Professional Networked Audio over DECT NR+: Feasibility and Performance

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***Abstract*—DECT NR+, also known by its official designation DECT-2020 NR, is a new radio standard developed and maintained by the European Telecommunications Standards Institute (ETSI). It is the world's first non-cellular 5G technology and fulfils the IMT-2020 requirements for both massive Machine Type Communications (mMTC) and Ultra-Reliable and Low Latency Communications (URLLC). The latter makes it a promising candidate for applications from the Programme Making and Special Events (PMSE) sector. In this paper, we investigate the feasibility of using DECT NR+ as a cable replacement solution for networked audio systems, with Dante serving as an example. Dante is an AV connectivity platform developed by Audinate that has become the de facto industry standard for digital audio networking. Using an open-source Software-Defined Radio (SDR) implementation of DECT NR+, we set up a wireless link between two Dante-enabled devices. Based on the metrics reported to Dante Controller, we assess the performance of Dante in the DECT NR+ wireless network and demonstrate that an average one-way latency of 1.7 ms can be achieved.**

***Index Terms*—DECT-2020, NR+, 5G, URLLC, PMSE, Dante.**

I. Introduction and Related Work

Digital audio networks are an essential part of professional sound systems and have become indispensable for a wide range of Programme Making and Special Events (PMSE) activities. By enabling the real-time transport of high-fidelity audio over ubiquitous Ethernet networks, they are prevalent in environments such as concert halls, theme parks, stadiums, and conference venues [1]. Audio over IP (AoIP) protocols such as Dante [2], developed by Sydney-based Audinate and widely adopted across the industry, rely on a deterministic communication behavior to meet the strict Quality of Service (QoS) requirements of professional audio applications. For example, in live stage performances, where artists are equipped with In-Ear Monitoring (IEM) systems, the Round-Trip Time (RTT) or so-called "mouth-to-ear latency"—measured between analog input (microphone, instrument) and analog output (IEM)—must not exceed 4 ms. Considering the input signals are sent to a mixing desk for audio processing, which can take up to 2 ms, professional live productions typically have a latency budget of no more than 1 ms per audio link [3], [4].

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In addition to the requirement of low latency, digital audio networks demand a precise synchronization of the media clocks to allow a sample-aligned audio reproduction across networked devices. Modern IP-based systems usually employ the IEEE 1588 Precision Time Protocol (PTP) or the IEEE 802.1AS generalized PTP (gPTP) profile for this purpose [2]. PTP-based clocking mechanisms are susceptible to network jitter, which becomes apparent when time-synchronizing devices over a Wide Area Network (WAN) [5]. In large networks, the PTP messages pass through multiple hops, including non-PTP-aware switches, and experience variations in packet delay. Such dynamic network conditions impair the receiving device's ability to adjust its clock to that of the transmitter, ultimately causing the receiver's Phase-Locked Loop (PLL) to unlock. Therefore, spatially distributed production scenarios such as in [6], where performance, production, and audience are spread across different geographical locations, demand a different solution. Synchronizing devices between geographically separated areas is typically achieved by having one device in each local network derive its clock from the Global Positioning System (GPS) and serve as time master for the other devices on that network. This way, Dante was successfully deployed for a long-distance audio transmission between Berlin and Edinburgh [7], and for a Networked Music Performance (NMP), spanning New York City, Washington, D.C., and Nashville, in collaboration with the Blue Note Entertainment Group [8]. In both cases, rather than running PTP across the WAN, PTP domains were synchronized using GPS.

AoIP solutions, such as AES67 and Dante, are designed for use in locally confined, Ethernet-based IP networks. Professional applications require reliable, low-latency operation, making a deployment in wireless networks challenging—

primarily due to the often non-deterministic behavior of wireless links. In [9], the authors evaluated AES67 over IEEE 802.11 (i.e., Wi-Fi) and observed significantly reduced performance compared to an Ethernet connection. To ensure reliable operation, they identified two options: Either send only the audio data over Wi-Fi while maintaining PTP and discovery messaging via Ethernet, or employ SMPTE 2022-7 to redundantly transmit over two Wi-Fi networks on separate frequency bands (2.4 GHz and 5 GHz). Whereas the former approach does not constitute a fully wireless solution, the latter requires twice the radio hardware and results in inefficient spectrum utilization.

The limited suitability of Wi-Fi for professional live sound and studio applications is largely attributable to the way it manages access to the wireless medium. IEEE 802.11 implements Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), a contention-based Medium Access Control (MAC) mechanism which uses a random backoff process to prevent collisions [10]. The contention window (i.e., backoff interval) increases exponentially with each retransmission attempt until either the packet is successfully transmitted or the window reaches its maximum size. Subsequent retransmissions use random backoff values within this maximum window until the packet is delivered or the retry limit is exceeded, at which point the packet is dropped. Consequently, CSMA/CA inherently introduces random access delays, resulting in jitter and non-deterministic packet delivery. Compounding this issue is the fact that Wi-Fi operates in the unlicensed Industrial, Scientific, and Medical (ISM) frequency bands (2.4 GHz, 5 GHz, and increasingly 6 GHz), which are shared with other technologies (e.g., Bluetooth) and often subject to significant congestion. A shared radio environment increases the likelihood of interference and collisions, further exacerbating the limitations of the CSMA/CA mechanism.

Similarly, 3rd Generation Partnership Project (3GPP) 5th Generation (5G) New Radio (NR) cellular mobile communication networks have been explored for professional live audio productions. However, like Wi-Fi, they fall short of meeting the stringent technical requirements of audio PMSE use cases and remain some distance from being fully viable for such applications [3]. A 3GPP Release 15-compliant 5G NR testbed operating in Standalone (SA) mode, implemented using a combination of open-source and Commercial Off-The-Shelf (COTS) components, was presented in [4] and evaluated in the context of a professional live audio production scenario. With an audio packet periodicity of 2.5 ms, the system achieved an overall transmission latency of approximately 25.5 ms at the 99.9999th percentile, exceeding the latency requirements of the application and highlighting a key limitation of current 3GPP technology for such use cases. In the context of NMPs, the authors of [11] analyzed the behavior of a pre-commercial public 5G Non-Standalone (NSA) network and a private 5G SA network with Multi-access Edge Computing (MEC) infrastructure. Neither setup achieved an end-to-end latency below 21 ms. On top of these technical constraints, there are also architectural and operational aspects to consider. 5G NR requires centralized infrastructure, which is difficult to align with the nomadic, ad hoc nature of typical PMSE deployments. Moreover, the involvement of a Mobile Network Operator (MNO) can reduce flexibility, increase operational complexity, and raise costs.

As of today, the shortcomings of standard technologies such as Wi-Fi and 5G NR necessitate the use of highly optimized proprietary RF solutions in professional wireless audio equipment. DECT NR+, an open, non-cellular 5G standard developed in Europe, has emerged as a notable alternative to traditional 3GPP cellular networks and is considered a strong candidate for PMSE applications [12]. First measurements evaluating the DECT NR+ physical layer performance for professional live audio—using the only commercially available modem to date (i.e., the nRF91 series by Nordic Semiconductor)—show promising results [13].

In this paper, we assess the feasibility of DECT NR+ as a cable replacement solution for professional audio networks. Using an open-source Software-Defined Radio (SDR) implementation of DECT NR+, we set up an experimental wireless link between two Dante-enabled devices and conduct measurements of latency and clock synchronization performance.

The remaining part of the paper is organized as follows: Section II gives an overview of DECT NR+ and Dante technologies, and explains why DECT NR+ suits the needs of wireless professional audio. In Section III, we describe the architecture of the experimental setup and its implementation. First technical measurement results are then presented in Section IV for different DECT NR+ transmission configurations. The results are discussed and compared to the performance of a typical wired link. Section V concludes the paper, summarizes the findings, and gives ideas on how to extend and continue the work in future research endeavors.

II. Technical Background

In this section, we begin with an introduction to DECT NR+ and its role in wireless professional audio, followed by an overview of Audinate's Dante to establish the application context.

A. DECT-2020 NR

DECT-2020 New Radio (NR) is a novel Radio Interface Technology (RIT) standardized by the European Telecommunications Standards Institute (ETSI) within its Technical Committee (TC) Digital Enhanced Cordless Telecommunications (DECT). It is also referred to as NR+, the brand name promoted by the DECT Forum—an industry association that advocates for DECT technologies and spectrum access [14]. The standard is concise, with the core Technical Specification (TS) comprising five documents published to date. Part 1 provides an overview of the technology [15], while Part 2 establishes the minimum radio reception and transmission requirements for DECT-2020 Radio Devices (RDs) [16]. Parts 3 and 4 contain the physical and MAC layer specifications, respectively [17], [18]. Part 5 describes the Data Link Control (DLC) and Convergence (CVG) layers. The DLC layer provides services to the MAC layer at each radio link, including segmentation and packet routing, while the CVG layer operates end-to-end, offering services such as multiplexing of higher-layer data [19]. A sixth part, detailing the DECT-2020 security architecture and defining an authentication procedure based on pre-shared keys, is currently in preparation [20].

In the fall of 2021, the International Telecommunication Union Radiocommunication Sector (ITU-R) approved DECT-2020 to be included in Recommendation M.2150, making it the world's first non-cellular 5G technology [21]. Fulfilling the International Mobile Telecommunications-2020 (IMT-2020) requirements for massive Machine Type Communications (mMTC) as well as Ultra-Reliable and Low Latency Communications (URLLC), DECT-2020 allows large-scale network deployments with high device density and qualifies as a cable-replacement technology.

Although standardized by the same ETSI committee and related by name, DECT-2020 and "classic" DECT—which dates back to the early 1990s and became widely associated with cordless landline telephony—have little in common. That said, both technologies employ Frequency-Division Multiple Access (FDMA) and Time-Division Multiple Access (TDMA) in a Time Division Duplex (TDD) configuration, which helps facilitate coexistence within the technology-exclusive, license-exempt frequency spectrum ranging from 1880 MHz to 1930 MHz (depending on the region). Radio frames have a length of 10 ms and are divided into 24 slots, resulting in a duration of 416.67 µs per slot. The frequency grid is organized into channels, each occupying a basic width of 1.728 MHz.

Aside from these shared characteristics, the technologies differ substantially, with DECT-2020 being a complete re-design aimed at addressing the demands of modern local-area wireless applications. At the physical layer, DECT-2020 uses Orthogonal Frequency Division Multiplexing (OFDM) with a Cyclic Prefix (CP), which is more robust against multipath distortion and frequency-selective fading than the single-carrier modulation employed in classic DECT [22]. For Forward Error Correction (FEC), DECT-2020 adopts turbo coding with Hybrid Automatic Repeat reQuest (HARQ), utilizing an encoder similar to that of Long Term Evolution (LTE) [23]. It also features an adaptive Modulation and Coding Scheme (MCS), with variable code rates and modulation formats spanning BPSK to 1024-QAM, enabling efficient operation under diverse channel conditions. Additionally, it supports antenna diversity and up to 8×8 Multiple Input Multiple Output (MIMO) configurations.

The transmission numerology is defined by two parameters, namely the subcarrier scaling factor μ and the Fourier transform scaling factor β. For a given pair of parameters μ and β, the OFDM signal comprises $\beta \cdot 64$ subcarriers spaced $\mu \cdot 27\,\text{kHz}$ apart, resulting in a nominal bandwidth of $\mu \cdot \beta \cdot 1.728\,\text{MHz}$. With $\mu \in \{1, 2, 4, 8\}$ and $\beta \in \{1, 2, 4, 8, 12, 16\}$, the bandwidth can be scaled up to $221.184\,\text{MHz}$. However, EU regulations restrict the nominal channel bandwidth of RDs operating in the 1880 MHz to 1900 MHz band to a maximum of 6.912 MHz, thus requiring the use of spectrum outside the DECT band for higher bandwidths [24]. DECT-2020 inherits the frame structure from classic DECT (i.e., 24 slots per 10 ms radio frame), but subdivides each slot into $\mu \cdot 2$ subslots, each containing five OFDM symbols—resulting in $\mu \cdot 10$ OFDM symbols per slot. Subslot durations between approximately 26 µs and 208 µs enable DECT-2020 to support short transmission intervals, emphasizing its suitability for low-latency applications. To meet the reliability demands of URLLC scenarios, DECT-2020 supports scheduled access for deterministic and periodic transmissions. Although random access via Listen Before Talk (LBT) is also available, its contention-based nature introduces uncertainty, making it less appropriate for use cases sensitive to latency and reliability.

Unlike 3GPP cellular communication networks, DECT-2020 does not impose fixed roles on RDs. Instead, RDs can dynamically adapt their operational mode based on contextual factors. When operating in Fixed Termination point (FT) mode, an RD coordinates local radio resources and advertises connectivity information that other RDs can use to establish communication. RDs in Portable Termination point (PT) mode can then act on this information and associate with a suitable FT-mode RD on the network. In DECT-2020 mesh networks, which follow a clustered tree topology, RDs may simultaneously operate in both FT and PT modes—acting as cluster heads while also participating in a parent cluster. The absence of centralized infrastructure enables nomadic ad hoc deployments, aligning with the vision of wireless systems that can be deployed anywhere, by anyone, at any time.

B. Audinate Dante

Dante is a system for digital media transport and management over computer networks. It replaces hundreds of point-to-point audio and video connections with a single, reliable network link. Beyond enhanced reliability, Dante simplifies cabling, patching, and routing of media signals, and enables monitoring and control of connected devices.

Dante uses the Precision Time Protocol (PTP, IEEE 1588) to synchronize time between all nodes in a local network, ensuring sample-accurate media transport and presentation. Within the local network, the time synchronization and media transport are both over IEEE 802.3 (wired or optical) Ethernet, which leads Dante to make certain design assumptions. That is, data packet delivery is very reliable, and timing—while affected by network topology and traffic—is predictable and fairly stable.

Officially, Dante does not support wireless networks. While it often just "works" in a controlled setting, Dante depends heavily on predictable timing. Wireless networks tend to have failure recovery mechanisms that prioritize bandwidth over (worst-case) latency. Dante also makes extensive use of multicast transmissions, both for time synchronization and media transport, a communication mode that can also conflict with wireless error recovery.

While the Dante media transport itself is less sensitive to this typical behavior of wireless networks, PTP time synchronization is likely the first to fail in the face of unpredictable

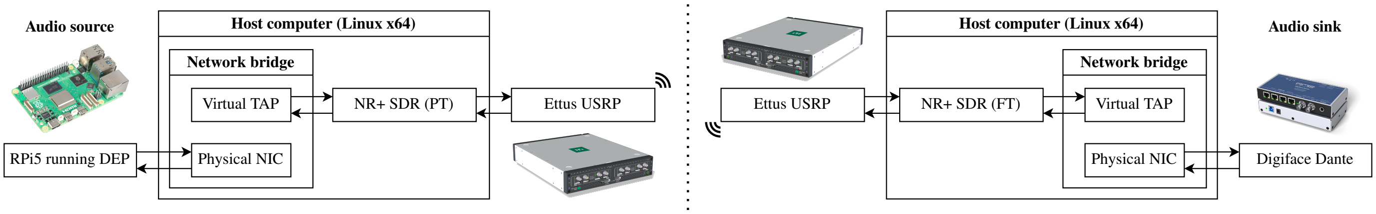

Fig. 1. Block diagram of the experimental setup connecting two Dante-enabled devices over a DECT NR+ wireless link. At both ends of the link, an open-source SDR implementation of DECT NR+ is used in conjunction with Ettus USRPs as radio hardware. The SDR receives application data from a virtual TAP device, which is bridged to the physical network interface connected via Ethernet to the Dante-enabled device. In this configuration, DECT NR+ acts as a transparent Layer 2 wireless bridge.

and unexpected network latencies. The media transport itself is vulnerable to packet loss and excessive delays but will merely fill in the missing data with "mute" samples.

Unmodified, off-the-shelf Dante products were used in this study. It is therefore important to note that some wireless network behavior can be "out of spec" for Dante devices, and experiments along these lines are likely to reveal behavioral differences between different Dante products. In particular, the PTP servo on DEP (Dante Embedded Platform) was designed to be more forgiving of timing uncertainty than older hardware Dante devices.

With that in mind, here are the factors that are likely affecting Dante operation and performance with radio networks. As the radio channel is often noisy and unreliable, wireless networks employ various mitigation strategies to improve reliability. There are two main strategies that deal with data loss—either acknowledge/re-send mechanisms or forward error correction. The former avoids upfront redundancy and prioritizes reliability at the cost of latency; the latter has bounded correction capacity and sacrifices some bandwidth to achieve predictable timing.

Whereas Wi-Fi mandates acknowledge/re-send at the MAC layer, DECT NR+ can refrain from using retransmissions by not requesting HARQ feedback. This makes it a more promising candidate for Dante, as the Dante system is most sensitive to timing unpredictability.

Further, in acknowledge/re-send networks, multicast transmission poses extra challenges. As the transmission is intended to be received by multiple receivers, it is not easy to determine who shall perform the acknowledge, as it is unpredictable which receivers would fail due to noise. Conventional Wi-Fi networks either run multicast without retransmits at the lowest, most robust bit rate, or convert it into individual unicast packets over the air. Forward error correction in multicast is not inherently more complex than in unicast, as it does not rely on receiver feedback or retransmissions.

A second important aspect of network-induced latency variability is the scheduling of transmissions in radio networks. Traditional radio networks do not entirely own the channel and will often hold back the next transmission in the face of other channel activity. Here too, DECT NR+ is expected to have better control over the channel and thus act more predictably.

Dante expects some latency irregularities from Ethernet too, although for modern Ethernet this is mostly reduced to packet egress contention at individual switch ports. Larger irregularities are prevented by traffic prioritization for the PTP packets. If a wireless network has a similar quality-of-service mechanism, it will likely benefit Dante to prioritize PTP event traffic (UDP port 319) over other packet types.

III. Experimental Setup

A high-level block diagram of the experimental setup is shown in Fig. 1. For the wireless link, an open-source Software-Defined Radio (SDR) transceiver implementation of DECT NR+, published on GitHub under the AGPL-3.0 license, was used [25]. Compared to the only commercially available integrated modem from Nordic Semiconductor, which is limited to a transmission bandwidth of 1.728 MHz and restricts operation to the DECT frequency band, the SDR offers increased flexibility by supporting the full feature set of the DECT NR+ physical layer. In addition to the 1880 MHz to 1930 MHz range of frequencies, the standard generally allows for operation below 6 GHz—subject to regional regulatory approval—including license-exempt ISM bands as well as licensed IMT-TDD spectrum.

The SDR is used on two identical general-purpose computers running Ubuntu 24.04 LTS with the 6.8.0 Linux low-latency kernel, one operating as an FT-mode RD and the other as an PT-mode RD. Each machine is equipped with an Intel Core Ultra 9 285K CPU, 64 GB of RAM, and an Intel X710 Network Interface Card (NIC), which provides a high-speed connection to an Ettus USRP via dual 10 Gbit Ethernet links. In the experiment, two USRP N310 were used with version 4.8.0 of the USRP Hardware Driver (UHD), and each device's FPGA was loaded with the XG image flavor [26]. To enhance the performance of IQ sample streaming between the USRP and the host system, UHD was configured to utilize the Data Plane Development Kit (DPDK) for network transport. DPDK facilitates direct control over packet I/O by allowing network drivers to bypass the kernel and operate in user space. With I/O threads pinned to dedicated CPU cores, DPDK helps avoid context switches and scheduler-induced latency spikes in the transport layer of UHD [27].

The USRP N310 supports only certain master clock rates, which are not integer multiples of the DECT NR+ sample rate. This incompatibility presents two options: either apply computationally intensive digital resampling, or operate with a mismatched sample rate, resulting in a subcarrier spacing that deviates from the normative value of $\mu \cdot 27\,\mathrm{kHz}$. To avoid the computational overhead associated with digital resampling, it was disabled for the experiment. With the USRP master clock rate set to 122.88 MHz and a subcarrier spacing of $\mu \cdot 30\,\mathrm{kHz}$, the nominal bandwidth increases to $\mu \cdot \beta \cdot 1.92\,\mathrm{MHz}$.

To receive application data, we configured the SDR to create a virtual TAP network device that operates at Layer 2 of the Open Systems Interconnection (OSI) model and handles raw Ethernet frames. The TAP interface is bridged to the host's physical network interface, which directly connects to the Dante-enabled device. In this configuration, the SDR transmits and receives raw Ethernet frames over the air, functioning as a transparent Layer 2 wireless bridge. As a result, the Dante device can communicate with remote wireless nodes as if they were on the same Layer 2 broadcast domain or Ethernet segment.

For each interface on the bridge (both the virtual TAP and the physical NIC), we configured a priority queueing discipline (prio qdisc) to prioritize packets based on their Differentiated Services Code Point (DSCP) values. Giving time-sensitive PTP events higher priority over audio and other traffic types helps ensure QoS for Dante.

For the experiment, we used two Dante nodes: an RME Digiface Dante and a Raspberry Pi 5 with Dante Embedded Platform (DEP). The Raspberry Pi 5 supports hardware timestamping, which enhances the accuracy of packet timestamps and improves synchronization performance. The Dante-enabled Raspberry Pi 5 was connected to the computer running the SDR in PT mode, serving as the audio source and PTP follower clock. Using the dplay example application provided with DEP, we played back audio decoded from an MP3 file (2 channels, 48 kHz). DEP provides the option to log comprehensive PTP diagnostic information, which help in analyzing synchronization performance and network behavior.

On the receiving end, an RME Digiface Dante was connected to the computer running the SDR in FT mode, serving as the audio sink and PTP leader clock. It was configured to output the received audio to headphones. To monitor clock status and packet latency in real time, a Laptop running Dante Controller was connected to the Digiface via Ethernet.

In this setup, the PT-mode SDR transmits audio data in the uplink direction to the FT-mode SDR. Since PTP packets are sent over the wireless link, it is critical that the link behavior aligns with the assumptions inherent in the PTP protocol. Specifically, packet delay variation (jitter) must be minimized, and the path delay should be as symmetric as possible between uplink and downlink directions to facilitate accurate time synchronization. With that in mind, we employed a transmission scheme with a fixed scheduling pattern and evenly allocated time resources between uplink and downlink. Each 10 ms radio frame begins with a downlink transmission from the FT, which serves as a beacon to provide frame and slot timing information to the PT. This is followed by an uplink transmission from the PT, with downlink and uplink transmissions alternating thereafter in a fixed, repeating sequence.

TABLE I
DECT NR+ TRANSMISSION CONFIGURATIONS

	μ	β	N_{bps}	R	$N_{\mathrm{subslot}}^{\mathrm{PACKET}}$	$N_{\mathrm{byte}}^{\mathrm{TB}}$	T_{packet}
(A)	1	8	6	3/4	2	1805	416.667×10^{-6}
(B)	2	8	4	3/4	3	1821	312.500×10^{-6}
(C)	2	8	6	3/4	2	1552	208.333×10^{-6}

As Ethernet frames become available in the TAP device's receive buffer, their transmission is scheduled for the next available opportunity according to assigned time resources. Rather than aggregating multiple Ethernet frames into a single transmission packet, each packet carries only one Ethernet frame to minimize delay variation caused by variable queuing delays. For the downlink transmission at the beginning of each radio frame, the FT multiplexes the beacon message and an Ethernet frame into a single packet.

Since each transmission packet carries no more than one Ethernet frame, the number of transmission opportunities must be sufficient to accommodate the rate of Ethernet frames received by the TAP interface. The rate of Ethernet frames carrying audio IP packets is determined by the audio sampling rate, the number of channels, and the number of audio samples per channel contained in each packet. Additional network traffic includes PTP messages, as well as control and monitoring data. In the experiment's configuration, the Dante nodes exchange in the order of 1500 packets per second. To carry a full-sized Ethernet frame, including its header, a transport block—that is, the payload of a DECT NR+ physical layer packet—must accommodate at least 1514 bytes, not accounting for MAC headers or other Service Data Units (SDUs) also present in the MAC Protocol Data Unit (PDU).

Table I contains the DECT NR+ transmission configurations that were used in the experiment. As detailed in Section II-A, the subcarrier width scaling factor μ and the Fourier transform scaling factor β define the transmission numerology. The Modulation and Coding Scheme (MCS) is specified by the parameters N_{bps} and R, where N_{bps} denotes the number of bits per QAM symbol and R denotes the coding rate. For example, if N_{bps} is equal to 6, then the modulation scheme is 64-QAM. T_{packet} and $N_{\mathrm{subslot}}^{\mathrm{PACKET}}$ indicate the length of a transmission packet in seconds and subslots, respectively. With five OFDM symbols per subslot, a transmission packet contains $N_{\mathrm{subslot}}^{\mathrm{PACKET}} \cdot 5$ OFDM symbols. The resulting transport block size in bytes is given by $N_{\mathrm{byte}}^{\mathrm{TB}}$.

In all configurations, turbo code blocks are segmented to a maximum size of 2048 bits. The system operates in Single Input Single Output (SISO) mode, with both transmitter and receiver using a single antenna supporting one spatial stream. Each of the three configurations allows the transport block to carry a full-sized Ethernet frame plus additional DECT NR+ MAC layer overhead. The key difference between configurations lies in the packet duration, which directly affects packet throughput.

In configuration A, each transmission packet spans a full slot, which corresponds to two subslots ($\mu = 1$). With 24 slots per 10 ms radio frame, this results in 1200 transmission opportunities per second per radio device, assuming time resources are evenly allocated between uplink and downlink.

In configuration B, the packet length is three-quarters of a slot, giving each radio device 16 transmission opportunities per radio frame, or equivalently, 1600 per second.

A packet duration of half a slot in configuration C yields a total of 48 transmission opportunities per radio frame, which allows each radio device to transmit up to 2400 packets per second, or one packet every 416.67 µs. Shorter transmission intervals, which increase packet throughput, provide a safety margin for accommodating the rate of user-plane data and are likewise expected to reduce latency.

IV. Results and Discussion

To obtain results independent of the wireless channel, measurements were conducted under ideal conditions by connecting the USRP antenna ports using coaxial cables and a 30 dB attenuator. This setup not only avoids multipath and fading effects but also eliminates the risk of interference with other DECT equipment, such as cordless telephones, when operating within the DECT frequency band. Moreover, with a nominal bandwidth of up to 30.72 MHz, the transmission would occupy more spectrum than is available in the DECT band. Accordingly, the results presented in this section reflect a best-case scenario and may be considered an upper bound on the performance of an SDR-based system. In practical deployments, a true wireless setup would be subject to channel impairments, interference, and regulatory constraints on spectrum availability.

In addition to the configurations listed in Table I, we performed a reference measurement over Ethernet, subsequently referred to as configuration W (for "wired"). For comparability, the Dante nodes were not connected directly, but rather through two intermediary computers acting as switches. Each Dante node was connected to one computer, and the two computers were linked via Ethernet. On each computer, the network interfaces connecting to the Dante node and to the other computer were bridged. Similar to the SDR setup, priority queueing disciplines were configured on the interfaces to prioritize time-critical traffic and ensure QoS.

Each configuration used a different Dante latency setting, which controls the amount of time received audio data is buffered before playback, balancing between low delay and reliable audio streaming to accommodate different network conditions. For configuration A, the latency was set to 10 ms, while configurations B and C utilized a 5 ms latency setting. A lower value of 2 ms was used for configuration W.

During the measurements, the Dante-enabled Raspberry Pi repeatedly played back audio decoded from an MP3 file for a duration of 30 minutes. Dante Embedded Platform (DEP) logs comprehensive PTP diagnostic information, which

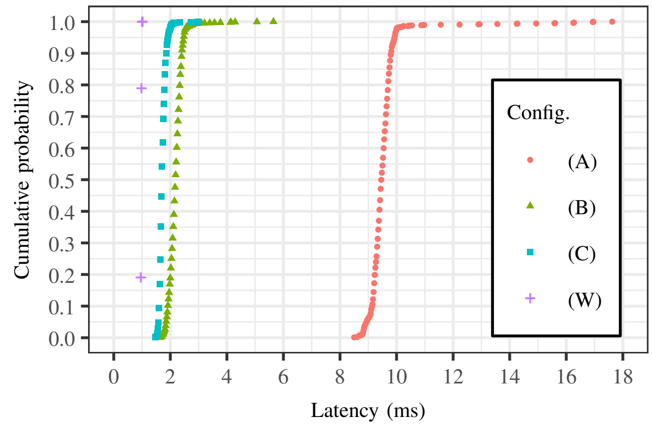

Fig. 2. Empirical CDFs of the latencies measured across the different configurations. The x-axis indicates latency in milliseconds, while the y-axis represents the proportion of packets with latency less than or equal to the corresponding x-axis value. A steep, left-shifted curve indicates better performance.

were recorded to get insight into the network behavior and clock synchronization performance. A laptop was connected to the Digiface to capture the Dante Control and Monitoring (ConMon) status events also reported to Dante Controller. This includes latency values and clock status, as well as counts of late, early, and out-of-order packets. Packets arriving after the Dante latency cut-off are considered "late", while packets that appear to arrive before they were sent—typically due to improper device time synchronization—are considered "early".

No early or out-of-order packets were observed for any of the configurations. However, late packets were reported for configurations A and B. Over the 30-minute measurement period, 20 073 packets exceeded the 10 ms cut-off value in configuration A, while 20 packets exceeded the 5 ms threshold in configuration B. No late packets were reported for the other configurations.

Fig. 2 shows the empirical Cumulative Distribution Functions (CDFs) of the latencies measured across the different configurations. The x-axis indicates latency in milliseconds, while the y-axis represents the proportion of packets with latency less than or equal to the corresponding x-axis value. A steep, left-shifted curve indicates better performance. Summary statistics, including the mean, maximum, and selected quantiles of the measured latencies for each configuration, are provided in Table II.

As expected, the reference measurement over Ethernet (i.e., configuration W) performs best and exhibits the lowest latencies, with a mean value of 980 µs. Also, the variability of the latency values is lower compared to the other configurations, considering the difference between the mean and the maximum value of 1021 µs is less than 50 µs. Among the DECT NR+ transmission configurations, configuration C achieves the lowest latency, with a mean value of 1.7 ms, followed by B (2.2 ms) and A (9.5 ms). This was also to be expected, since shorter transmission intervals help reduce queueing delays in the transmit buffer. The notably higher latency in configuration A is due to the fact that each RD has only 1200 transmission opportunities per second, which is less than the rate at which packets arrive at the TAP interface. This also explains the high number of late packets (20 073) in that configuration.

To evaluate and compare clock synchronization performance across configurations, we utilized the PTP diagnostic information provided by DEP. As follower clock, DEP employs a dual-loop PTP servo structure to align the local clock to that of the PTP leader. A narrow-bandwidth rate control loop compensates for frequency (rate) differences between the clocks, while a wide-bandwidth phase control loop adjusts the phase offset. The wide bandwidth enables the phase loop to respond rapidly to transient changes, making it more sensitive to network asymmetry, delay variation, and measurement noise. As a result, the adjustments made to correct phase offsets can serve as a useful indicator of network-induced timing variability affecting the PTP packets.

Fig. 3 shows violin plots of the phase offset adjustment values, which help compare clock synchronization performance across configurations. The values are given in parts per million (ppm). A wide and short violin plot indicates a more stable synchronization performance, whereas a narrow and tall violin plot suggests higher variability in the phase offset adjustment values, likely caused by network-induced delay variations affecting the PTP packet timing. To exclude the initial phase offset and early adjustments immediately after synchronization begins, the first 30 seconds of measurement data were discarded prior to plotting. Packets that were not used to adjust the clock phase offset, such as those subject to PTP outlier filtering, were also removed from the measurement data.

The box plots indicate that all configurations have a median at or near zero, suggesting no prominent systematic bias or directional skew due to asymmetrical network delay characteristics. While configurations A and B exhibit similar distributions of phase offset adjustments, configuration C shows lower variability and values more tightly clustered around the median, reflecting improved synchronization performance. This is further supported by the narrower interquartile range

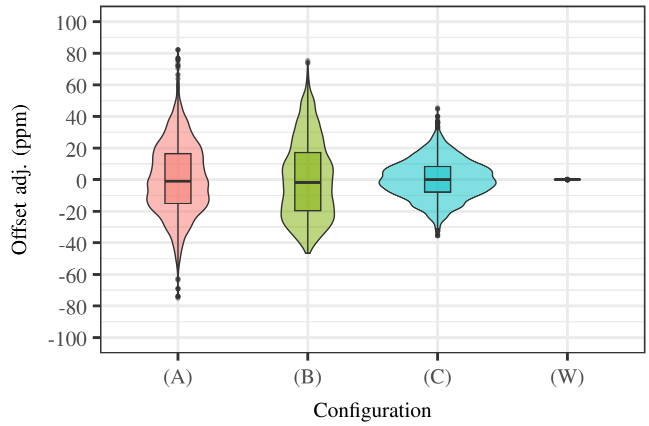

Fig. 3. Violin plots of the phase offset adjustment values for the different configurations in parts per million (ppm). A wide and short violin plot indicates a more stable synchronization performance, whereas a narrow and tall violin plot suggests a higher amount of network-induced timing variability affecting the PTP packets.

observed in configuration C (see also Table III). As expected, the Dante-over-Ethernet link yields a significantly more stable performance; however, the scale of the vertical axis conceals any minor variability that may be present.

The better performance of configuration C, relative to A and B, can be attributed to its more frequent scheduling of transmission opportunities. Since slot timing is not aligned with PTP packet generation, denser scheduling effectively results in a finer sampling of PTP packet transmission times, thereby reducing latency jitter.

Aligning transmission opportunities with PTP packet generation times is not feasible in a scheduled channel access system, since delay request messages are deliberately randomized to avoid introducing measurement bias. A more viable approach is to exploit the inherent synchronization mechanism of DECT NR+. This could be achieved by implementing a PTP boundary clock that uses the radio's synchronized timing to re-time PTP packets across the link, effectively hiding radio-induced delay variation. Alternatively, a word clock could be derived from the radio synchronization—such as DECT NR+ frame or slot timing—to synchronize Dante media clocks and eliminate the need for PTP over the radio link entirely.

An alternative synchronization mechanism, that does not require dense transmission opportunities to allow the transport of PTP packets over the air, could also improve channel uti-

TABLE II
LATENCY IN MILLISECONDS

	Quantile	(A)	(B)	(C)	Wired (W)
Median	0.5	9.479	2.188	1.708	0.979
Mean	N/A	9.526	2.192	1.730	0.980
	0.99	11.271	2.771	2.052	1.000
	0.999	16.943	4.484	2.807	1.000
	0.9999	17.508	5.544	2.996	1.017
	0.99999	17.613	5.636	3.000	1.020
	0.999999	17.624	5.645	3.000	1.021
Maximum	1	17.625	5.646	3.000	1.021

TABLE III
PHASE OFFSET ADJUSTMENT IN PARTS PER MILLION

	(A)	(B)	(C)	Wired (W)
Median	-0.945	-1.842	-0.066	0.027
Interquartile range	31.524	36.817	16.159	0.045
Standard deviation	22.352	24.337	12.124	0.053

lization. For example, in configuration C, the uplink capacity is 29.798 Mbit/s, yet on average, less than 10 % was utilized during the experiment. Since the audio packets can tolerate packet delay variation to some degree, they can be aggregated into DECT NR+ transmission packets and sent at a lower rate, provided the delay stays below the configured Dante latency.

V. Conclusion and Future Work

In this paper, we investigated the feasibility using DECT NR+ as cable replacement solution for professional networked audio systems such as Dante. To set up an experimental Dante wireless link, we utilized an open-source Software-Defined Radio (SDR) implementation of DECT NR+ as a transparent Layer 2 wireless bridge. Based on PTP diagnostic information provided by Dante Embedded Platform (DEP) and metrics reported to Dante Controller, we assessed latency and clock synchronization performance for different DECT NR+ transmission configurations.

With transmission opportunities scheduled every 416.67 μs, an average network latency of 1.73 ms was achieved. While clocks remained synchronized over the DECT NR+ wireless link, the phase offset adjustment values of the PTP follower clock indicated that the performance does not yet match that of a conventional Dante-over-Ethernet connection.

Future work should focus on leveraging the radio synchronization mechanism inherent to DECT NR+ to implement a PTP boundary clock that bridges the radio link. Since the measurements in this study were conducted under ideal channel conditions and for a limited duration (30 minutes), further measurements should be performed over longer periods and in representative environments such as stadiums, arenas, and concert halls. Additionally, it would be valuable to investigate whether Hybrid Automatic Repeat reQuest (HARQ) can be used to mitigate packet loss while still maintaining operation within the configured latency budget. Beyond HARQ, redundancy-based error mitigation techniques such as application-layer Forward Error Correction (FEC) might be worth exploring.

Overall, as the world's first non-cellular 5G technology, DECT NR+ remains a promising candidate for professional networked audio applications, thanks to its scheduled channel access, support for ultra-short transmission intervals, and technology-exclusive frequency spectrum.

References

[1] N. Bouillot *et al.*, "AES White Paper: Best Practices in Network Audio," *J. Audio Eng. Soc.*, vol. 57, no. 9, pp. 729–741, September 2009.

[2] K. Walsh, "Dante and AVB Networking," in *AES UK 24th Conference: The Ins & Outs of Audio*, June 2011, paper 9, pp. 1–7.

[3] J. Pilz, B. Holfeld, A. Schmidt, and K. Septinus, "Professional Live Audio Production: A Highly Synchronized Use Case for 5G URLLC Systems," *IEEE Netw.*, vol. 32, no. 2, pp. 85–91, 2018.

[4] J. Dürre, N. Werner, P. Matzakos, R. Knopp, A. Garcia, and C. Avellan, "A Disaggregated 5G Testbed for Professional Live Audio Production," in *2022 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB)*, 2022, pp. 1–6.

[5] H. Bruckmeyer, J. Bühlmeyer, T. Ackermann, and G. Fischer, "Time Synchronization of Spatial Separated Areas for AV-Production," in *2021 Joint Conference of the European Frequency and Time Forum and IEEE International Frequency Control Symposium (EFTF/IFCS)*, 2021, pp. 1–6.

[6] S. Giacomelli, C. Centofanti, J. Santos, M. Galbiati, T. Salvi, F. Graziosi, and C. Rinaldi, "Remote Immersive Audio Production: State of the Art Implementation, Challenges, and Improvements," in *2024 IEEE 5th International Symposium on the Internet of Sounds (IS2)*, 2024, pp. 1–10.

[7] P. Ferguson, C. Chafe, and S. Gapp, "Trans-Europe Express Audio: Testing 1000 mile low-latency uncompressed audio between Edinburgh and Berlin using GPS-derived word clock, first with jacktrip then with Dante," in *AES 148th Convention*, May 2020, paper 605, pp. 1–5.

[8] Dante Over Distance: Technical Dive. Audinate. [Online]. Available: https://www.getdante.com/wp-content/uploads/2025/02/DanteOverDistance-TechnicalDive-20241129.pdf

[9] M. Henry and W. Aubry, "Setup and First Experimentation Over an AES67 Over 802.11 Network," in *AES 146th Convention*, March 2019, paper 510, pp. 1–5.

[10] C. Chousidis, R. Nilavalan, and A. Floros, "Enhancement of IEEE 802.11 in Handling Multiple Broadcasting Audio Data in Wireless Ad-Hoc Networks," *J. Audio Eng. Soc.*, vol. 61, no. 4, pp. 165–173, April 2013.

[11] L. Turchet and P. Casari, "Assessing a Private 5G SA and a Public 5G NSA Architecture for Networked Music Performances," in *2023 4th International Symposium on the Internet of Sounds*, 2023, pp. 1–6.

[12] A. Poets, M. Waßmann, M. Penner, S. Preihs, and J. Peissig, "DECT NR+: A new radio standard to benefit professional audio applications," in *Fortschritte der Akustik - DAGA 2024, 50. Jahrestagung für Akustik*, March 2024, pp. 52–55.

[13] J. Dürre and N. Werner, "DECT-2020 NR for Professional Live Audio: Design Space Exploration and Practical Evaluation," in *2024 IEEE 5th International Symposium on the Internet of Sounds (IS2)*, 2024, pp. 1–7.

[14] DECT Forum. [Online]. Available: https://www.dect.org/

[15] *DECT-2020 New Radio (NR); Part 1: Overview; Release 2*, ETSI TS 103 636-1, Rev. 2.1.1, October 2024.

[16] *DECT-2020 New Radio (NR); Part 2: Radio reception and transmission requirements; Release 2*, ETSI TS 103 636-2, Rev. 2.1.1, October 2024.

[17] *DECT-2020 New Radio (NR); Part 3: Physical layer; Release 2*, ETSI TS 103 636-3, Rev. 2.1.1, October 2024.

[18] *DECT-2020 New Radio (NR); Part 4: MAC layer; Release 2*, ETSI TS 103 636-4, Rev. 2.1.1, October 2024.

[19] *DECT-2020 New Radio (NR); Part 5: DLC and Convergence layers; Release 2*, ETSI TS 103 636-5, Rev. 2.1.1, October 2024.

[20] *DECT-2020 New Radio (NR); Part 6: Security; Release 2*, ETSI TS 103 636-6, Rev. 2.0.1, June 2025, Early draft.

[21] *Detailed specifications of the terrestrial radio interfaces of International Mobile Telecommunications-2020 (IMT-2020)*, ITU-R Recommendation M.2150-1, February 2022.

[22] M. Waßmann, A. Poets, J. Peissig, and J. Pilz, "Performance of DECT-2020 NR in an industrial environment for varying RF bands," in *2024 Joint European Conference on Networks and Communications & 6G Summit (EuCNC/6G Summit)*, 2024, pp. 610–615.

[23] *Evolved Universal Terrestrial Radio Access (E-UTRA); Multiplexing and channel coding*, 3GPP TS 36.212, Rev. 18.1.0, December 2024.

[24] *Digital Enhanced Cordless Telecommunications (DECT); Harmonised Standard for access to radio spectrum; Part 2: DECT-2020 NR*, ETSI EN 301 406-2, Rev. 3.1.1, August 2023.

[25] M. Penner. (2025, March) DECT-NR-Plus-SDR. License: AGPL-3.0. [Online]. Available: https://github.com/maxpenner/DECT-NR-Plus-SDR

[26] USRP N3xx Series. USRP Hardware Driver and USRP Manual. [Online]. Available: https://uhd.readthedocs.io/en/uhd-4.8/page_usrp_n3xx.html

[27] N. Temple, A. Williams, W. Fife, M. Prost, and M. Dickens. Getting Started with DPDK and UHD. Ettus Knowledge Base. [Online]. Available: https://kb.ettus.com/Getting_Started_with_DPDK_and_UHD