International Journal of Internet, Broadcasting and Communication Vol.16 No.2 10-21 (2024)
http://dx.doi.org/10.7236/IJIBC.2024.16.2.10

IJIBC 24-2-2

Implementation of Tone Control Module in Anchor System for Improved Audio Quality

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 are changing the configuration of conventional sound reinforcement (SR) systems and public address (PA) systems by using audio over IP (AoIP), a technology that can transmit and receive audio signals based on internet protocol (IP). With the advancement of IP technology, AoIP technologies are leading the audio market and various technologies are being released. In particular, audio networks and control hierarchy over peer-to-peer (Anchor) technology based on AoIP is a system that transmits and receives audio signals over a wide bandwidth without an audio mixer, creating a novel paradigm for existing audio system configurations. Anchor technology forms an audio system by connecting audio sources and output equipment with On-site audio center (OAC), a device that can transmit and receive IP. Anchor's receiving OAC is capable of receiving and mixing audio signals transmitted from different IPs, making it possible to configure a novel audio system by replacing the conventional audio mixer. However, Anchor technology does not have the ability to provide audio effects to input devices such as microphones and instruments in the audio system configuration. Due to this, when individual control of each audio source is required, there is a problem of not being able to control the input signal, and it is impossible to individually affect a specific input signal. In this paper, we implemented a tone control module that can individually control the tone of the audio source of the input device using the audio processor core in the audio system based on Anchor technology, tone control for audio sources is possible through a tone control module connected to the transmitting OAC. As a result of the study, we confirmed that OAC receives the signal from the audio source, adjusts the tone and outputs it on the tone control module. Based on this, it was possible to solve problems that occurred in Anchor technology through transmitting OAC and tone control modules. In the future, we hope that the audio system configuration using Anchor technology will become established as the standard for audio equipment.

Keywords: *Anchor system, Audio over IP, Audio system, Equalizer, OAC, Tone control*

Manuscript Received: March. 3, 2024 / Revised: March. 12, 2024 / Accepted: March. 18, 2024
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

1. Introduction

Advances in information and communication technology have brought audio over IP (AoIP) technology, which is currently leading the audio industry, to the market. [1, 2]. AoIP technology is a method of transmitting audio signals using a local area network (LAN) cable instead of transmitting audio signals through a conventional audio cable. Audio facilities using AoIP technology have the advantage of enabling high-quality listening and reducing facility costs [3-5]. Recently, audio networks and control hierarchy over peer-to-peer (Anchor) technology based on AoIP is receiving a positive response in the audio market as a technology that can replace the conventional audio system configuration by enabling the configuration of a broadband audio system through IP without an audio mixer [6]. In particular, Anchor technology uses IP to transmit and receive audio signals, and the receiving side can select and mix signal coming from different IPs, enabling the configuration of a novel audio system [7, 8]. A conventional audio system configuration typically consists of an input device, an audio mixing device, an equalizer device, and an output device. [1, 9, 10]. An audio system using Anchor technology consists of an input device, a transmitter and receiver (On-site audio center: OAC), and an output device. Anchor technology's OAC includes conventional audio mixing functions. In this way, the anchor-based audio system configuration is simpler and more efficient than the conventional audio system configuration [7, 8, 11]. However, Anchor technology does not have the ability to provide tone control or effects to input devices. Therefore, when individual control and specific effects of input devices are required, problems arise such as having to use external equipment. This problem makes installation and maintenance difficult due to complex wiring between input devices and control equipment. Additionally, effects on audio sources cannot be controlled remotely. As a result, when using Anchor technology to configure an audio system, the audio source requires a device that can control the tone. This paper is a study to implement a tone control module that can control the tone of the audio source in an audio system based on anchor technology. Section 2 analyzed the audio system configuration and PA and SR systems based on Anchor technology, and described the equipment configuration for sound settings. Section 3 describes the design of the tone control module. Section 4 describes the implementation of the tone control modules, and Section 5 discusses conclusions and the construction of a new audio system using Anchor technology in the future.

2. Audio System Configuration

AoIP-based Anchor technology was launched in 2017 and has continued to evolve to the present. Anchor technology not only transmits audio signals over IP, which is a feature of AoIP technology, but also enables novel audio system configuration by configuring a broadband audio system without an audio mixer [6, 7, 11]

2. 1 Audio System Configuration Based on Anchor Technology

Anchor technology consists of OAC, which transmits and receives audio signals, and network-based audio/video integrated control system (NAVICS), which can control OAC [12]. Anchor's receiving OAC can select signals coming from different transmission IPs and mix them or control the signal level [8, 11]. Figure 1 shows the systems of OAC and NAVICS constituting the Anchor system. The transmitting OAC receives the analog signal from the input device and transmits it through the network, and the receiving OAC selects and controls audio signals coming from different IPs and outputs them. The transmitting and receiving OAC is controlled by NAVICS through the network. Anchor's receiving OAC can replace the traditional audio mixer function. As a result, the Anchor-based audio system configuration became possible to create a novel audio system configuration that was different from the previous one [6, 12].

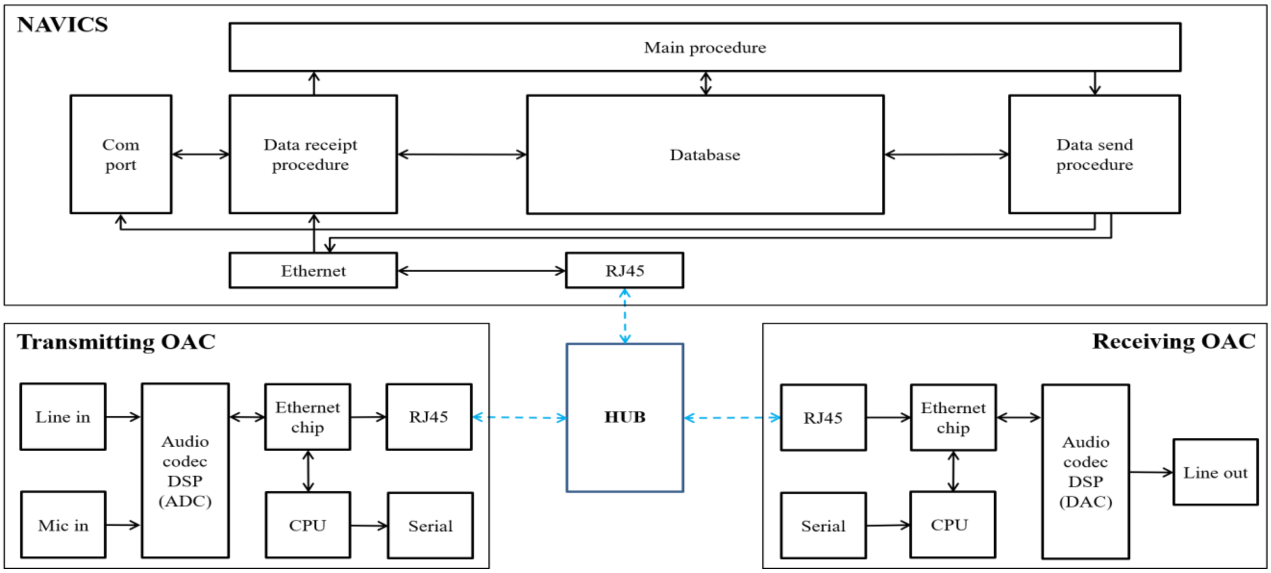


Figure 1. Anchor system configuration diagram

Figure 2 shows the audio system configuration using Anchor technology. Input source (PC, W/L Mic., Wired Mic., Audio source) in the input section of Figure 2 is transmitted to the network through the transmitting OAC. The receiving OAC in the Output section selects the transmitting OAC and mixes and outputs the corresponding signal. In the control section of Figure 2, NAVICS communicates with transmission and reception OAC and protocol through the network. NAVICS can control OAC by receiving control data through the Web or graphical user interface (GUI) [13].

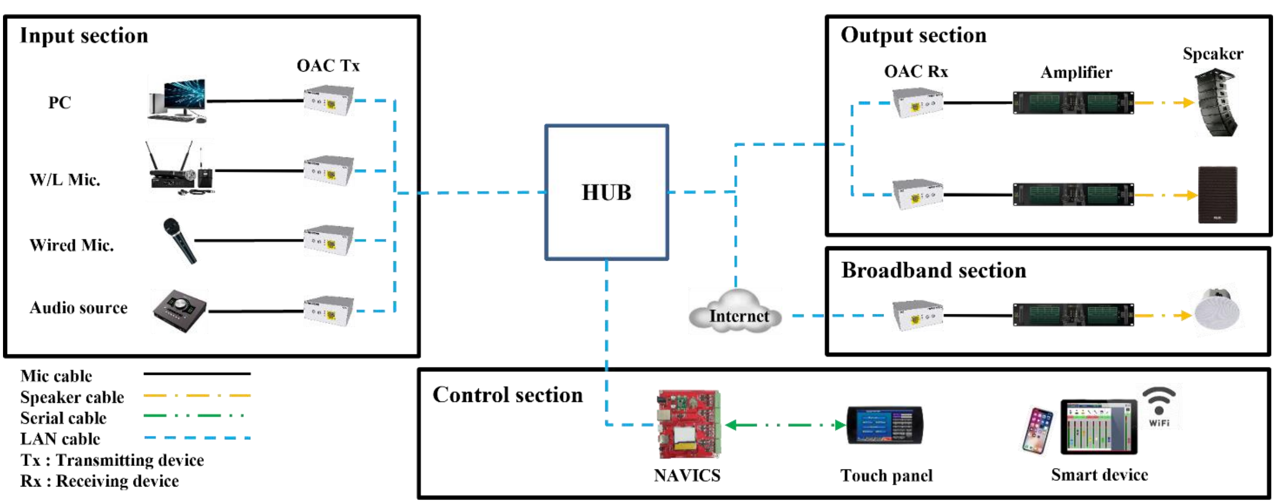
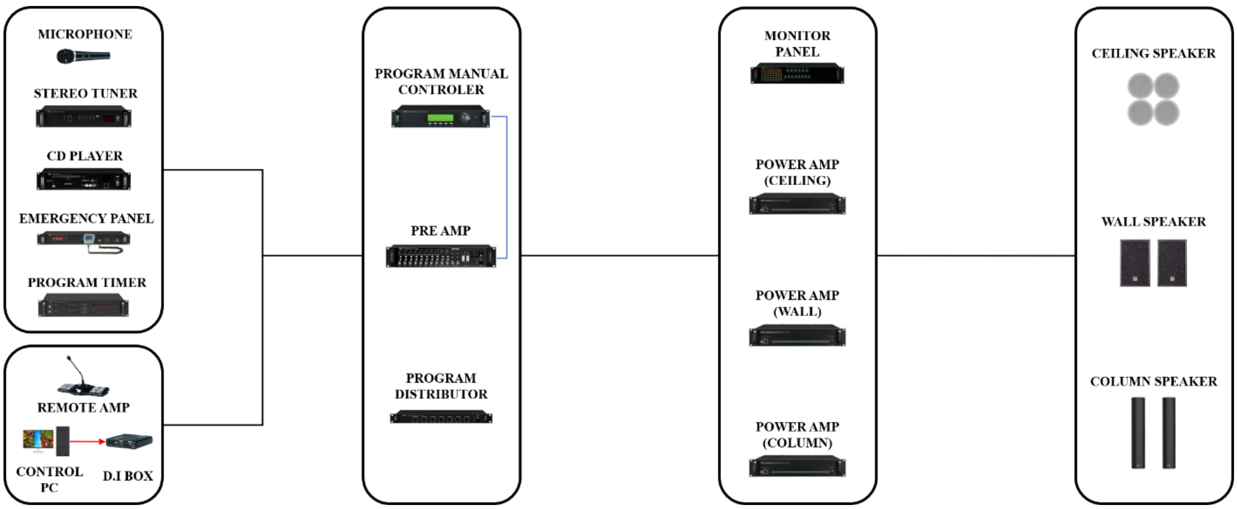
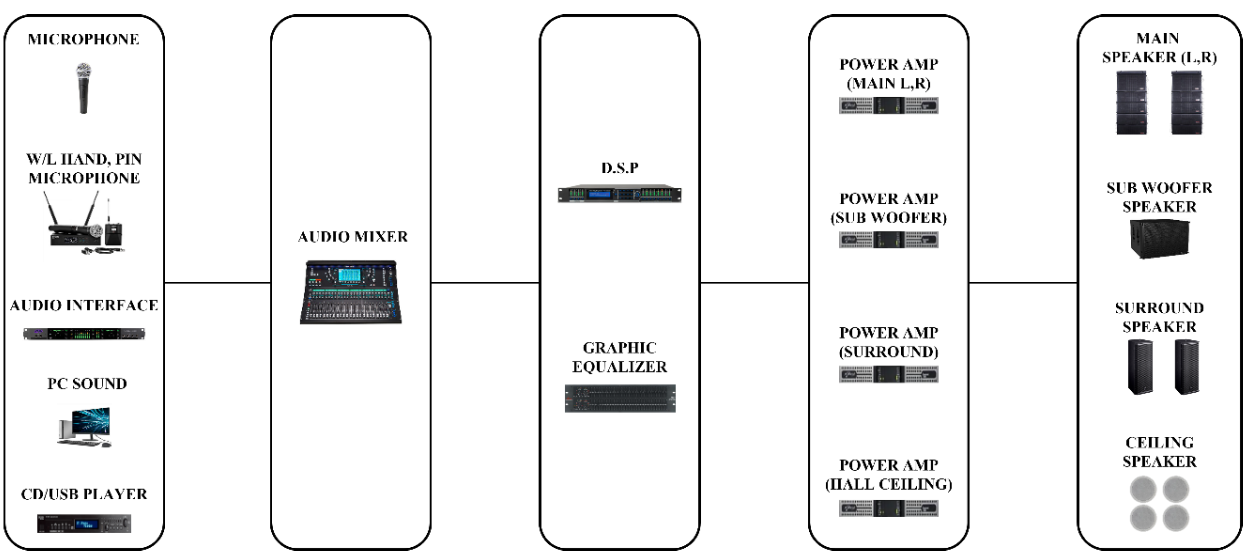


Figure 2. Audio system configuration based on Anchor technology

Figure 3 compares the configuration of the public address (PA) system and sound reinforcement (SR) system, which are audio system systems. Figure 3(a) shows the configuration of the PA system. The PA system transmits audio signals through a pre-amp, amplifier and speaker, and has the feature of being able to receive fire reception signals and make guidance broadcasts in the event of a fire [14-16]. Figure 3(b) shows the configuration of the SR system. In the SR system configuration, speakers are selected through speaker system design in consideration of the purpose of use of the audio system and the characteristics of the building. Once the speaker system is selected, the power amplifier that can drive the speaker system is determined. The audio source and audio mixer are determined depending on the size and purpose of the audio system. [17-19]. The tone control device such as an equalizer may be configured without being applied to an audio system. However, the tone control device that can adjust the balance between the architectural environment and the audio system has the function of improving the quality of the audio equipment in the SR system.



(A) PA system configuration

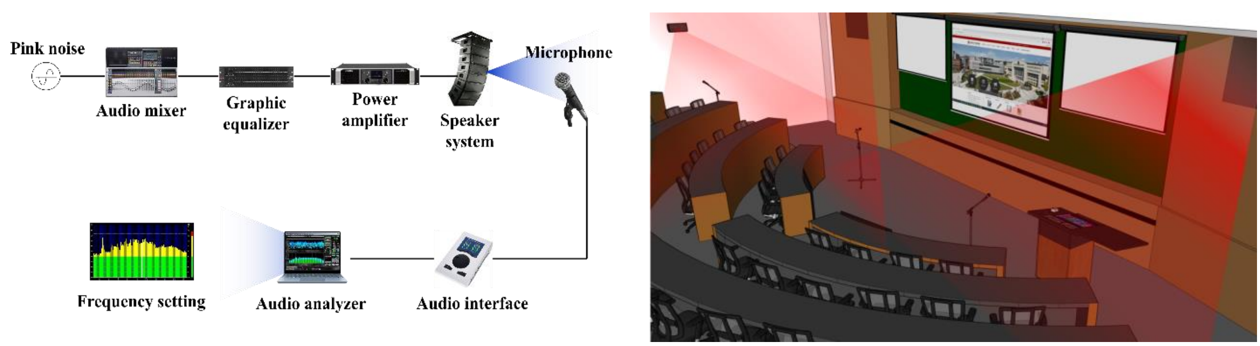


(b) SR system configuration

Figure 3. Audio system configuration comparison

2. 2 Audio System Configuration for Sound Setting

Tone control devices generally include a graphic equalizer for the house curve, a time delay for time difference adjustment, and a crossover that separates each frequency band and allows it to be played on a speaker. In particular, the graphic equalizer makes it possible to hear notes that have been altered by the architectural environment (structure, finishes) as close to the original sound as possible. The graphic equalizer sets the house curve through 31 bands at 1/3 octave of the audible frequency (20 Hz ~ 20 kHz) [20, 21]. 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 [22]. 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) [23, 24].

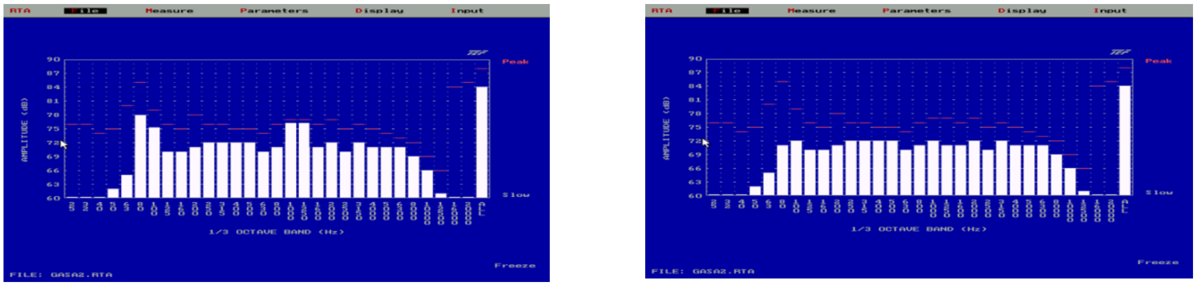


(a) Sound setting configuration diagram for adjusting the house curve

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

Figure 4. Equalizer setting to adjust tone loss

Figure 5 shows a comparison of before and after the equalizer setting. Figure 5 (a) shows before equalizer setting, and high-level sounds are being picked up at 80 Hz, 100 Hz, 1000 Hz, and 1250 Hz. Figure 5 (b) shows the equalizer setting to reduce high-level sounds and listen to the sound played in pink noise. The audio facility can be configured without Equalizer setting according to the purpose of use. The PA system is aim to emergency broadcast in accordance with disasters and emergency. Therefore, in the PA system, it is important to clearly convey it with loud volume inside and outside the room rather than with high-quality sound through Equalizer setting [25]. The SR system aims to enhance live or pre-recorded sound and distribute it to the audience. In the SR system, Equalizer setting is important to improve the quality of the audio system, and Equalizer setting can control the tone for sound reflection and distortion [26]. SR system configuration using Anchor technology does not have the function of controlling tone such as Equalizer, so when sound problems occur due to the architectural environment, high-quality audio system configuration cannot be made. Therefore, this study enables the equalizer function in Anchor technology, so that audio quality problems that may occur when configuring an SR system can be controlled through the tone control module.



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

Figure 5. Comparison before and after equalizer setting

3. Tone Control Module Design

Audio effects can be implemented using a digital signal processor (ADAU1701) chip and ARM Cortex M4. ADAU1701 is a chip designed to process audio signals, and is mainly used for audio effects using filters such as equalizer, time delay, and crossover in audio systems. This paper describes the implementation of equalizer using ADAU1701. ADAU1701 has a programmable digital signal processor (DSP) core, allowing user programming and detailed processing of audio signals. In addition, external audio signals can be converted into digital signals or digital signals can be converted into analog signals. ADAU1701 is capable of inter-integrated circuit (I2C), serial peripheral interface (SPI), and universal asynchronous receiver/transmitter (UART) interfaces, allowing it to communicate with devices and run user-created programs. ADAU1710 can be programmed using tools provided by the chip manufacturer. To use ADAU1710, design a block diagram for the audio signal processing algorithm using a tool and upload the generated hex file to ADAU1710. Figure 6 shows the configuration diagram for the tone control module. The tone control module receives the output of the transmitting OAC, adjusts and outputs the sound desired by the user. The output signal of the tone control module is sent to the receiving OAC through the transmitting OAC, and then sent to the power amplifier to amplify the audio signal and amplify it through the speaker system. Protocol control of ADAU1710 is possible through ARM Cortex M4. The tone control module is installed in the same place as the transmitting OAC and can perform serial communication with the transmitting OAC.

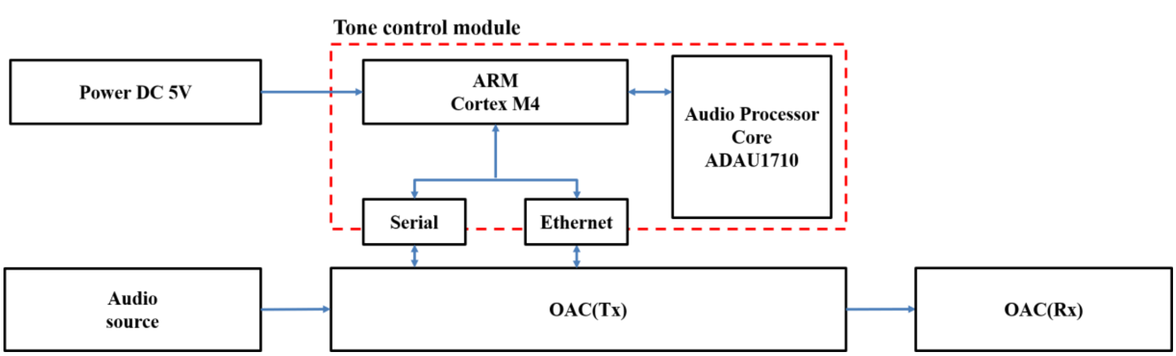


Figure 6. Data processing configuration diagram of tone control module

Figure 7 shows a schematic diagram for controlling the tone control module. In order to access the tone control module, the control device is used to access NAVICS through the network. NAVICS and the transmitting OAC can communicate with each other through a network, and the transmitting OAC has a built-in serial port and has the ability to communicate with surrounding communication devices. Therefore, the transmitting OAC communicates with the ARM Cortex M4 of the tone control module and controls the ADAU1710.

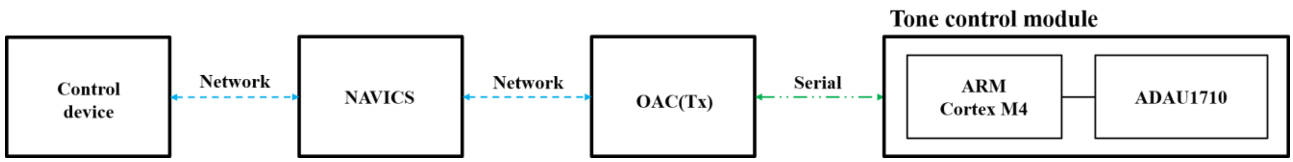


Figure 7. Schematic diagram for accessing and controlling the tone control module

Figure 8 shows the relationship between ARM Cortex M4 and ARM Cortex A7 data processing to control ADAU1710. ARM Cortex A7 of NAVICS stores and extracts information from the receiving OAC and ADAU1710 in the database and controls ADAU1710 through ARM Cortex M4. The ARM Cortex M4 of the tone control module receives and controls the hex file uploaded to the ADAU1710 from the ARM Cortex A7. Section 4 enables equalizing through NAVICS and the ARM Cortex of the tone control module.

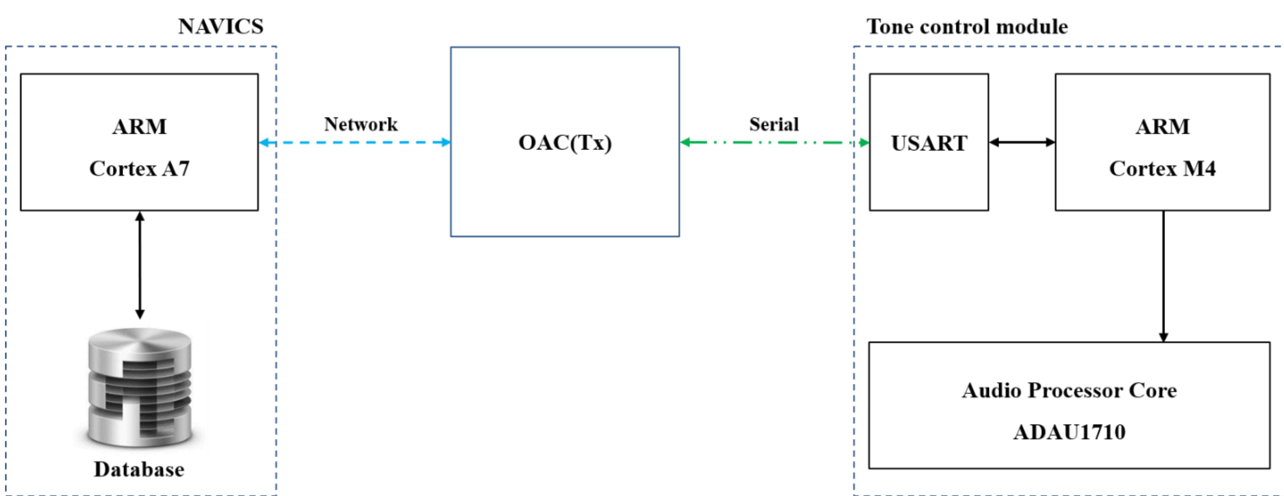
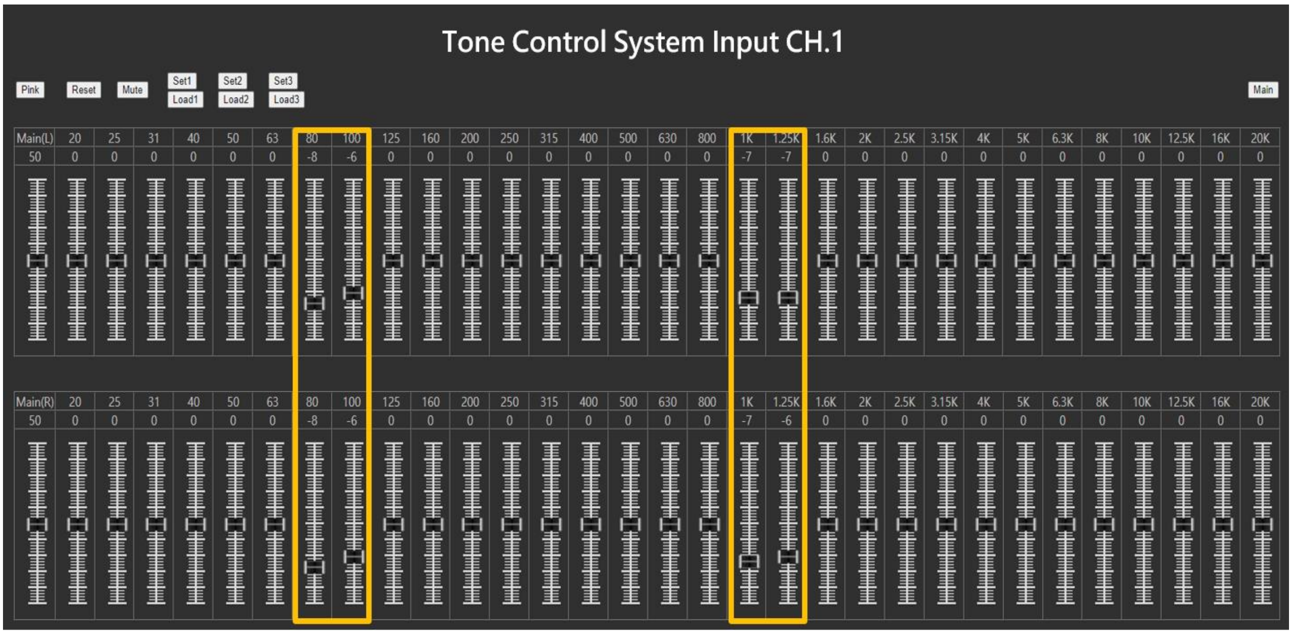


Figure 8. ARM Cortex M4 and ARM Cortex A7 data processing diagram

4. Implementation of Tone Control Module

Tone control module implementation through ADAU1710 is a chip designed to process audio signals. The ADAU1710 chip is capable of processing audio signals through audio signal block diagram design. To control the tone control module, firmware the audio signal data designed as a block diagram to the ADAU1710, and control the ADAU1710 through the ARM Cortex A7 of NAVICS and the ARM Cortex M4 of the tone control module using the web. Figure 9 shows the web-based 1/3 octave band graphic equalizer GUI. Users can adjust the graphic equalizer of the tone control module through a web browser. Figure 9(a) and 9(b) show the purpose of uniformizing the protruding frequency in Figure 5. Data controlled through the GUI of the web browser controls the ADAU1710 through the ARM Cortex A7 of NAVICS and the ARM Cortex M4 of the tone control module.



(a) **(b)**

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

Table 1 to Table 3 show the protocol for controlling ADAU1710 through GUI. Table 1 shows the basic protocol for equalizer control, allowing reset and mute by clicking a button. Table 2 is the protocol regarding the level of equalizer. The Main volume in Table 2 can adjust the house curve output level and range from 0 to 30. EQ_Left and EQ_Right in Table 2 are protocols for controlling the slide bar. Table 3 shows the protocol corresponding to each frequency band that can control the sound range. The level for each frequency can be adjusted from 0 to ±15 through 1/3 octave 31 band. Protocol data in Table 1 to Table 3 is firmware data in ADAU1710 and can be controlled directly through ARM Cortex M4.

Table 1. Equalizer basic protocol

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

Table 2. Equalizer level protocol

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

Table 3. Equalizer frequency protocol

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

Algorithm 1 shows the method for controlling ADAU1710 in ARM Cortex M4 in pseudo code. ARM Cortex M4 can receive data coming from NAVICS and equalizing through Algorithm 1. When ARM Cortex M4 receives “Reset” data, it initializes the parameter values for the volume and turns Mute On. When “Mute” data is received, the output of the equalizer is muted on or off depending on the parameters. “Volume” data is for controlling main volume and is controlled through parameters for channel and volume. “EQ” data controls ADAU1710 through parameters for channel, frequency, and value (-15 to +15). Tone control module implementation for tone control of Anchor's transmitting OAC is 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 "L":
                Control left master volume with value;
            Else if channel is "R"
                Control right master volume with value;
            Else return: // data is false

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

        Else:
            Pass
```

5. Conclusion

In this paper, we conducted research to apply the tone control function to each input device in the Anchor-based audio system configuration and came to the following conclusions. First, we implemented a tone control module to enable sound quality adjustment in Anchor technology, enabling a high-quality the audio system to be achieved. Second, NAVICS enables remote control of the tone control modules, enabling ubiquity without restrictions on time and place. The results of this study confirm that the tone control module enabled an audio system based on Anchor technology to be implemented as a high-quality audio 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 a tone control module 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 tone control module is complex, thus requiring simplification. Second, because adjusting the sound quality of the tone control module 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 audio system by implementing the tone control module capable of tone control. We hope that the Anchor-based audio system configuration is expected to be a future standard in the audio industry.

Acknowledgement

The present research has been conducted by the excellent researcher support project of Kwangwoon University in 2024. And, this work was supported by the Institute of Information & communications Technology Planning & Evaluation (IITP) grant funded by the Korea government (MSIT) (No. 2020-0-01846, Research and development of realistic content device technology).

References

[1] C. Y. Kang, S. J. Lee, K. Y. Jo, and S. H. Lee, "Audio Signal processing and System Design for improved intelligibility in Conference Room," *The Journal of the Internet, Broadcasting and Communication*, Vol. 17, Issue. 2, pp. 225–232, Apr. 2017. DOI: 10.7236/JIIBC.2017.17.2.225.

[2] J. He, and W. H. Zhang, "Digital audio broadcast over IP: Problems and solutions," *Wireless Communication and Sensor Network*, pp. 821–828, 2016. DOI: 10.1142/9789813140011_0096.

[3] S. D. Yang, and J. K. Choi, "Low-latency implementation of multi-channel in AoIP/UDP-based audio communication," *The journal of the institute of internet, broadcasting and communication*, Vol. 23, No. 3, pp. 59–64, Jun. 2023. DOI: 10.7236/JIIBC.2023.23.3.59.

[4] C. Chafe, S. Wilson, R. Leistikow, D. Chisholm, and G. Scavone, "A simplified approach to high quality music and sound over IP," *Proceedings of the COST G-6 Conference on Digital Audio Effects* (DAFX-00), Dec. 2000.

[5] M. S. Kang, S. W. Lee, and Y. S. Park, "Implementation of local distribution audio based on AoIP," *Journal of the Korea Institute of Information and Communication Engineering*, Vol. 12, Issue. 12, pp. 2165–2170, 2008. DOI: 10.6109/jkiice.2008.12.12.2165.

[6] J. H. Lee, J. W. Park, J. Y. Lee, S. C. Kwon, and S. H. Lee, "Implementation of independent audio mixing control using AoIP," *Applied Acoustics*, Vol. 163, Art. No. 107235, June, 2020. DOI: 10.1016/j.apacoust.2020.107235.

[7] J. H. Lee, H. J. Jeon, P. H. Choi, S.C. Kwon, and S. H. Lee, "Implementation of a Novel Audio Network Protocol," *ARCHIVES OF ACOUSTICS*, Vol. 43, No. 4, pp. 637–645, 2018. DOI: 10.24425/aoa.2018.125157.

[8] J. H. Lee, A. Hamacher, S. C. Kwon, and S. H. Lee, "System Integration for the Operation of Unmanned Audio Center based on AoIP," *The International Journal of Advanced Smart Convergence*, Vol. 6, No. 2, pp. 1–8, 2017. DOI: 10.7236/IJASC.2017.6.2.1.

[9] K. W. Kim, and J. P. Cho, "Development of digital/analog hybrid redundancy system for audio mixer," *The Journal of the Institute of Internet, Broadcasting and Communication*, Vol. 16, No. 5, pp. 63–68, Oct. 2016. DOI: 10.7236/JIIBC.2016.16.5.63.

[10] K. W. Kim and J. P. Cho, "The development of Intuitive User Interface and Control Software for Audio Mixer in Digital PA System," *Journal of Korea institute of information, electronics, and communication technology*, Vol. 11, No. 3, pp. 307–312, 2018. DOI: 10.17661/jkiiect.2018.11.3.307.

[11] J. H. Lee, S. C. Kwon, and S. H. Lee, "Implementation of on-site audio center based on AoIP," *International journal of advanced smart convergence*, Vol. 6, No. 2, pp. 51–58, 2017. DOI: 10.7236/IJASC.2017.6.2.51.

[12] S. W. Lee, S. C. Kwon, and S. H. Lee, "A study on the audio/video integrated control system based on network," *The International Journal of Advanced Smart Convergence*, Vol. 11, No. 4, pp. 1–9, 2022. DOI: 10.7236/IJASC.2022.11.4.1.

[13] S. W. Lee, S. C Kwon, and S. H. Lee, "Study on eliminating delay and noise in on-site audio center of Anchor technology," *IEEE Access*, Vol. 11, pp. 80081–80088, 2023. DOI: 10.1109/ACCESS.2023.3294561.

[14] J. S. Kim, and C. W. Song, "Development of integrated public address system for intelligent building," *Journal of Korean Institute of Intelligent Systems*, Vol. 21, No. 2, pp. 212–217, 2011. DOI: 10.5391/JKIIS.2011.21.2.212.

[15] S. K. Kim, M. Y. Lee, and C. H. Lee, "A Study on the Necessity of an Addressable Emergency Broadcasting System to Improve Evacuation Performance," *Korean Institute of Fire Science and Engineering*, Vol. 29, No. 5, 2015, pp.

14–22. DOI: 10.7731/KIFSE.2015.29.5.014.

[16] S. W. Lee, S. C. Kwon, and S. H. Lee, "Implementation of Public Address System Using Anchor Technology," *The International Journal of Advanced Smart Convergence*, Vol. 12, No. 3, 2023, pp. 1–12. DOI: 10.7236/IJASC.2023.12.3.1

[17] K. Eggenschwiler, "Lecture halls – Room acoustics and sound reinforcement," *Forum Acusticum, Budapest*, 2005.

[18] R. L. Bleidt, D. Sen, A. Niedermeier, B. Czelhan, S. Füg, S. Disch, J. Herre, J. Hilpert, M. Neuendorf, H. Fuchs, J. Issing, A. Murtaza, A. Kuntz, M. Kratschmer, F. Küch, R. Füg, B. Schubert, S. Dick, G. Fuchs, F. Schuh, E. Burdiel, N. Perers, and M. Y. Kim, "Development of the MPEG-H TV audio system for ATSC 3.0," *IEEE Transaction on Broadcasting*, Vol. 63, No. 1, Mar., pp. 202–236, 2017. DOI: 10.1109/TBC.2017.2661258.

[19] M. Karjalainen, and T. Paatero, "Equalization of loudspeaker and room responses using Kautz filters: direct least squares design," *EURASIP Journal on Advances in Signal Processing*, Vol. 2007, No. 1, pp. 1–13, 2006. DOI: 10.1155/2007/60949.

[20] Y. Zhu, S. W. Fan, L. Cao, K. Donda, and B. Assouar, "Acoustic meta-equalizer," *Phys. Rev. Applied*., Vol. 14, No. 1, pp. 1–11, 2020. DOI: 10.1103/PhysRevApplied.14.014038.

[21] P. Z. Kozlowski, "How to adjust room acoustics to multifunctional use at music venues," *Joint Conference – Acoustics*, Ustka, Poland, 2018, pp. 1–6. DOI: 10.1109/ACOUSTICS.2018.8502383.

[22] Y. K. Jang, M. W. Kang, and Y. K. Oh, "A study on the sound insulation and absorption design plan which affect the noise propagation from household appliances and facilities in a house using room acoustic simulation," *Journal of Korean Institute of Architectural Sustainable Environment and Building Systems*, Vol. 14, Issue. 6, pp. 767–778, 2020. DOI: 10.22696/jkiaebs.20200065.

[23] C. J. Park, and C. H. Haan, "Effects of the Complexity of 3D Modeling on the Acoustic Simulations and Auralized Sounds," *The Acoustical Society of Korea*, Vol. 30, No. 1, 2011, pp. 22–32. DOI: 10.7776/ASK.2011.30.1.022.

[24] Y. H. Kim, and J. O. Yeon, "Effect of acoustic calibration factors of measurement equipments on test results of floor impact noises," *Journal of Korean Institute of Architectural Sustainable Environment and Building Systems*, Vol. 16, No. 2, pp. 158–172, 2022. DOI: 10.22696/jkiaebs.20220014.

[25] J. H. Jeong, and S. C. Lee, "Comparison of sound pressure level and speech intelligibility of emergency broadcasting system at longitudinal corridor," *Fire Science and Engineering*, Vol. 32, No. 4, pp. 42–49, 2018. DOI: 10.7731/KIFSE.2018.32.4.042.

[26] F. M. Heuchel, E. F. Grande, F. T. Aferkvist, and E. Shabalina, "Active room compensation for sound reinforcement using sound field separation techniques," *J. Acoust. Soc. Am*., Vol. 143, No. 3, pp. 1346, 2018. DOI: 10.1121/1.5024903.