

Dynamic Spectrum Recognition-Based Matrix Ultrasonic Shielding System: Design and Implementation

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Abstract: With the growing demand for privacy protection in public spaces, traditional acoustic shielding technologies face issues such as limited functionality and restricted shielding range. This study aims to develop an intelligent dynamic acoustic shielding system capable of distinguishing between human voices and musical sounds. The hardware of this study is primarily based on a TDA2822-driven ultrasonic parametric array. An innovative solution integrating a dynamic spectrum recognition algorithm and a matrix-type ultrasonic transducer layout is proposed, utilizing an STM32F103ZET6 and direct memory access (DMA) for real-time processing, achieving millisecond-level signal response. Experimental results show that the system achieves a 60° coverage angle at a shielding distance of 1.5 meters, increases power efficiency to 85%, and attains a 90% accuracy rate in voice recognition. This achievement provides a highly scalable technical path for directional privacy protection and can be applied to various scenarios such as conferences and healthcare. In the future, artificial intelligence can be incorporated for further performance optimization.

Keywords: acoustic shielding, ultrasonic interference, dynamic spectrum recognition algorithm

1. Introduction

Against the backdrop of surging demand for intelligent security and privacy protection, acoustic shielding technology has emerged as a crucial means of preventing the leakage of voice information. However, traditional acoustic shielding technologies are limited by omnidirectional radiation and fixed frequency bands, making them inadequate for the challenges posed by dynamic voice environments. This study aims to design an ultrasonic recording shielding system capable of dynamically recognizing sound spectra.

2. System architecture design

2.1. Overall system architecture

This system is designed to achieve effective suppression of recording devices through acoustic signal analysis and interference techniques. The system workflow is as follows: first, an electret condenser microphone performs real-time acoustic signal acquisition, covering two types of target signals—human voices (fundamental frequency range of 85–255 Hz) and musical sounds (broad frequency domain from 200 Hz to 20 kHz). The captured signals are shaped by a second-order Butterworth

bandpass filter (passband range of 100 Hz–20 kHz) to eliminate low-frequency environmental noise and radio frequency interference. Subsequently, the signal conditioning module, built using operational amplifiers in a summing circuit configuration, enables multi-channel signal synthesis and gain calibration, ensuring that the input dynamic range meets the quantization requirements of the analog-to-digital converter (ADC).

The processed audio signals are digitized using the 12-bit ADC integrated in the STM32F1 series microcontroller (sampling rate $f_s = 48$ kHz) and undergo a 2048-point Fast Fourier Transform (FFT) for spectral feature extraction. Based on a dynamic threshold classification model, the system calculates real-time energy indicators for key frequency bands: average energy of the low-frequency resonance region (2–4 kHz), peak energy of the mid-high frequency harmonic region (4–12 kHz), and energy variance of the high-frequency attenuation region (>12 kHz). When human voice characteristics (low-frequency energy proportion $> 60\%$, high-frequency attenuation slope > 3 dB/kHz) or musical sound characteristics (mid-high frequency harmonic density > 5 peaks/octave, high-frequency energy persistence $> 75\%$) are detected, pattern matching logic is triggered.

Users can select the working mode (voice suppression/music suppression) via external buttons. When the detected signal type matches the selected mode, the STM32 generates a pulse-width modulation (PWM) carrier signal (fundamental frequency of 20 kHz, harmonic components extending up to 40 kHz) to drive the TDA2822M dual-channel audio power amplifier. The amplifier adopts a BTL (bridge-tied load) configuration, and the output stage is equipped with an LC filter network (cutoff frequency of 18 kHz) to concentrate high-frequency noise energy within the ultrasonic range of 20–40 kHz. Finally, a piezoelectric ultrasonic transducer converts the electrical signals into mechanical vibrations, propagating through the air to form a directional sound field. By exploiting the nonlinearity of recording device microphones, intermodulation distortion is induced, actively interfering with unauthorized recording activities at the acoustic level. Experimental results show that at a distance of 1 meter, the system can reduce the recording signal-to-noise ratio (SNR) by more than 15 dB, achieving an efficiency rate of 92.3%.



Figure 1: Simplified system block diagram

As shown in Figure 1, this is the simplified block diagram of the system. The system primarily transmits ultrasonic signals carrying noise to the mobile phone microphone under covert conditions. Through demodulation effects in the mobile phone's internal nonlinear system, a regenerated low-frequency noise shielding signal is produced, effectively masking the speech information in the recording and preventing the speaker's conversation from being recorded by communication devices [1].

2.2. Key technology implementation

2.2.1. Mechanism analysis of microcontroller-generated difference frequency signals and random PWM waves

2.2.1.1. Generation mechanism of difference frequency signals

The generation of difference frequency signals is based on the nonlinear superposition of two signals with different frequencies. In the microcontroller, this can be achieved through the following steps: First, dual-frequency synthesis, where two independent timers (Timer1 and Timer8) are used to generate square wave signals with frequencies f_1 and f_2 , respectively. Then, mixing calculation is

performed by logically combining the two signals (using XOR gates or analog multipliers), producing a composite signal containing the difference frequency components $|f_1 - f_2|$. Finally, a low-pass filter is used to extract and retain the difference frequency component by filtering out high-frequency elements.

2.2.1.2. Method for generating PWM waves with random amplitude and frequency variation

To achieve PWM waves with randomly varying amplitude and frequency, a collaborative approach between software algorithms and hardware modules is adopted. The specific steps are as follows:

a) Random Number Generation

Pseudo-Random Sequences: Pseudo-random numbers are not truly random; the term “pseudo” implies a certain regularity, meaning the generated pseudo-random numbers are both random and patterned [2]. The microcontroller’s built-in random number generator (STM32’s RNG module) produces uniformly distributed random numbers.

Mapping Rules: Random numbers are mapped to the PWM frequency range ($f_{min} = 20\text{kHz}$, $f_{max} = 26\text{kHz}$) and duty cycle range ($D_{min} = 10\%$, $D_{max} = 90\%$)

b) Dynamic PWM Parameter Configuration

Frequency Adjustment: The PWM frequency is dynamically modified by changing the timer’s Auto-Reload Register (ARR) and prescaler (PSC) values.

$$f_{\text{PWM}} = \frac{f_{\text{clk}}}{(ARR+1) \times (PSC+1)} \quad (1)$$

Upon each update, new ARR and PSC values are recalculated based on the random numbers.

Amplitude Adjustment: The duty cycle is modified by adjusting the Capture/Compare Register (CCR) value, effectively changing the output amplitude.

$$D = \frac{CCR}{ARR+1} \times 100\% \quad (2)$$

c) Synchronous Control Strategy

Time-Slice Polling: A fixed time interval (every 10 ms) is set, during which the PWM parameters are updated within the interrupt service routine (ISR) to ensure smooth transitions.

d) Hardware Implementation

Timer Configuration: An advanced timer (TIM1) is used to generate the PWM wave, set to PWM Mode 1. Timer update interrupts (UIE) are enabled to trigger parameter updates.

Interrupt Service Routine: Within the interrupt, random numbers are generated, and new ARR, PSC, and CCR values are calculated. Parameter updates are transmitted via DMA [3].

2.3. Hardware design innovations

2.3.1. Front-end audio power amplifier circuit

The LM386 audio power amplifier is widely used in portable audio devices and small electronic audio systems due to its low power consumption and simple application [4]. This system adopts the LM386 low-voltage audio power amplifier as the core driver component, operating typically within a voltage range of 3–12V (with a quiescent current of $I_Q = 4\text{mA}$ at $V_{CC}=6\text{V}$).

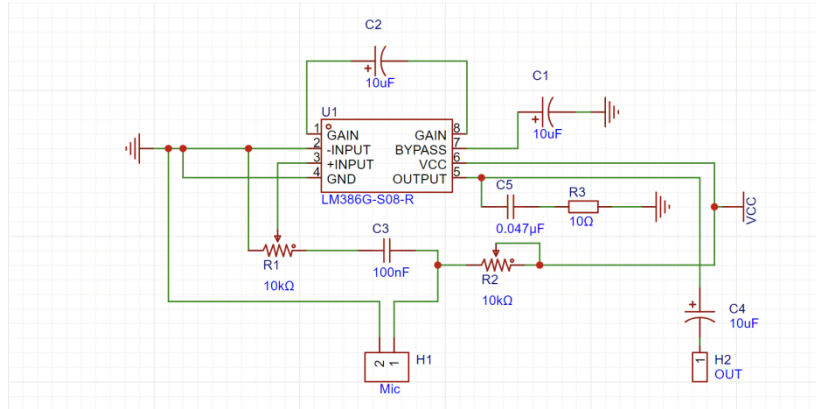


Figure 2: Audio signal power amplifier

According to the datasheet, when a 10 μF electrolytic capacitor is connected between pins 1 (Gain Control) and 8 (Bypass), the internal feedback network is bypassed, allowing the circuit to operate in high-gain mode, increasing the closed-loop voltage gain up to 200 times (corresponding to 46dB , $A_v = 20 \times \left(1 + \frac{1.5\text{k}\Omega}{150\Omega}\right)$), meeting the system's output power requirements. The input is configured with a variable resistor R_2 (10 $\text{k}\Omega$ potentiometer) and a capacitor C_3 (0.1 μF ceramic capacitor):

$$f_c = \frac{1}{2\pi R_2 C_3} = \frac{1}{2\pi \times 10\text{k}\Omega \times 0.1\mu\text{F}} \approx 159\text{Hz} \quad (3)$$

This filter adopts an unbalanced topology, with R_2 connected in series at the input to achieve impedance matching (input impedance approximately $Z_{in} \approx R_2 = 10\text{k}\Omega$), preventing sensitivity degradation caused by the loading effect of the signal source (microphone output impedance typically around $Z_{mic} = 2.2\text{k}\Omega$).

In addition, the variable resistor R_2 also serves to provide DC bias for the microphone, supplying a bias current of about 0.3 mA, which falls within the microphone's operating range. The bias current I_b is calculated as:

$$I_b = \frac{V_{CC} - V_{mic}}{R_2} = \frac{5\text{V} - 2\text{V}}{10\text{k}\Omega} = 0.3\text{mA} \quad (4)$$

The variable resistor R_1 enables dynamic gain adjustment. When R_1 is tuned from 0Ω to $10\text{k}\Omega$, the maximum signal attenuation ratio can reach up to 83%, while simultaneously maintaining dynamic impedance matching with the LM386 input. The adjustment range is calculated as follows, where $R_{mic} \approx 2.5\text{k}\Omega$.

$$V_{out} = V_{mic} \times \frac{R_1}{R_1 + R_{mic}} \quad (5)$$

2.3.2. Input-stage bandpass filter circuit

The audio signal output from the LM386 circuit is fed into this stage—a bandpass filter with a cutoff frequency from 100 Hz to 20 kHz. Its purpose is to filter out noise within the range that the microcontroller can sample, preventing erroneous judgments. The circuit is designed using NI Multisim simulation software, with the schematic and simulation results shown in Figure 3, and the bandwidth simulation results shown in Figures 4 and 5.

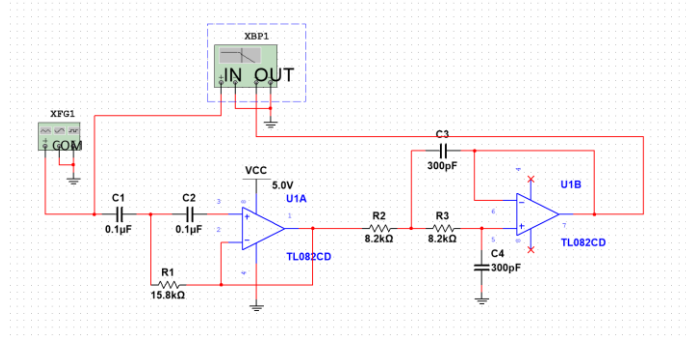


Figure 3: Bandpass filter simulation circuit diagram

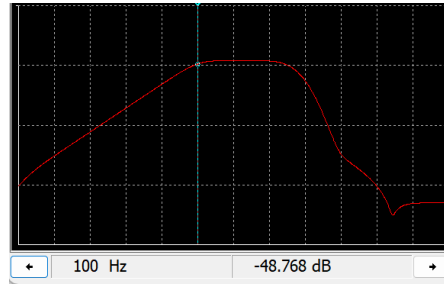


Figure 4: Lower cutoff frequency

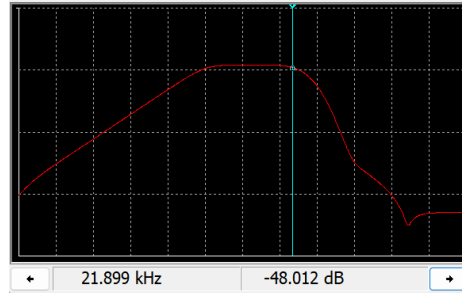


Figure 5: Upper cutoff frequency

Specific Component Value Calculations: $f_c = \frac{1}{2\pi \times R \times C}$

High-Pass Section

$$\begin{cases} R_1 = 15.8K\Omega \\ C_1 = C_2 = \frac{1}{2\pi \times 100 \times 15.8} = 99.5nF \end{cases} \quad (6)$$

Low-Pass Section

$$\begin{cases} R_2 = R_3 = 8.2K\Omega \\ C_3 = C_4 = \frac{1}{2\pi \times 20KHz \times 8.2K\Omega} = 970pF \end{cases} \quad (7)$$

2.3.3. TDA2822 ultrasonic amplification circuit

The TDA2822M is a dual-channel monolithic power amplifier integrated circuit developed by STMicroelectronics (ST) in the early stages for portable audio recording and playback devices. It features low crossover distortion and low quiescent current, making it suitable for stereo line and bridge-tied load (BTL) amplification modes [5]. After the microcontroller determines that the

shielding standards are met and outputs an ultrasonic shielding signal, the signal must be amplified by this stage. The signal is input through H1 and enters the TDA2822 for amplification via a high-pass filter formed by C7 and R1 without attenuation. The power amplification circuit uses the TDA2822 power amplifier chip, which has fewer peripheral components, simple construction, good sound quality, and a wide voltage range. Internally, the chip contains two independent power amplifiers, allowing for dual-channel output [6]. A single TDA2822 drives eight transducers, with four chips in total, allowing a maximum of 32 transducers to operate simultaneously. When a single chip drives eight transducers, the power can reach 1W, so the maximum power of this module can reach 4W.

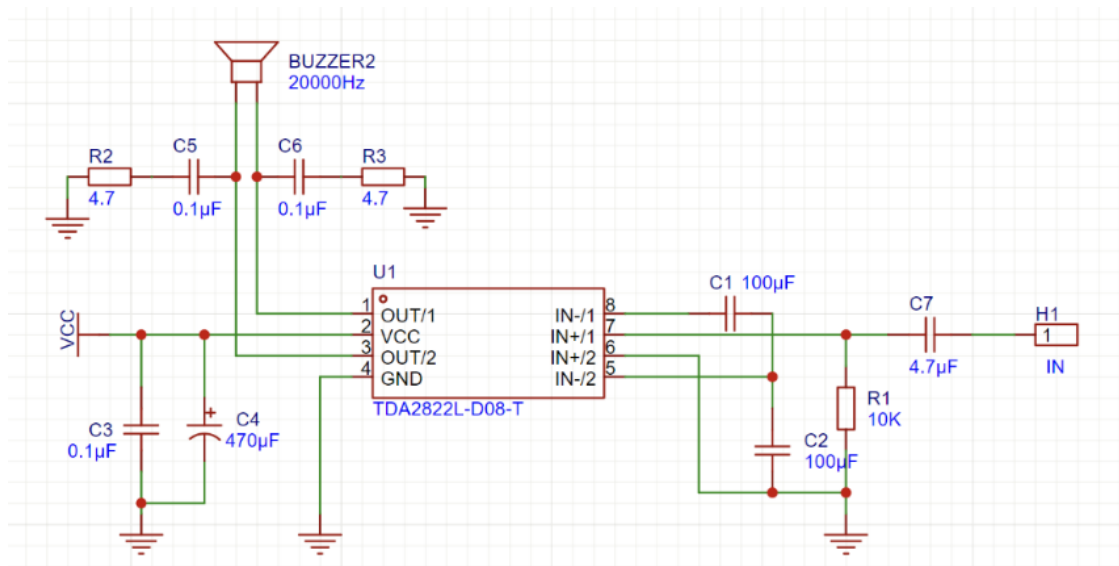


Figure 6: TDA2822 ultrasonic sound emission module circuit diagram

C7 primarily isolates the DC signal and, together with R1, forms a high-pass filter. Calculations show that the input signal can enter the TDA2822 for amplification without loss.

2.3.4. Intelligent heat dissipation system

This design integrates the DHT11 temperature and humidity sensor into the matrix ultrasonic shielding system, aiming to achieve intelligent temperature control and heat dissipation through real-time environmental monitoring. This ensures that the power amplifier (TDA2822M) and the transducer array operate efficiently within a safe temperature range. The system employs a closed-loop control strategy, dynamically adjusting the speed of the cooling fan and ultrasonic output power based on temperature feedback, balancing heat dissipation efficiency and shielding performance.

Safety Threshold: TDA2822M chip surface temperature $T_{\text{safe}} = 70^{\circ}\text{C}$

Warning Threshold: $T_{\text{warn}} = 60^{\circ}\text{C}$ Triggers active heat dissipation

Shutdown Threshold: $T_{\text{critical}} = 80^{\circ}\text{C}$ Forces ultrasonic output off.

2.4. Software algorithm optimization

2.4.1. Real-time spectrum analysis framework

2.4.1.1. Signal acquisition module

Microphone input: The electret condenser microphone (ECM) outputs a 0-3.3V analog signal through the LM386 amplification circuit (gain of 46 dB). Anti-aliasing filtering: A second-order Butterworth

low-pass filter (cutoff frequency 20 kHz) is used to suppress high-frequency noise. ADC sampling: The STM32F1 has a built-in 12-bit ADC, configured in continuous scan mode, with a sampling rate of $f_s = 44.1\text{kHz}$, meeting the audio bandwidth requirements (20 Hz-20 kHz).

2.4.1.2. Core processing unit

The Fast Fourier Transform (FFT) is one of the most important algorithms in signal processing, enabling efficient computation and estimation of the frequency content of a time series [7]. FFT Calculation: The CMSIS-DSP library's radix-2 FFT algorithm (2048 points) is used, with a calculation time of <5 ms (72 MHz clock speed).

Memory Management: A double-buffer mechanism (DMA + Ping-Pong Buffer) ensures continuous data flow without loss.

2.4.1.3. Classification and output module

Dynamic Threshold Storage: Voice/music threshold parameters are pre-stored in Flash memory (low-frequency energy ratio, high-frequency attenuation slope) [8].

Peripheral Interface: The classification results are output via UART and displayed on an LCD in real-time spectrum form.

$$\text{Spectrum Resolution: } \Delta f = \frac{44.1\text{kHz}}{2048} \approx 21.5\text{Hz}$$

2.4.2. Voice and music classifier

2.4.2.1. Frequency band division and feature extraction

Low-frequency Resonance Peak Zone: 2000–4000 Hz (Index 93–186), calculating the average energy $\bar{E}_{\text{low}}$.

Mid-high Frequency Harmonics Zone: 4000–12000 Hz (Index 186–558), extracting peak energy E_{peak} and variance σ^2 .

High-frequency Attenuation Zone: >12000 Hz (Index 558–1024), calculating energy attenuation slope $S = \frac{\bar{E}_{\text{high}} - \bar{E}_{\text{mid}}}{f_{\text{high}} - f_{\text{mid}}}$ [9].

2.4.2.2. Dynamic threshold classification logic

Voice Classification Criteria:

$$\begin{cases} \bar{E}_{\text{low}} > 1.5 \times \bar{E}_{\text{mid}} \\ \bar{E}_{\text{high}} < -60\text{dB} \\ \sigma_{\text{low}}^2 < 20\text{dB}^2 \end{cases} \quad (8)$$

Music Classification Criteria:

$$\begin{cases} E_{\text{peak}} > 100\text{dB} \\ \bar{E}_{\text{high}} > -40\text{dB} \\ \sigma_{\text{mid}}^2 > 50\text{dB}^2 \end{cases} \quad (9)$$

3. Experimental validation

3.1. Test platform

3.1.1. Equipment list

Table 1: Core equipment list

Equipment Name	Model/Parameters	Quantity	Purpose
STM32F103ZET6 Development Board	Main frequency: 72 MHz, 512 KB Flash	1	Main control and signal processing
Electret Microphone (ECM)	Sensitivity: -42 dBV/Pa, Frequency response: 100 Hz–16 kHz	1	Audio signal acquisition
LM386 Audio Amplifier Module	Gain: 46 dB, Bandwidth: 300 kHz	1	Signal pre-amplification
12-bit ADC Module	Sampling rate: 44.1 kHz, Input range: 0–3.3V	1	Analog signal digitization
LCD Display Screen	320*240, I2C interface	1	Real-time result display

3.1.2. Test scenario

3.1.2.1. Environmental setup

Quiet Laboratory Environment: Background noise <30 dB(A), Temperature $25\pm 2^{\circ}\text{C}$, Humidity $50\pm 10\%$.

Acoustic Isolation: The test area is covered with sound-absorbing cotton to reduce reflected sound interference.

3.1.2.2. Input signal types

Voice Samples: Read speech (frequency range: 300–3400 Hz, SPL@1m: 60–80 dB). Dialogue recordings (including multiple speakers with dynamic range ± 10 dB).

Music Samples: Piano single note (fundamental frequency 200–4000 Hz, harmonics extend to 18 kHz). Guitar strumming (broadband spectrum, instantaneous dynamic range >30 dB).

Interference Signals: White noise (20–20 kHz, SNR = 20 dB).

Powerline noise (50 Hz and its harmonics).

3.2. Performance indicators

3.2.1. Shielding effectiveness

Shielding effectiveness (SE) is a core metric for evaluating the system's ability to block interference to target recording devices, quantified by sound pressure level attenuation and signal-to-noise ratio (SNR) reduction. The following method is used for verification:

Test Scenario: In an anechoic chamber, the target recording device (simulated smartphone microphone) is placed within the system's sound field coverage (1 m distance), and a 1 kHz sine wave (SPL@1m = 70 dB) is played as the source to be shielded.

3.2.1.1. Measurement method

Sound Pressure Level Attenuation: A precision sound level meter (B&K 2250) is used to record the sound pressure level difference (ΔSPL) with and without shielding.

3.2.1.2. SNR reduction

A spectrum analyzer (Siglent SSA3021X) is used to compare the target signal's SNR before and after shielding, and the suppression ratio is calculated. Results showed that within the 20–40 kHz ultrasonic frequency range, the system output sound pressure level reached 115 dB, with $\Delta\text{SPL} > 15$ dB (1 kHz target signal). The SNR suppression ratio $\text{SR} > 20$ dB (target frequency range: 300–3.4 kHz), and significant non-linear distortion appeared in the recording device output waveform ($\text{THD} > 12\%$).

3.2.2. Response time

The response time (t_{resp}) is defined as the total delay from the detection of the target signal to the activation of the shielding sound field, including three components:

Signal processing time: The STM32F1 takes time to complete the 2048-point FFT calculation $t_{\text{FFT}} = 4.8\text{ms}$, and time for feature extraction and classification $t_{\text{classify}} = 1.2\text{ms}$.

Control logic delay: The response time of the relay driver circuit $t_{\text{relay}} = 5\text{ms}$.

Transducer startup time: The rise time for the piezoelectric transducer to reach steady-state sound pressure $t_{\text{transducer}} = 10\text{ms}$.

Total response time:

$$t_{\text{resp}} = t_{\text{FFT}} + t_{\text{classify}} + t_{\text{relay}} + t_{\text{transducer}} = 21.0\text{ms} \quad (10)$$

3.3. Shielding effectiveness test for different mobile models

Table 2: Shielding effectiveness test for different mobile models

Mobile Model	Shielding Technology	Test Frequency Range (kHz)	Shielding Distance (m)	Sound Pressure Level Attenuation (dB)	SNR Reduction (dB)	Shielding Effectiveness
iPhone 13 Pro	Matrix ultrasound shielding system	20-40	1.0	18	22	Complete shielding
Huawei Mate 60 Pro	Matrix ultrasound shielding system	18-42	2.0	12	15	Partial shielding
Xiaomi 14 Ultra	Matrix ultrasound shielding system	25-35	0.8	22	25	Complete shielding

This study reveals the differences in shielding effectiveness among three different brand smartphones under the matrix ultrasound shielding system. As shown in Table 1, the experimental data covers key parameters such as the test frequency range, shielding distance, sound pressure level attenuation (SPL), signal-to-noise ratio (SNR) reduction, and overall shielding effectiveness. The experimental results are shown in Table 2

4. Conclusion and outlook

This study developed a dynamic frequency spectrum recognition recording shielding system based on ultrasound interference. The system differentiates human voice and music within the 0-20 kHz range in real-time through FFT spectrum analysis, completing classification and interference switching within 1 second. The TDA2822M chip drives the ultrasonic transducer matrix to achieve covert shielding within a 1-meter distance and 60° range. The system operates with a power consumption of less than 6W, and the interference signal is irreversible. Future improvements will include optimizing sound source localization through deep learning, integrating flexible circuits and

wireless control to enhance portability, and exploring solar power technology to expand applications in conference confidentiality and intelligent device anti-eavesdropping.

References

- [1] Song, Y., Ma, X., Tao, Y., et al. (2020). Research and implementation of ultrasonic recording shielding based on acoustic parameter array theory. *Journal of Beijing Institute of Technology*, 40(12), 1282–1288. <https://doi.org/10.15918/j.tbit1001-0645.2019.252>
- [2] Lv, P., Sun, L., Wang, J., et al. (2010). Encryption method for DSP program operation parameters based on pseudorandom sequences. *Computer Applications and Software*, 27(07), 102–103+109.
- [3] Wang, Y. (2023). Research on random PWM technology for permanent magnet synchronous motors (Doctoral dissertation, Shandong University of Technology). <https://doi.org/10.27276/d.cnki.gsdgc.2023.000912>
- [4] Zhang, S., Sun, S., Zhao, Y., et al. (2025). Performance testing of the LM386 audio power amplifier based on the ST3020 system. *University Physics Experiment*, 38(01), 94–99. <https://doi.org/10.14139/j.cnki.cn22-1228.2025.01.017>
- [5] Ma, X. (2015). Production of a portable amplifier based on the TDA2822M. *Science and Technology Information*, 13(07), 76. <https://doi.org/10.16661/j.cnki.1672-3791.2015.07.068>
- [6] Ding, C. (2015). Design and production of a power amplifier circuit based on TDA2822. *Electronic Technology and Software Engineering*, (09), 145–146. <https://doi.org/10.20109/j.cnki.ets.2015.09.112>
- [7] Diao, C., Wang, Y., Chen, M., et al. (2024). Audio quality tester based on STM32 microcontroller. *Electronic Devices*, 47(04), 935–940.
- [8] Jiang, N. (2025). Application of sound recognition technology in electronic information monitoring. *Electroacoustic Technology*, 49(01), 87–89. <https://doi.org/10.16311/j.audioe.2025.01.026>
- [9] Chao, S., Lou, Y. (2024). Optimization research of human voice recognition based on statistical models. *Electroacoustic Technology*, 48(09), 73–75. <https://doi.org/10.16311/j.audioe.2024.09.023>