Application of Photoelectric Conversion to Dentary Bone Conduction Device Using Analog Signals

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Abstract

It is attempt that uses dental bones to receive audio signals are rare in the past. Therefore, the main contribution of this article is to understand and analyze the audio signal through the teeth. Our homemade laser pointer has a wavelength of 630-650 nm and a maximum output of 5 milliwatts. It will light up when the smartphone's music starts playing at a music frequency that matches the light frequency. The frequency signal of the light received by the solar panel is converted into an electrical analog signal, and then output to the DC motor. The motor shaft will not rotate under a small current, but will only slightly vibrate according to the analog frequency of the current. By biting the vibration axis with teeth, a person can transmit audio to the ossicles (ie malleus, incus and bone) through the dental bones. This device can enable people with congenital or acquired hearing impairment to access external audio.

Keywords: dentary bone conduction; photoelectric conversion; auditory ossicles

Introduction

In general, the auditory system involves two means of sound transmission. The first uses air as the transmission medium. In the second process, sound is transmitted through the bones. The auditory path begins with bone vibration and passes through the inner ear via the auditory nerve to the auditory center, without passing through the outer and middle ear. Sound is thus transmitted directly to the cochlea in the inner ear through the bone. The major function of this auditory path is to cause the cochlear wall to vibrate, allowing the inner ear sensors to receive the signal, which is then transmitted through the auditory nerve to the auditory center in the brain, which generates an auditory sensation. Without passing through the auditory structures before the oval window, bone conduction directly transmits sound through bone vibration. In this case, an auditory sensation travels by the following path: skull → cochlea → auditory nerve → brain. People with hearing loss who have a normal auditory nerve can still utilize the bone conduction path to receive an auditory sensation [1-2]. The signal converter receives the ambient sound signal, converting it into a corresponding electrical signal. A vibrating sheet is then driven by the electrical signal to generate a vibration. Users would simply press the vibrating sheet against the bones in their middle ear area to utilize the bone as a medium to transmit the vibration frequency of the vibrating sheet to the user's cochlear structure and auditory nerve, enabling perception of the sound received by the signal converter.

Materials and Methods

The motor shaft does not rotate under the effect of a small current, but only vibrates slightly according to the magnitude of current's analog frequency. In this study, the proposed device for transmitting sound to the auditory ossicles utilizes dentary bone conduction [3]. The device can receive an optical signal that carries auditory information, but only allows the auditory information within the signal to be received, thus improving transmission accuracy. In addition, because the proposed device [4-5] receives an optical rather than an audio signal, the defect of signal attenuation caused by distance is substantially reduced, thereby improving usability.

A. Apparatus.

The power amplifier is a low-voltage LM386 chip produced by National Semiconductor, which has pin functions (Fig 1). Such chips are usually used in low-voltage consumer products to minimize peripheral chip components. The voltage gain of the LM386 chip is usually set to 20. However, by adding an external resistor and a capacitor between pins 1 and 8, the voltage gain can be arbitrarily adjusted to a maximum of 200 (Fig 2). The power amplifier can increase the output power of the signal. It obtains the energy source through the power supply to control the waveform of the output signal to be consistent with the input signal, but with a larger amplitude. In this way, the amplifier circuit can also be regarded as an adjustable output power source to obtain a stronger output signal than the input signal.

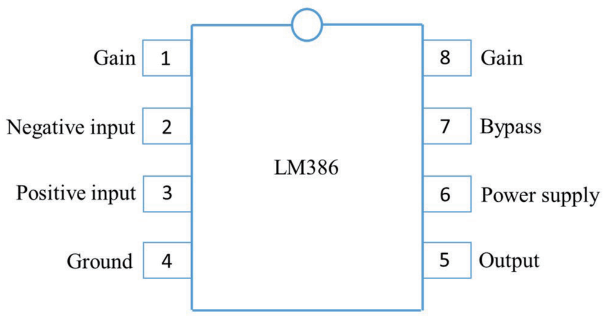

Fig. 1 LM386 Chip function pin.

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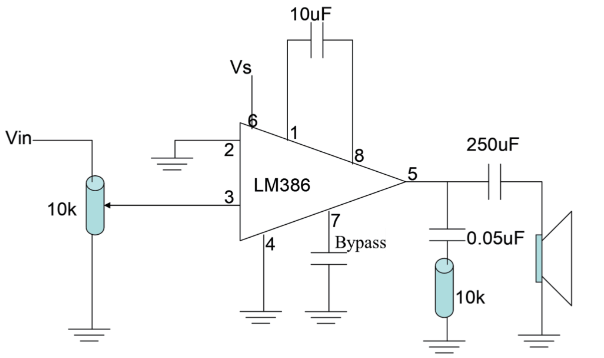

Fig. 2 Circuit diagram with a gain of two hundred.

B. Magnetic field measurement

Fig. 3 illustrates the magnetic field system framework. The impedance spectrum is measured through the following process. The signal generator produces a frequency; one end of the wire of the signal generator is connected to the vibrator (coil), with the other end connected to an ammeter for voltage measurement. The oscilloscope receives the voltage signal generated by the vibrator and measures the voltage. A magnetic field sensor placed on the section of the vibrator coil measures the magnetic field variation. Voltage and current values are input at various frequencies to determine the output impedance. Regarding additional input frequency, V1 and V2 are the voltage values measured by the oscilloscope, and A is the current value measured by the volt-ohm-milliammeter. Resistivity (R) can be derived using a conversion formula. The audio signal is first output from the signal generator, and the signal power is increased by an amplifier. The signal generator is cascaded to an alternating current (AC) meter and then connected to the vibrator.

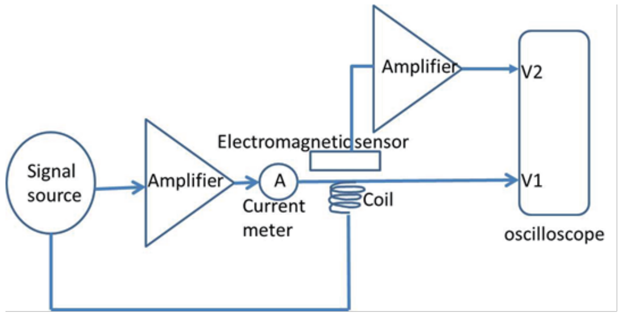

Fig. 3 Magnetic field measurement system

C. Vibration measurement

Fig. 4 illustrates the vibration measurement system. The vibration measurement process is as follows. The signal generator produces a frequency signal, which is augmented by an amplifier. One end of the wire of the signal generator is connected to the vibrator and the other end to an ammeter for current measurement. The vibration circuit is detected using a laser, which projects a beam onto the test object. When the system vibrates, the laser detection vibration circuit receives the reflected light and converts the vibration signal into an electrical one. The oscilloscope receives the voltage signal generated by the vibrator and displays its value on the screen. A tissue phantom is placed on the section of the vibrator coil to determine variations in the vibration at different hardnesses. The voltage and current values are entered at different frequencies to measure the vibration magnitude. The audio signal is first output by the signal generator, and its power is increased by an amplifier. The signal generator is cascaded to an AC current meter and then connected to the vibrator.

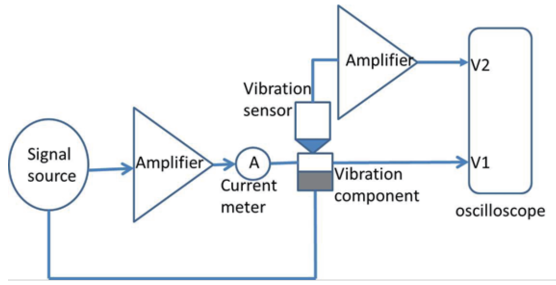

Fig. 4 Vibration measurement system

D. Spectrum measurement for inductive reactance

Fig. 5 illustrates the spectrum measurement system for inductive reactance, which uses a spectrum analyzer to generate signals of various frequencies. These signals are input to the vibrating earphone and connected to pin 2 of the operational amplifier, with a resistance (R1) of 10 Ω. Pin 6 of the operational amplifier is for output voltage; pins 7 and 4 are connected to V+ and V-, respectively; and pin 3 is the grounding pin. When the spectrum analyzer generates signals at various frequencies and inputs them to the earphone, the coils generate inductive reactance due to current conversion, with high frequency generating high inductive reactance. Finally, the output signal is sent to the spectrum analyzer to detect changes in earphone impedance and output gain at different frequencies. The frequency of this architecture ranges from 20 Hz to 20k Hz, with 1000 sample points, 1V input voltage, and Swept Sine as the measurement module. The architecture features a wide bandwidth (ranging from 20 Hz to 110 MHz) and a large input impedance range; the output signal features reverse gain.

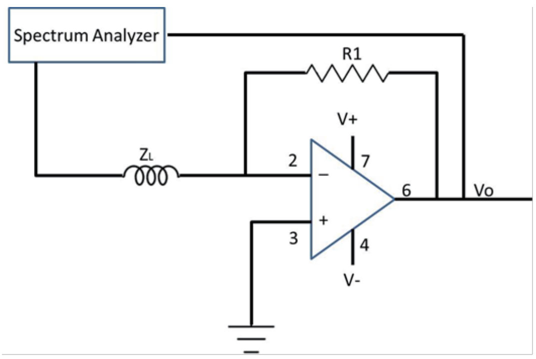

Fig. 5 Frequency spectrum measurement system for inductive reactance.

E. Magnetic distance measurement

A cobalt–nickel magnet with a diameter of 10 mm and thickness of 2 mm was used to measure magnetic force (Fig. 6), together with an electronic scale to detect the magnetic value. The magnetic distance measurement system is made of acrylic, with a length of 15 cm, width of 16 cm, and height of 13 cm. A cylindrical tube is inserted through 0.5-cm hole in the center of the top plate to finely adjust the distance, and a square flat plate connected under this tube serves to attach the magnet to the plate. The bottom magnet is glued to the weight indicator to detect the magnitude of force on the double magnet. The upper part of the cylindrical tube is connected to an acrylic

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plate fixed to the fine adjustment base as a support rod for adjusting the distance between the double magnets.

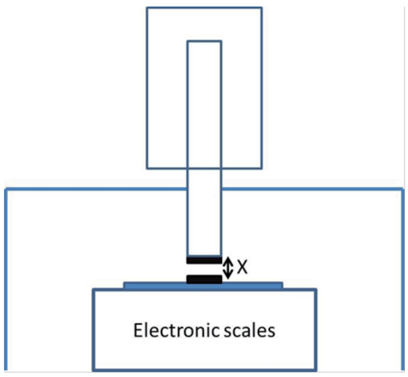

Fig. 6. Magnetic distance measurement system

Results

A. Speech analysis

Before feature extraction, audio signals undergo preprocessing through an analog-to-digital converter, using software to easily and accurately obtain their characteristics. After the speech frame is determined through normalization, the initial and final positions of the sound are estimated using the zero crossing rate (ZCR) of the time domain endpoint detection algorithm. Subsequently, effective frames are extracted, and feature extraction is performed using Fast Fourier transform (FFT). Fig. 7 presents the voice signal processing procedure.

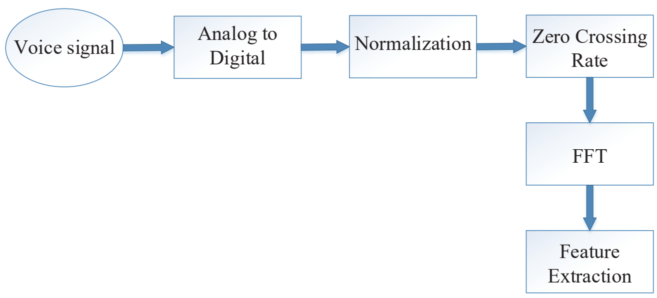

Fig. 7 Flowchart of voice signal processing

Firstly, the analog signal of speech is converted into a digital signal using the Audacity software program. The sample points obtained per second are expressed in Hz, and this is called the sample rate (fs). Common sample rates include 8 kHz (telephone), 16 kHz (general speech recognition), and 44.1 kHz (CD player). A voice-receiving channel can be divided into mono and stereo. A mono system is similar to a point source, which only replays the original sound but cannot represent its directionality and spatial sense, whereas the stereo system uses linear noise composed of both left and right channels to represent horizontal sound in its directionality and spatial sense. The different sound volume of each person generates a distinctive amplitude. To unify the amplitudes for subsequent signal processing, they are normalized as shown in (1).

$$P_{max} = max\left[P(n)\right]$$
$$y(n) = \frac{P(n)}{P_{max}} \qquad (1)$$

Zero crossing is defined as the number of times the amplitude of a signal crosses the zero level. The aspiration or friction sound in human speech requires low energy but features higher frequency, shorter cycle length, and higher ZCR than sound in general. These features can assist in identifying the location of aspiration or friction sound sections in the audio signal. This can be expressed in the following equation:

$$Z(n) = \frac{1}{N}\sum_{k=-N/2}^{N/2} s\left|sgn(x(c+k)\times x(c+k-1))\right| \qquad (2)$$

In (2),

$$s(x) = \begin{cases} 1, x \geq 0 \\ 0, x < 0 \end{cases} ; \quad sgn(x) = \begin{cases} 1\ , x \geq 0 \\ -1, x < 0 \end{cases}$$

where *Z (n)* is the ZCR of the *n*th frame; *s(x)* denotes the step function; *sgn(x)* represents sign function; *x(…)* is the amplitude information of the signal; *c* denotes the center of the frame; and *N* is the frame width, generally 256 or 512. In (2), when two adjacent signal samples of the same sign multiply, the result must be positive; conversely, when their signs are opposite, the result must be negative. For every frame, the frequency at which each sample point crosses zero is calculated. (Fig. 8).

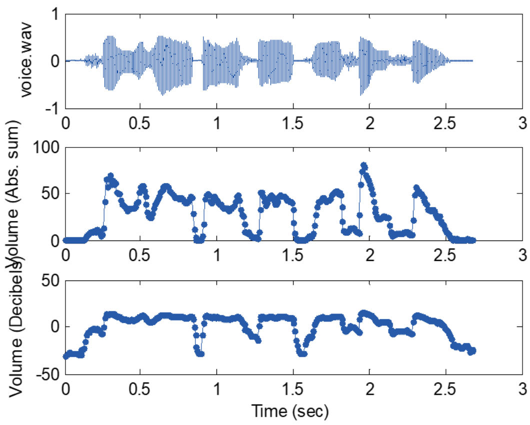

Fig. 8 Sampled voice signals after ZCR calculation.

B. Feature extraction

Signal features are troublesome to be cognized by perceiving changes of the scope value of an audio signal in the time domain (Fig. 9). Transforming the audio signal into a spectrogram (Fig. 10) permits sonic features to be identified. Since this procedure seems fingerprint contrast, it is also familiar as voiceprint recognition. The scope value of an audio signal in the time domain is often transformed to an energy allocation in the frequency domain for inspection. Various energy allocations in the frequency domain imply different speech features. A signal in the time domain transforms rapidly and constantly, guiding to inexact observation. Thence the most frequently used skill is transposition of the audio signal from the time to the frequency domain, accordingly define the spectral features of various sounds through their energy allocation. The spectrum is a expression of a time domain signal in the frequency domain and can be acquired by performing FFT on the signal. The result is expressed as a spectrogram, with the amplitude or phase as the vertical axis and the frequency as the horizontal axis (Fig. 11).

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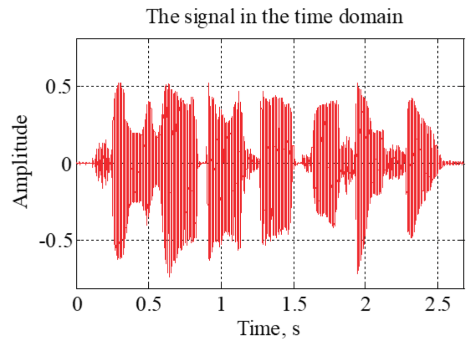

Fig. 9 A Scope value of audio signal, where fs (44100) is the sampled frequency

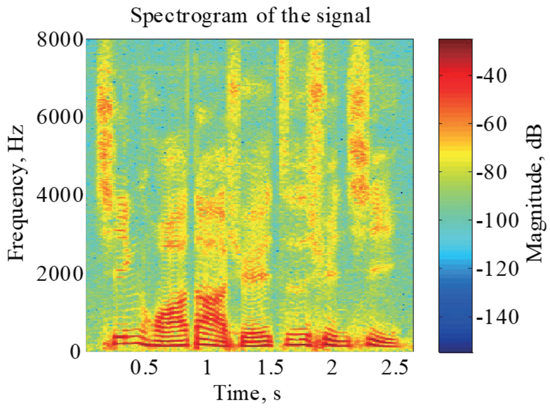

Fig. 10 Audio signal spectrogram.

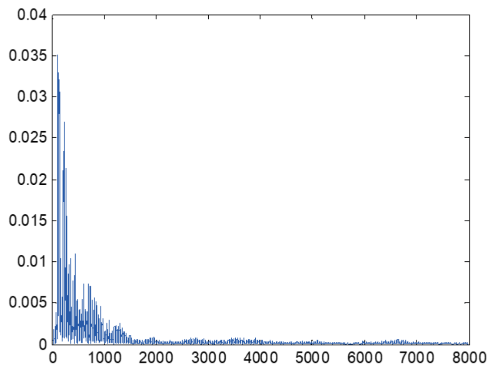

Fig. 11 Audio signal spectrum.

C. Filter Design

Fig. 12a describes the original signal of the audio. A low-pass filter (LPF) is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The exact frequency response of the filter depends on the filter design. The filter is sometimes called a high-cut filter, or treble-cut filter in audio applications. A low-pass filter is the complement of a high-pass filter. The low-pass filter [6] functions to pass the low-frequency signal and weaken the high-frequency signal, which is appropriate for high-frequency noise as shown in Fig 12b. For example, if the sensors have low frequency, the low-pass filter can be used to eliminate electrical noise [11].

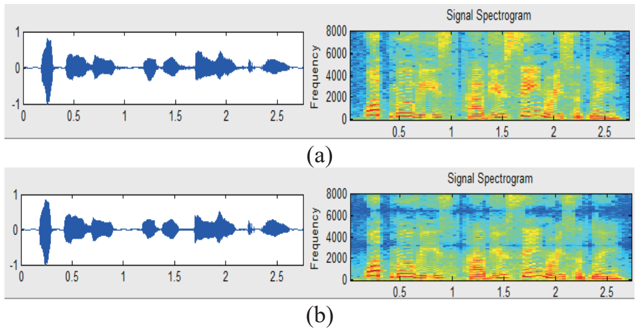

Fig. 12 (a) Original audio. (b) The output signal passes through a low-pass filter.

Discussion

The objective of this study was to enable patients with congenital hearing loss or acquired tympanic membrane perforation to receive music therapy for psychological healing. Although the theoretical basis and experimental design have been satisfactorily developed (Fig. 13), the proposed design still lacks funding for its realization. The experimental work has not been fully commercialized, but is expected to be adopted by skilled enterprises through a technology transfer to improve the design and turn it into a final product (Fig. 14) that benefits people with hearing loss who have the need for psychological rehabilitation. This study aims at assisting people with hearing loss in regaining their auditory sense by designing audio transmission devices such as hearing aids, which is a simple and common method. Various hearing aids have been designed for people with different hearing loss conditions. Those with an intact middle ear structure can use general air conduction hearing aids, whereas those whose middle ear structure prevents this must use bone conduction hearing devices to perceive sound in their surroundings through vibrations of the auditory ossicles.

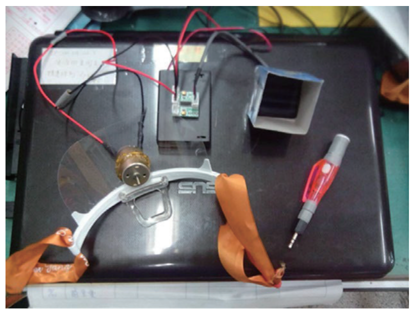

Fig. 13 Design of the experimental dentary bone conduction device.

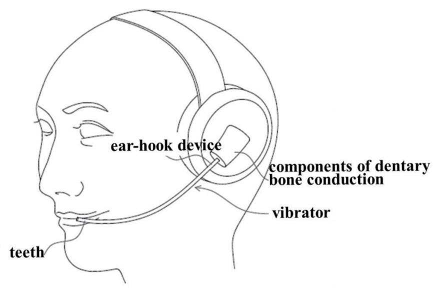

Fig. 14 Final dentary bone conduction device.

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