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Implementation of On-site Audio Center based on AoIP

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Abstract

Recently, rapid advances of Ethernet and IP technology have brought many changes in the sound industry. In addition, due to AoIP-based audio transmission technology, various problems of the acoustic system (sound quality deterioration due to long distance transmission, complicated wiring) have improved dramatically. However, when many distributed audio systems are connected with AoIP equipment, if there is a problem in the equipment, it is impossible to operate the connected system. AoIP equipment only can transmit audio signals but cannot adjust the system for acoustic environment. In this paper, AoIP equipment is to be installed with sound equipment on a one-to-one basis, so that various existing problems can be solved and adjustment of sound quality (reverberation, echo, delay and EQ) can be possible by AoIP-based OAC (On-site Audio Center) with built-in DSP function. As a result, uncompressed real-time transmission by distributed transmission/receipt module in OAC (On-site Audio Center) and high quality sound by adjustment of sound quality with built-in DSP can be acquired. It is expected that OAC based sound system will be the industry standard in ubiquitous environment.

Key words: *AoIP, DANTE, CobraNET, SR System*

1. Introduction

The industrial sound system includes a PA (public address) system for inducing announcements in the event of a fire or an emergency, and SR (sound reinforcement) system for performances and lectures [1]. The sound system is installed according to various purpose and environment. In case of long distance transmission, the sound quality is deteriorated by the loss of the line. Various problems related to the line have been solved by the advent of AoIP technology [2-4]. Sound transmission systems based on LAN cables include CobraNET, LiveWire, Dante, Q-LAN, and RAVENNA [3]. Especially, the installation cost of sound system has been decreased due to the configuration of AoIP-based sound system. In this paper, we examine the traditional sound technology based on AoIP, and propose a new OAC (on-site audio center) system. The proposed system is compared with the existing method. The composition of this paper is as follows. In

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Chapter 2, we analyze network-based audio transmission technology. In Chapter 3, we compare OAC and DANTE system. Chapter 4 explains OAC technology and Chapter 5 discusses conclusions and future challenges.

2. AoIP Technology

The International Standardization Organization has divided the network standard model into seven layers. The layer 1 is physical connection layer, the layer 2 is data link layer, the layer 3 is IP network layer and over the layer 4 is not related to audio transmission. CobraNET sends and receives signals on layer 2 and AoIP sends and receives signals on layer 3. In the early days of network-based sound transmission technology, AoE (Audio over Ethernet) has been studied mainly by the audio-related companies. Recently, various AoIP technologies have been applied to the protocol of its products.

2.1 CobraNet

CobraNet is a technology that enables real-time transmission of sound signals at the data link layer. It was developed by Cirrus Logic in the US in 1996 and is still in use today. Especially, it is possible to transmit 48 kHz, 20 bit, and 64 channels of sound signals at the same time. It is also robust to electrical interference, high frequency attenuation and acoustic signal degradation due to long distance transmission. Although CobraNet is installed for large-scale commercial facilities such as large stadiums, airports, theme parks, international conferences, and concert halls, there is a problem in that it requires payment of the license fee or purchase of interface equipment required for ADC and DAC [3].

2.2 AoIP

AoIP is a technology that transmits 44.1 khz CD sound quality based on IP network. AoIP is different from VoIP technology [5-6]. While VoIP is used for transmission of telephone quality signal, AoIP can be implemented using current network. The delay time of AoIP is less than 0.01 second. AoIP based uncompressed real-time audio transmission technologies are released in the market. Table 1 shows the status of AoIP technology.

Table 1. AoIP Technology

Technology	*Launch date*	*Synchronization scheme*	*Protocol*
Live-Wire	2004	IEEE1588-(Developing)	RTP
DANTE	2006	IEEE1588-2002	UDP
Q-LAN	2009	IEEE1588-2002	UDP
RAVENNA	Developing	IEEE1588-2002	RTP

2.3 DANTE

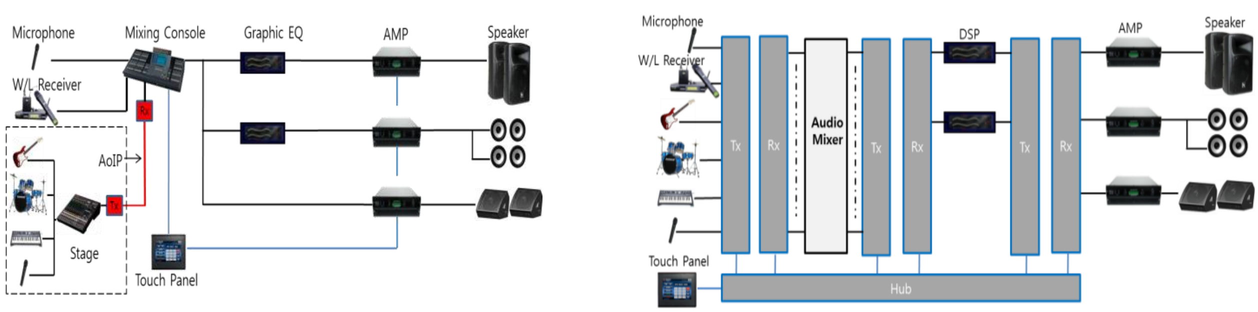
DANTE transmits sound signals over IP-based network. Developed by Australia's Audinate, it is used in the audio industry because of its transmission speed, number of transmission channels, and ease of operation, flexibility and scalability. It can also send and receive audio signal from other Dante devices by local network using internet standard protocols (TCP/IP, UDP/IP, etc). Dante is capable of transmitting and receiving Gigabit 24-bit 48kHz data in 512×512 channels and 24-bit 96 kHz data in 256×256 channels. In addition, 100Mbps 24bit, 48kHz data in 48×48 channels and 24bit 96KHz data in 24x24 channels are possible. Dante has the advantage of running on an IP network. However, this presents a problem with sound loss in complex networks. Nevertheless, it is regarded as the best system for single connection between the audio mixer in the sound room and the system on the stage [7-8].

3. Compared with OAC and DANTE

The SR system consists of a sound source, an audio mixer, a tone correction device, an amplifier, and a speaker system. All components except the power amplifier and speaker connection section transmit signals based on the audio cable. AoIP can be applied for audio cable section and high quality data can be transmitted in real time.

3.1 Dante-based sound system configuration

DANTE sends and receives audio signals based on the OSI Layer 4 IP network. In addition, man y audio companies use DANTE protocol for their products due to transmission speed, multi-channel transmission and scalability. Especially, as shown in Figure 1 (a), sound transmission between the main mixer in sound room and the sound system in the stage makes optimized acoustic environment. Figure 1 (b) shows a diagram of sound system consisting of all the equipment connected to the DANTE system. D ANTE receives and transmits signals not on a one-to-one basis but on two or more channels. Some of DANTE's products (Audio mixer, DSP, Amp) include network function and do not require separate transceiver equipment. DANTE transmits audio signals, controls network connection, and monitor status and signal delay. However, it does not control the gain control or house curve of the transmission si gnal.



(a) (b)

Figure 1. Sound system diagram based DANTE
(a) Transmission of several sections on multiple channels
(b) System configuration based on Tx/Rx equipment

3.2 OAC-based sound system configuration

The acoustic system based on OAC sends and receives sound by one-to-one connection with each acoustic equipment. It also broadcasts PCM data using UDP protocol and can receive signal at a specific point or at multiple points. OAC is compact, lightweight, easy to install and equipped with DSP function. The Reverberator, Delay, Graphic Equalizer and Pink noise functions of the DSP make it possible to set and adjust the sound corresponding to the change of the room acoustic environment. In addition, I/O signals of all devices can be monitored and managed at desired points [10-11].

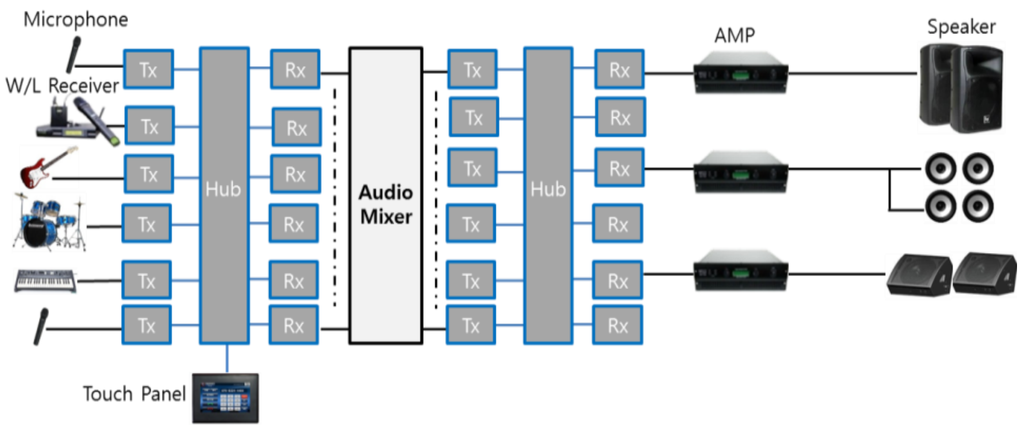


Figure 2. Sound system diagram based OAC

OAC and Dante are AoIP-based technologies. However, the operating program, control and DSP functions are different. Dante system has features for transmitting audio signals and performs network control and status monitoring associated with it. However, OAC can provide user with optimum sound environment because OAC control I/O sound and sound environment by DSP [12]. Table 2 shows Dante and OAC comparisons.

Table 2. DANTE vs. OAC

Feature	*DANTE*	*OAC*	*Remark*
OS	Windows, Mac	Windows, Mac, Linux, Android	
Control	IP connectivity and state control	IP connectivity, state control and Audio gain control	
DSP	X	O	

4. Implementation of OAC Technology

OAC (On-site audio center) is a compact system for real-time transmission and reception of audio signals without compression. The input level deviation of sound source is divided into Microphone input and Line input. In addition, OAC can control sound equipment and surrounding environment using serial communication. The input sound signal is digitized and modified to the desired sound, and it is transmitted to the designated IP through Ethernet. The signal transmitted by the LAN cable receives data through the Ethernet of the receiving device. The DAC chip processed the notes so that the adjusted data could be converted into audio signals. The transmitted audio data can send on a point-to-point basis or one-to-many. The input signal is processed by the CPU and is sent by Ethernet. At the remote site, efficient management and operation of the acoustic system can be achieved.

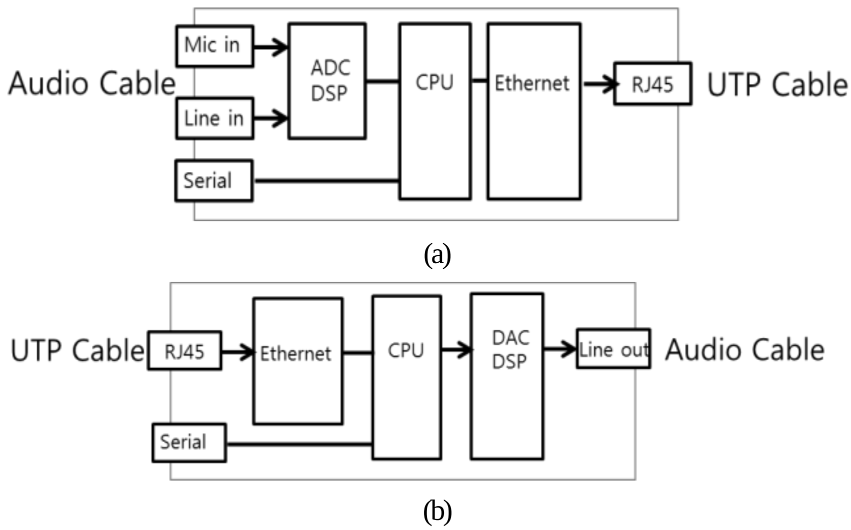


Figure 3. OAC Block diagram
(a) Transmitting device (b) Receiving device

The sound system installed by AoIP based OAC or Dante system can't guarantee the sound quality. Sound is much affected by various environments. Especially seasonal factors such as sound and temperature and humidity depend on the characteristics of the building. Therefore, setting the sound in general requires much effort and time. There is a function to adjust each frequency gain in DSP of OAC. Using this technology, optimum acoustic environment can be realized by automatically adjusting the sound setting suitable for the room environment.

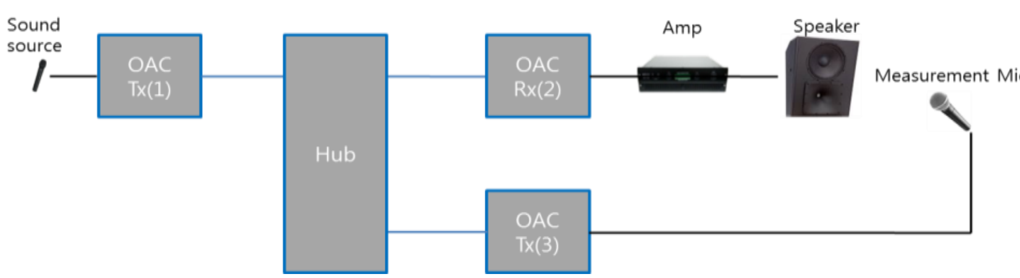


Figure 4. House Curve Setting Based OAC

Figure 4 shows the system configuration for automatic sound setting. The DSP of OAC Tx (1) generates Pink noise, and then the sound from speaker is picked up by the measurement microphone by setting the graphic EQ of OAC Rx (2) to the OAC Tx (3) calculated data. The Sound setting is intended to automatically adjust the House Curve to match the room characteristics. The measuring microphone or House Mic picks up the pink noise. It is possible to automatically adjust to the average value only for frequencies out of 3dB from the average of each frequency level. The automatic adjustment of the sound by the DSP enables the audio frequency 20 to 20 kHz to be set based on the 31 frequency of the 1/3 octave band. The algorithm for this is as follows. Frequencies (20, 25, 31, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1k, 1.25k, 1.6k, 2k, 2.5k, 3.15k, 4k, 5k, 6.3k, 8k, 10k, 12.5k, 16k, 20k) was averaged in Equation (1). OAC Tx (3) samples Pink Noise Data 3 times and applies the average value to the EQ of OAC Rx (2). Sampling has a time interval. The sampling interval may be different depending on the construction environment, but the sampling time interval is 5 seconds. Equation (1) (2) (3) is the first sampling, second sampling, and third sampling data. It is the dB value of 31 frequencies.

$$\mathrm{Av}(1) = \frac{1}{31}\sum_{k=1}^{31} \mathrm{dB1(k)} \tag{1}$$

$$\mathrm{Av}(2) = \frac{1}{31}\sum_{k=1}^{31} \mathrm{dB2(k)} \tag{2}$$

$$\mathrm{Av}(3) = \frac{1}{31}\sum_{k=1}^{31} \mathrm{dB3(k)} \tag{3}$$

Equation (4) is the average value of dB for each frequency of the 1st, 2nd and 3rd order. This value is the reference level for adjusting the Graphic EQ.

$$\mathrm{Avf} = \frac{1}{3}\sum_{k=1}^{3} \mathrm{Av(k)} \tag{4}$$

Figure 5 shows a method for reducing the highest frequency level of each frequency by 3, based on the Avf value of Equation (4), so that it can be within ± 3 of the average level.

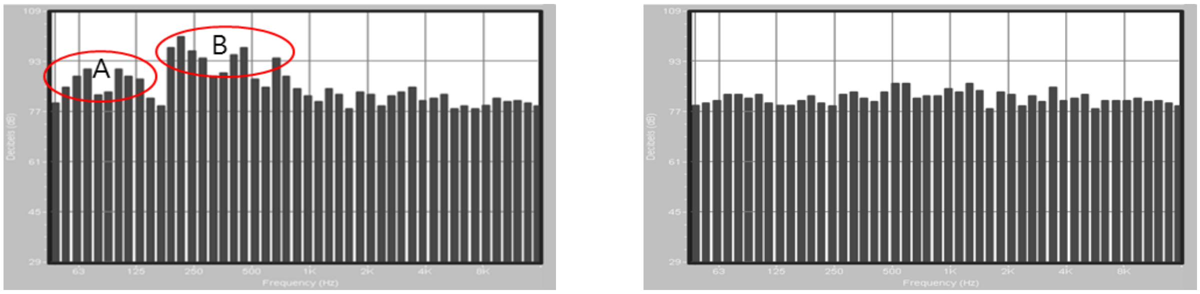
```
eq_Val=House Curve Value
Void a()
{
val = 0

For f=1 to 31
    Avp(f) = 1/3 dB1(f) + dB2(f) + dB3(f)]
    if  Avp(f) > val :
         val = Avp(f)
         i = f
     Avp(i)  = val
}

If ( |Avf – Avp(i)| > 3 ) :
         eq_Val = Avp(i) – 3
         a()
else ( |Avf – Avp(i)| =< 3 ) :
         pass
```

Figure 5. Coding for House Curve

Eq_Val is the graphical EQ adjustment value of the OAC Rx (2) DSP in Figure 6. This algorithm makes it easy to adjust the sound according to the environment change.



(a) (b)

Figure 6. Spectrum analysis using pink noise
(a) Before DSP application (b) After DSP application

Figure 6 (a) shows the analysis of the data picked up through the measurement microphone by Spectrum Analysis. Peaks and valleys are formed at points A and B, so you hear the original sound and other distorted sounds at the middle bass. Figure 6 (b) shows the result of automatic data setting by OAC DSP through the above algorithm.

5. Conclusion

The results of this study are summarized as follows. First, the proposed system avoids wiring of equipment by audio cable. In addition, the OAC can be made compact and lightweight so that it can send and receive acoustic signals based on the LAN cable. Second, OAC has built-in audio DSP to adjust and set the sound signal for each channel, so that sound suited to room acoustic conditions can be ensured. Third, I / O signals can be operated and operated remotely or locally, providing convenient operation. The results of this study will be able to reduce the economic loss due to the installation and management of the acoustic system and to minimize the human loss due to the operation and management.

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