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Implementation of Audio Effect Device for Anchor System

Seungwon Lee[1], Soonchul Kwon [2], Seunghyun Lee [*]

[1]*Doctoral student, Department of Immersive Content Convergence, Kwangwoon University, Korea*
[2]*Associate professor, Graduate School of Smart Convergence, Kwangwoon University, Korea*
[*]*Professor, Ingenium College Liberal Arts, Kwangwoon University, Korea*
E-mail : [1]audio@kw.ac.kr, [2]ksc0226@kw.ac.kr, [*]shlee@kw.ac.kr

Abstract

Recently, Audio systems transform the configuration of conventional sound reinforcement and public address systems using audio over internet protocol (AoIP), whereby audio signals are transmitted and received based on internet protocol (IP). Currently, AoIP technologies are leading the audio market, and various technologies have been released. Audio networks and the control hierarchy over peer-to-peer (Anchor) technology based on AoIP transmit and receive audio signals over a wide bandwidth without an audio mixer. Audio system based on Anchor technology is constructed by connecting the on-site audio center (OAC), a device that can transmit and receive audio sources and output equipment over IP. Receiving OAC of the Anchor technology can receive and mix audio signals transmitted from different IPs; consequently, novel audio systems can be configured by replacing conventional audio mixers. However, the Anchor technology does not have an equalizer function for improving the quality of audio equipment. Therefore, tone distortion may occur owing to signal loss between equipment, poor audio-signal clarity, and howling due to audio deformation according to different architectural structures and environments. In this study, we implemented an audio effect device capable of tone control using the Audio Processor Core. Using Anchor technology, tone control was realized through an audio effect device in the receiving OAC. The output of the incoming OAC was received by the audio effect device, which adjusted the tone and then outputted it. Thus, the tone issues in Anchor technology were overcome by the receiving OAC and audio effect devices. In future, audio system configurations using Anchor technology could be the standard for audio equipment.

Keywords: *Anchor system, AoIP, Audio effect, Graphic equalizer, PA system, SR system*

1. Introduction

Advances in information and communication technology have resulted in audio over internet protocol (AoIP) technology, which is currently a leading standard in the audio industry [1, 2]. The AoIP technology transmits audio signals using a local area network (LAN) cable instead of a conventional audio cable. Audio

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Corresponding Author: shlee@kw.ac.kr
Tel: +82-2-940-5290, Fax: +82-2-909-5290
Professor, Department of Immersive Content Convergence, Kwangwoon University, Korea

facilities using AoIP technology have the advantages of enabling a high listening quality and reducing facility costs [3–5].

Recently, audio networks and the control hierarchy over peer-to-peer (Anchor) technology based on AoIP have received a positive response in the audio market. Anchor technology can replace the conventional audio system configuration by enabling broadband audio system configuration through internet protocol (IP) without an audio mixer [6]. Anchor technology uses IP to transmit and receive audio signals. The receiving side can select and mix signals from different IPs, thus enabling the configuration of a novel audio system [7, 8].

Conventional audio systems typically consist of an input device, audio mixing device, equalizer device, and output device [1, 9, 10]. An audio system using Anchor technology consists of an input device, a transmitter and receiver at an on-site audio center (OAC), and an output device. The OAC of the Anchor technology includes conventional audio-mixing functions. Thus, Anchor-based audio system configurations are simpler and more efficient compared with conventional audio system configurations [7, 8, 11]. However, when constructing large-scale audio systems, the Anchor technology is subject to problems that depend on the architectural environment. Given that the OAC of Anchor technology does not have a tone control function similar to an equalizer, problems may arise in large-scale audio system configuration environments, essentially requiring tone control [6, 12]. A large-scale audio system cannot solve problems when the audio signal is unclear or when delays and howling occur, as the output of the audio can be modified owing to the architectural structure, characteristics of each piece of equipment, and signal loss between the equipment [13-15]. Therefore, when constructing large-scale audio systems using the Anchor technology, a device that can correct the tone is required. This study compensates the tone problems that may arise in an audio system based on Anchor technology by implementing an audio effect device that can correct the audio quality and connect it to the Anchor receiving OAC. The remainder of this paper is organized as follows. Section 2 presents an analysis of the audio system configuration and Section 3 describes the audio effect device design. Section 4 details the implementation of audio effect devices and Section 5 presents the conclusions and scope of future research.

2. Audio System Configuration

The AoIP-based Anchor technology was launched in 2017, followed by significant advancements. Anchor technology transmits audio signals over IP, which is a feature of AoIP technology, and allows for the realization of a novel audio system configuration by configuring a broadband audio system without an audio mixer [7, 8, 11]

2. 1 Anchor Technology-based Audio System Configuration

The Anchor technology consists of an OAC, which transmits and receives audio signals, and a network-based audio/video integrated control system (NAVICS), which controls the OAC [16]. The receiving OAC of Anchor can select incoming signals from different transmission IPs, and mix them or control the signal level [8, 11]. Figure 1 shows the OAC and NAVICS systems constituting the Anchor system. The transmission and reception of the OAC are controlled by NAVICS through the network. The transmitting OAC receives the analog signal from the input device and transmits it through the network. The receiving OAC selects and controls audio signals from different IPs and outputs them. The receiving OAC of Anchor technology can replace the traditional audio mixers. Consequently, a novel audio system configuration that is different from previous configurations can be realized [6, 16].

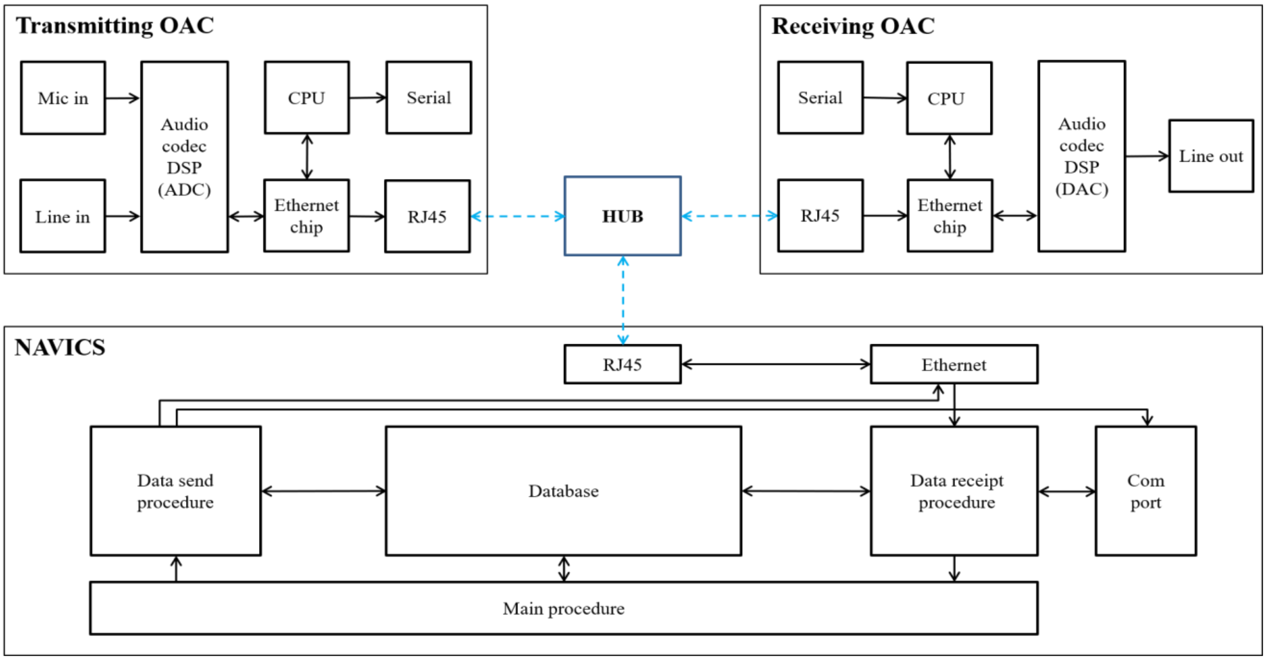


Figure 1. Anchor system configuration

Figure 2 shows the audio system configuration based on the Anchor technology. The audio source (i.e., wireless microphone, computer sound, wired microphone, and instrument audio source) in the input section of Figure 2 is transmitted to the network through a transmitting OAC. The receiving OAC in the output section selects the transmitting OAC, mixes it, and outputs the corresponding signal. In the control section shown in Figure 2, NAVICS communicates with the transmission and reception of the OAC and protocols via the network. In particular, NAVICS controls the OAC by receiving control data through the web or a graphical user interface (GUI) [17].

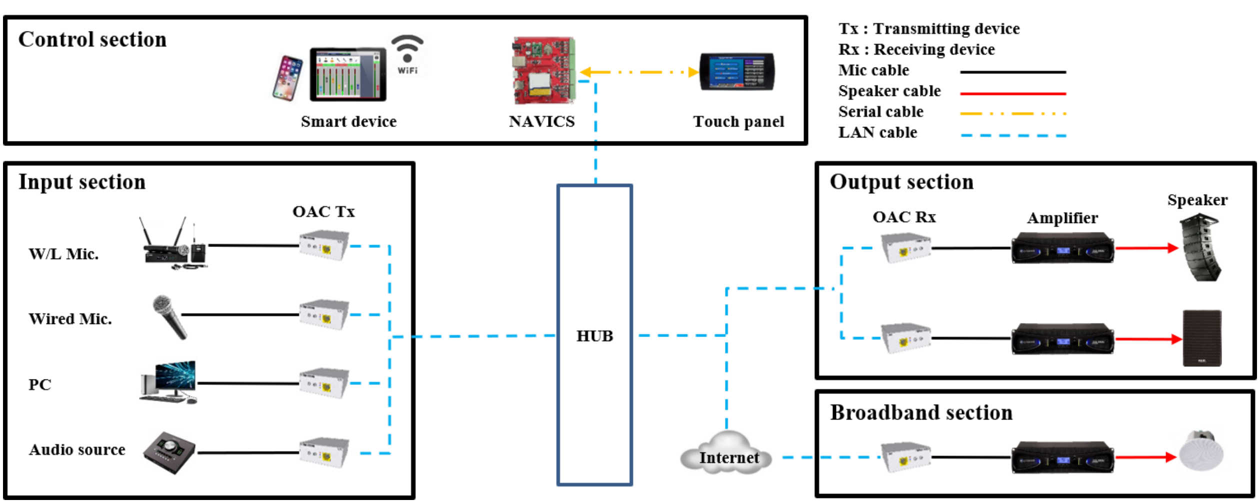
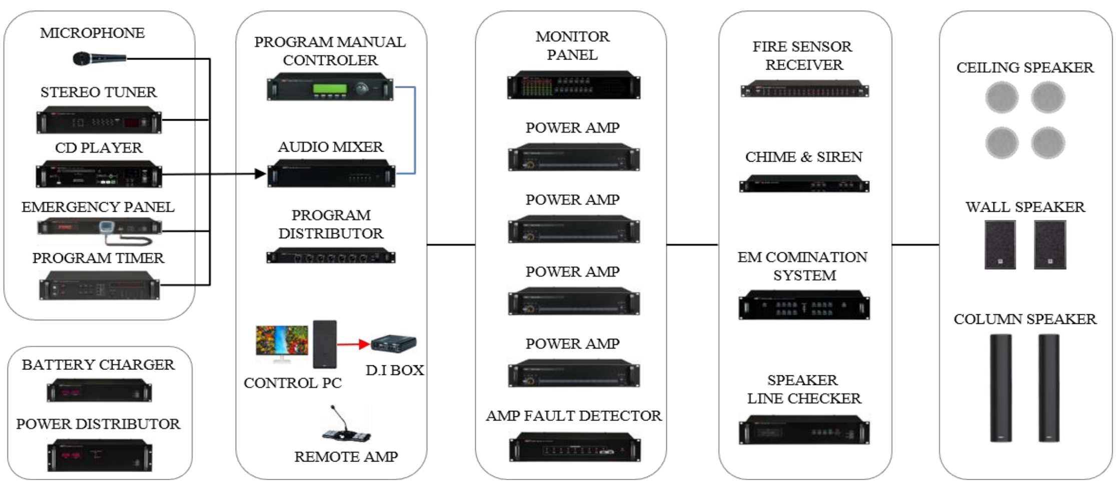
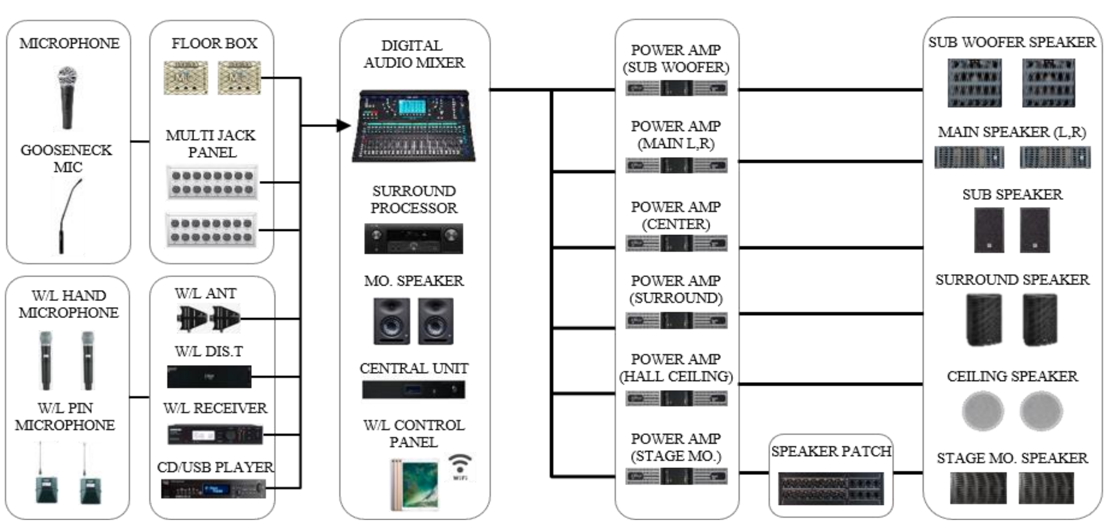


Figure 2. Audio system configuration based on Anchor technology

Figure 3 compares the configurations of the public address (PA) and sound reinforcement (SR) systems, which are audio systems. Figure 3(a) shows the configuration of the PA system. The PA system transmits audio signals via a pre-amp, power amplifier, and speaker; further, it can receive fire reception signals and broadcast guidance in the event of a fire [18–20]. Figure 3(b) shows the configuration of the SR system. In the SR system configuration, speakers are selected through a speaker system design by considering the objective of the audio system and characteristics of the building. Once the speaker system is selected, the power amplifier that can drive the speaker system is determined. Next, the audio source and mixer are selected depending on the size and objective of the audio system [21–23]. Tone-control devices such as equalizers can be configured without application to audio systems. Particularly, a tone-control device that can strike a balance between the architectural environment and audio system can improve the quality of the audio equipment in the SR system.



(A) Public address system configuration



(b) Sound reinforcement system configuration

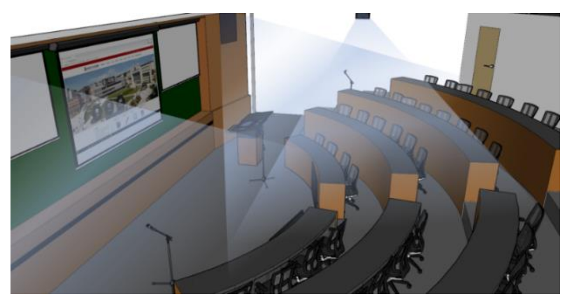
Figure 3. Audio system configuration comparison

2. 2 Audio System Configuration for Sound Setting

Audio effect devices generally include a graphic equalizer for the house curve, time delay for time-difference adjustment, and crossover, which separates each frequency band and allows it to be played on a speaker. In particular, the graphic equalizer enables the modification of sounds altered by the architectural environment (structure and finishes) with maximal similarity to the original sound. The graphic equalizer sets the house curve through 31 bands at 1/3 oct of the audible frequency (20–20000 Hz) [24, 25]. Figure 4 shows the equalizer setting to adjust tone loss and distortion in the audio system. Figure 4(a) shows a schematic diagram of the sound setting for adjusting the house curve using the Equalizer. Figure 4(b) shows a three-dimensional (3D) model depicting the method of measuring pink noise for equalizer setting in the field based on the schematic in Figure 4(a).

A house curve utilizing an equalizer achieves sound balance. Essentially, audio systems installed in buildings distort the original sound, depending on the structure or finishing materials of the building [26]. An equalizer adjusts the distorted sound such that it approximates the original sound. As shown in Figure 4, the pink noise reproduces the entire audible frequency range (20–20000 Hz) as a sound of uniform size. This reproduced sound is detected by a microphone and transmitted to the audio analyzer. An equalizer adjusts the sound deviation with respect to frequency, as shown in the audio analyzer. This equalizing action adjusts the house curve depending on the use case of the sound system (speech, live music, and theater) [27, 28].

(a) Sound setting configuration diagram for adjusting the house curve

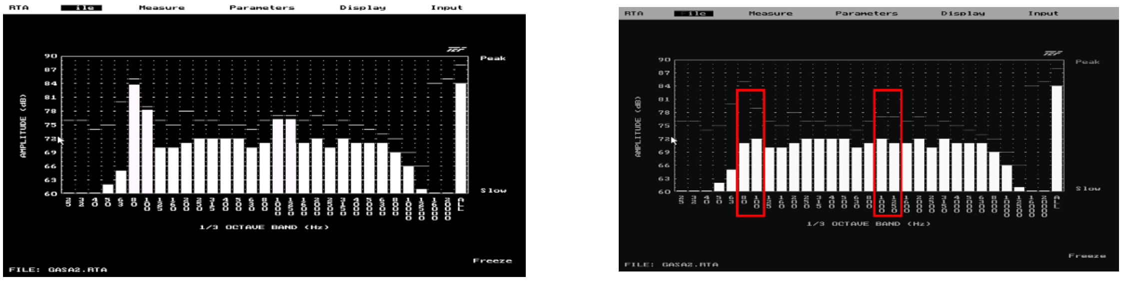


(b) 3D model for measuring pink noise in the field

Figure 4. Equalizer setting to adjust tone loss

Figure 5 shows a comparison between before and after the equalizer setting. Figure 5(a) reveals that before setting the equalizer, high-level sounds are detected at 80, 100, 1000, and 1250 Hz. Figure 5(b) shows the equalizer setting used to reduce high-level sounds and assess the sound played in the pink noise. The audio facility can be configured without an equalizer setting according to the use case. The objective of the PA system is emergency broadcasting in the events of disasters and emergencies. Therefore, in a PA system, sound must be clearly transmitted at a high volume inside and outside the room, rather than with high quality through an equalizer setting [29]. The SR system aims to enhance live or pre-recorded sounds and distribute them to the audience. In SR systems, the equalizer setting is critical for improving the quality of the audio system, as it can control the tone for sound reflection and distortion [30].

The SR system configuration using Anchor technology does not exhibit the function of tone control; thus, when sound problems due to the architectural environment occur, a high-quality audio system configuration cannot be realized. Therefore, the proposed system contains the equalizer function in the Anchor technology to control the audio-quality problems that may occur when configuring an SR system through an audio effect device.



(a) Before equalizer setting **(b) After equalizer setting**

Figure 5. Comparison before and after setting the equalizer

3. Audio Effect Device Design

Audio effects can be implemented using a digital signal processor (ADAU1701) and ARM Cortex M4. The ADAU1701 chip was designed to process audio signals, and is mainly utilized for audio effects using filters such as equalizers, time delays, and crossovers in audio systems. This study describes the implementation of an equalizer using the ADAU1701 chip. The ADAU1701 chip has a programmable digital signal processing core that enables user programming and detailed processing of audio signals. Additionally, external audio signals can be converted into digital signals, which can then be converted into analogous signals.

The ADAU1701 chip exhibits functionality across various communication protocols such as inter-integrated circuits (I2C), serial peripheral interfaces (SPI), and universal asynchronous receiver/transmitter (UART) interfaces, thus allowing it to communicate with devices and run user-created programs. Moreover, it can be programmed using tools provided by chip manufacturers. To use the ADAU1710 chip, a block diagram can be designed for the audio signal processing algorithm using a tool, and the generated hex file can be uploaded to the ADAU1710 chip. Figure 6 shows a configuration diagram of the audio effect device. The audio effect device receives the OAC output, and adjusts and outputs the sound desired by the user. The output signal from the audio effect device is then transmitted to a power amplifier, which amplifies the audio signal and outputs it through the speaker system. Protocol control of ADAU1710 can be realized using an ARM Cortex M4 processor. The audio effect device is installed at the same position as the receiving OAC and can perform serial communication with the receiving OAC.

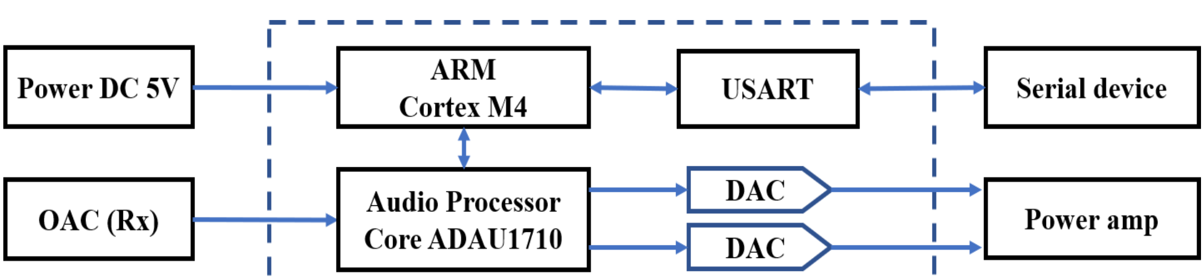


Figure 6. Data processing configuration diagram of audio effect device

Figure 7 shows the configuration diagram for the control of audio effect devices. To access the audio effect device, a control device is used to access NAVICS through the network. NAVICS can communicate with the receiving OAC over the network. The receiving OAC has a built-in serial port and can communicate with the surrounding communication devices. Therefore, NAVICS and the receiving OAC communicate with the ARM Cortex M4 of the audio effect device and control the ADAU1710 chip.

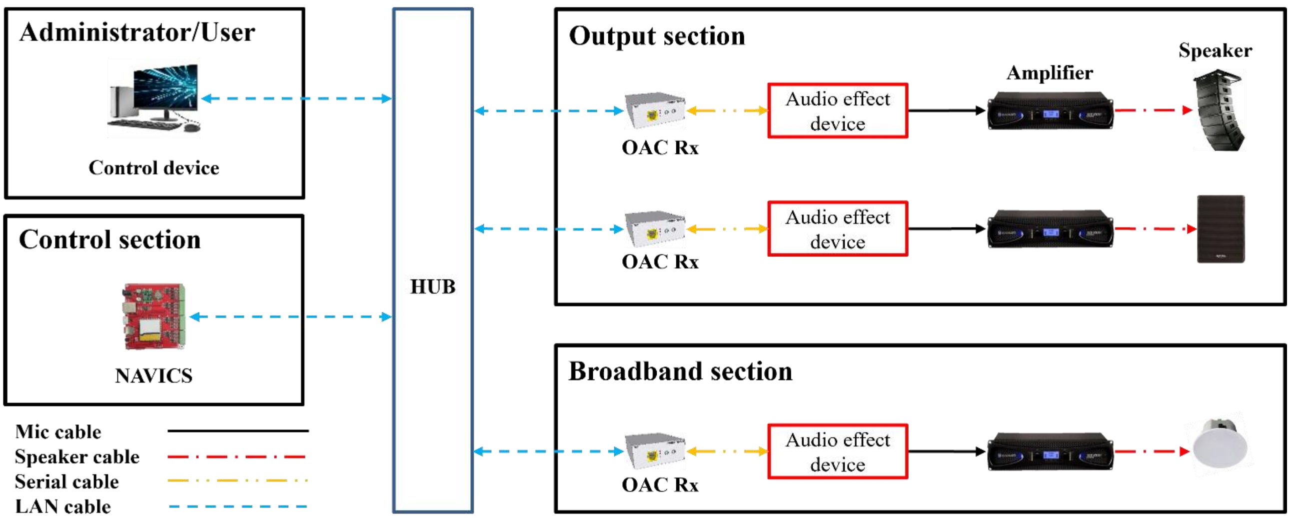


Figure 7. Configuration diagram for the control of audio effect device

Figure 8 shows the relationship between the ARM Cortex M4 and ARM Cortex A7 data processing for the control of the ADAU1710 chip. The ARM Cortex A7 of NAVICS stores and extracts information from the receiving OAC and ADAU1710 chip in the database and controls the ADAU1710 chip through ARM Cortex M4. The ARM Cortex M4 audio effect device receives and controls the hex file uploaded to the ADAU1710 chip from an ARM Cortex A7. Section 4 describes the equalization through NAVICS and the ARM Cortex of the audio effect device.

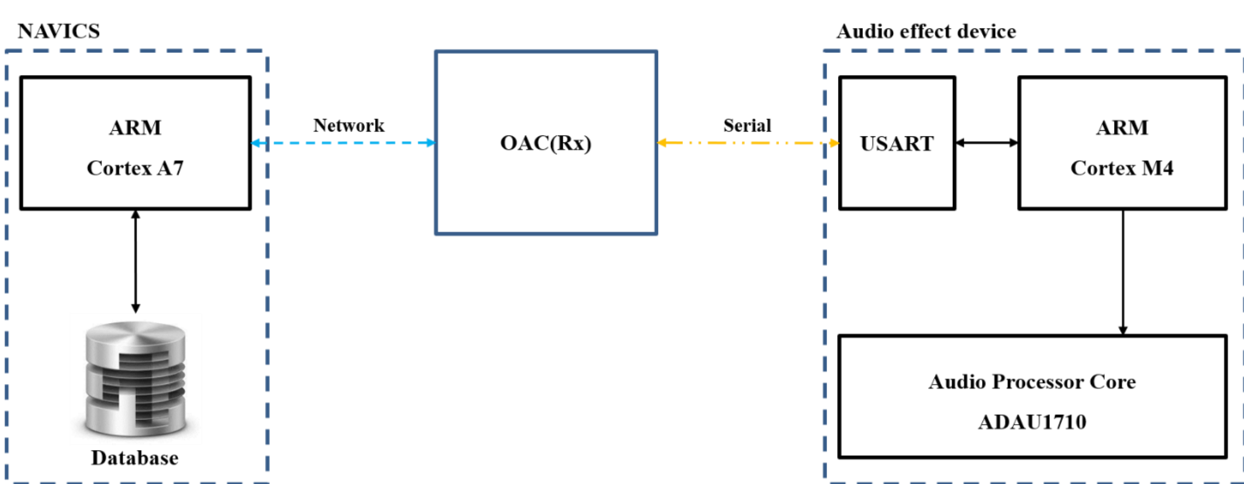


Figure 8. ARM Cortex A7 and ARM Cortex M4 data processing diagram

4. Implementation of Audio Effect Device

The audio effect device implemented using the ADAU1710 chip is a chip designed to process audio signals. The ADAU1710 chip can process audio signals using an audio signal block diagram design. In this study, to control the audio effect device, the audio signal data were designed as a block diagram to the ADAU1710 chip, and the ADAU1710 was controlled through the ARM Cortex A7 of NAVICS and ARM Cortex M4 of the audio effect device via the web. Figure 9 shows the web-based 1/3 OCT band-graphic equalizer GUI. Users can adjust the graphic equalizer of the audio effect device via a web browser. Figure 9(a) and 9(b) reveal the objective of uniformizing the protruding frequency shown in Figure 5. Data controlled through the GUI of the web browser controls the ADAU1710 chip through the ARM Cortex A7 of NAVICS and ARM Cortex M4 of the audio effect device.

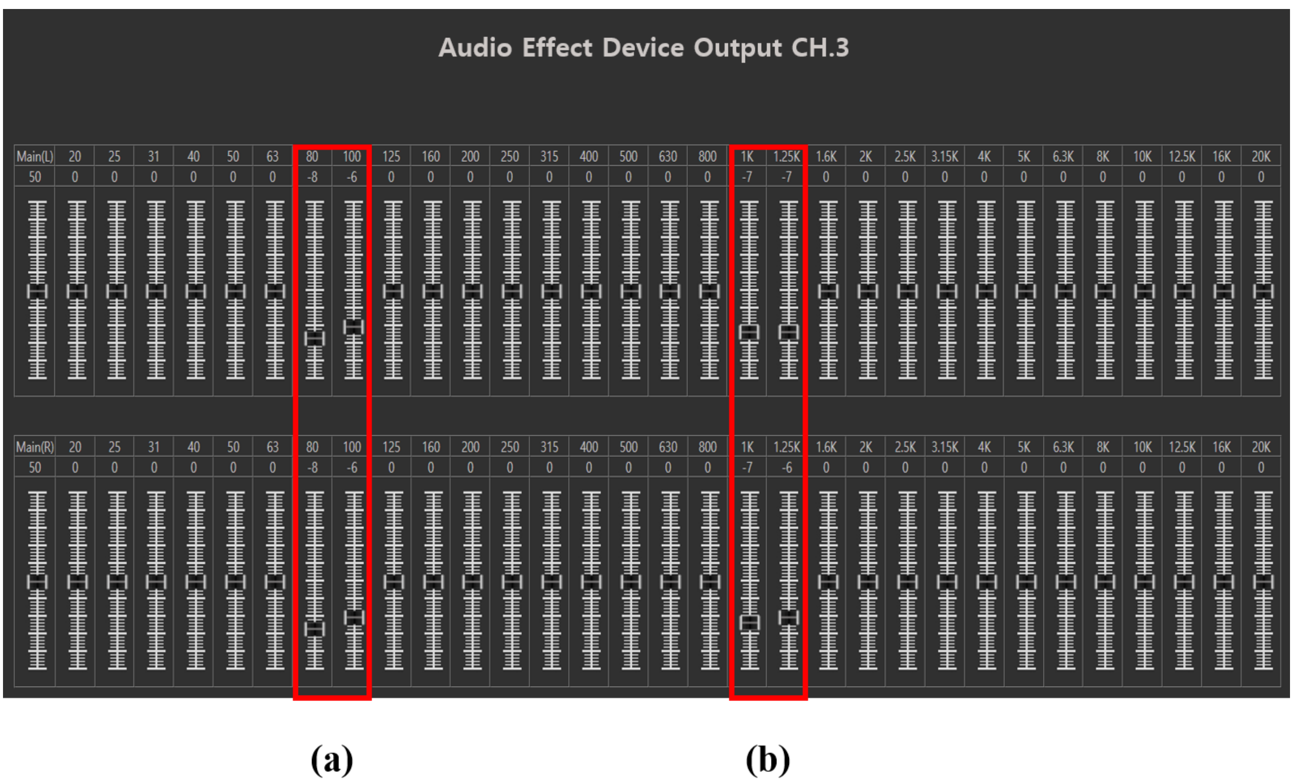


Figure 9. Web-based 1/3 octave band graphic equalizer GUI

Table 1 to Table 3 present the protocol for controlling the ADAU1710 chip using a GUI. Table 1 lists the basic protocol for equalizer control, which allows for resetting and muting via the click of a button. Table 2 lists the protocols for the equalizer levels. The "Main volume" function in Table 2 adjusts the house curve output level and ranges from 0 to 30. Moreover, "EQ_Left" and "EQ_Right" functions in Table 2 are the protocols for controlling the sliding bar. Table 3 lists the protocols corresponding to each frequency band used to control the sound range. The level for each frequency can be adjusted from 0 to ±15 through the 1/3 OCT 31 Band. The protocol data in Tables 1 to Table 3 are firmware data from the ADAU1710 chip and can be controlled directly through ARM Cortex M4.

Table 1. Equalizer basic protocol

	Header	Data length	Command	Data	Checksum
Reset	0X55	0X03	0X02	0X63	0X68
Mute On	0X55	0X03	0X07	0X01	0X0B
Mute Off	0X55	0X03	0X07	0X00	0X60A

Table 2. Equalizer level protocol

	Header	Data length	Command	Channel	Frequency	Data	Checksum
Main Volume(L)	0X55	0X05	0X03	0X00	0X1F	0X01	0X28
Main Volume(R)	0X55	0X05	0X03	0X01	0X1F	0X01	0X29
EQ_Left (+)	0X55	0X05	0X01	0X00	0X00	0X10	0X16
EQ_Left (0)	0X55	0X05	0X01	0X00	0X00	0X0F	0X16
EQ_Left (–)	0X55	0X05	0X01	0X00	0X00	0X1E	0X16
EQ_Right (+)	0X55	0X05	0X01	0X01	0X00	0X10	0X16
EQ_Right (0)	0X55	0X05	0X01	0X01	0X00	0X0F	0X16
EQ_Right (–)	0X55	0X05	0X01	0X01	0X00	0X0E	0X16

Table 3. Equalizer frequency protocol

Frequency	20 Hz	25 Hz	31 Hz	40 Hz	50 Hz	63 Hz	80 Hz	100 Hz
Protocol	0X00	0X01	0X02	0X03	0X04	0X05	0X06	0X07
Frequency	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz	630 Hz
Protocol	0X08	0X09	0X0A	0X0B	0X0C	0X0D	0X0E	0X0F
Frequency	800 Hz	1 kHz	1.25 kHz	1.6 kHz	2 kHz	2.5 kHz	3.15 kHz	4 kHz
Protocol	0X10	0X11	0X12	0X13	0X14	0X15	0X16	0X17
Frequency	5 kHz	6.3 kHz	8 kHz	10 kHz	12.5 kHz	16 kHz	20 kHz	
Protocol	0X18	0X19	0X1A	0X1B	0X1C	0X1D	0X1E	

Algorithm 1 expresses the method for controlling the ADAU1710 chip in an ARM Cortex M4 using the pseudocode. The ARM Cortex M4 can receive data from NAVICS and equalize it using Algorithm 1. When ARM Cortex M4 receives “Reset” data, it initializes the parameter values for volume and turns “Mute” on. When “Mute” data are received, the output of the equalizer is muted on or off depending on the parameters. “Volume” data correspond to controlling the main volume and are controlled through parameters for the channel and volume. “EQ” data control the ADAU1710 chip via parameters for the channel, frequency, and value (from –15 to +15). Thus, the implementation of an audio effect device for tone control in Anchor's receiving OAC was possible through ARM Cortex M4, ADAU1710 Design, and Algorithm 1.

Algorithm 1 Algorithm for controlling ADAU1710 on ARM Cortex M4

```
Void ADAU1701():
    While True:
        If Received data through Serial port:

            If data is “Reset” then:
                Reset parameter values such as volume, mute_On

            Else if data is “Mute(val)” then:
                If val is “On” then
                    Execute mute left & right channel;
                Else if val is “Off” then
                    Unmute left & right channel;
                Else return: // data is false

            Else if data is “Volume(channel, value)” then:
                If channel is “R”:
                    Control left master volume with value;
                Else if channel is “L”
                    Control right master volume with value;
                Else return: // data is false

            Else if data is “EQ(channel, frequency, value)” then:
                If channel is “Right”:
                    Control frequency value of the right channel;
                Else if channel is “Left”
                    Control frequency value of the left channel;
                Else return: // data is false

            Else:
                Pass
```

5. Conclusion

In this study, tone control in an audio system configuration was applied to Anchor technology, and the following conclusions were obtained. First, we implemented an audio effect device to enable tone control in Anchor technology, enabling a high-quality SR system to be achieved. Second, NAVICS enables remote control of audio effect devices, enabling ubiquity without restrictions on time and place. The results of this study confirm that the audio effect device enabled an audio system based on Anchor technology to be implemented as a high-quality SR system, and it was confirmed that control of the audio system without restrictions in time and place was possible through IP. However, tone control through an audio effect device has a complicated control process and requires expert knowledge, so there are improvements to be made for non-experts to use it conveniently. First, the firmware process for the ARM Cortex A7 of NAVICS and ARM Cortex M4 of the audio effect device is complex, thus requiring simplification. Second, because the tone control of audio effect devices is only possible on Windows-based PC, tone control must be enabled on various platforms for user convenience. It is expected that these problems can be improved through algorithms. Through this study, Anchor technology enabled a novel paradigm in the SR system by implementing an audio effect device capable of tone control. We hope that the Anchor-based audio system configuration is expected to be a future standard in the audio industry.

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