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Development of the Mobile Application for Equalization of Digital Audio System

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Abstract

Digital audio is technology that can be used to record, store, generate, manipulate, and reproduce sound using audio signals encoded in digital form. A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. Graphic equalizers are often included in consumer audio equipment and software which plays music on home computers. In the graphic equalizer, the input signal is sent to a bank of filters. The digital audio system for concert in opera theater or concert hall must be installed a lot of speaker units. Also, the speaker units are installed at intervals in a theater. And the audio systems must be controlled by the kinds of concert for the best sound quality. As that result, the user needs the remote control in order to play the audio system according to the kinds of concert in opera theater or concert hall efficiently in real-time. In this paper, we implement the mobile application for the remote control using wireless communications to control the digital audio system. Also, we implement the crossover and equalizer of the digital audio system. Each GEQ (Graphic EQualizer) Vertical Seek Bar is used. It has the frequency range value from -15.0 to 15.0 and the frequency range value is divided in units of 0.5 such as -15.0, -14.5, -14.0 …. 14.0, 14.5, and 15. And the 31Vertical Seek Bars are used for controlling each frequency range of volume. Also they are represented by graphic user interface for easy operation of user.

Keywords*: Band Pass Filter, Equalization, Digital Audio System, Mobile Application, Remote Control, Graphic Equalizer*

1. Introduction

Digital audio is technology that can be used to record, store, generate, manipulate, and reproduce sound using audio signals encoded in digital form. Following significant advances in digital audio technology during the 1970s, it rapidly replaced analog audio technology in most areas of sound production, sound engineering and telecommunications. A microphone converts sound to an analog electrical signal, then an analog-to-digital converter (ADC)—typically using pulse-code modulation—converts the analog signal into a digital signal. The amplitude passed by each filter is adjusted using a slide control to boost or cut frequency components passed by that filter. A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor-inductor capacitor circuit). These filters can also be created by combining a low-pass filter with a high-pass filter. Equalization is the process of adjusting the balance between frequency components within an electronic signal. The most well known use of equalization is in sound recording and reproduction but there are many other applications in electronics and telecommunications. The circuit or equipment used to achieve equalization is called an equalizer. These devices strengthen (*boost*) or weaken (*cut*) the energy of specific frequency bands. In sound recording and reproduction, equalization is the process commonly used to alter the frequency response of an audio

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system using linear filters. The vertical position of each slider thus indicates the gain applied at that frequency band, so that the knobs resemble a graph of the equalizer's response plotted versus frequency. Room acoustics describes how sound behaves in an enclosed space. The sound wave has reflections at the walls, floor and ceiling of the room. The incident wave then has interference with the reflected one. This action creates standing waves that generate nodes and high pressure zones. After determining the best dimensions of the room, using the modal density criteria, the next step is to find the correct reverberation time. The most appropriate reverberation time depends on the use of the room. Times about 1.5 to 2 seconds are needed for opera theaters and concert halls. For broadcasting and recording studios and conference rooms, values under one second are frequently used. The recommended reverberation time is always a function of the volume of the room. Several authors give their recommendations. A good approximation for Broadcasting Studios and Conference Rooms is: TR[1 kHz] = [0,4 log (V+62)] – 0,38 TR in seconds and V=volume of the room in m^3. The ideal RT60 must have the same value at all frequencies from 30 to 12,000 Hz. Or, at least, it is acceptable to have a linear rising from 100% at 500 Hz to 150% down to 62 Hz [. Analog audio signals are susceptible to noise and distortion, due to the innate characteristics of electronic circuits and associated devices. Disturbances in a digital system do not result in error unless the disturbance is so large as to result in a symbol being misinterpreted as another symbol or disturb the sequence of symbols. It is therefore generally possible to have an entirely error-free digital audio system in which no noise or distortion is introduced between conversion to digital format, and conversion back to analog. A digital audio signal may be encoded for correction of any errors that might occur in the storage or transmission of the signal, but this is not strictly part of the digital audio process. This technique, known as channel coding, is essential for broadcast or recorded digital systems to maintain bit accuracy. The discrete time and level of the binary signal allow a decoder to recreate the analog signal upon replay. Eight to Fourteen Bit Modulation is a channel code used in the audio Compact Disc (CD) [1-7]. As a result, a digital audio system is developing by many manufacturers for sale. But, the price of digital audio system is very expensive and the cheap product is necessary. The digital audio system for concert in theater must be installed a lot of speaker units. They are connected by networks each other. If the size of theater for playing the concert is big then many speaker units are needed. Also, the speaker units are installed at intervals in a theater. And the audio systems must be controlled by the kinds of concert for the best sound quality. As that result, the user needs the remote control in order to play the audio system according to the kinds of concert in opera theater or concert hall efficiently in real-time. In this paper, we implement the digital audio system with remote control using wireless communications for controlling the audio system. Also, we implement he graphic equalizer and crossover of digital audio system.

The organization of this paper is as follows. Section 2 describes the related works and section 3 explains the system structure of digital audio system with the crossover and graphic equalizer for remote control on android and Section 4 presents the development of mobile application and results. Finally, conclusions and future research presented in Section 5.

2. Related Works

2.1. Audio Crossover

Audio crossovers are a class of electronic filter used in audio applications. Most individual loudspeaker drivers are incapable of covering the entire audio spectrum from low frequencies to high frequencies with acceptable relative volume and absence of distortion so most hi-fi speaker systems use a combination of multiple loudspeaker drivers, each catering to a different frequency band. Crossovers split the audio signal into separate frequency bands that can be separately routed to loudspeakers optimized for those bands. Active crossovers are distinguished from passive crossovers in that they divide the audio signal prior to amplification. Active crossovers come in both digital and analog varieties. Digital active crossovers often include additional signal processing, such as limiting, delay, and equalization. Signal crossovers allow the audio signal to be split into bands that are processed separately before they are mixed together again. Some examples are: multiband dynamics (compression, limiting, de-essing), multiband distortion, bass enhancement, high frequency exciters, and noise reduction such as Dolby A noise reduction. The definition of an ideal audio crossover changes relative to the task at hand. If the separate bands are to be mixed back together again (as in multiband processing), then the ideal audio crossover would split the incoming audio signal into separate bands that do not overlap or interact and which result in an output signal unchanged in frequency, relative levels, and phase response. This ideal performance can only be approximated. How to implement the best approximation is a matter of lively debate. On the other hand, if the audio crossover separates the audio bands in a loudspeaker, there is no requirement for mathematically ideal characteristics within the crossover itself, as the frequency and phase response of the loudspeaker drivers within their mountings will eclipse the results. Satisfactory output of the complete system comprising the audio crossover *and* the loudspeaker drivers in their enclosure(s) is the design goal. Such a goal is often achieved using non-ideal, asymmetric crossover filter characteristics [3].

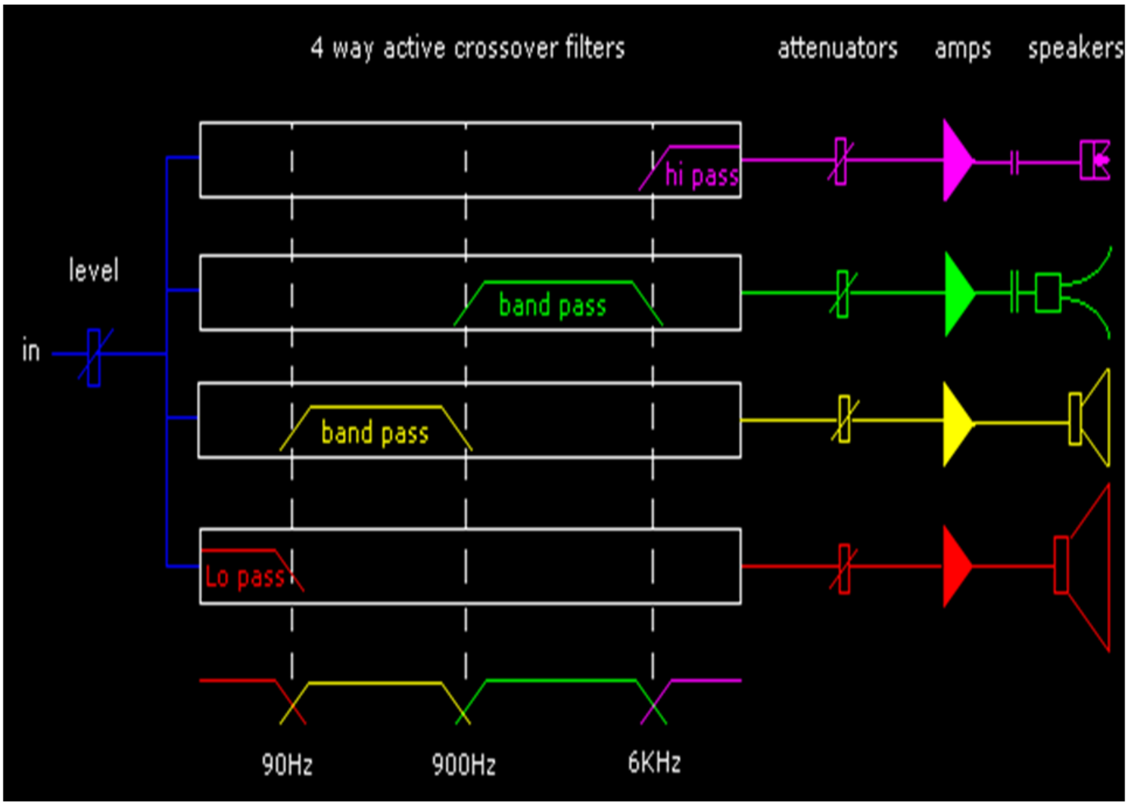

Figure 1. Crossover Diagram

2.2. Equalizer

Equalization is the process of adjusting the balance between frequency components within an electronic signal. The most well known use of equalization is in sound recording and reproduction but there are many other applications in electronics and telecommunications. The circuit or equipment used to achieve equalization is called an equalizer. These devices strengthen (*boost*) or weaken (*cut*) the energy of specific frequency bands. In sound recording, equalization is used to improve an instrument's sound or make certain instruments and sounds more prominent. For example, a recording engineer may use an equalizer to make some high-pitches in a vocal part louder while making low-pitches in a drum part quieter. Eualization is commonly used to increase the 'depth' of a mix, creating the impression that some sounds in a mono or stereo mix are farther or closer than others, relatively. Equalization is also commonly used to give tracks with similar frequency components complementary spectral contours, known as mirrored equalization. Select components of parts which would otherwise compete, such as bass guitar and kick drum, are boosted in one part and cut in the other, and vice versa, so that they both stand out. Equalizers can correct problems posed by a room's acoustics, as an auditorium will generally have an uneven frequency response especially due to standing waves and acoustic dampening. The frequency response of a room may be analyzed using a spectrum analyzer and a pink noise generator for instance. Then a graphic equalizer can be easily adjusted to compensate for the room's acoustics. Such compensation can also be applied to tweak the sound quality of a recording studio in addition to its use in live sound reinforcement systems and even home hi-fi systems. During live events where signals from microphones are amplified and sent to speaker systems, equalization is not only used to "flatten" the frequency response but may also be useful in eliminating feedback. When the sound produced by the speakers is picked up by a microphone, it is further reamplified; this recirculation of sound can lead to "howling" requiring the sound technician to reduce the gain for that microphone, perhaps sacrificing the contribution of a singer's voice for instance. Even at a slightly reduced gain, the feedback will still cause an unpleasant resonant sound around the frequency at which it would howl. But because the feedback is troublesome at a particular frequency, it is possible to cut the gain only around that frequency while preserving the gain at most other frequencies. This can best be done using a parametric equalizer tuned to that very frequency with its amplitude control sharply reduced. By adjusting the equalizer for a narrow bandwidth (high Q), most other frequency components will not be affected. The extreme case when the signal at the channel's center frequency is completely eliminated is known as a notch filter [4].

Figure 2. Sample of Graphic Equalizer

3. System Structure

A digital audio system is composed of two parts. The one is hardware and the other is software part. The hardware system has the functions; 1) input: 4 channels which are the analog inputs and 4 channels which are the digital inputs using AES/EBU simultaneous; 2) output: 8 channels according to setting value of a DSP. The software has the functions; 1) sound processing: Cross-over, EQ (Equalizer), Delay, and Limiter; 2) applications for controlling the audio system. The following figure 3 shows the software system structure of the digital audio system.

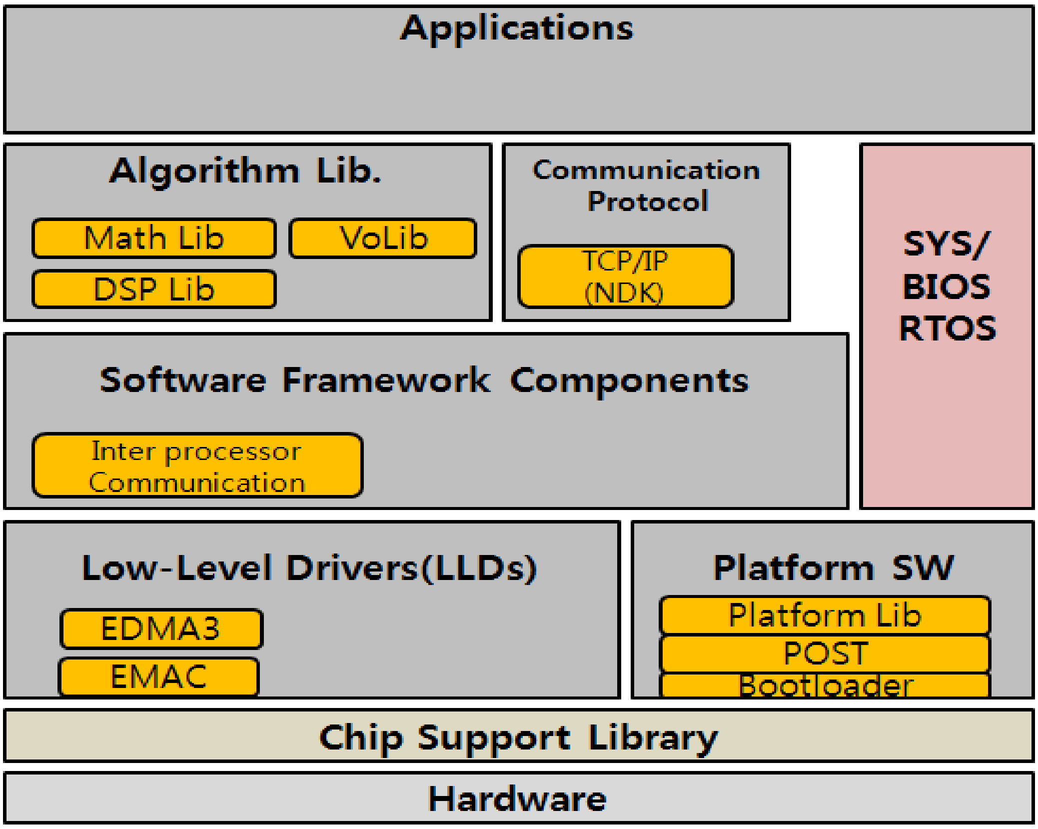

Figure 3. Software Structure

4. Development of Mobile Application for Remote Control

We are implementing the mobile application of digital audio system for remote control using java language on the android environment. And we used the TCP/IP socket interface in order to connect client and server for remote control. In sound recording and reproduction, equalization is the process commonly used to alter the frequency response of an audio system using linear filters. Most hi-fi equipment uses relatively simple filters to make bass and treble adjustments. Analog signals that have not already been bandlimited must be passed through an anti-aliasing filter before conversion, to prevent the distortion that is caused by audio signals with frequencies higher than the Nyquist frequency, which is half of the system's sampling rate. An imageView class is used for representing the circles on a screen and onTouchListener is used for finding the axis of an ID that is an audio system. And setOnseekBarChanged is used for getting the changed value of seekBar which is implemented using a VerticalSeekBar class. Graphic and parametric equalizers have much more flexibility in tailoring the frequency content of an audio signal. Graphic equalizers are often included in consumer audio equipment and software which plays music on home computers. In the graphic equalizer, the input signal is sent to a bank of filters. Each filter passes the portion of the signal present in its own frequency range or band. The amplitude passed by each filter is adjusted using a slide control to boost or cut frequency components passed by that filter. The vertical position of each slider thus indicates the gain applied at that frequency band, so that the knobs resemble a graph of the equalizer's response plotted versus frequency. In the mobile application, each GEQ (Graphic EQualizer) Vertical Seek Bar is used. It has the frequency range value from -15.0 to 15.0 and the frequency range value is divided in units of 0.5 such as -15.0, -14.5, -14.0 …. 14.0, 14.5, and 15. And the 31Vertical Seek Bars are used for controlling each frequency range of volume. Also they are represented by graphic user interface for easy operation of user. Analog signals that have not already been band limited must be passed through an anti-aliasing filter before conversion, to prevent the distortion that is caused by audio signals with frequencies higher than the Nyquist frequency, which is half of the system's sampling rate. A band-pass filter is a device that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range. An example of an analogue electronic band-pass filter is an RLC circuit (a resistor-inductor capacitor circuit). These filters can also be created by combining a low-pass filter with a high-pass filter [3].

4.1. Design of the Packet and Commands

The TCP/IP socket interface is used in order to connect from client and to server vice versa for remote control. And the network protocol and data packet must be design. The figure 4 shows a sequence diagram to connect between server and client and table 1 and 2 shows the network protocol, data packets, and commands .

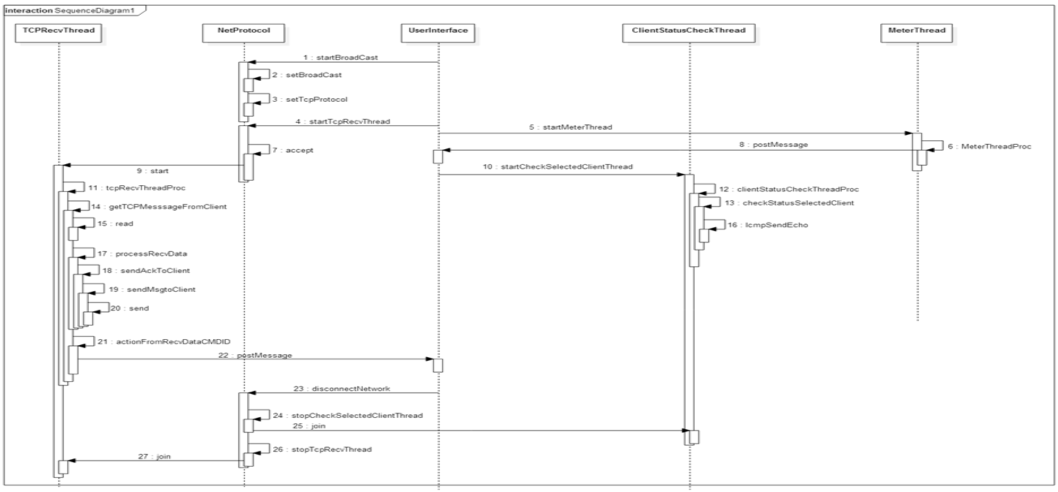

Figure 4. Sequence Diagram

- UART / SPI / Ethernet Protocol

  [UART]

  Baud rate : 115200
  Data bits : 8 Bits
  Parity : None
  Stop bits : 1 Bit

  [SPI]

  <Read>

  1. CS start + read cmd'R'(M) + cmd_id(M) + sub_para(M) + data0(M) + CS end
  2. 5msec delay

3. CS start + Length(S) + CS end
4. CS start + Length(S) + Command ID(S) + Sub Parameter(S) + Data n(S) + Check sum(S) + CS end

<Write>
1. CS start + write cmd('W') + cmd_id(M) + sub_para(M) + data0(M) + CS end
2. CS start + Length(M) + CS end
3. CS start + Length(M) + Command ID(M) + Sub Parameter(M) + Data n(M) + Check sum(M) + CS end

- **Packet Structure**

Table 1. Packet

Length	Command Id	Sub Parameter	Data	Check Sum
1	1	1	N	1

Table 2. Extension Packet

*Length	Command Id	Sub Parameter	Data	Check Sum
3	1	1	N	1

- Command: Input G-EQ(Command ID = 0x71('q'))

3.4.1 Channel No.
Sub Para = '1'[0x31], '2'[0x32], '3'[0x33], '4'[0x34]
Data 0 = Freq Hz(float to hex - 0 byte)
Data 1 = Freq Hz(float to hex - 1 byte)
Data 2 = Freq Hz(float to hex - 2 byte)
Data 3 = Freq Hz(float to hex - 3 byte)
Data 4 = Gain dB(float to hex - 0 byte)
Data 5 = Gain dB(float to hex - 1 byte)
Data 6 = Gain dB(float to hex - 2 byte)
Data 7 = Gain dB(float to hex - 3 byte)
//**[-40~+15] 0.1db step
Data 8 = On/Off switch(0:off, 1:on)
Data 9 = Link Value (-1(0xFF) == is Not Link

- Command: Iutput BPF(Command ID = 0x58('X'))

3.8.1 output Channel No. Sub Para = '1'[0x31], '2'[0x32], '3'[0x33], '4'[0x34], '5'[0x35], '6'[0x36], '7'[0x37], '8'[0x38]
Data 0 = Filter select (0 : IIR FILTER , 1 : FIR FILTER)
Data 1 = Filter type Low (depend on e_DSPFilterTypeInfo)
Data 2 = Filter type Hi (depend on e_DSPFilterTypeInfo)
Data 3 = Fir Filter Tab(unsigned short Value -LO byte)
Data 4 = Fir Filter Tab(unsigned short Value -HI byte)
Data 5 = Freq Low Cutoff Hz(float to hex - 0 byte)

Data 6 = Freq Low Cutoff Hz(float to hex - 1 byte)
Data 7 = Freq Low Cutoff Hz(float to hex - 2 byte)
Data 8 = Freq Low Cutoff Hz(float to hex - 3 byte)
**[20~22000] step = 0.5
Data 9 = Freq Hi Cutoff Hz(float to hex - 0 byte)
Data 10 = Freq Hi Cutoff Hz(float to hex - 1 byte)
Data 11 = Freq Hi Cutoff Hz(float to hex - 2 byte)
Data 12 = Freq Hi Cutoff Hz(float to hex - 3 byte)
**[20~22000] step = 0.5
Data 13 = Gain dB(float to hex - 0 byte)
Data 14 = Gain dB(float to hex - 1 byte)
Data 15 = Gain dB(float to hex - 2 byte)
Data 16 = Gain dB(float to hex - 3 byte)
**[-60~25] step = 0.5
Data 17 = Phase (float to hex - 0 byte)
Data 18 = Phase (float to hex - 0 byte)
Data 19 = Phase (float to hex - 0 byte)
Data 20 = Phase (float to hex - 0 byte)
**[0~180] step = 5 degree
Data 21 = Freq (float to hex - 0 byte)
Data 22 = Freq (float to hex - 0 byte)
Data 23 = Freq (float to hex - 0 byte)
Data 24 = Freq (float to hex - 0 byte)
**[20~20000] step = 0.5
Data 25 = Invert(0:off["-"], 1:on["+"])
Data 26 = On/Off switch(0:off, 1:on)
Data 27 = Link Value (-1(0xFF) == is Not Link)

4.2. Band Pass Filter

An imageView class is used for representing the circles on a screen and onTouchListener is used for finding the axis of an ID that is an audio system. And setOnseekBarChanged is used for getting the changed value of seekBar which is implemented using a VerticalSeekBar class. As you can see in figure 5, a PC version of band pass filter is shown.

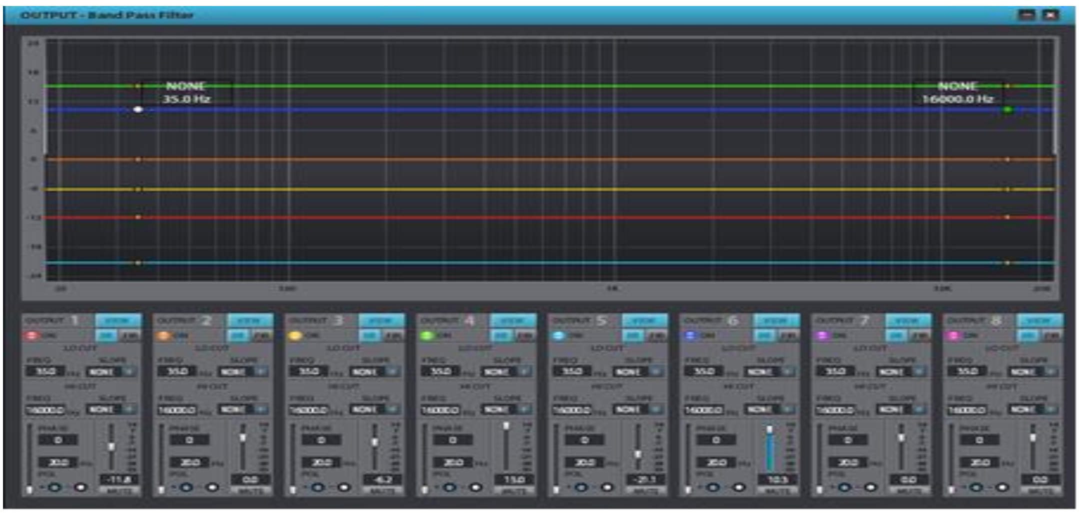

Figure 5. Band Pass Filter on PC

The following figure 6 shows the band pass filter on android.

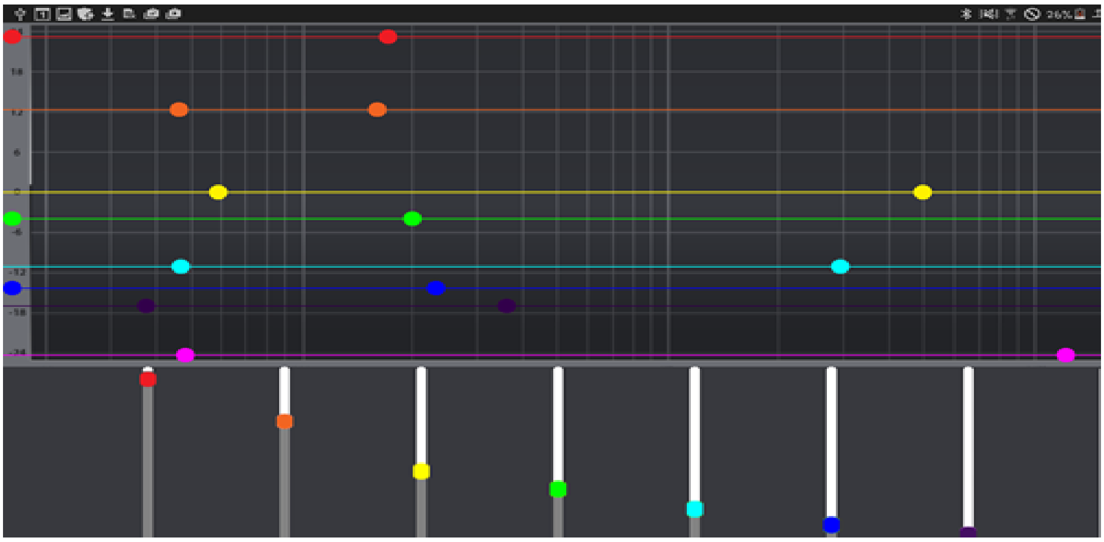

Figure 6. Band Pass Filter on Android

4.3. Graphic Equalizer

Each GEQ (Graphic EQualizer) Vertical Seek Bar is used. It has the frequency range value from -15.0 to 15.0 and the frequency range value is divided in units of 0.5 such as -15.0, -14.5, -14.0 …. 14.0, 14.5, and 15. And the 31Vertical Seek Bars are used for controlling each frequency range of volume. Also they are represented by graphic user interface for easy operation of user. The following figure 7 shows GEQ on PC.

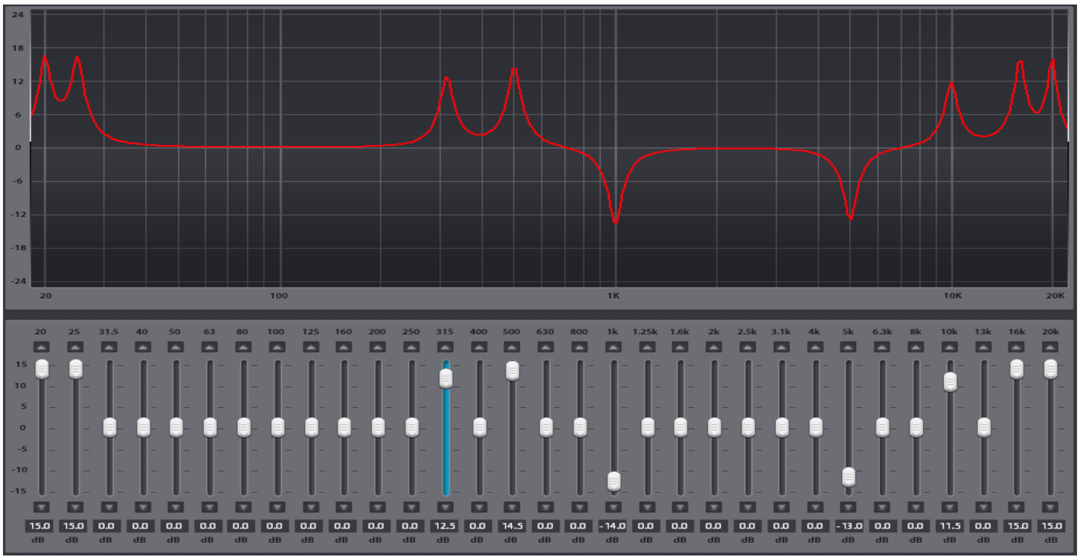

Figure 7. Graphic Equalizer on PC

The following figure 8 shows the result of graphic equalizer on the android.

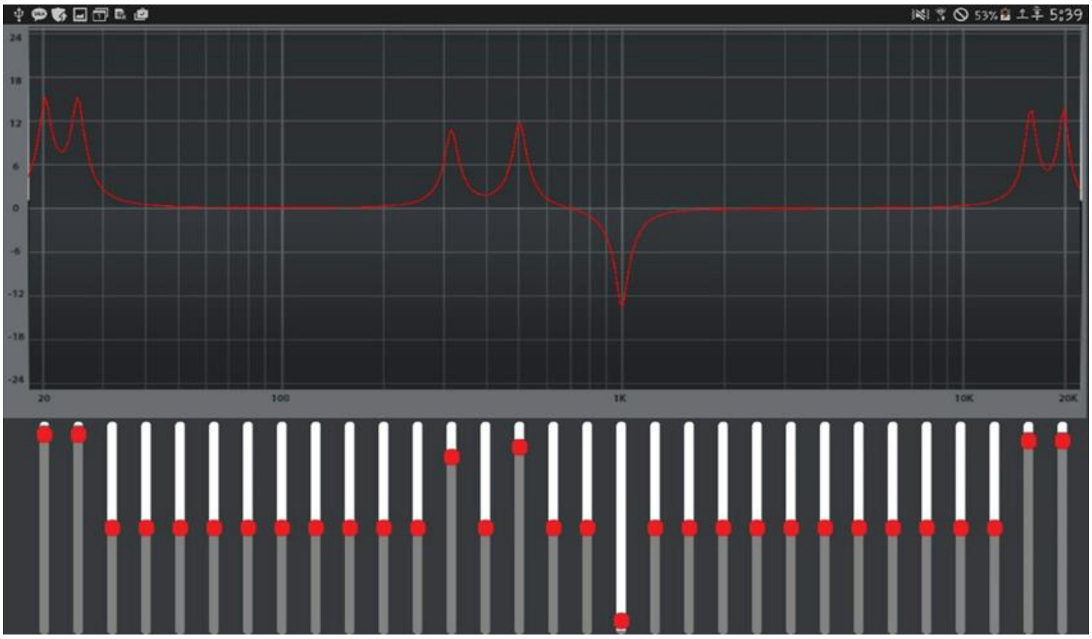

Figure 8. Graphic Equalizer on Android

5. Conclusions and Future Work

A digital audio system for concert in opera theater or concert hall must be installed a lot of speaker units. They are connected by networks each other. If the size of opera theater for playing the concert is big then many speaker units are needed. Also, the speaker units are installed at intervals in a theater. And the audio systems must be controlled by the kinds of concert for the best sound quality. Aa that result, the user needs the remote control in order to play the audio system according to the kinds of concert in opera theater or concert hall efficiently in real-time. In this paper, we implement the mobile application with remote control using wireless communications for controlling the audio system. Also, the equalizer and crossover were implemented. Graphic equalizers are often included in consumer audio equipment and software which plays music on home computers. In the graphic equalizer, the input signal is sent to a bank of filters. In order to implement, an imageView class is used for representing the circles on a screen and onTouchListener is used for finding the axis of an ID that is an audio system. And setOnseekBarChanged is used for getting the changed value of seekBar which is implemented using a VerticalSeekBar class. Each GEQ (Graphic EQualizer) Vertical Seek Bar is used. It has the frequency range value from -15.0 to 15.0 and the frequency range value is divided in units of 0.5 such as -15.0, -14.5, -14.0 …. 14.0, 14.5, and 15. And the 31-Vertical Seek Bars are used for controlling each frequency range of volume. Also they are represented by graphic user interface for easy operation of user. In the future, we will implement the digital audio system functions for sale.

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