A Feasibility Study of Analog Audio Transmission over Near-Ultrasonic Frequency Bands Using Consumer Audio Equipment

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Abstract

Near-ultrasonic frequency bands provide an underutilized portion of the acoustic spectrum with relatively low perceptual salience[1, 2] while remaining potentially usable with conventional audio hardware. This paper investigates the feasibility of transmitting analog audio over near-ultrasonic frequency bands using consumer-grade loudspeakers, microphones, and sound cards. A single-sideband (SSB) modulation scheme is employed to shift baseband audio into the near-ultrasonic range, where it is transmitted through an acoustic channel and subsequently demodulated at the receiver. The system is evaluated under various transmission conditions using commonly available audio equipment. Signal quality is assessed through spectral analysis and objective similarity metrics between the original and recovered audio signals. Experimental results demonstrate that consumer audio devices are capable of supporting near-ultrasonic analog audio transmission within the limits imposed by their frequency response. Although attenuation and distortion increase toward the upper end of the transmission band, the original audio can still be recovered with acceptable quality under suitable conditions. These findings suggest that near-ultrasonic acoustic channels could provide a low-cost communication medium for specialized applications without requiring dedicated ultrasonic transducers.

Keywords — Computational Audio, Near-ultrasound, Audio Transmission

1 Introduction

Generally speaking, mobile devices use Wi-Fi or Bluetooth to transmit data, but in some cases, using sound as a communication medium also has its advantages. (1) Establishing a Bluetooth or Wi-Fi connection may require additional time for device discovery, pairing, or association, whereas sound playback or recording can begin within tens of milliseconds. (2) Unlike Bluetooth and Wi-Fi, high-frequency sound is strongly attenuated by walls and other obstacles, which can be useful for copresence applications by naturally constraining the transmitter and receiver to approximately the same room or acoustic environment. (3) Bluetooth is often disabled and is otherwise problematic[3]

Therefore, this paper proposes a new scheme aimed at utilizing a near-ultrasonic frequency band where human auditory sensitivity is generally reduced, but which can still be played and received by consumer-grade audio devices, to transmit analog audio. The main advantages of the proposed scheme are summarized as follows:

1. Low perceptual salience - The proposed system places the modulated signal in a high-frequency, near-ultrasonic region, where auditory sensitivity is generally reduced compared with the mid-frequency range.[4]

2. Conceptual and implementation simplicity - Unlike conventional wireless digital communication systems, the proposed approach does not require packet-based transmission, device pairing, or a dedicated communication protocol. The audio waveform is directly translated into a high-frequency carrier band and recovered at the receiver.

3. No dedicated radio frequency transceiver is required - The system can potentially be implemented using conventional audio input and output hardware rather than dedicated radio-frequency transceivers.

4. No occupation of conventional RF spectrum - Since the proposed system uses acoustic rather than radio-frequency propagation, it does not require the use of conventional RF spectrum.

2 Principle

2.1 Bandwidth Reduction by Pitch Scaling

A further bandwidth-reduction technique is employed before SSB modulation. Let $x(t)$ denote the original audio signal,and $x_{\mathrm{br}}(t)$ denote the bandwidth-reduced audio.

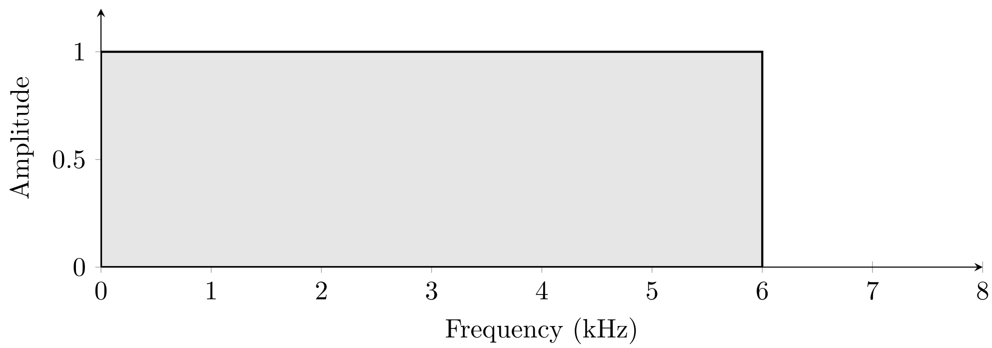


Figure 1: Idealized spectrum of the original audio signal.

In the proposed system, the original audio is pitch-shifted downward by one octave, corresponding to a frequency scaling factor of 1/2.[5] Thus, each frequency component f of the original signal is mapped to

$$f' = \frac{f}{2}.$$

For example, an audio signal with a bandwidth of approximately 6 kHz can be transformed into a signal with a bandwidth of approximately 3 kHz:

$$\text{0–6 kHz} \quad \longrightarrow \quad \text{0–3 kHz.}$$

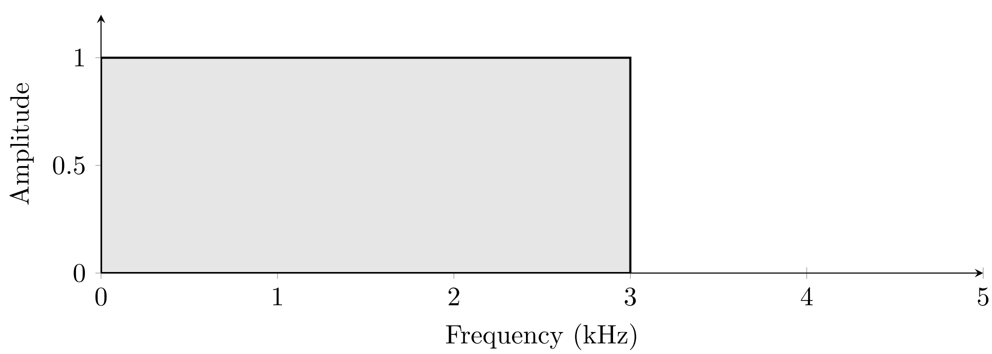


Figure 2: Idealized spectrum after pitch scaling by -12 semitones.

This operation reduces the bandwidth that must subsequently be transmitted through the high-frequency acoustic channel. When combined with SSB modulation, the resulting signal can therefore be placed within a relatively narrow frequency interval near the upper limit of a 44.1-kHz audio system.

The frequency scaling can be regarded as a reversible transformation rather than a permanent loss of the original audio information. At the receiver, the inverse frequency scaling can be applied after SSB demodulation:

$$f_{\text{original}} = 2f'.$$

Consequently, the complete signal-processing chain can be represented as

$$\text{Original audio} \xrightarrow{\times 1/2} \text{Bandwidth-reduced audio} \xrightarrow{\text{SSB}} \text{Near-ultrasonic signal.}$$

At the receiver, the inverse operations are performed:

$$\text{Near-ultrasonic signal} \xrightarrow{\text{SSB demodulation}} \text{Bandwidth-reduced audio} \xrightarrow{\times 2} \text{Recovered audio.}$$

The main purpose of this additional transformation is to allow a wider original audio bandwidth to be represented within the limited bandwidth available in the 19–22.05 kHz region. In particular, reducing a 6-kHz audio bandwidth to 3 kHz makes it possible to fit the signal into the approximately 3-kHz bandwidth available to the proposed SSB channel.The bandwidth-reduced signal $x_{\mathrm{br}}(t)$ is then used as the input to the SSB modulator as $m(t)$.

2.2 Modulation and Frequency Translation

The proposed approach shifts the audio spectrum into a high-frequency, near-ultrasonic region between 19 and 22.05 kHz. The upper boundary of 22.05 kHz is determined by the Nyquist frequency of a 44.1-kHz sampling system.[6] This frequency range is theoretically representable by audio hardware operating at a 44.1-kHz sampling rate, although the actual frequency response of the transducers may limit practical performance. By employing single-sideband modulation, the bandwidth-reduced audio spectrum can be translated into this region without occupying the bandwidth of both sidebands. The resulting system can therefore be implemented using conventional audio input and output hardware, without requiring a dedicated RF transceiver or a packet-based communication protocol.

Let $m(t)$ denote the bandwidth-reduced analog audio signal, and let B denote its bandwidth. In order to transmit the audio signal in the proposed near-ultrasonic frequency band, the spectrum of $m(t)$ must first be translated from the baseband to a higher-frequency region.

A conventional method of frequency translation is double-sideband suppressed-carrier (DSB-SC) modulation. Let the carrier signal be[6]

$$c(t) = \cos(2\pi f_c t),$$

where f_c is the carrier frequency. The DSB-SC signal is obtained by multiplying the audio signal by the carrier:

$$s_{\mathrm{DSB}}(t) = m(t)\cos(2\pi f_c t).$$

In the frequency domain, multiplication by a sinusoidal carrier produces two frequency-shifted copies of the bandwidth-reduced spectrum. If $M(f)$ denotes the Fourier transform of $m(t)$, the resulting spectrum can be expressed as

$$S_{\mathrm{DSB}}(f) = \frac{1}{2}\left[M(f - f_c) + M(f + f_c)\right].$$

The two terms correspond to the upper and lower sidebands, respectively. For a baseband signal with bandwidth B, the positive-frequency component of the DSB-SC signal occupies approximately the frequency range

$$f_c - B \le f \le f_c + B.$$

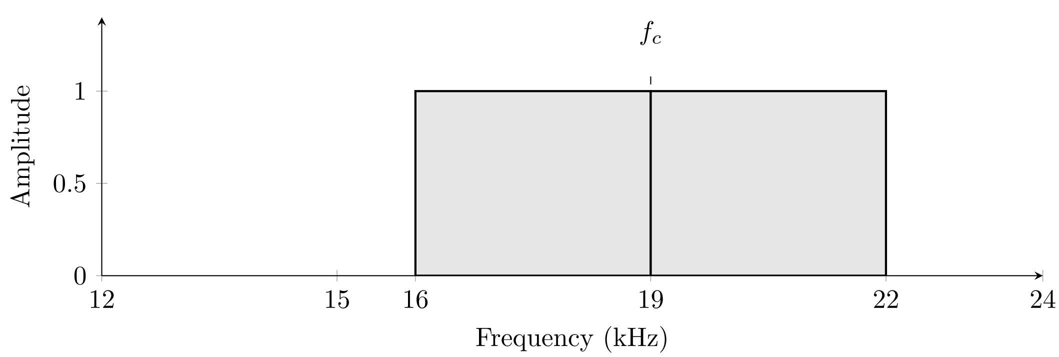


Figure 3: Idealized DSB-SC spectrum for a 3-kHz bandwidth signal with a 19-kHz carrier.

Therefore, the required transmission bandwidth is approximately[6]

$$B_{\mathrm{DSB}} = 2B.$$

For a real-valued baseband signal, the upper and lower sidebands are related by spectral symmetry and therefore contain redundant information. Consequently, one sideband can be suppressed while retaining the information necessary for signal reconstruction. Single-sideband (SSB) modulation therefore removes one of the two sidebands while retaining the information required to reconstruct the original signal.[6] As a result, the required transmission bandwidth is reduced by approximately one half:

$$B_{\mathrm{SSB}} \approx B.$$

For the proposed system, the remaining sideband is positioned within the 19–22.05 kHz near-ultrasonic region. For example, suppose that the bandwidth-reduced audio signal occupies the baseband from 0 to 3 kHz. By selecting an appropriate carrier frequency and sideband, the spectrum can be translated so that the resulting SSB signal occupies approximately 19–22 kHz. For example, using the upper sideband with a carrier frequency of $f_c = 19\text{kHz}$, an audio signal occupying 0–3 kHz is translated to

$$19 \text{ kHz} \leq f \leq 22 \text{ kHz}.$$

In this case, the lower sideband is not transmitted. The resulting signal therefore occupies only approximately 3 kHz of bandwidth rather than the 6 kHz that would be required by DSB-SC. This frequency translation can be represented schematically as

$$\underbrace{\text{0–3 kHz}}\text{ bandwidth-reduced audio} \xrightarrow{\text{SSB modulation}} \underbrace{\text{19–22 kHz}}\text{ transmission band.}$$

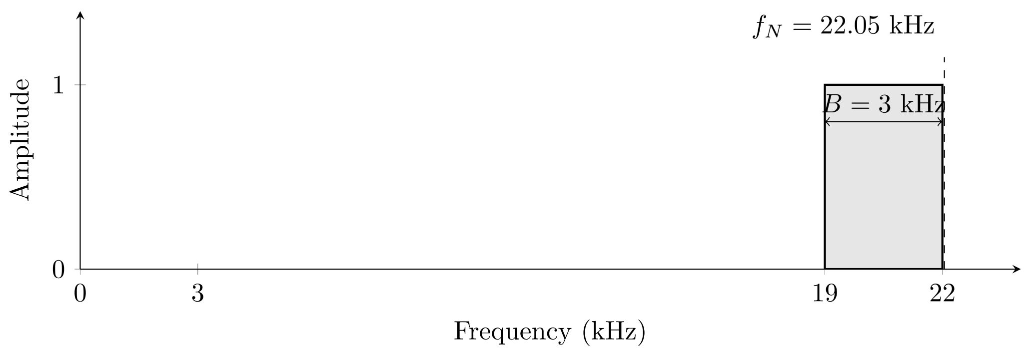


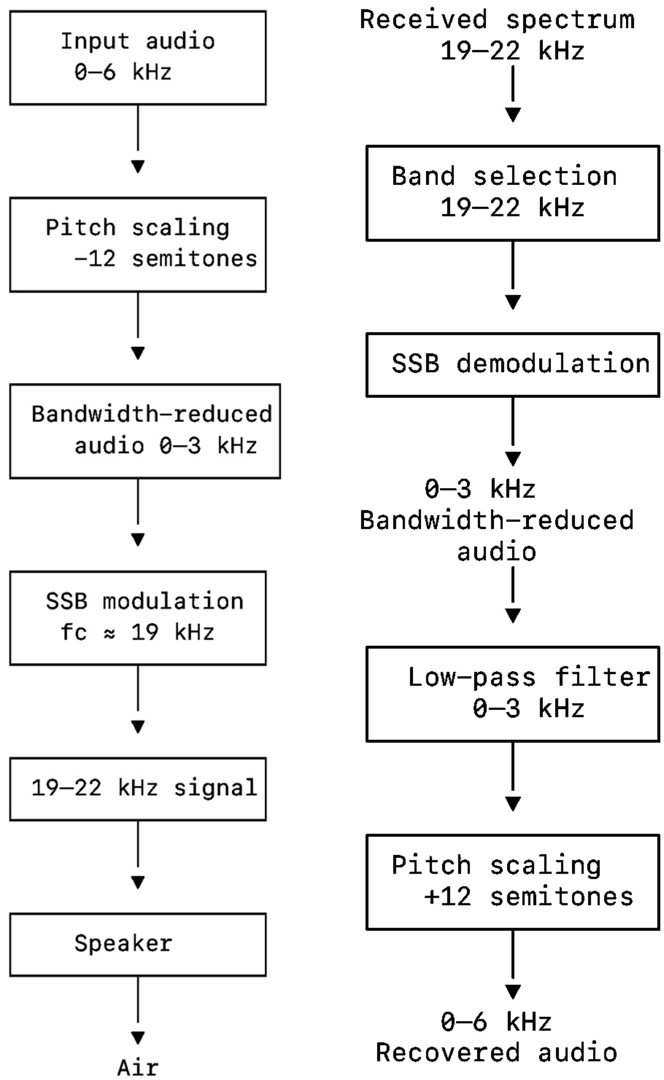
Figure 4: The proposed 19–22 kHz transmission band within the Nyquist limit of a 44.1-kHz sampling system.

At the receiver, the process can be reversed by selecting the desired high-frequency band and applying coherent SSB demodulation. The recovered baseband signal is then passed through a low-pass filter and subsequently pitch-shifted upward by one octave to reconstruct the original audio signal.

3 Implementation

3.1 Transmitter

The transmitter converts the input audio into an acoustic signal suitable for transmission in the proposed near-ultrasonic frequency band. An overview of the complete transmitter and receiver signal-processing chain is shown in Fig. 5a.



(a) transmitter and receiver

Figure 5: Block diagram of the proposed transmitter and receiver.

For the proposed implementation, the input audio is assumed to have a bandwidth of approximately 6 kHz. The signal is then pitch-shifted downward by 12 semitones, corresponding to a one-octave reduction in frequency. This reduces the effective bandwidth from approximately 6 kHz to 3 kHz while preserving the temporal structure of the audio signal.

The resulting bandwidth-reduced audio is subsequently processed using single-sideband modulation. With the carrier frequency selected at approximately 19 kHz, the desired sideband occupies the frequency range from approximately 19 to 22 kHz. This places the modulated signal below the 22.05-kHz Nyquist frequency of a 44.1-kHz sampling system.

The resulting waveform is then played through the transmitting audio device. In this way, the digital audio output of the device is converted into an acoustic waveform that propagates through the air toward the receiver.

The audio processing was implemented using Audacity, with the individual processing stages performed sequentially. The resulting acoustic waveform is then transmitted through the air, forming the acoustic channel described in the following section.

3.2 Acoustic Channel

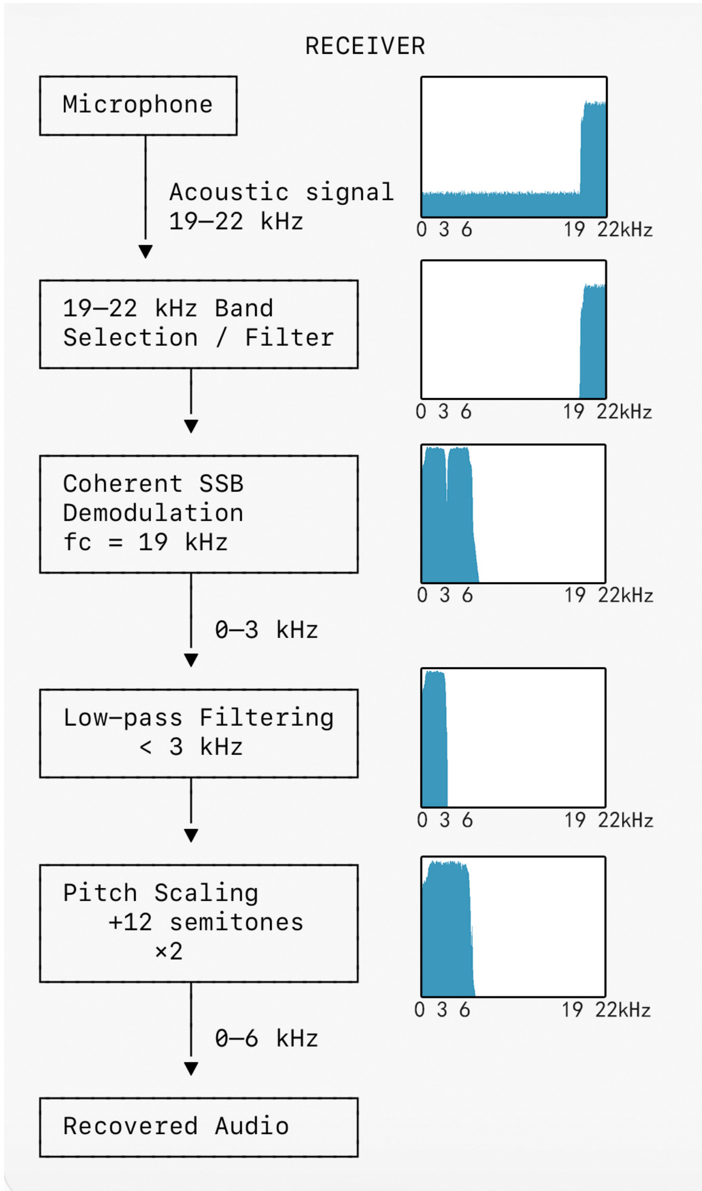
The acoustic channel consists of the propagation path between the transmitting loudspeaker and the receiving microphone. After modulation, the 19–22 kHz signal is converted into an acoustic waveform by the loudspeaker and propagates through the surrounding air. The receiving microphone subsequently converts the acoustic signal back into an electrical audio signal for further processing.

Unlike a conventional radio-frequency communication channel, the proposed system uses air as the transmission medium without requiring a dedicated RF transceiver. The characteristics of the acoustic channel are therefore determined by factors such as the distance between the transmitter and receiver, the frequency response of the loudspeaker and microphone, background noise, and reflections from surrounding surfaces. This frequency-selective propagation characteristic is also relevant to the intended copresence property of the system, since the signal is expected to be most readily received within the acoustic range of the transmitting device.

In particular, the high-frequency nature of the proposed signal makes the system sensitive to the frequency response and directivity of consumer-grade audio hardware. Atmospheric attenuation and reflections may also affect the received signal, especially as the transmission distance increases. High-frequency acoustic signals are subject to frequency-dependent atmospheric absorption, which can contribute to increased propagation loss at higher frequencies.[7]

For the experiments, the experimental setup and the specific transmitter–receiver distances are described in Section 4. Unless otherwise specified, the received signal was recorded by the microphone and subsequently processed by the receiver described in Section 3.3.

3.3 Receiver



(a) receiver

Figure 6: Diagram of the proposed receiver.

The acoustic signal is captured using a consumer-grade microphone connected to an audio input device. The recorded signal is sampled at 44.1 kHz, providing a Nyquist frequency of 22.05 kHz. This allows the proposed 19–22 kHz transmission band to be represented in the sampled signal, although the actual usable bandwidth is limited by the frequency response of the microphone and audio input chain.

The recorded signal may contain components outside the intended transmission band, including audible-frequency content and other noise introduced by the recording environment. Therefore, the receiver first selects the frequency range corresponding to the transmitted SSB signal. In the proposed system, the nominal passband is approximately 19–22 kHz. In the frequency domain, components outside the selected band are attenuated or removed, leaving the 19–22 kHz portion for subsequent demodulation.

After band selection, coherent SSB demodulation is applied to translate the received signal back to baseband. A locally generated carrier at the same frequency as the transmitter carrier is used for coherent detection. For the configuration described above, the carrier frequency is $f_c = 19$ kHz.

If the received upper sideband can be simplified as[6]:

$$s_{\mathrm{USB}}(t) = m(t)cos(2pif_ct) - \hat{m}(t)sin(2pif_ct),$$

then multiplying it with the local carrier:

$$s_{\mathrm{USB}}(t)cos(2pif_ct)$$

will produce baseband and high-frequency components. After low-pass filtering, we obtain:

$$m(t)$$

The output of the coherent demodulator contains the desired baseband component as well as higher-frequency mixing products. A low-pass filter is therefore applied to suppress these unwanted components and retain the bandwidth-reduced audio signal. The nominal cutoff frequency is approximately 3 kHz.

$$19-22\text{ kHz} \xrightarrow{\text{SSB demodulation}} 0-3\text{ kHz}$$

The resulting baseband signal corresponds to the bandwidth-reduced audio produced at the transmitter. Since the transmitter lowered the pitch by 12 semitones before modulation, the receiver applies the inverse pitch transformation by shifting the recovered signal upward by 12 semitones.

$$f_{\mathrm{out}} = 2f_{\mathrm{in}}$$

The complete receiver processing chain can therefore be summarized as

$$19\text{–}22\text{ kHz} \to 0\text{–}3\text{ kHz} \to 0\text{–}6\text{ kHz},$$

corresponding to band selection and SSB demodulation, low-pass filtering, and inverse pitch scaling, respectively. The final output is a reconstructed version of the original audio signal, subject to the attenuation, noise, and frequency-response limitations of the acoustic channel and recording hardware.

3.4 Signal Processing

The recorded and reconstructed signals are processed using the following steps before quantitative evaluation:

1. **Signal acquisition and segmentation.** The transmitted and recovered signals are recorded at a sampling rate of 44.1 kHz. The relevant portions of the recordings are segmented to exclude silence and unrelated background signals.

2. **Preprocessing.** The signals are converted to a common format and normalized before further processing. Any necessary DC offset removal and amplitude normalization are performed at this stage.

3. **Frequency-domain analysis.** The discrete Fourier transform (DFT) is calculated using the FFT algorithm[8]. The resulting magnitude spectra are used to examine the transmitted frequency band and the spectral characteristics of the recovered signal.

4. **Signal alignment.** The original and recovered signals are temporally aligned before calculating waveform-based similarity measures.

5. **Normalization.** The aligned signals are normalized to a common amplitude reference so that differences in overall recording level do not dominate the comparison.

6. **Quantitative evaluation.** The reconstructed signal is evaluated using image-based spectrogram analysis. The mean absolute grayscale difference between the original and recovered spectrograms is calculated over the full spectrogram and separately for individual frequency bands.

The main parameters used for audio acquisition and preprocessing are summarized in Table 1.

Table 1: Audio signal processing parameters.

Parameter	Value / Description
Sampling rate	44.1 kHz
Audio channel	Mono(1)
Bit depth	16-bit
Normalization	Applied
DC offset removal	Applied
Signal segmentation	Effective transmission interval

4 Experimental Results

4.1 Experimental Setup

Experiments were conducted to evaluate the feasibility and practical performance of the proposed near-ultrasonic acoustic transmission system. The experiments focused on the frequency response of the acoustic link, the effects of transmission distance and device orientation, and the quality of the recovered audio signal.

Table 2: Experimental equipment and signal parameters.

Parameter	Value / Description
Sampling rate	44.1 kHz
Audio channel	Mono
Bit depth	16-bit
Transmission band	19–22 kHz
Reduced audio bandwidth	0–3 kHz
Original audio bandwidth	0–6 kHz
Modulation	SSB-AM
Carrier frequency	19 kHz
Playback device	Apple iPad 9
Microphone	Apple iPhone 17e
Processing software	Audacity 3.0.0[9]

The transmitter and receiver were implemented using consumer-grade audio hardware. The transmitted signal was reproduced through a built-in loudspeaker, while the acoustic signal was captured by a microphone and recorded through an audio input interface.

Several test signals were used to evaluate different aspects of the system. A sinusoidal signal was used for frequency-response measurements, while speech and music signals were used to evaluate the reconstruction quality of practical audio.

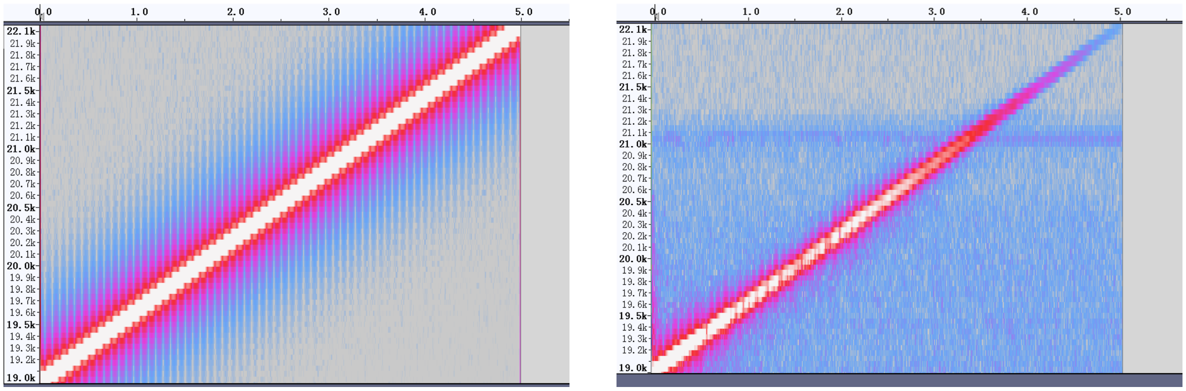
A frequency sweep from 19 to 22 kHz was used to characterize the high-frequency response of the complete acoustic transmission path. The sweep was generated at the transmitter and recorded by the receiver under the same experimental conditions used in the subsequent measurements. Figure 7 shows an example of the transmitted and recorded spectra of the transmitted and recorded sweep signals.

As shown in Fig. 7, the received sweep preserves the general frequency progression of the transmitted signal, while its amplitude decreases toward the upper end of the band.

4.2 Frequency Response and Transmission Feasibility

The first experiment was conducted to determine whether the proposed 19–22 kHz transmission band can be reliably reproduced and received using consumer-grade audio hardware. A frequency sweep covering the nominal transmission band was transmitted through the loudspeaker and recorded by the microphone under fixed experimental conditions.

The transmitted and received spectra are shown in Fig. 7. The received signal follows the frequency progression of the transmitted sweep, indicating that the acoustic link is capable of transmitting



(a) Transmitted signal.

(b) Received signal.

Figure 7: Comparison of the transmitted and received 19–22 kHz frequency spectra.

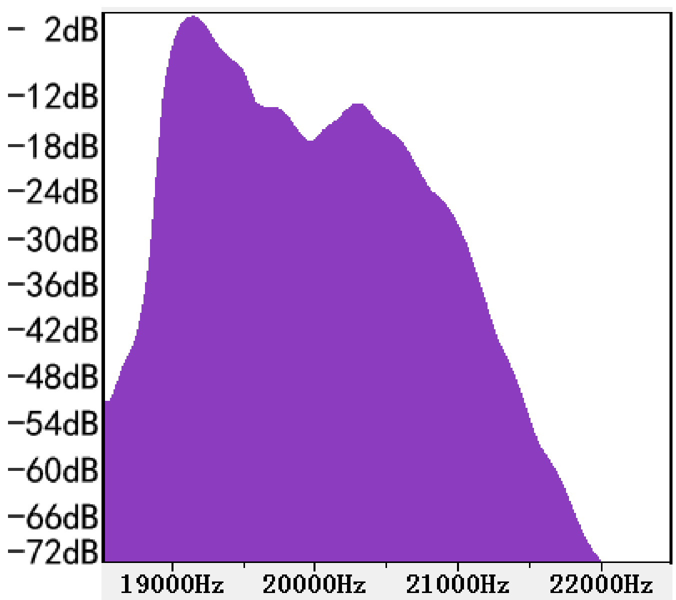
components throughout a substantial portion of the proposed frequency band.

To quantify the frequency-dependent attenuation, the amplitude of the received sweep was compared with that of the transmitted signal. The relative magnitude response was calculated as

$$H(f) = \frac{A_{\mathrm{rx}}(f)}{A_{\mathrm{tx}}(f)},$$

and expressed in decibels as

$$H_{\mathrm{dB}}(f) = 20\log_{10}\left(\frac{A_{\mathrm{rx}}(f)}{A_{\mathrm{tx}}(f)}\right).$$



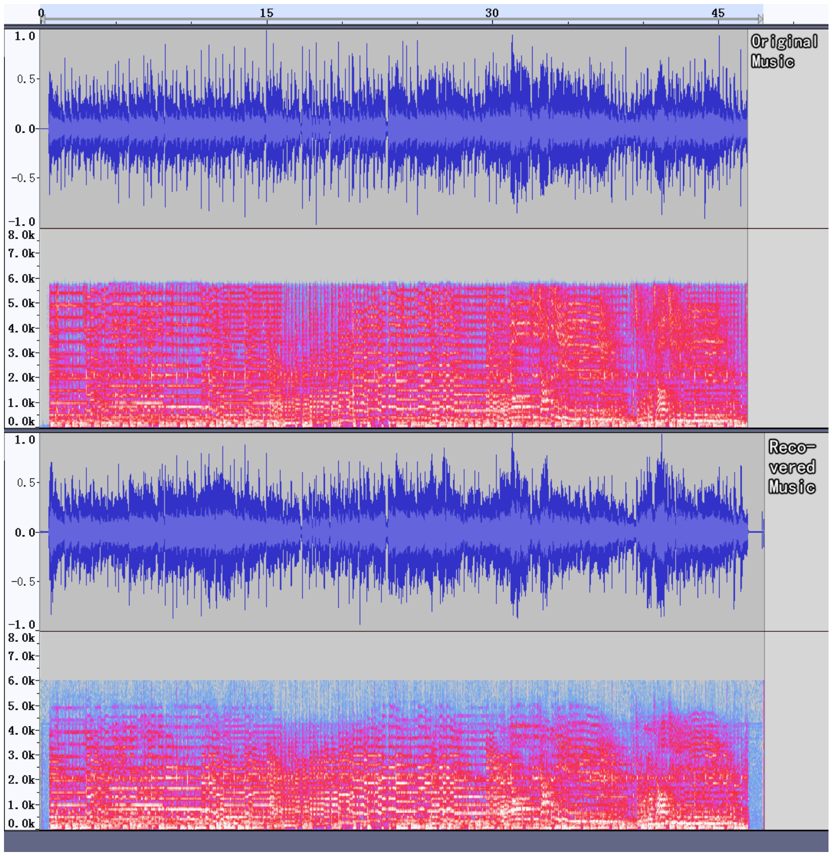
(a) Frequency response curve

Figure 8: Frequency response[8] curve from a microphone.

The measured response decreases progressively above 21 kHz, indicating that the upper portion of the nominal transmission band is more strongly attenuated. The measured response also illustrates the practical limitation of using the upper portion of the 44.1-kHz audio bandwidth. Although the Nyquist frequency is 22.05 kHz, the usable acoustic bandwidth is not necessarily flat up to this theoretical limit. The measured attenuation toward the upper end of the band is primarily determined by the frequency response of the loudspeaker, microphone, and associated audio hardware.

Despite the frequency-dependent attenuation, the received signal remained detectable over the tested range, and the transmitted sweep could be recovered after demodulation. This demonstrates the basic feasibility of near-ultrasonic analog audio transmission using the tested consumer-grade hardware.

To further verify end-to-end transmission, a test music was processed using the complete transmitter and receiver chains. The resulting recovered waveform was compared with the original signal after inverse pitch scaling.



(a) Original(top) vs. recovered(bottom) test audio spectrum

Figure 9: Comparison of the original and recovered music waveform diagram and audio spectrum[9]

Overall, the experimental results demonstrate that the proposed 19–22 kHz acoustic channel can transmit signals using consumer-grade audio hardware. Although the channel exhibits non-uniform frequency response and increased attenuation toward portions of the upper frequency range, the received signal remains sufficient for end-to-end analog audio reconstruction under the tested conditions.

4.3 Spatial and Environmental Effects

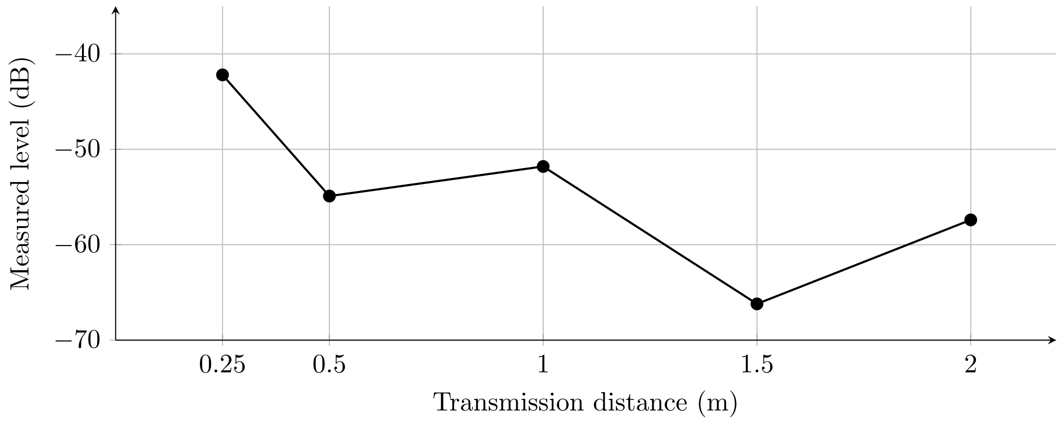


Figure 10: Measured signal level as a function of transmission distance.

The measured level does not exhibit a strictly monotonic decrease with distance. Instead, considerable fluctuations are observed, particularly at intermediate distances. This behavior may be attributed to reflections and multipath interference in the indoor acoustic environment.

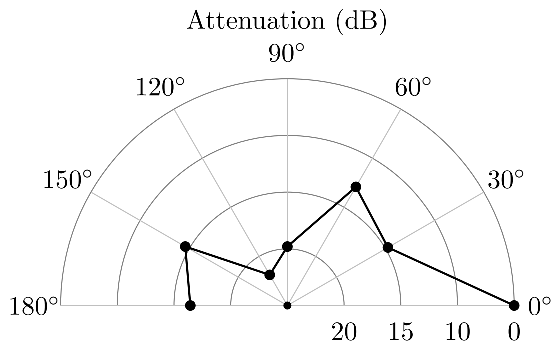


Figure 11: Measured attenuation as a function of transmitter orientation, relative to the 0° reference orientation.

The strongest received signal was observed at 0°, where the transmitter was oriented directly toward the receiver. The received level decreased substantially at larger angular offsets, with the largest measured attenuation of approximately 21.1 dB occurring at 120° relative to the 0° reference.

The response was not strictly monotonic, indicating that room reflections and the directional characteristics of the audio hardware may both contribute to the observed angular dependence.

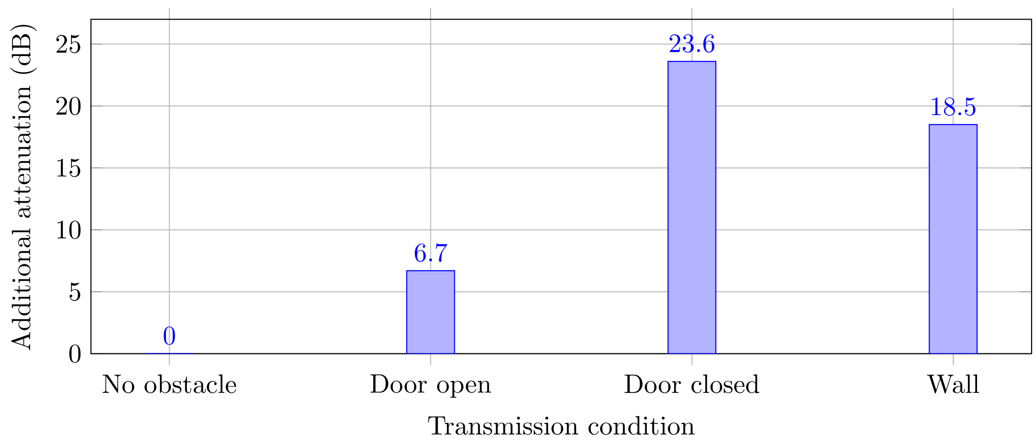


Figure 12: Additional attenuation introduced by different acoustic obstacles relative to the unobstructed condition.

The presence of obstacles produced substantially greater attenuation than the unobstructed condition. Closing the door resulted in an additional attenuation of approximately 23.6 dB, while transmission through the wall resulted in approximately 18.5 dB of attenuation.

4.4 Audio Quality Evaluation

To further evaluate the quality of the recovered audio, spectrograms were compared between the original and recovered signals. Unlike a single Fourier spectrum, a spectrogram provides a time-frequency representation and therefore allows changes in the spectral structure throughout the audio signal to be observed.

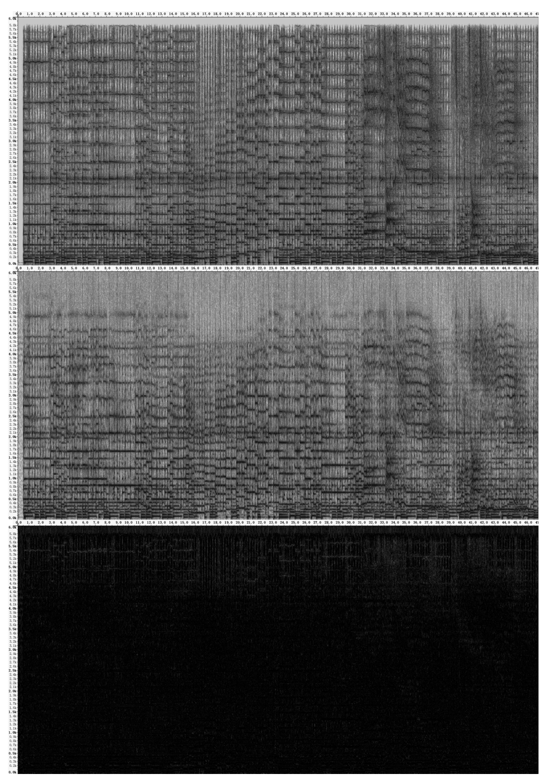


Figure 13: Spectrograms of the original audio, recovered audio, and their difference.

The spectrogram of the recovered signal retains the major time-frequency structures present in the original signal, indicating that the principal spectral characteristics of the audio are preserved after acoustic transmission and subsequent demodulation. However, differences can be observed in several regions of the spectrogram, with relatively larger differences observed in some higher-frequency components.

To visualize these differences, a difference map was generated by comparing the original and recovered spectrograms under identical spectrogram settings. Darker regions indicate greater spectral similarity, while brighter regions indicate larger differences. The difference map shows that the transmission process does not affect all frequency components equally, suggesting that the frequency response of the acoustic transmission path contributes to the observed distortion.

These results indicate that, although the recovered waveform may not be identical to the original waveform at the sample level, the major time-frequency characteristics of the audio can still be preserved. This observation is particularly relevant to the proposed system, where the objective is to transmit analog audio through a bandwidth-limited near-ultrasonic acoustic channel rather than to reproduce the original waveform exactly.

To further evaluate the spectral fidelity of the recovered audio, spectrograms of the original and recovered signals were compared using identical display settings. A difference map was generated to visualize the time-frequency regions affected by the transmission process.

The mean grayscale difference over the analyzed spectrogram region was 20.76 on a 0–255 scale, with a median of 16 and a standard deviation of 17.00. Approximately 7.52% of the pixels exhibited a grayscale difference of at least 50. These results indicate that the majority of the time-frequency representation remains relatively similar, although localized differences are present. To further investigate the frequency dependence of the distortion, the spectrogram was divided into six 1-kHz frequency bands. The mean grayscale difference was 15.65, 15.53, 15.22, 17.11, 26.05, and 35.03 for the 0–1, 1–2, 2–3, 3–4, 4–5, and 5–6 kHz bands, respectively. The difference remains relatively stable below 4 kHz but increases noticeably above 4 kHz, with the largest difference observed in the 5–6 kHz band. This frequency-dependent increase in spectrogram difference suggests that the higher-frequency components of the original audio are more strongly affected by the transmission system. This observation is consistent with the frequency-response limitations observed in the acoustic transmission channel.

The mean grayscale differences were normalized to the full grayscale range (0–255) and expressed as percentages.

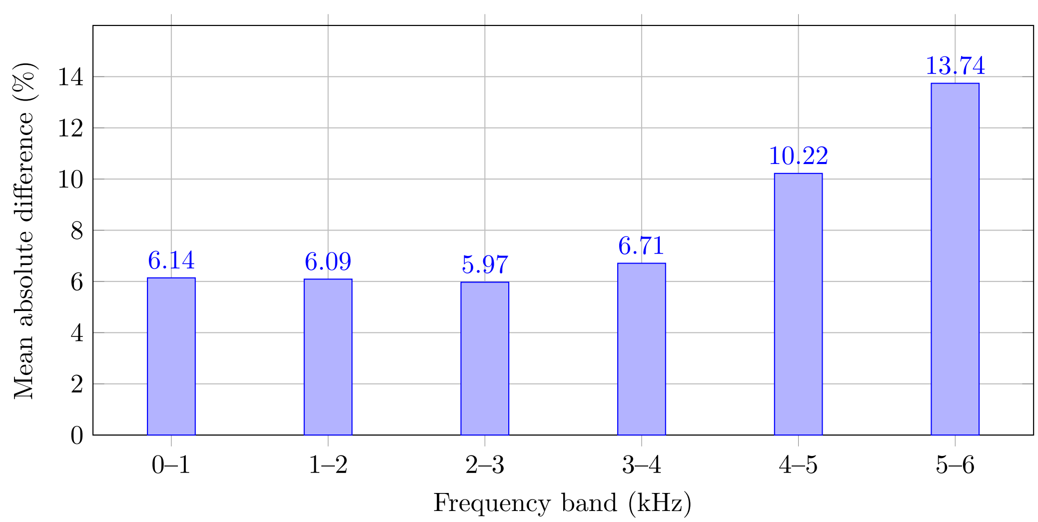


Figure 14: Mean absolute spectrogram difference across different frequency bands, expressed as a percentage of the full grayscale range.

5 Discussion

5.1 Limitations

Although the proposed scheme demonstrates a certain degree of feasibility, its performance relies on consumer-grade devices having sufficient transmission and reception capabilities near the upper end of their supported frequency ranges. The substantially larger spectrogram difference observed in the 5–6 kHz band suggests that the upper portion of the audio bandwidth is more strongly affected by the frequency response of the acoustic hardware. Consequently, the effective usable bandwidth is relatively narrow and may vary considerably between different devices.

In addition, the experimental results show that transmission distance, relative orientation, and acoustic obstacles can significantly affect the received signal strength. These factors limit the robustness and practical range of the proposed system. Therefore, the proposed scheme is not intended to replace conventional wireless communication technologies such as Wi-Fi or Bluetooth, but is instead more suitable for short-range experimental or other specialized applications.

5.2 Overall Performance

Overall, the experimental results support the feasibility of using consumer-grade audio devices for near-ultrasonic analog audio transmission. Audio can be transmitted within the proposed 19–22 kHz frequency range and subsequently recovered at the receiver. However, the quality of the higher-frequency components is more strongly affected by the frequency response of the hardware, while transmission distance, orientation, and acoustic obstacles also influence reception performance.

The main value of the proposed approach lies in its ability to utilize existing consumer-grade audio hardware to establish an acoustic communication link without requiring dedicated ultrasonic transducers. Potential applications include experimental acoustic communication, low-cost short-range analog signal transmission using existing audio hardware, audio hardware frequency-response testing, and educational or experimental demonstrations.

6 Conclusion

This paper presented a near-ultrasonic analog audio transmission system based on single-sideband amplitude modulation and consumer-grade audio hardware. By reducing the audio bandwidth through pitch scaling, the resulting signal could be translated into the 19–22 kHz region and transmitted through an acoustic channel.

Experimental results demonstrated that the proposed system can recover intelligible audio using commonly available speakers, microphones, and sound cards. The performance was affected by transmission distance, orientation, acoustic obstacles, and the frequency response of the hardware. Spectrogram analysis further indicated that the higher frequency components of the recovered audio were more strongly affected than the lower-frequency components.

Overall, the results demonstrate the feasibility of using the near-ultrasonic portion of conventional audio systems as a low-cost acoustic communication channel, while also highlighting the practical limitations imposed by consumer audio hardware.

References

[1] P. G. Stelmachowicz, K. A. Beauchaine, A. Kalberer, and W. Jesteadt, "Normative thresholds in the 8- to 20-kHz range as a function of age," *The Journal of the Acoustical Society of America*, vol. 86, no. 4, pp. 1384–1391, 1989, doi: 10.1121/1.398698.

[2] M. Wang, Y. Ai, Y. Han, Z. Fan, P. Shi, and H. Wang, "Extended high-frequency audiometry in healthy adults with different age groups," *Journal of Otolaryngology–Head & Neck Surgery*, vol. 50, 2021, doi: 10.1186/s40463-021-00534-w.

[3] O. Riva and J. Kangasharju, *"Challenges and lessons in developing middle-ware on smartphones,"* Computer,vol.41,no.10,pp.23–31,Oct.2008.

[4] "The frequency range of human hearing is approximately 20 Hz to 20 kHz.", *Designing Sound*, Chapter 1 "Foundations of Audio"

[5] J. Driedger and M. Müller, "A Review of Time-Scale Modification of Music Signals," *Applied Sciences*, vol. 6, no. 2, p. 57, 2016, doi: 10.3390/app6020057.

[6] B. P. Lathi and Z. Ding, *Modern Digital and Analog Communication Systems*, 4th ed., Oxford University Press, New York, 2009.

[7] International Organization for Standardization, *ISO 9613-1:1993: Acoustics—Attenuation of Sound During Propagation Outdoors—Part 1: Calculation of the Absorption of Sound by the Atmosphere*, International Organization for Standardization, Geneva, 1993.

[8] J. W. Cooley and J. W. Tukey, "An Algorithm for the Machine Calculation of Complex Fourier Series," *Mathematics of Computation*, vol. 19, no. 90, pp. 297–301, Apr. 1965.

[9] Audacity Team, "Audacity 3.0.0" *https://www.audacityteam.org/*