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Analysis of Ultra-Wire Band Wireless Audio Transceiver for High-Resolution Audio Transmission

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ABSTRACT

This paper presents a comprehensive evaluation of a novel Ultra-Wideband (UWB) wireless audio transceiver (SPARK Microsystems SR1020), which achieves lossless transmission of high-resolution audio up to (96 kHz/24-bit) with latencies below 10 milliseconds matching the quality of traditional wired connections. The evaluation utilizes a UWB-enabled USB dongle connected to a USB audio-compliant host for managing both the transmission and reception of audio data. Additionally, the system incorporates a printed circuit board assembly (PCBA) configured as a headset reference design. This PCBA receives audio from the dongle and outputs it via I2S, S/PDIF, or a Digital-to-Analog Converter (DAC) for analog reproduction.

To ensure the accuracy of the audio quality assessment, various audio signals—including pure tones, multi-tones, and non-sinusoidal waveforms designed to challenge the audio transmission path—are generated on a personal computer and processed through the evaluation platform. The audio output is then captured from the digital outputs to avoid potential quality degradation from the DAC. Audio fidelity is quantitatively analyzed using an Audio Precision measurement apparatus, a standard in audio analysis, to compare the performance of the UWB system against a conventional USB-to-S/PDIF converter, which serves as the wired reference. Additionally, data is collected and presented from commercially available Bluetooth wireless streaming modules supporting APT-X HD, APT-X LL (low latency), and LDAC.

The experimental setup focuses on tests conducted within a 1-meter line-of-sight environment, reflecting common use cases such as in wireless studio and gaming headsets. The frequency domain characteristics of the audio output are analyzed using a fast Fourier transform (FFT). Results from the tests confirm that the UWB transceiver reliably delivers artifact-free audio, matching the quality of traditional wired connections while maintaining consistent sub-10-millisecond latencies. This validation highlights the potential of UWB technology as a superior alternative for high-fidelity wireless audio applications.

Further research is proposed to compare this UWB technology against other advanced wireless audio transmission technologies. This future work aims to elaborate on the comparative audio quality and latency metrics, providing a deeper insight into the distinctive performance advantages of UWB technology.

Introduction

The pursuit of optimal sound fidelity combined with minimal latency in wireless audio transmission remains a significant challenge within the field of communication technologies. Conventional wireless technologies, such as Bluetooth, Bluetooth Low Energy (BLE) and Wi-Fi, encounter notable limitations that affect their efficacy in transmitting high-resolution audio formats. These limitations are intrinsically linked to the design constraints of these technologies, which hinder their ability to deliver both high audio quality and low latency concurrently [1].

Bluetooth and Bluetooth Low Energy (BLE) are primarily engineered for intermittent data transfer with a focus on power efficiency, which imposes significant constraints on bandwidth and, consequently, ability to support high-resolution audio streaming. Despite advancements in BLE 5.0 and subsequent iterations, which offer enhanced data rates, these improvements are insufficient for the transmission of high-resolution audio (e.g., 96 kHz/24-bit) without introducing compression artifacts. Additionally, BLE's latency, influenced by connection intervals and data packet sizes, often exceeds the stringent latency requirements essential for real-time audio applications, including professional audio production and interactive gaming. Several commercially successful audio compression technologies have been applied to Bluetooth audio transmission, such as APT-X and LDAC, in an attempt to deliver High-Resolution audio. But these all deploy algorithms that must sacrifice some data in order to fit within the Bluetooth data pipe [2,3].

Wi-Fi technology, while capable of supporting higher data transfer rates relative to Bluetooth and BLE, also presents several limitations in the context of high-resolution audio transmission. The performance of Wi-Fi networks is susceptible to interference from other devices, network congestion, and variations in signal strength. These factors contribute to latency and jitter, which can adversely affect audio playback quality, particularly in scenarios requiring precise synchronization, such as interactive gaming and professional audio production. Furthermore, the latency inherent in Wi-Fi protocols, including the time required for acknowledgments and retransmissions, often surpasses the low-latency thresholds necessary for a seamless audio experience [4].

UWB technology offers a compelling alternative due to its unique advantages. UWB operates over a broad frequency spectrum, which allows sufficiently high data transfer rates and the ability to transmit high-resolution audio without degradation. The wide bandwidth of UWB enables it to support uncompressed audio transmission and maintain exceptionally low latency, essential for real-time applications such as professional audio production, interactive gaming, and high-end consumer audio systems. Additionally, UWB technology is less susceptible to interference from other wireless devices due to its use of a wide frequency range. This characteristic contributes to stable and reliable performance, even in environments with high wireless device density. The low latency of UWB also enables accurate synchronization of audio signals with video or other media elements, further enhancing the user experience.

Achieving latencies below 10 milliseconds is imperative for several reasons. In professional audio production, such low latency is essential for real-time audio monitoring and accurate synchronization of audio signals with video or other media elements, thereby avoiding perceptible delays that could disrupt the production process. In interactive gaming, reduced latency is crucial for ensuring that audio cues are delivered synchronously with on-screen actions, thereby enhancing the immersive experience and maintaining responsiveness in dynamic environments. For high-end consumer audio systems, minimal latency ensures that audio and visual elements are perfectly aligned, thereby improving the overall user experience by eliminating perceptible lag and ensuring seamless integration. Thus, the development of a UWB transceiver with latencies below 10 milliseconds addresses these critical requirements, presenting a robust wireless alternative to traditional wired connections [5, 6].

This work presents a comprehensive evaluation of an advanced low-power UWB wireless audio transceiver (SPARK Microsystems SR1020), demonstrating its capability to achieve lossless transmission of high-resolution audio with latencies under 10 milliseconds. The work will provide an in-depth discussion of the experimental setup used for characterization, the hardware implemented, and the results obtained. This detailed analysis highlights the transceiver's performance and reliability, showcasing its potential as a leading solution for high-fidelity wireless audio applications.

Methods

The primary aim of this work is to evaluate the latency and capability of the UWB wireless audio transceiver in transmitting high-resolution audio. The evaluation focuses on the transmission and reception processes under controlled conditions, specifically targeting the delivery of high-fidelity audio signals with latencies below 10 milliseconds.

This work does not address packet loss due to channel impediments or interference. Moreover, the objective is to measure the system's latency in optimal conditions where retransmissions do not occur, thereby providing a clear view of the UWB transceiver's true capabilities.

Figure 1 illustrates the experimental setup for the reference performance system, which will serve as a benchmark for comparison. Figure 2 shows the evaluation system that consists of a UWB-enabled USB dongle connected to a USB audio-compliant host, which manages both the transmission and reception of audio data. The dongle pairs with a PCBA configured as a headset reference design. This PCBA is responsible for receiving audio from the dongle and outputting it through I2S, S/PDIF, or a digital-to-analog converter (DAC) for subsequent analog reproduction.

To ensure a comprehensive analysis of the audio transmission quality, various stimulus audio test signals—including pure tones, multi-tones, and complex non-sinusoidal waveforms—are generated using a digital audio workstation (DAW) on a personal computer. These signals are designed to rigorously test the integrity and performance of the transmission path. The output is captured directly from the digital S/PDIF interface to eliminate any potential distortion introduced by the digital-to-analog conversion processes.

For precision, the methodology relies heavily on high-resolution FFT analysis using an industry-standard Audio Precision system. This allows for a

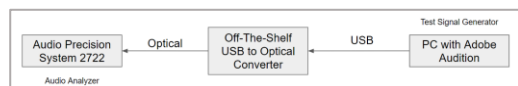


Figure 1. Test set up for reference comparison

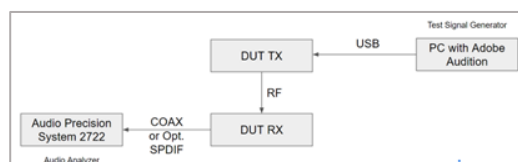


Figure 2. Test set up for DUTs

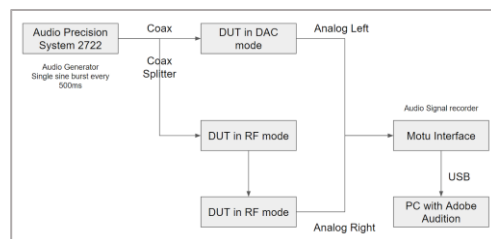


Figure 3. Latency measurement of wireless DUTs

detailed visual analysis of any distortion components within the frequency domain.

The USB interface was selected as the input method for all devices under test (DUTs) because the UWB tested device uses USB exclusively for audio input. To ensure uniformity across all tests, the other DUTs were also chosen for their support of a USB input interface. Additionally, all devices, including the UWB DUT, were equipped with an S/PDIF output. This consistent configuration guaranteed that testing conditions were the same for each device.

Test signals, generated at the native sample rates and bit depths appropriate to each test, were sent to the DUTs via USB. The DUTs wirelessly transmitted these signals to their respective companion receivers, which then output the audio through S/PDIF for analysis by the Audio Precision system. A USB-to-S/PDIF converter was used as a reference for comparison, ensuring that the audio data remained uncompressed and was transmitted without using a wireless link, serving as a "straight wire" benchmark.

Due to the availability constraints of the UWB DUT hardware and software, as well as the limitations of commercially available modules at the time of this study, the initial data sets were compiled using 48 kHz at 16-bit and 96 kHz at 16-bit audio formats.

Figure 3 illustrates the test setup used for evaluating latency with the LDAC and APT-X wireless transmission DUTs.

The retransmission impact on the latency will depend on the mechanism used and the implementation being not related directly to the physical layer used.

Results

All subsequent plots are presented on a uniform scale with the vertical axis ranges from -180 dBFS to $+6$ dBFS, while the horizontal axis spans from 0 Hz to 24 kHz in a linear format.

48 KHz/16 bits Audio Measurements

Figure 4 illustrates the FFT analysis of the output from the 'Straight Wire' reference USB-to-S/PDIF converter. The input signal was a pure tone sine wave at 1 kHz, with a level of -1 dBFS, sampled at 48 kHz with 16-bit resolution, and applied via PC USB. It is important to note that the FFT computations employed very narrow frequency bins, resulting in a noise floor significantly below the theoretical minimum for 16-bit audio. This observation is typical for such computations.

Figure 5 illustrates the FFT analysis of the output signal from the UWB DUT pure tone sine wave at 1 kHz. The input signal, set at a level of -1 dBFS, was sampled at 48 kHz with 16-bit resolution and transmitted from a PC via USB to the UWB dongle. Notably, a slight difference in the noise floor is observed when compared to the reference system.

While the precise cause of this variation remains undetermined, it is speculated that the discrepancy may result from the use of a coaxial interface on the UWB DUT as opposed to the optical interface employed in the reference system.

Nevertheless, this variance is deemed negligible. Crucially, the UWB DUT does not introduce any detectable harmonics into the audio signal, maintaining parity with the reference system.

Figure 6 presents the FFT analysis of the output signal from the LDAC DUT when a 1 kHz tone at 48 kHz, 16-bit resolution is applied.

The analysis indicates that LDAC exhibits negligible deviation from the reference signal, maintaining high fidelity in the output.

Figure 7 displays the FFT analysis for the APT-X HD DUT under the same conditions.

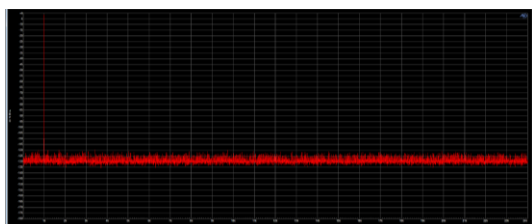


Figure 4. Reference 1kHz tone at 48kHz, 16-Bit

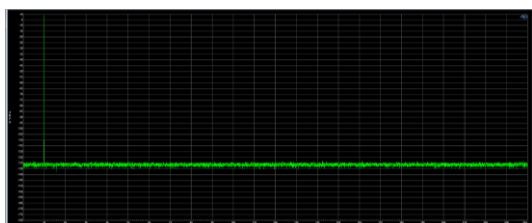


Figure 5. UWB DUT 1kHz tone at 48kHz, 16-Bit

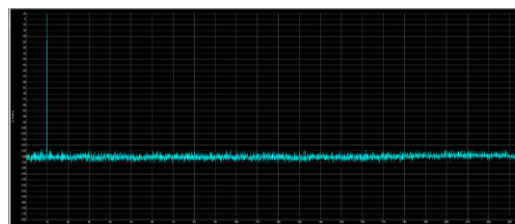


Figure 6. LDAC DUT 1kHz tone at 48kHz, 16-Bit

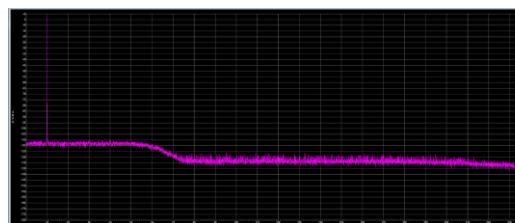


Figure 7. APT-X HD DUT 1kHz tone at 48kHz, 16-Bit

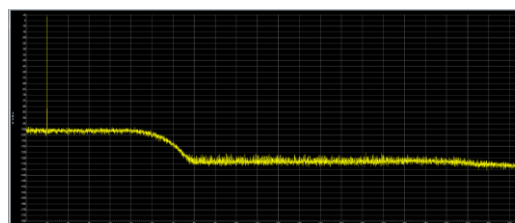


Figure 8. APT-X LL DUT 1kHz tone at 48kHz, 16-Bit

Here, the analysis reveals a noticeably elevated noise floor compared to the reference, suggesting that the APT-X HD codec introduces additional noise into the audio signal.

Figure 8 shows the FFT analysis for the APT-X LL DUT, which similarly demonstrates an elevated noise floor. This increase in the noise floor in both APT-X variants highlights potential compromises in audio quality, particularly in scenarios requiring high fidelity and minimal noise interference.

Figure 9 presents the FFT analysis of the Reference system's output when subjected to a multi-tone input signal, consisting of four distinct frequencies—1 kHz, 2.5 kHz, 11.5 kHz, and 15 kHz—all of equal amplitude, combined to a total level of -1 dBFS, and sampled at a rate of 48 kHz with 16-bit resolution. The analysis reveals a clean spectrum with no added harmonics, serving as the benchmark for comparison.

Figure 10 shows the FFT analysis of the UWB DUT under the same conditions. The UWB DUT demonstrates excellent performance, with no additional harmonics detected, thus maintaining spectral purity comparable to that of the Reference system. This indicates the UWB DUT's capability to transmit complex audio signals without introducing significant distortion, making it suitable for high-resolution audio applications.

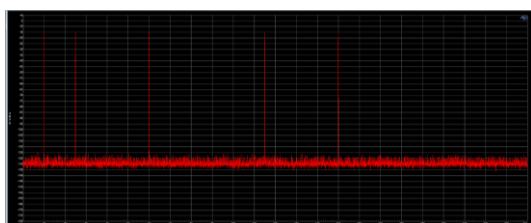


Figure 9. Reference, Multitone Output, 48kHz, 16-Bit

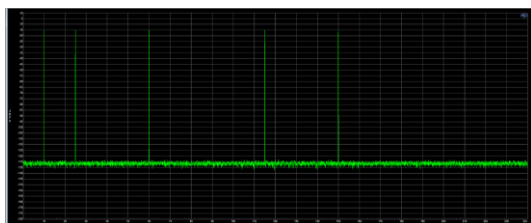


Figure 10. UWB DUT, Multitone Output, 48kHz, 16-Bit

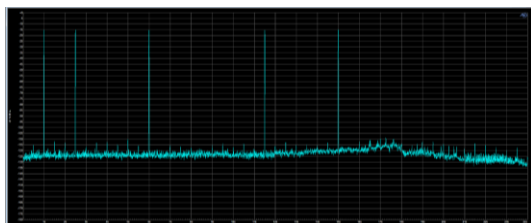


Figure 11. LDAC, multi-tone output, 48kHz, 16-Bit

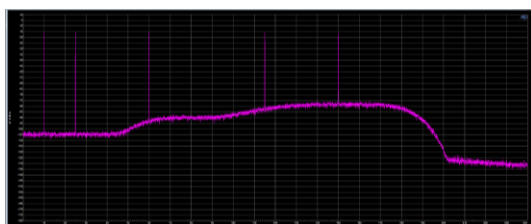


Figure 12. APT-X HD, multi-tone output, 48kHz, 16-Bit

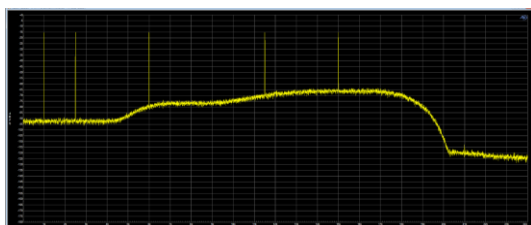


Figure 13. APT-X LL, multi-tone output, 48kHz, 16-Bit

In contrast, Figure 11 depicts the FFT analysis for the LDAC codec. Here, the analysis reveals an elevated noise floor in the higher frequency ranges, particularly above 10 kHz. This suggests the onset of codec-induced noise that could compromise audio fidelity under certain conditions. While subtle, this noise elevation could affect perceived audio quality, especially in applications demanding ultra-low noise performance.

Figure 12 and Figure 13 present the FFT analyses for the APT-X HD and APT-X LL codecs, respectively. Both codecs display more pronounced issues, with continued noise masking evident across the spectrum. This noise masking, characterized by a broad increase in the noise floor, could obscure finer details in the audio signal, particularly in complex musical passages. The analysis highlights potential limitations in the APT-X codecs when it comes to maintaining audio transparency in high-fidelity contexts, where even minor deviations from the reference signal can be perceptible to discerning listeners.

96 KHz/16 bits Audio Measurements

The subsequent data and figures present results derived from audio signals sampled at 96 kHz with a 16-bit resolution (96k/16). It is pertinent to note that while the LDAC codec is included in these analyses, the APT-X codec is excluded due to the incompatibility of the DUTs with 96 kHz sampling rates in APT-X mode. All subsequent plots are standardized on a consistent scale: the vertical axis ranges from -180 dBFS to $+6$ dBFS, and the horizontal axis spans from 0 Hz to 48 kHz on a linear scale.

Figure 14 illustrates the FFT analysis of the Reference system's output with a 1 kHz pure tone input signal, sampled at 96 kHz with a 16-bit depth.

Figure 15 depicts the FFT analysis of the output from the UWB DUT under the same conditions. The analysis shows how the UWB DUT performs with the 1 kHz pure tone input at 96 kHz sampling rate and 16-bit resolution. No harmonics are seen and signal fidelity is maintained.

Figure 16 presents the FFT analysis of the LDAC DUT's output, again with a 1 kHz pure tone input sampled at 96 kHz and 16-bit depth. LDAC exhibits negligible deviation from the reference signal.

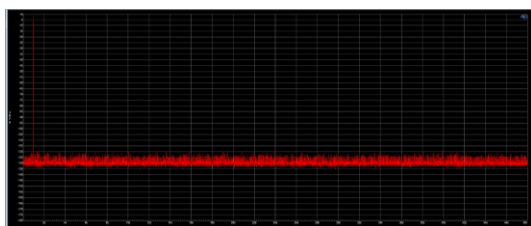


Figure 14. Reference, FFT of output with 1kHz pure tone input produced at 96kHz sample-rate, 16-bit depth

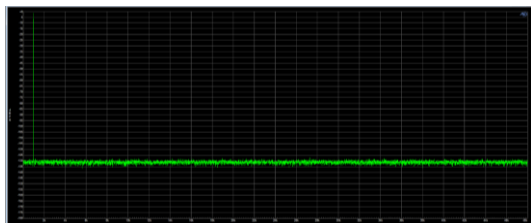


Figure 15. UWB DUT FFT output with 1kHz pure tone input produced at 96kHz sample-rate, 16-bit depth

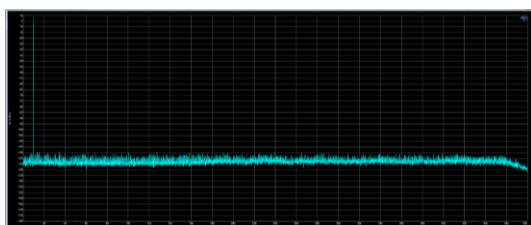


Figure 16. LDAC DUT output with 1kHz pure tone input produced at 96kHz sample-rate, 16-bit depth

Figure 17 provides the FFT analysis of the Reference system's output when subjected to a multi-tone sine wave input consisting of frequencies at 10 kHz, 20 kHz, 30 kHz, and 40 kHz, all at equal amplitude, with a combined total level of -1 dBFS, sampled at 96 kHz with a 16-bit depth and applied via PC USB.

Figure 18 shows the FFT analysis for the UWB DUT under identical input conditions. This figure demonstrates that the UWB DUT maintains spectral purity, as it does not introduce any additional harmonics or distortion compared to the Reference system. This is notable considering the elevated frequencies utilized in this test, and clearly exposes the benefits of the throughput capabilities of the UWB DUT utilized.

In contrast, Figure 19 illustrates the FFT analysis for the LDAC DUT. This analysis reveals a significant increase in noise and artifacts at higher frequencies, which compromises audio fidelity. Such artifacts are likely to be perceptible to listeners who can discern differences in 96 kHz sampled audio.

Figure 20 provides the FFT analysis of the reference system's output when subjected to a custom multi-tone inverted-sine wave input. The input signal consists of multiple tones with a combined total amplitude of -3 dBFS, sampled at 96 kHz with a 16-bit resolution, and transmitted via PC USB.

Figure 21 presents the corresponding FFT analysis for the UWB DUT. The results demonstrate that the UWB DUT maintains spectral integrity by not introducing any measurable harmonics or additional distortion, thereby again performing comparably to the Reference system, regardless of the spectral complexity of the a custom multi-tone inverted-sine wave input.

In contrast, Figure 22 shows the FFT analysis for the LDAC DUT. This analysis reveals the presence of significant artifacts and the absence of several harmonics from the applied signal, indicating a notable deterioration in signal fidelity relative to both the Reference and UWB DUT.

As can be inferred from these tests, the UWB DUT exhibits superior performance by preserving signal integrity and avoiding additional harmonic

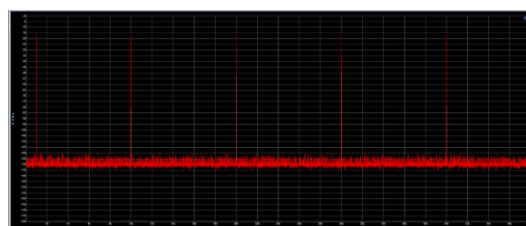


Figure 17. Reference FFT multi-tone sine at 96kHz, 16-bit

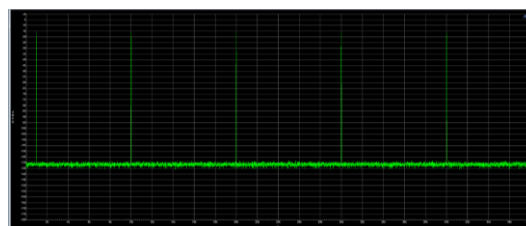


Figure 18. UWB DUT FFT multi-tone sine at 96kHz, 16-bit

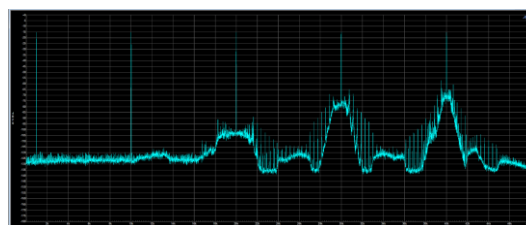


Figure 19. LDAC DUT FFT multi-tone sine at 96kHz, 16-bit

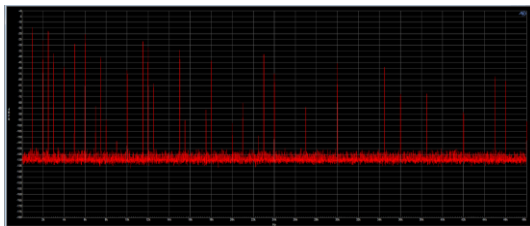


Figure 20. Reference FFT multi-tone inverted-sine at 96kHz, 16-bit

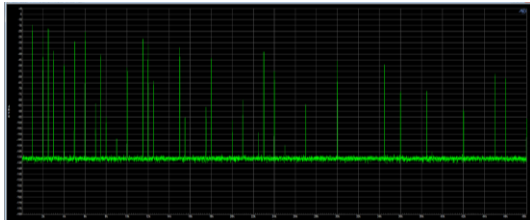


Figure 21. UWB DUT FFT multi-tone inverted-sine at 96kHz, 16-bit

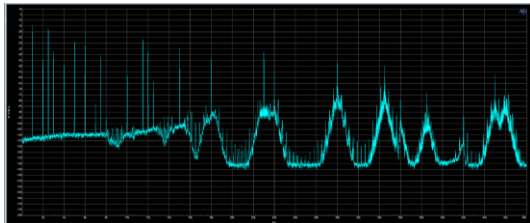


Figure 22. LDAC DUT FFT multi-tone inverted-sine at 96kHz, 16-bit

distortion, in stark contrast to the LDAC DUT, which introduces considerable artifacts and missing harmonics. These findings underscore the importance of codec selection in maintaining high audio fidelity. The results prove that for applications demanding precise wireless audio reproduction, the UWB DUT is the preferred choice over LDAC.

Latency Measurements

Latency measurements were conducted using the experimental setup illustrated in

Figure 3. This configuration was employed to systematically evaluate and compare the latency performance of the UWB DUT, LDAC, APT-X HD, and APT-X LL related to the straight wire.

Figure 23 presents the latency measurement for the LDAC codec, which was approximately 200 milliseconds. Figure 24 details the latency for the APT-X HD codec, recorded at approximately 280 milliseconds. In contrast, Figure 25 depicts the latency for the APT-X LL codec, which was measured to be approximately 30 milliseconds.

Figure 26 illustrates the latency measurement of the UWB DUT, which was evaluated at a sampling

rate of 96 kHz and a resolution of 24 bits. The results demonstrate a latency of approximately 7 milliseconds, which is substantially lower than the latency observed in the other tests conducted under different configurations. Notably, this low latency was achieved while maintaining superior audio quality as per the spectral plots previously shown, making it a highly favorable outcome in comparison to the other measurements performed, which

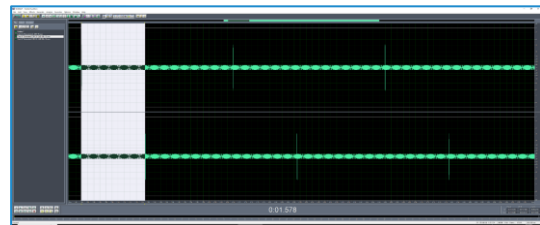


Figure 23. LDAC latency measured approximately 200 ms at 48 kHz/16bits

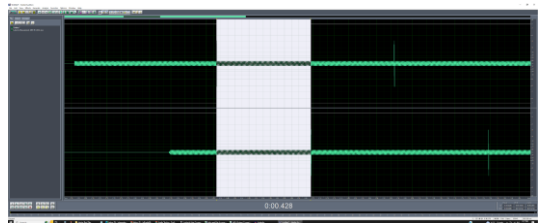


Figure 24. APT-X HD latency measured approximately 280 ms at 48 kHz/16bits

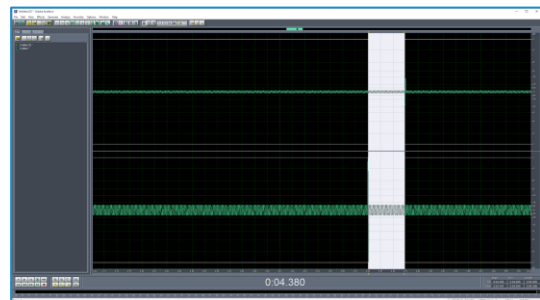


Figure 25. APT-X LL latency measured approximately 30 ms at 48 kHz/16bits

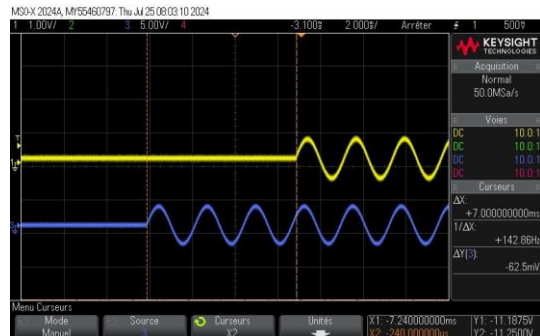


Figure 26. UWB latency measured of 7 ms at 96 kHz/24bits

exhibited both higher latency and lower sound quality.

These measurements were carefully conducted to accurately quantify the latency introduced by various audio transmission technologies. By analyzing and comparing the latency values of the UWB DUT, LDAC, APT-X HD, and APT-X LL, this study provides a comprehensive evaluation of their performance. Such an analysis is crucial for applications that demand precise synchronization and minimal delay, such as real-time audio processing, interactive gaming and multimedia synchronization.

Conclusions

The evaluation of the UWB wireless audio transceiver highlights its superior performance in delivering high-resolution, lossless audio with consistently low latencies, well below 10 milliseconds. Utilizing a UWB-enabled USB dongle and a headset reference design, this study comprehensively assessed the system's capabilities across multiple test scenarios, including rigorous FFT analysis of audio signals sampled at 48 kHz and 96 kHz. The UWB transceiver maintained exceptional audio fidelity, comparable to wired connections, while outperforming Bluetooth-based codecs like LDAC, APT-X HD, and APT-X LL, which showed elevated noise floors and artifacts that compromised audio quality.

Latency measurements further emphasized the UWB transceiver's advantages, with sub-10-millisecond latencies proving crucial for real-time applications such as professional audio production and interactive gaming. In contrast, the LDAC and APT-X HD codecs exhibited latencies of 200 ms and 280 ms, respectively, while the APT-X LL codec, though designed for low latency and not high quality audio transport, recorded 30 ms, still lagging behind UWB's performance.

In conclusion, the UWB transceiver's ability to consistently deliver high-fidelity audio with minimal latency positions it as a leading solution for high-resolution wireless audio applications. Future research will focus on expanding the dataset and comparing UWB with other emerging wireless audio technologies to further investigate its performance advantages in a broader range of use cases.

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