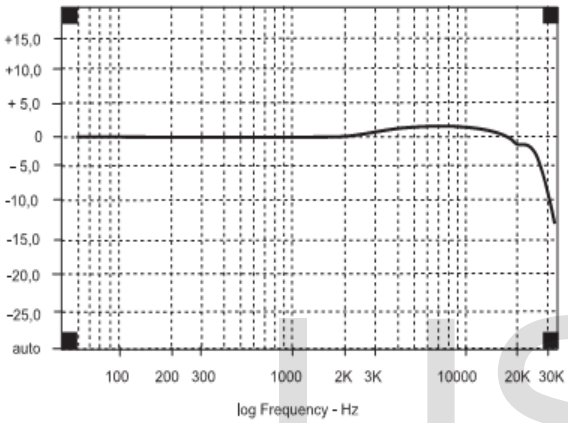


Figure 2.d: Behringer ECM8000 Frequency Response.[7]

Figure 2.e shows the comparative frequency response of Shure SM58 and Behringer ECM8000. The lower graph is that of Shure SM 58 whereas the graph on the top is that of Behringer. From the comaparison we observe that the gain of Shure SM58 at lower frequencies needs to be boosted for a better performance. And the gain of Shure SM58 at higher frequencies needs to be attenuated. On the other hand the gain of Behringer ECM8000 till 3KHz is quite flat, it does not require any boosting or attenuating. But the gain at much higher frequencies needs to be boosted slightly.

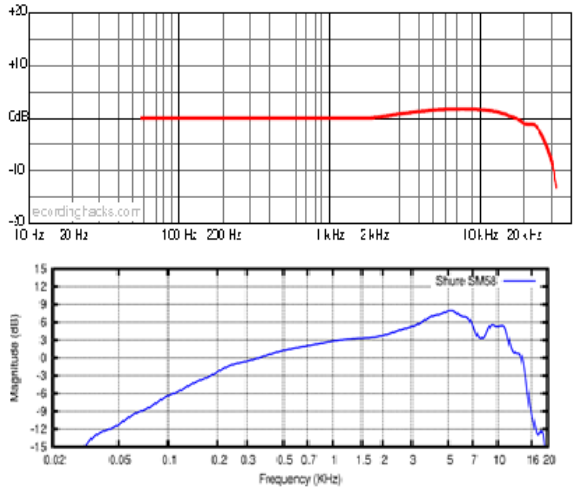


Figure 2.e: Comparison of Shure SM58 and Behringer ECM8000.

Frequencies above 15Khz will not be boosted as usually no useful data lies at such high frequencies. What lies however in these frequencies is the noise which will be boosted unnecessarily.

3.2 Equalization Based On Vocal Range

Vocal Range is the region in the frequency domain where a person's vocals lie. This parameter changes from person to person. Some people are more easily able to sing the bass notes whereas some other people are more easily able to sing the higher frequency notes. There are 6 types of vocal ranges that are defined:[8]

1. Soprano: C4–C6
2. Mezzo-soprano: A3–A5
3. Contralto: F3–F5
4. Tenor: C3–C5
5. Baritone: F2–F4
6. Bass: E2–E4

So a person, whose vocal range is Soprano, sings best between the notes C4 to C6. So, it is important that the frequencies between that corresponding to C4 (261Hz) and that corresponding to C6 (1047Hz) need to be highlighted. For every vocal range, certain frequencies sound better than the others and if boosted can give a shine to the vocals. Hence depending on the vocal range certain frequencies can be boosted and cut for best quality vocals.

There will be an option for the user to input the lowest note used in the song and the highest note used in the song. Depending on these two inputs the computer will determine an approximate range in which the vocals lie. Now the corresponding boosts and cuts will be applied to the vocals

ISSN 2229-5518

depending on the vocal range such that the quality of the voice gets enhanced.

Also, when the user inputs the lowest note, it is clear that any frequency below the frequency of that note is redundant and if significant can affect the voice quality. So we pass the vocal signal through a high pass filter with a cutoff frequency equal to that of the lowest note.

3.3 Auto equalization of the voice and the instrument signals:

In sound recording and production, equalization is used to alter the frequency response of an audio system using linear filters. A typical song is made up of the voice signals as well as the instrument signals. The vocals in a song are the most important feature of the song. They must be such as to be able to cut through the instrument signals, so that the vocals can be heard in front of the music. Thus, for the best mix, it is essential that equalization takes place.

Auto equalization as the name suggests is automatic equalization of the voice signals and the music signals. We are making use of Short Time Fourier Transform for obtaining the frequency response. To test how auto equalization takes place, we make use of MATLAB. The very first step while operating on MATLAB is to import the voice as well as the instrument signals, which on mixing make up for the song. Stereo is the commonly used method of sound production now-a-days. Hence we need to take into account both the left and the right channels of both the signals. Thus, we need to create two arrays for individual channels in MATLAB. The content in the arrays undergoes Short Time Fourier Transform[9]. It is used to determine sinusoidal frequency and phase content of local sections of a signal as it changes over time. In case of continuous STFT, the signal to be transformed is multiplied by a window function which is non-zero only for a short period of time.[10] The window used is a Hamming window because it reduces harmonic distortion. This results into frame blocking. Working with frames rather than the whole signal is essential because the audio signals are non-stationary signals, i.e. frequency components change with time. So it becomes necessary to consider smaller frames in which the signals are fairly stationary. STFT is taken for all these frames of both channels of both the signals.

Mathematically, it is represented as,

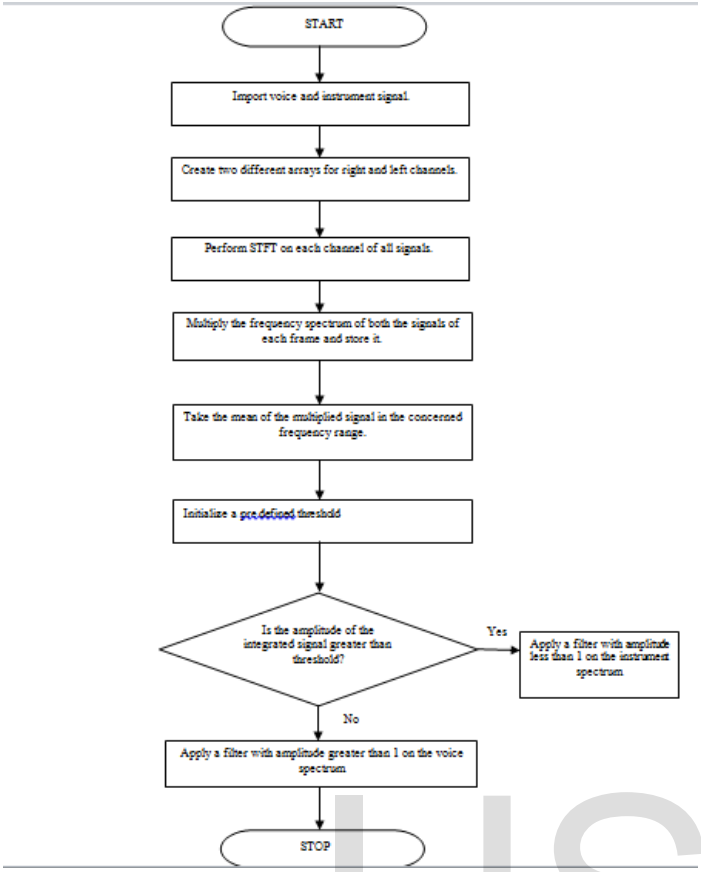
$$\mathbf{STFT}\{x(t)\}(\tau,\omega) \equiv X(\tau,\omega) = \int_{-\infty}^{\infty} x(t)w(t-\tau)e^{-j\omega t}\,dt$$

where w(t) is the window function and x(t) is the signal to be transformed, in our case the voice and the instrument signals.

Now we carry out equalization frame by frame. Essentially what is done is, STFT of a frame of the left channel of the voice signal is multiplied with the STFT of the corresponding frame of left channel of the instrument signal and the same is done for the right channel as well. It is important to note here that the size of STFT has to be same for all frames. Now after the multiplications, the dominant frequencies that are common in both voice and instrument signals will get boosted and the other frequencies will get attenuated. The prominent frequencies are the frequencies where we need to work on individually in voice and instrument signals. A threshold is set to find out the common frequencies. If a frequency is common in both signals and has a high enough gain, then the voice does will not have space to cut through the music. Hence, the frequency component is attenuated from the instrument signal's frame so that the frequency gets more emphasis in voice signal. If the frequency is not common in both signals and the frequency lies below 5 Khz, the frequency is accentuated from the voice signal if it has a minimum gain(defined by a threshold) because it can be a harmonic of the fundamental frequency of a formant which can improve the voice quality. Here a minimum threshold is kept because we don't want to increase harmonic distortion which can happen. We can't afford to boost these frequencies much because of possible harmonic distortion. An important aspect of sound signals is that they are most audible when they are in the range of 2 kHz-4 kHz.

Thus gain variations will be very subtle in this range to avoid overall degradation of the sound quality.

ISSN 2229-5518



4 Results

We performed the 2nd part of the equalization which concentrates on making space for the vocals. We took a

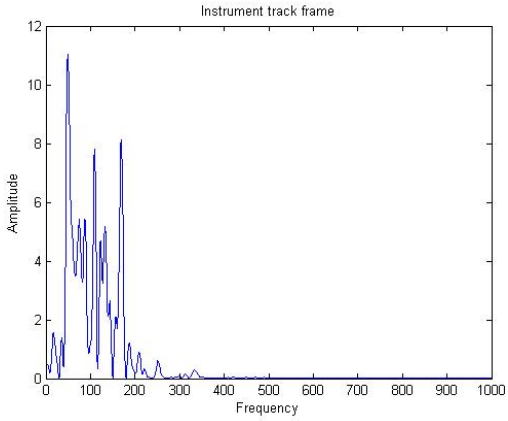
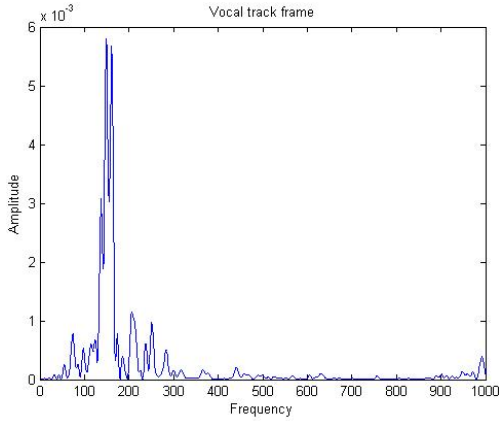


Figure 3.a : Frequency Spectrum of the Instrument Track

sample instrument signal and a voice signal and performed STFT on that signal. Then we took one frame of the STFT of 30ms. The frequency spectrum of the instrument and the voice signals is shown in the figures Figure 3.a and Figure 3.b respectively.

Figure 3.b : Frequency Spectrum of the Vocal Track



Note that these spectrums are only of one time frame of 30 ms out of the several frames obtained after performing STFT.

These two frequency spectrums were then multiplied. The frequency spectrum of the multiplied signal is as shown in Figure 3.c.

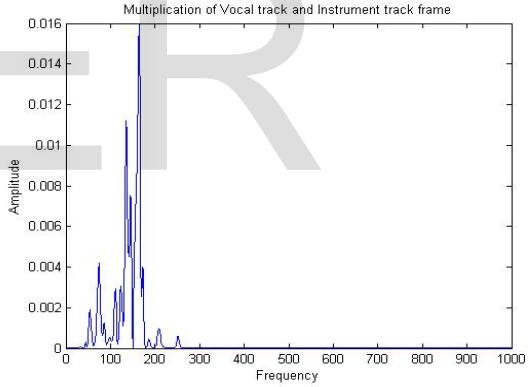


Figure 3.c : Multiplication of frequency response of Vocal track and Instrument track

Now, we considered only the Bass Range and took a mean of the values of the multiplied signal in the bass region that is between 80Hz and 250Hz. We compared this to a threshold, in this case 0.008. Since the mean of the multiplied values in the bass range is greater than 0.008 we attenuated the frequency response of the instrument signal by a factor of 0.95 by using a band pass filter centered at a frequency of 165 Hz. It is important to note here that the attenuating factor is 0.95 which is not very less than 1. This is because equalization

always has to be subtle and large changes may affect the quality of the sound vastly.

The final equalized frequency response of the instrument signal is as shown in Figure 3.d.

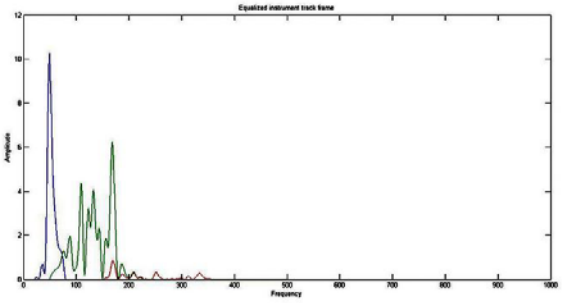


Figure 3.d : Frequency Response of Equalized Instrument Track

5 Future Scope

The proposed methods in this paper find their way naturally into a Virtual Studio Technology(VST) plugin .These plugins are developed for Digital Audio Stations(DAW) on which sound Engineers work for music production.DAWs basically are high end softwares for music production. We plan on optimizing and giving variety of features with our plugin so that even an enthusiast with little working knowledge can mix vocals easily. We plan on making a VST as soon as our results are confirmed by few sound engineers because they can judge the plugin better.

Real time equalization is also an area we look to explore soon. In live concerts there is a sound engineer who constantly monitors all the channels and does equalization based on what he listens to. Making an auto equalizer for such a scenario is a complex task and needs to take into account all the above mentioned parameters. In addition to all this, the acoustics of the place where concert is being held also plays a big role here. A feedback from the auditorium will be used to constantly work on equalizing various tracks.

We also plan on automating the step where user needs to manually input the lowest frequency in the vocal range

6 Conclusion

This paper covers various aspects of voice equalization. Also, proposals at auto equalization have been made by taking into consideration several aspects of equalization.

Acknowledgement

We would like to thank the respected principal Dr. Hari Vasudevan of Dwarkadas J. Sanghvi College of Engineering. We would also like to thank Dr. Sunil Karamchandani, Prof. Shivali Kulkarni and the HOD of Electronics Department at D.J Sanghvi College of Engineering Prof. Prasad Joshi for guiding us and supporting us whilst we undertook the writing of this paper. We would also like to thank S.V.K.M. for encouraging us in such co-curricular activities.

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