Sound-Wave Transmission System in Mobile Device

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Abstract. Based on equipments of speaker and microphone in mobile device, we develop a sound wave transmission system. The digital data is encoded in the analogy sound wave, transmitted by speaker, received by microphone, and decoded by the mobile device to derive the original data. For enhance the accuracy, the functionalities of error correction and adaptive frequency response are consideration in the system. The throughput of the system is 72 bps with accuracy rate 90% under transmission distance 100 cm.

Keywords: Sound-wave transmission system · Fast Fourier Transform Hamming code

1 Introduction

With the advance of VLSI design and communication technique, the smart phone and tablet PC are widely accepted as the major platform for mobile communication. Up to now, the major communication interfaces in mobile devices includes Bluetooth, WiFi, and NFC. The NFC only supports close communication. It needs pairing operations in Bluetooth to make a connection before data communication. In the WiFi, the IP address or some of setup operations for the Direct WiFi is required. These pairing or set up operations are difficult for general users. For providing a friendly user interface in mobile device communication, this paper develops a sound-wave transmission system based on the general speaker and microphone. If we want to start a communication between mobile devices, we can first send the pairing/setup information or IP address via the acoustics transmission system to the peer mobile device. Once the peer mobile device receives the configuration information, based on the received information, a connection between mobile devices via Bluetooth or WiFi interface can be established automatically for further data communication.

The sound-wave transmission system can be classified into digital audio transmission and analog audio transmission, as shown in Fig. 1. Most of the literatures target on the digital audio signal processing for embedding a copyright watermarking

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[3, 4]. Most of researches in analog audio communication focus on the topics of underwater communication. The properties of underwater transceiver are very different from general speaker and microphone. Considering the human audible frequency, the frequency response of speaker and microphone is generally designed to less than 20 kHz [1]. Xiang et al. [2] consider the analog audio communication, such Fig. 1(a) depicts. In [2], it focuses on study the effect of AD/DA equipments. Thus, the wire-line is used to connect the line-out of transmission side to the line-in of receiving side for excluding the sound-wave distortion in the air communication.

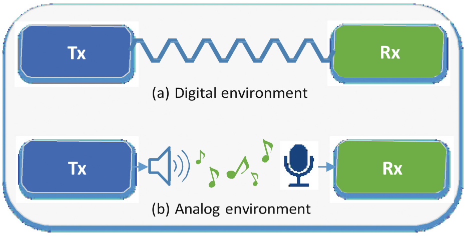

Fig. 1. The concept of sound-wave transmission system

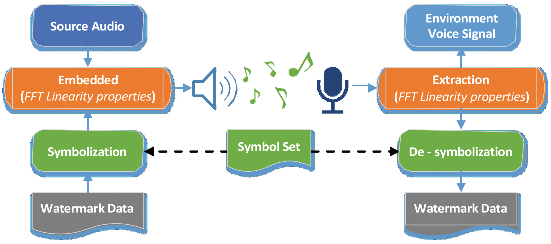

Fig. 2. Sound-wave transmission system architecture

Difference from the [2] consideration, we target on the sound-wave transmission system design based on the speaker/microphone transceiver, as Fig. 1(b) depicts.

2 The Design of Sound-Wave Transmission System

Figure 2 illustrates the sound-wave transmission system architecture. The digital data is embedded in the background audio under frequency domain. An inverse Fast Fourier Transform (IFFT) converts the frequency domain sound wave to the time domain and playout to the speaker. In the receiver side, the sound wave is received by the microphone and digitalized by the A/D converter. The digital sound waves are converted to the frequency domain by FFT for extracting the transmitted digital data.

2.1 Protocol Stack

For reducing the interference, we choose the frequencies of 17 kHz, 18 kHz, and 19 kHz as the elementary frequencies. Based on these three frequencies, we design four type of symbols, called *Bit0* symbol, *Bit1* symbol, *Gain* symbol, and *Guard* symbol. Figure 3 illustrates the protocol stacks of the proposed sound-wave transmission system. Herein, the application layer generates the transmission data called watermarking data which relies sound-wave to transmit. The data link layer composes the application data with *Sync-code* as data link frame. The purpose of *Sync-code* is used for frame delimitation. The pattern of *Sync-code* is binary pattern 0110. At the same time, the data link layer implements hamming code in application data for 1-bit error correction. The codec function of physical layer converts the analogy sound-wave to digital binary bits. Due to the frequency response in speaker/microphone is difference. This means the attenuation a sound frequency may difference in different speaker/microphone. Figure 3 depicts the attenuation of the three elementary frequencies against to distance. Thus, we design a *Gain* symbol which consists all of sound frequencies used in the system for amplitude adaption in receiving side. In the transmission side, the *Gain* symbol is transmitted with the fix power. In the receiving side, based on the receiving power of *Gain* symbol, we understand the attenuation rate of the used speaker/microphone pair. Therefore, depending on the attenuation rate of the *Gain* symbol, the receiver compensates the attenuation in the following *sync-code* and *data symbol*. The *Guard* symbol uses for avoiding the interference between *Gain* symbol and *Sync-code* (Fig. 4).

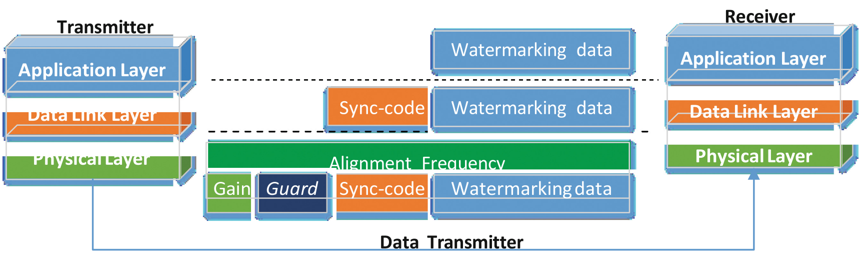

Fig. 3. The protocol stack of sound-wave transmission system

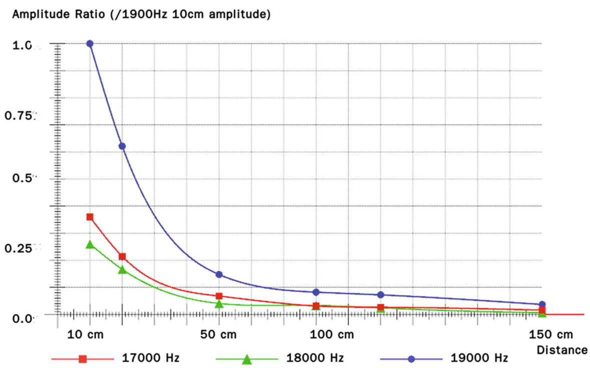

Fig. 4. The relationship between attenuation and distance

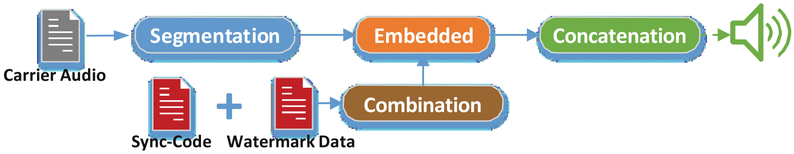

Fig. 5. The transmission of data link layer

2.2 System Implementation

Figure 5 shows the transmission of data link layer. First the *Sync-code* denoted as D_{sync} precedes the application data denoted as D_{app} to form the data sequence denoted as $\boldsymbol{D}$. The data sequence D is represented in Eq. (1).

$$D = (D_{sync} \ll Len(D_{app}) \,|\, H_c(D_{app}). \tag{1}$$

Where the functions of $\boldsymbol{Len(D_{app})}$ and $\boldsymbol{Hc(D_{app})}$ represent the length of D_{app} and the D_{app} with 1-bit error correction hamming code, respectively.

The carrier audio (background sound) in Fig. 5 denoted as $\boldsymbol{A_{sound}}$, is divided into frames denoted as $\boldsymbol{x_i}$ which represents the i'th frame. Each frame have $\boldsymbol{s}$ samples which must be the 2 power for FFT operations. In this implementation the $\boldsymbol{s}$ is 512.

The data sequence D must be embedded into carrier audio $\boldsymbol{A_{sound}}$. Each frame of $\boldsymbol{A_{sound}}$ is embedded one symbol. The embedded processes is represented in Eq. (2).

$$F(SoundWave) = \begin{cases} F(x_i) + F(GainSymbol), & if\ i = 0, \\ F(x_i) + F(GuardSymbol), & if\ i = 1, \\ F(x_i) + F(Bit0Symbol), & if\ \mathrm{D}[i-2] = 0, \\ F(x_i) + F(Bit1Symbol), & if\ \mathrm{D}[i-2] = 1, \end{cases} \tag{2}$$

Where the function $\boldsymbol{F(x)}$ mean the FFT operations of x. Finally, the frequency domain embedded sound frame $\boldsymbol{F(SoundWave)}$ is inverted to the time domain sound wave via inverse FFT and playout to the speaker.

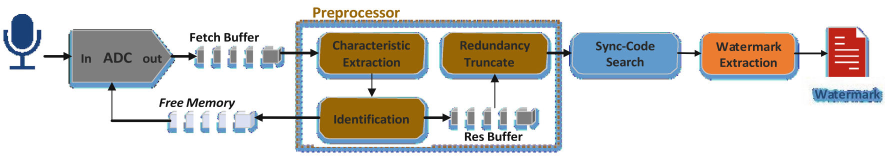

Fig. 6. The receiver of data link layer

The receiving process of the acoustics Transmission System is shown in Fig. 6. The sound wave signal is sensing by the microphone and digitalizing by ADC. The time domain sound wave data is converted to the frequency domain by FFT. In the pre-processor phase, we only find the basic frequency to identify whether the sound wave has embedded the digital data. If we found the basic frequency in preprocessor phase, we further do the sync-code search for delimitating the starting point of the digital data and extract the digital data.

3 Performance Evaluation

To evaluate the performance of sound-wave transmission system, an experimental model is proposed as Fig. 7 illustrates. The device noise of microphone and A/D converter is measured and built these noise model in Fig. 7. A de-noise model is applied to cancel the noise effect of microphone and A/D converter. The $\boldsymbol{DB_{env}}$ records four type environment background sounds which include road, peaceful, restaurant, and air condition. The spectrum of background sounds is shown in Fig. 8. We use HTC Bufferfly S as receiver and HTC Desire A818 as transmitter to measurement the performance. Table 1 shows the associated implementation parameters. Based on Table 1 parameter, the transmission rate of the system is 72.53 bps. The transmission accuracy is evaluated in the following.

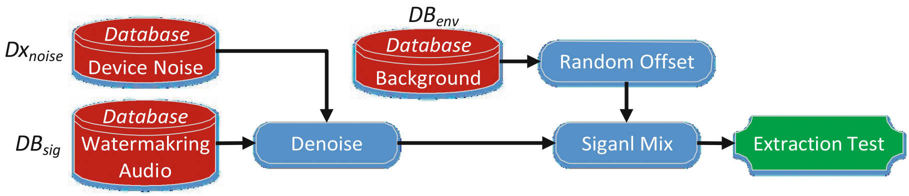

Fig. 7. The experimental model for performance evaluation

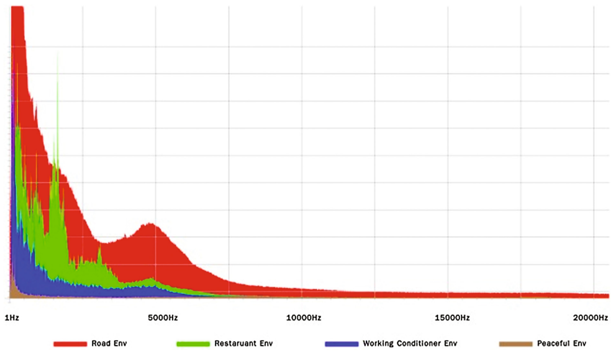

Fig. 8. The frequency spectrum of background sound

Table 1. The implementation parameters.

Parameters	Value
Element frequency	17 kHz 18 kHz 19 kHz
Sample/frame	512 samples
Sample rate	44100 Hz
Sync-code size	16 bits
Sync-code size	16 bits
D_{app}	96 bits

The comparison of average transmission accuracy under different distance and scheme is shown in Fig. 9. As Fig. 9 depicts, when the transmission distance increases, the scheme with hamming code and Gain symbol keep a better results than others. Under the transmission distance 100 cm, the scheme with hamming code and Gain symbol has 40% increasing than without any scheme.

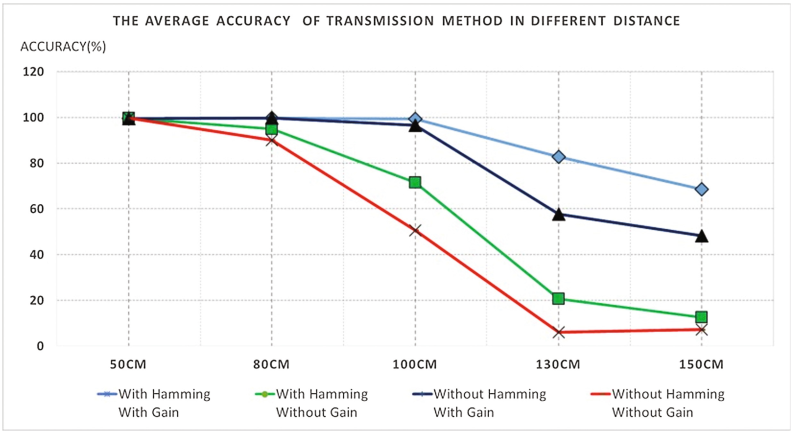

Fig. 9. Compare the average accuracy under difference distance and scheme

4 Conclusions

In this paper, based on the basic speaker and microphone, a analog sound-wave transmission system is design and implementation in mobile device. For increasing the transmission accuracy rate, a 1-bite error correction hamming code is applied in data link layer. To overcome the different frequency response of speaker/microphone, the physical layer adds Gain symbol to compensate sound wave attenuation of speaker/microphone. The performance measurement results reveal that the throughput of the system is about 72 bps. The average transmission accuracy in different environment can up to 90% under transmission distance 100 cm.

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