

AIRBORNE ACOUSTIC COMMUNICATION USING INAUDIBLE FREQUENCIES SUPPORTED BY SMART DEVICES

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ABSTRACT

Aerial acoustic communication enables low-rate data exchange using audible or inaudible acoustic waves and has the advantage of operating on virtually all smart devices without additional hardware, unlike NFC, whose adoption remains limited by hardware and platform constraints. Standard microphones and speakers can therefore be used for both transmission and reception, making the technology an appealing and practical complement to existing wireless methods. However, commodity devices primarily support the audible band, much of which overlaps with speech and ambient noise, leaving only a narrow portion suitable for reliable communication. To address this limitation, the proposed approach utilizes a frequency region that is broadly supported across devices yet minimally influenced by everyday acoustic environments, thereby enhancing overall stability and robustness. Furthermore, this paper introduces a Zoom-FFT-based narrow-band acoustic communication technique that improves robustness and frequency resolution within this constrained spectrum. By exploiting its high-resolution spectral analysis, the system can reliably extract communication signals even in noisy indoor settings, supporting practical short-range data exchange applications across diverse usage scenarios.

KEYWORDS

Acoustic Communication, FFT, Zoom FFT, NFC

1. INTRODUCTION

With the widespread adoption of smartphones, the demand for short-range communication technologies and related services has continued to grow across various application domains. Traditional short-range communication technologies, such as Bluetooth and NFC, are commonly used; however, these technologies are constrained by hardware requirements, platform fragmentation, and compatibility issues. As a result, their deployment is sometimes limited in real-world environments. Consequently, there has been a growing interest in acoustic-based communication technologies that can provide reliable short-range data exchange without relying on additional or specialized hardware components [1–8]. Many studies have focused on improving the performance of aerial acoustic communication (AAC). ChaoCai et al. propose new quaternary CSS symbols, a low-power demodulation method, and a frame combining technique to address noise, frequency selectivity, and high power consumption in AAC on commercial devices[1]. Jihwan Lee et al. employ multiple carriers with loose orthogonality, range and Doppler-aware beacon detection, and rate adaptation techniques to substantially increase data transmission rates [2]. Some papers introduce chirp signalbased AAC technologies [3-4]. Such acoustic communication techniques have been applied to short-range device-to-device

communication scenarios, including user identification in smart devices [5]. Such techniques are attractive because they can operate on virtually any device equipped with a speaker and microphone, enabling broader accessibility and ease of integration.

General-purpose audio devices such as televisions, laptops, and smartphones are primarily designed to process sound within the human audible frequency range (20 Hz–20 kHz). These devices exhibit excellent response characteristics in this band, allowing them to reproduce music, speech, and media content effectively. However, their responsiveness to acoustic signals in the near-ultrasonic or inaudible region is uncertain and often limited by hardware variability. For this reason, current acoustic communication systems typically rely on audible frequencies when implemented on consumer devices. Indeed, most existing or publicly available acoustic communication mechanisms adopt this approach due to its predictable performance and ease of deployment.

Nonetheless, certain practical environments restrict the use of the full audible spectrum. For instance, in television broadcasting or digital signage environments, additional information must be transmitted through acoustic signals without disturbing the main audio content. Although the audible range spans from 20 Hz to 20 kHz, human speech predominantly occupies only the 500 Hz–3,000 Hz region, and the sensitivity of human hearing decreases significantly near both ends of the spectrum. This characteristic makes the upper boundary of the audible range, particularly frequencies near 20 kHz, suitable for embedding data in a minimally intrusive manner. By utilizing these high-frequency components, information can be transmitted in parallel with existing audio content while remaining imperceptible to most listeners.

However, the use of this narrow near-20-kHz band introduces several technical challenges. The limited bandwidth severely constrains achievable data rates, and conventional demodulation techniques often struggle to accurately reconstruct the received signal under such tight spectral constraints, especially in noisy indoor environments. To overcome these limitations, this paper proposes an acoustic communication system that employs MFSK modulation at the transmitter and incorporates Zoom FFT processing at the receiver. The combination of MFSK's robustness and Zoom FFT's high-resolution spectral analysis enables stable and reliable data transmission within a narrow frequency band close to the inaudible region, thereby providing an effective communication framework suitable for real-world applications.

2. HIGH-RESOLUTION FREQUENCY ANALYSIS USING THE ZOOM FFT

2.1. Theoretical Background

The Fast Fourier Transform (FFT) is an essential tool in digital signal processing (DSP) for efficiently analyzing the frequency spectrum of a signal. The frequency resolution (Δf) of the FFT is a critical factor determining the precision of the analysis results, and it is defined by the following fundamental Discrete Fourier Transform (DFT) equation:

$$\Delta f = \frac{f_s}{N} \quad (1)$$

Here, f_s is the Sampling Frequency, and N is the Sampling Length or the number of samples used for the FFT. Theoretically, improving frequency resolution is straightforward: decrease f_s or increase N .

However, in practical systems, both variables face significant practical constraints. The constraints on improving FFT frequency resolution ($\Delta f = f_s/N$) are twofold: the sampling

frequency (f_s) is limited by the Nyquist-Shannon theorem ($f_s \geq 2f_{max}$) to avoid aliasing and often requires a higher margin (3-5 times f_{max}) for the anti-aliasing filter's guard band, while the sampling length (N) is constrained by the resulting increase in data acquisition time (latency) and the rising computational complexity of the FFT, which is proportional to $O(N \log_2 N)$.

2.2. Principle of Zoom FFT

The basic principle of Zoom FFT is illustrated in Figure 1. The Zoom FFT technique strategically addresses the limitations of standard FFT by enabling selective frequency resolution enhancement for a specific Region of Interest (ROI) in the signal spectrum through a combined process of Frequency Shifting and Decimation: first, the continuous-time input signal $x(t)$ is modulated by multiplying it with a complex exponential function $e^{-j2\pi f_c t}$ whose frequency f_c is the center of the ROI, a time-domain operation known as Complex Heterodyning that shifts the entire spectrum by $-f_c$ to center the ROI at the DC component (0 Hz); subsequently, because the bandwidth B of the now baseband-centered ROI allows for a drastically lower minimum sampling rate ($f'_s \geq 2B$), digital filtering and downsampling (Decimation) are applied to reduce the effective sampling frequency to f'_s , thereby minimizing the number of samples and the subsequent computational burden; finally, performing the FFT on this decimated data yields a significantly improved high-resolution $\Delta f' = f'_s / N$ compared to the original resolution ($\Delta f = f_s / N$) due to $f'_s \ll f_s$, with a final Frequency Modification step adjusting the output spectrum back to its correct absolute frequency location by adding f_c . Zoom FFT successfully avoids the two main hurdles—the Nyquist constraint and the increase in computational complexity—by utilizing complex heterodyning and decimation to enhance the resolution of a specific frequency band of interest. This makes it an indispensable technique in various applications, such as vibration analysis, communication signal monitoring, and ultrasonic measurement, where precise frequency information is required within a specific narrow bandwidth.

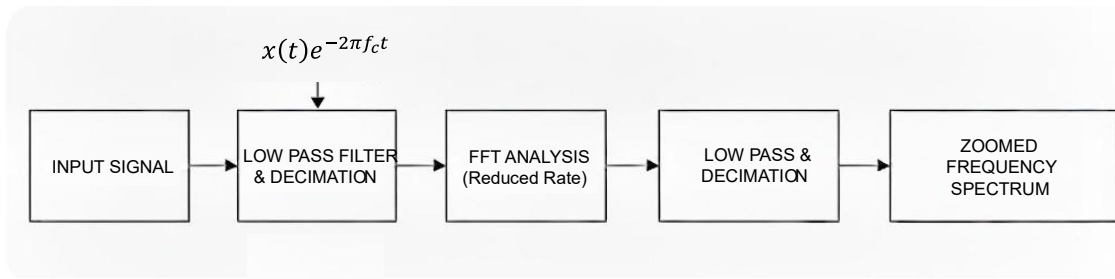


Figure 1. Zoom FFT

3. ACOUSTIC COMMUNICATION TECHNOLOGY BASED ON ZOOM FFT

3.1. The Fundamental Concept of the Proposed Algorithm

Methods for recovering data transmitted using audible frequencies can be broadly classified into two approaches: time-domain signal analysis and frequency-domain signal analysis. In the time-domain approach, the received signal is analyzed directly to extract temporal features and reconstruct the transmitted information, whereas in the frequency-domain approach, the spectral characteristics of the signal are examined to identify the transmission frequencies and recover the corresponding data. While these conventional methods are effective for simple signals, they exhibit significant limitations when multiple frequencies are transmitted within a narrowband. Specifically, in such scenarios, accurately distinguishing between closely spaced frequencies becomes challenging, and achieving sufficient frequency resolution often requires using only a

small subset of available frequencies. This restriction reduces the data rate and limits the flexibility of the communication system. Consequently, conventional techniques struggle to support high-density frequency multiplexing within narrow frequency bands, which is a critical requirement for efficient and reliable acoustic communication systems operating in real-world environments.

In this paper, we propose a novel receiver-side technique for narrowband acoustic communication that leverages the Zoom Fast Fourier Transform (Zoom FFT) to overcome the inherent limitations of conventional signal reception schemes, such as Quadrature Receivers and standard FFT-based approaches. The fundamental principle of the proposed methodology is depicted in Figure 2. Specifically, to facilitate data transmission within the audible frequency spectrum while minimizing interference with concurrent audio signals, we consider a narrowband transmission range, for instance, from 19.5 kHz to 20.0 kHz. Given a conventional sampling rate of 44.1 kHz and a 1024-point FFT, the maximum resolvable frequency reaches approximately 43 kHz. Within the specified narrowband, the number of distinct transmission frequencies is limited to 11, which, when coupled with a conventional MFSK modulation scheme, permits only up to 3 symbols to be transmitted simultaneously, thereby imposing a significant constraint on spectral efficiency.

By contrast, the application of Zoom FFT with a decimation factor of 10 enables a tenfold enhancement in frequency resolution, substantially improving the system's ability to discriminate closely spaced spectral components. This refinement effectively doubles the maximum number of simultaneously transmittable symbols from 3 to 6, thereby markedly enhancing transmission efficiency within the narrowband. Beyond increased spectral utilization, Zoom FFT confers additional benefits, including precise detection of low-amplitude signals under high-noise conditions and improved robustness against multipath fading and other channel impairments frequently encountered in practical acoustic communication environments. Consequently, the proposed approach not only augments the data throughput and modulation efficiency of narrowband acoustic communication systems but also significantly strengthens signal reliability, enabling more resilient and high-fidelity transmission in realistic operational scenarios.

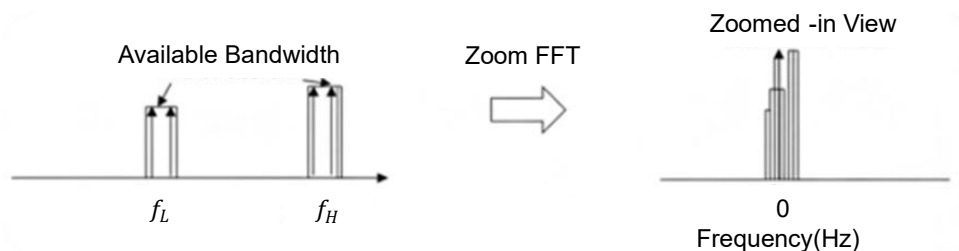


Figure 2. Frequency Band for Zoom FFT-Based Acoustic Communication

3.2. Operation Procedure of the Proposed Algorithm

The short-range acoustic communication system proposed in this paper, which operates within the audible frequency band, is primarily composed of a transmitter and a receiver. At the transmitter, the data to be transmitted undergoes a series of processing stages, including source coding, encryption, and channel coding, followed by modulation. The processed data is then converted into audible acoustic signals and emitted through the speaker. At the receiver, the acoustic signals are captured via a microphone and subsequently processed through demodulation, channel decoding, decryption, and source decoding to reconstruct the transmitted

information. This structured signal processing chain ensures reliable data recovery while maintaining compatibility with conventional consumer audio hardware.

At the transmitter, the system employs M-ary Frequency Shift Keying (MFSK) modulation to encode symbols composed of M bits into sinusoidal acoustic signals within the audible frequency range of 20 Hz to 20 kHz. Each symbol is mapped to a distinct carrier frequency within this band, ensuring that every transmitted symbol corresponds to a unique frequency. During each symbol period, the transmitter emits a sinusoidal waveform whose frequency is determined by the mapped symbol, thereby enabling the representation of digital information as frequency variations in the acoustic domain. This approach allows for simultaneous utilization of multiple discrete frequencies within the audible spectrum, facilitating robust symbol differentiation and efficient encoding of information while minimizing interference with other acoustic sources in the operating environment. The careful selection and mapping of frequencies also ensure that the transmitted signals remain within the capabilities of conventional audio hardware, such as consumer speakers, while preserving signal fidelity for subsequent demodulation at the receiver.

When employing a conventional FFT-based detection method at the receiver, the frequency resolution of an FFT is determined by the sampling frequency Δf and the number of FFT points N . In typical audio signal processing, a sampling frequency of 44.1 kHz is commonly used. For a 128-point FFT, the resulting frequency resolution is 345 Hz. Within the audible frequency range of 20 Hz to 20 kHz, this configuration allows for the decomposition of approximately 58 distinct frequency components. Consequently, only a maximum of 4-bit symbols can be transmitted per symbol period. However, when employing a high-order modulation scheme such as 256-MFSK to transmit 8-bit symbols simultaneously, the minimum frequency separation between adjacent symbol frequencies is approximately 78 Hz. Under these conditions, the frequency components of each symbol after FFT transformation become nearly indistinguishable, making reliable symbol differentiation impossible. This limitation highlights the inadequacy of conventional FFT-based detection for high-order MFSK modulation in narrowband acoustic communication systems, where closely spaced frequencies must be resolved accurately to achieve high spectral efficiency.

Therefore, this paper proposes a novel receiver-side method that enhances the frequency resolution of the received signal through a two-stage FFT process, thereby maximizing the transmission efficiency of the acoustic communication system. The signal processing at the receiver is structured into two primary stages. In the first stage, referred to as the frequency band identification stage, a conventional FFT is applied to the received signal to obtain its discrete spectral representation within the frequency band of interest. The frequency interval containing the spectral peak, corresponding to the dominant frequency component, is then determined.

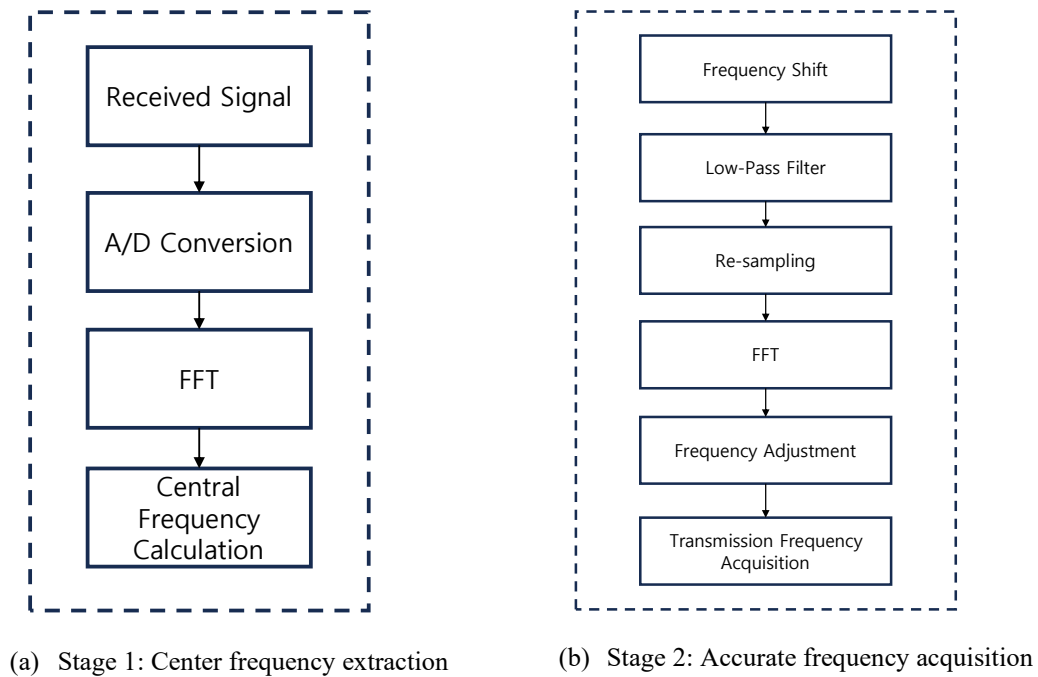


Figure 3. Two Stages of Proposed Algorithm

In the first stage, although the FFT transformation provides only low frequency resolution and therefore cannot yield the precise transmission frequency, it allows for the identification of the central frequency as well as the upper and lower bounds of the frequency interval in which the signal is located. This coarse estimation of the frequency range serves as a foundation for subsequent high-resolution processing in the second stage.

In the second stage, known as the frequency resolution enhancement stage, the identified frequency interval containing the dominant spectral component is shifted to baseband, and the signal is subsequently resampled at a lower sampling rate. The resulting frequency-domain data is then subjected to inverse frequency shifting to recover the precise transmission frequency of the original signal. This two-step approach allows for significantly improved frequency resolution, enabling the receiver to distinguish closely spaced MFSK symbols that would otherwise be unresolved using a conventional single-stage FFT. The processes corresponding to the first and second stages are illustrated in Figures (a) and (b), respectively, demonstrating the stepwise enhancement of frequency resolution and the associated improvement in symbol discrimination and system throughput.

Following this procedure, the received signal at the receiver is transformed as illustrated in Figure 4. As shown, the application of the Zoom FFT technique enables the conversion of the originally closely spaced and indistinguishable frequency components into a spectrum with sufficiently high frequency resolution. This enhanced resolution allows for precise identification and separation of individual symbol frequencies that could not be reliably distinguished using conventional FFT methods. By effectively concentrating the frequency components within a narrower band and increasing the sampling resolution, Zoom FFT provides a significant improvement in the receiver's ability to resolve multiple MFSK symbols, thereby enhancing symbol discrimination and overall system performance. This process ensures that even in narrowband acoustic communication scenarios, where frequency components are densely packed within the audible spectrum, the transmitted signals can be accurately recovered and demodulated, ultimately contributing to increased transmission efficiency and reliability.

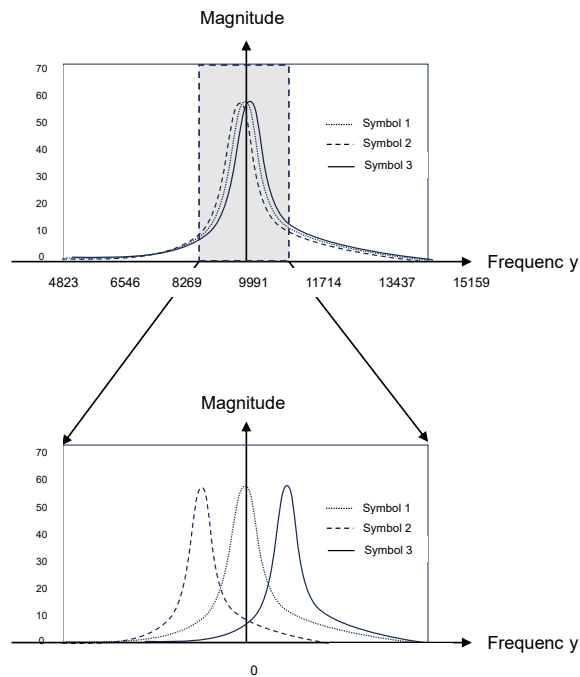


Figure 4. Received Signal Resolution

4. CONCLUSIONS

This paper proposes an acoustic communication system that employs M-ary Frequency Shift Keying (MFSK) within the audible band and utilizes an FFT-based receiver for demodulation. The receiver operates in two stages: a conventional FFT first provides a coarse estimate of the frequency band containing the transmitted signal, after which frequency shifting, resampling, and inverse shifting are applied to obtain a high-resolution estimate of the actual transmission frequency. By enhancing frequency resolution at the receiver, the method increases the number of resolvable frequency tones for MFSK, thereby improving symbol capacity, spectral efficiency, and overall data throughput. This approach mitigates the limitations of conventional FFT detection in narrowband scenarios with closely spaced frequencies.

While this study begins with the assumption that the acoustic bandwidth of devices used for acoustic communication is approximately similar, the usable bandwidth is not completely identical across all such devices. Therefore, future research should investigate methods for accurately characterizing each device's bandwidth response and determining the most optimal operational band accordingly. Furthermore, the performance of the proposed method will be evaluated through simulations or experiments under realistic conditions, including an analysis of potential impairments encountered in practical environments.

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REFERENCES

- [1] Chao Cai, Ruinan Jin, Jiangtian Nie, Jiawen Kang, Yang Zhang & Jun Luo, (2024) "Reliable High Throughput Aerial Acoustic Communication for Mobile Network", IEEE Transactions on Vehicular Technology, Vol. 73.
- [2] Jihwan Lee, Chulyoung Kwak, Seongwon Kim & Saewoong Bahk, (2020) "Reliable and Low-Complexity Chirp Spread Spectrum-Based Aerial Acoustic Communication", IEEE Access, Vol. 8.
- [3] Hyewon Lee, Tae Hyun Kim, Jun Won Choi & Sunghyun Choi, (2015) "Chirp signal-based aerial acoustic communication for smart devices", IEEE Conference on Computer Communications (INFOCOM).
- [4] Chao Cai, Zhe Chen, Jun Luo, Henglin Pu, Menglan Hu & Rong Zheng, (2022) "Boosting Chirp Signal Based Aerial Acoustic Communication Under Dynamic Channel Conditions", IEEE Transactions on Mobile Computing, Vol. 21.
- [5] Andrea Gabbriellini, Georg K. J. Fischer, Thomas Schaechtle, Wenxin Xiong, Dominik Jan Schott, Joan Bordoy, Johannes Wendeberg, Fabian Höflinger, Christian Schindelhauer & Stefan J. Rupitsch, (2023) "Airborne Acoustic Chirp Spread Spectrum Communication System for User Identification in Indoor Localization", IEEE Transactions on Instrumentation and Measurement, Vol. 72.
- [6] Henglin Pu, Xingqi Wu & Chao Cai, (2024) "High-Speed Hidden Aerial Acoustic Communication Exploiting the Whole Available Bandwidth", ICC 2024 - IEEE International Conference on Communications.
- [7] Xiao Zhang, Jiqiang Liu, Si Chen, Yongjun Kong & Kui Ren, (2019) "PriWhisper+: An Enhanced Acoustic Short-Range Communication System for Smartphones", IEEE Internet of Things Journal, Vol. 6.
- [8] Yann Hornych, Javier Cañada Toledo, Boyang Wang, Won-Jae Yi & Jafar Saniie, (2020) "Near-Ultrasonic Communications for IoT Applications using Android Smartphone", 2020 IEEE International Conference on Electro Information Technology (EIT).