Low Latency 5G for Professional Audio Transmission

Exploring the latency and jitter performance of isochronous audio transmission over 5G URLLC

White paper

5G Ultra-Reliable Low Latency Communications (URLLC) offers wireline-like reliable data transmissions for deterministic low-latency applications. This paper explores the option of using 5G networks for the professional transmission of music. 5G can deliver major benefits to the end user - simpler audio network deployment and more secure network operation.

Nokia and Sennheiser research teams have examined latency as one of the main challenges when using a wireless IP network for transmission of isochronous audio data within a professional production environment. A 5G URLLC testbed was set up to measure, evaluate and optimize the transmission latency. The measurements obtained demonstrate that 5G URLLC can meet the latency requirements for professional audio use.

Contents

Introduction

5G is expected to connect people, things, data, applications, transport systems and cities within smart networked communication environments[1]. 5G will provide much higher capacity and data rates, connect devices reliably with very low latency, and connect an extremely large number of devices with support for very long battery life. 5G networks will support three main application areas, namely enhanced mobile broadband (eMBB) for very high data rates, ultra-reliable low-latency communications (URLLC) for mission critical applications, and massive machine-type communications (mMTC) for huge numbers of connected devices. The key performance metrics for URLLC are latency and reliability.

Programme Making and Special Events (PMSE) applications, such as professional audio, involve four Key Performance Indicators (KPI) that 5G can address - reliability, latency, synchronicity and spectral efficiency. Latency for professional audio transmission is currently the biggest challenge for a standardized wireless technology to meet, creating high interest in whether 5G will fulfil this need.

5G URLLC for professional audio transmission

Programme Making and Special Events

PMSE encompasses all wireless applications used in professional audio/video productions such as concerts, musicals, theater shows or other staging of entertainment, as well as live sports events, Electronic News Gathering (ENG), film productions for cinema and TV, press conferences and the like. It covers not only entertainment, but also public events and various types of education and cultural activity.

PMSE applications can be divided broadly into three categories: audio PMSE, video PMSE and stage control / service PMSE. Typical PMSE equipment includes wireless microphones, wireless in-ear monitors, wireless video cameras, conference systems, intercom systems, wireless light and effect control. All these applications need dedicated and interference-free spectrum resources to meet their high demands for reliability and latency.

Currently, there is no harmonized frequency range for PMSE equipment. Every country allocates different frequency ranges, but mostly with some overlap. There is also no standardized wireless technology for PMSE. Even within the audio sector itself the technologies differ depending on the respective application, e.g. live music, presentation and so on. To avoid unacceptable audio disturbance and to guarantee interference-free operation, every audio production requires intensive frequency planning in advance, and an elaborate network setup on site.

To reduce this radio technology effort and enable high network availability, integration into a standardized, globally available technology such as 5G is a potential solution. However, the KPIs for reliability, latency, synchronicity and spectral efficiency of any proposed technology must align precisely with the requirements of professional audio transmission, which is the basis for professional audio use cases such as professional live audio production.

Requirements of professional live audio production

Today's typical professional live audio production combines several wireless and wired technologies to capture, produce, playback and distribute audio content, see figure 1. One of the most important wireless connections is used by artists on stage. Here, wireless audio transmitters for vocals or instruments are combined with wireless in-ear monitor systems to provide audio feedback for the user. Every wireless audio link is composed of one transmitter and its destined receiver, which provides the input data for further processing, or in the case of an in-ear monitor system the audio stream for the artist on stage. The requirements for the wireless link resulting from this setup can be summed up with four KPIs.

Spectral efficiency: Audio PMSE systems typically follow a link-based approach based on a dedicated transmitter-receiver pair for unidirectional audio transmission. Thus, the required resources in radio spectrum scales with the number of links to be used. Currently, audio PMSE equipment uses analog and digital narrowband modulation techniques with each audio link typically occupying a 200 kHz wide RF channel[2]. In larger live setups, spectrum is a limited resource, requiring detailed planning, often performed manually, to accommodate all wireless devices.

Reliability: Due to its live nature, reliability of the production network is essential. Every audible artefact will reduce the quality of the content, potentially reducing the profit of music producers. Reliability of the radio link is fundamental to audio PMSE. For professional (live) audio productions especially, the commercial pressure is significant as there is mostly no opportunity for recovery, so there is extremely low tolerance for disturbance to the quality of service.

Latency: Application latency describes the roundtrip time between audio acquisition (e. g. microphone) and audio playback (e.g. in-ear monitor (IEM)), which is mainly determined as double the transmission latency and the audio processing time (e.g. mixing and filtering). It is a critical parameter, because above a certain threshold, the artist will no longer be able to perform. Depending on the sort of music, the instrument, and the musician's skills this threshold varies widely, but in most cases, it must not exceed 4 milliseconds (ms). Mixing and filtering of audio signals can take up to 2 ms, leaving only about 1 ms for transmission in one direction. Currently there is no digital transmission-based commercial solution available that meets this latency requirement. As a result, combinations of digital (e.g. microphones) and analog (IEM) devices are used today to reduce latency.

Synchronicity: It is important to avoid quality-reducing resampling and to optimize latency in a production network. This calls for every audio device to precisely synchronize incoming and outgoing audio samples with a common reference. This requirement is called synchronicity and describes a new KPI for audio PMSE. In addition, this enables highly time-aligned applications, for example in immersive 3D audio setups and productions.

Table 1. Summarizes the key requirements and limits in professional audio transmission.

Figure 1. Exemplary professional audio setup

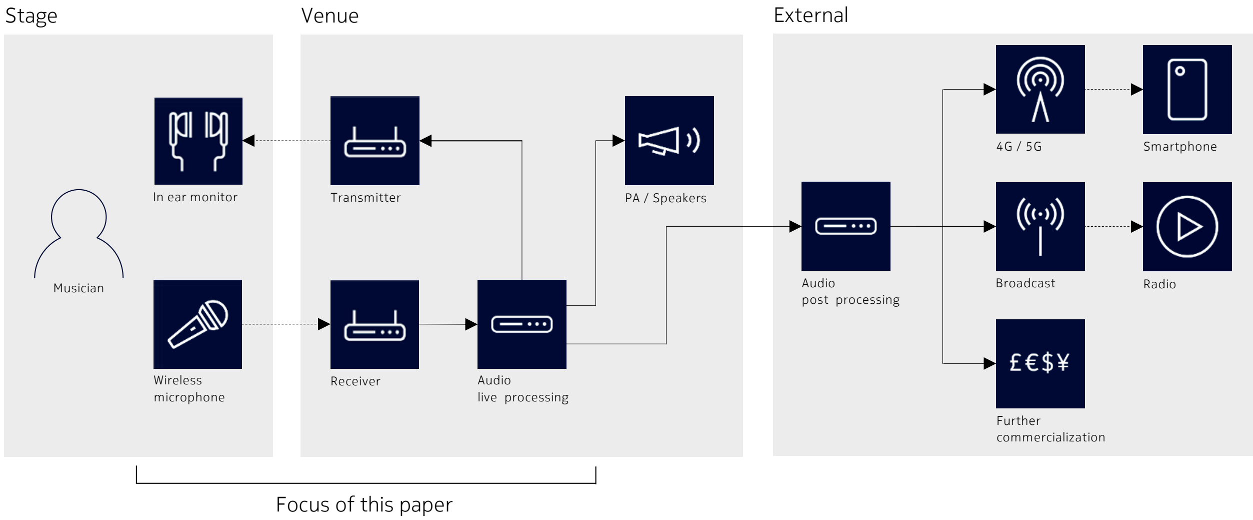


Table 1. Key requirements of the professional live audio production use case[3]

Requirement		Comment
Application latency	< 4 ms	Maximum allowable roundtrip time at application level, including interfacing, audio processing and AD / DA conversion
Wireless transmission latency	< 1 ms	Latency per link in the wireless communication system, including the transmission interval of the audio data. This is less than half the application latency to allow for additional audio processing, e.g. mixing
User data rate	150 kbit/s – 5 Mbit/s	Different user data rates per audio link need to be supported for different audio demands
Reliability	99.9999 %	The Packet Error Ratio (PER) of the system shall be below 10^{-6} for a packet size corresponding to 1 ms audio data
# of audio links	50 – 300	Simultaneous audio links in a single location: microphones and IEMs
Service area	≤ 10,000 m^2	Event area, indoor and outdoor
Synchronicity	≤ 1 µs	All wireless mobile devices of one local audio production network shall be synchronized at the application level within the specified accuracy

5G URLLC for PMSE

Tight latency control and synchronicity are key enablers for PMSE. 5G URLLC refers to communications with an end-to-end user plane latency of a few milliseconds or sub-millisecond with successful packet delivery of 99.999% or higher (99.9999% in 3GPP Release 16). 3GPP's work to set the URLLC standard to meet these requirements began in 3GPP Release 15 and is continuing in Release 16 and beyond[4].

3GPP standardized URLLC enables the KPI requirements of PMSE applications to be met:

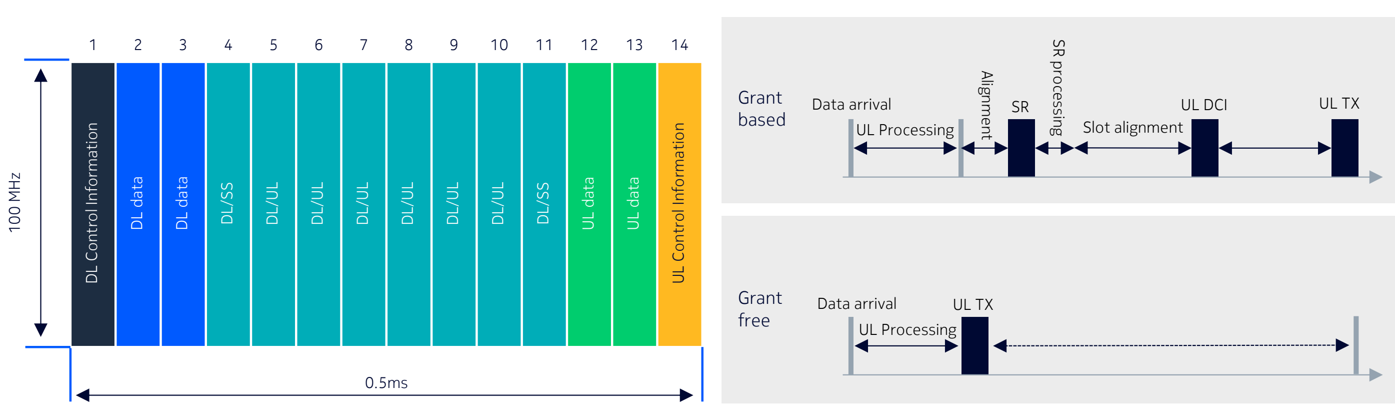
Reliability: High reliability is supported in 3GPP Release 15 through multi-slot transmission for the grant channel access, low rate Modulation and Coding Scheme (MCS) and further uplink multi-antenna techniques.

Latency: Technical enablers for reduced latency in 3GPP Release 15 include a flexible frame structure and scheduling of very short transmit time interval (TTI) to less than 100 microseconds, short uplink control channel, and edge computing. Furthermore, preemptive scheduling offers improved downlink multiplexing of eMBB and latency-critical traffic, where the gNB (5G base transceiver station) may partially overwrite (i.e. pre-empt) an ongoing eMBB transmission with a shorter, urgent URLLC transmission.

For the uplink, Release 16 enables the gNB to instruct eMBB UEs to cancel an ongoing transmission to quickly release radio resources for urgent URLLC transmissions. 5G New Radio (NR) URLLC comes with reduced processing times for UEs to decode transmissions (downlink) and preparation of new transmissions (uplink). To meet a <1ms requirement in the uplink and to mitigate the scheduling request delay from a UE, the gNB may configure periodic transmission resources for a UE (grant-free scheduling). The UE can immediately transmit to the gNB on those pre-configured resources whenever it has data available, thereby achieving lower latencies. In Release 16, enhancements to Semi Persistent Scheduling (SPS) tailored to fast isochronous traffic flows are also added.

Synchronicity: 3GPP Release 16 adds URLLC with deterministic communications through time synchronization by integrating Time Sensitive Networking (TSN) into the device and the core network. The end-to-end 5G system can be considered as an 802.1AS "time-aware system" enabling sub-microsecond device synchronization to any time-domain, either the native 5G time-domain or a time-domain used by the vertical application. With this framework, delay-jitter can also be controlled to a sub-microsecond level by hold-and-forward buffers integrated in 5G core network and devices supporting 5G TSN.

Figure 2. Latency reduction due to URLLC mini-slot structure and UL Grant free transmission



Spectrum

Current PMSE spectrum use and associated challenges

Frequency ranges for audio PMSE are mainly dependent on national allocations even though significant cross-border use of equipment takes place. Some frequency bands or parts of them are similar in International Telecommunication Union for Radio (ITU-R) regions and countries and therefore have become the predominant frequency ranges for audio PMSE use.

Looking at ITU Region 1, the 470 – 694 MHz frequency range is open for audio PMSE use based on secondary operation to broadcasting. Audio PMSE deployments make use of unused spaces in the radio spectrum within their service area. In the 694 - 790 MHz range the possible uses of audio PMSE radio systems are also in secondary operation to broadcasting and mobile. In parts of ITU Region 1 the duplex gap 823 – 832 MHz is used for audio PMSE, e.g. in the European Union this frequency band is harmonized for exclusive use by audio PMSE applications. Experience with the use of that band shows that it is not suitable for every audio PMSE application due to the potential harmful interference from unwanted emissions of IMT components into this duplex center gap.

Additionally, 863 - 865 MHz is harmonized within the European Union for audio transmission including wireless headphones, assistive listening devices and some wireless microphones. Also, the frequency range of 1785 - 1805 MHz is harmonized for exclusive use by audio PMSE applications. Additional frequency ranges in 470 - 960 MHz in Region 1 might be available for possible audio PMSE use on a national basis.

The variability of spectrum access (on a secondary basis or exclusively) and the different options for spectrum allocation are in deep contrast to the strict quality requirements of audio PMSE. To provide and guarantee a high quality of service, intensive frequency planning and monitoring is required before and during professional events. For touring events, each new venue requires fresh planning and setup, which is very time-consuming and expensive.

Benefits of 5G for spectrum use

As mentioned earlier, while audio PMSE requires interference-free spectrum to provide content capture at the highest possible quality, the lack of a globally harmonized frequency band increases the efforts and costs of professional audio productions for error-free operation. 5G is predominantly used in exclusively assigned spectrum and technically allows the prioritization of PMSE over other traffic to provide the required Quality of Service (QoS).

Unlike in Industrial, Scientific and Medical (ISM) bands, in exclusively assigned spectrum no interference from generally authorized other spectrum users is expected. Much of the spectrum for 5G is assigned to public network operators which could offer the desired QoS level thus easing the planning and deployment efforts for PMSE users. Some jurisdictions also provide spectrum access via local licensing for non-public networks which may provide opportunities for handling PMSE requirement e.g. in production studios or theaters.

Audio PMSE equipment includes handheld microphones or bodypack transmitters for guitar/bass or for clip-on microphones, which transmit audio data continuously over several hours indoors and outdoors. To support the low power transmission of these battery-driven portables and to overcome wireless channel effects, such as propagation losses arising from complex stage sets and the body loss when using body-worn equipment, the favorable propagation conditions of sub-1 GHz frequencies are employed currently. Which 5G spectrum band(s) might be used by audio PMSE in the future is a topic for discussion.

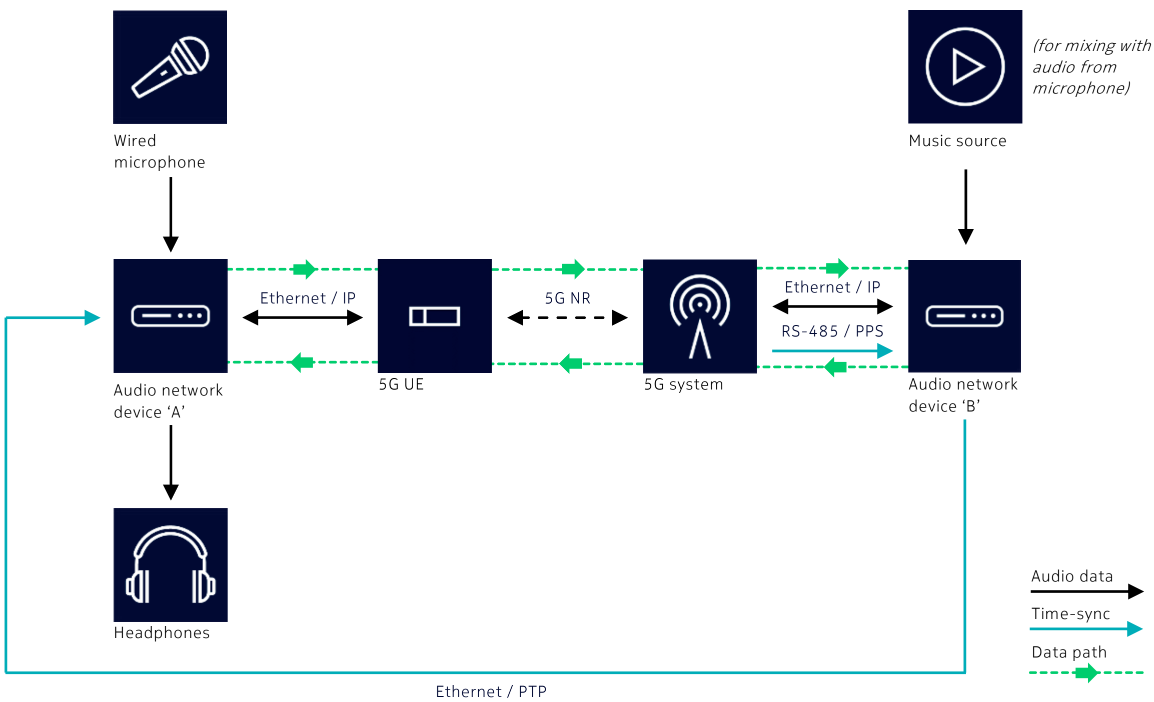
Optimizing the network latency with 5G URLLC

Sennheiser and Nokia researchers created a testbed to investigate the feasibility of early 5G concepts for professional audio transmission use cases.

Testbed for professional audio transmission based on 5G URLLC

The testbed was a simplified implementation of the professional live audio production use case as shown on the left side of Figure 1. A microphone was used to capture the audio from a musician and was connected to an audio network device 'A', which performed analog to digital conversion. This audio network device was connected to a 5G UE, which transmitted the audio data across the 5G NR air interface, through the core network to an audio network device 'B', which was connected to a music source. Audio network device 'B' mixed audio from the music source with the incoming audio from the microphone. The resulting mix was then transmitted back through the 5G network to audio device 'A' where the outgoing audio source could be heard through headphones (acting in place of an IEM).

Figure 3. Setup of testbed



As TSN based synchronization is a forthcoming 3GPP release16 feature and the implemented demo system was based on release 15, the setup included a method to synchronize the clocks between the 5G system and the audio devices. For that, the 5G base station provided a Pulse Per Second (PPS) signal over a dedicated wire-connection to audio network device 'B'. This time information was shared via Precision Time Protocol IEEE 1588-2008[5] (PTP) over a dedicated Ethernet connection between the two audio network devices. This allowed the synchronization of media clocks to avoid audio quality-reducing resampling, and the synchronization of sending audio IP packets to the transmit timing of the 5G system. In addition, the synchronization of clocks was the basis for high precision jitter and latency measurements.

5G URLLC Implementation

The URLLC testbed was implemented on a commercial Nokia AirScale radio platform. As commercial chipsets or terminals were not available, the testbed included a Software Defined Radio (SDR) implementation for the UE.

The testbed implemented 3GPP Release 15 URLLC features. A new numerology was used together with grant-free scheduling in the uplink. Scheduling was based on configured grant, which is "pre-scheduling" of transmission in the uplink. In the downlink, scheduling was according to Downlink Control Information (DCI) to enable the required system latency.

The packet core and application processing were co-located with the system module for baseband processing in order to minimize latency impacts.

URLLC key parameters used in this proof-of-concept were as follows:

- PDSCH/PUSCH coding rate 0.17
- 3.5 GHz, 30 kHz Sub Carrier Spacing (S numerology, 100 MHz bandwidth, TDD Duplex
- 2-symbols mini-slot support for downlink and uplink
- Mini slot based scheduling every 0.5 ms for downlink and uplink
- PDCCH aggregation level 16
- Uplink grant-free transmission

Integration of audio devices

An audio network device is a gateway combining the analog audio world of sound capture and audio playback with the world of IP networks. It has the following functionalities:

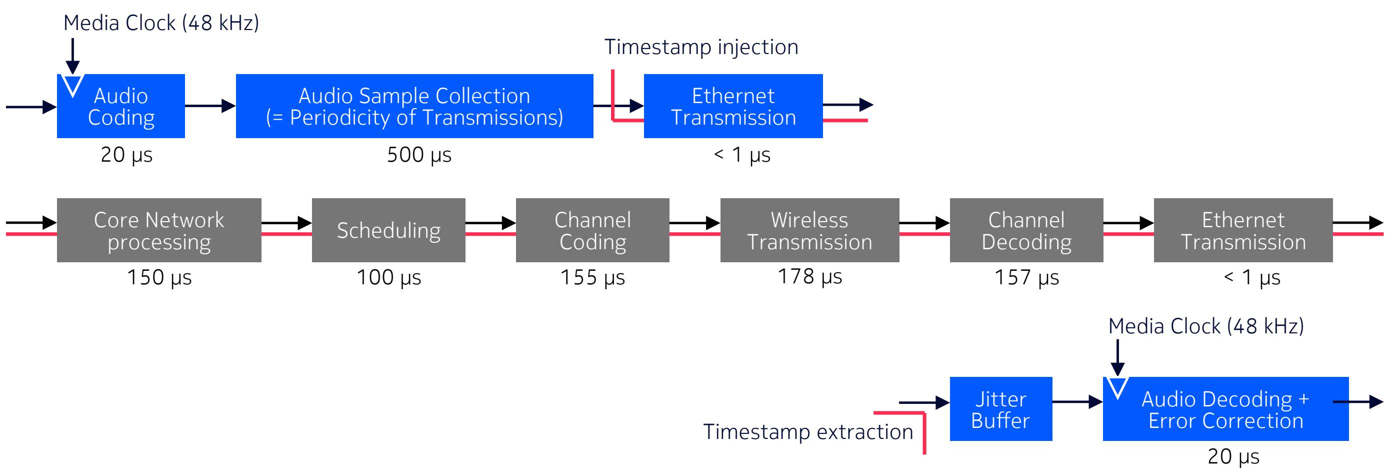
- Converting audio signals from analog to digital and vice versa, with adjustable sample rates and resolution
- Providing deterministic IP packet transmission including time stamps for latency measurement
- Adjusting the length of the outgoing IP packets according to RF frame structure
- Processing of audio codecs
- Low latency mixing of several audio input streams
- Synchronization of local sample clock to a global clock
- Converting PPS or GPS input to PTP output

The devices used in the test bed were optimized for the low-latency transmission of isochronous audio data transmission within an IP network, and high-precision time measurement of such transfers.

Measurement results

This section discusses the transmission latency and jitter measurements from the testbed. The audio applications' one-way transmission latency comprised multiple factors. Figure 4 depicts the path from audio sample compression in an audio sender to the availability of samples for further processing in an audio receiver. Only the components related to the wireless transmission are depicted. AD conversion and audio processing are omitted but must be taken into account when evaluating the application's user-experienced audio latency.

Figure 4. Components of the one-way latency related to wireless transmission in downlink (all values are typical)



The transmission latency can be separated into three sections, the audio packet sender, the packet-transferring 5G system, and the audio packet receiver.

Initially, the sender prepared audio samples for network transmission and sent them via Ethernet to the 5G system. Sample based processing in professional audio systems is typically done with 24 bit / sample resolution at 48 kHz sample rate. Using a proprietary method, individual samples were compressed to a 6.3 bit / sample while maintaining high-end audio quality.

After compression, multiple samples were combined into individual packets for wireless transfer. The size of such packets is determined by the periodicity of transfers. The rate at which an audio packet is sent out is a parameter not necessarily derived by the audio application and could be adjusted for optimized data transmission, e.g. for matching the wireless system transfer interval. As the 5G testbed was working with mini-slots of 0.5 ms the periodicity was set to this value, leading to 2,000 audio IP packets per second at the Ethernet interface at a data rate of 360 kbit/s including some overhead. The system allowed for each packet to be injected with a start timestamp just before sending it over Ethernet.

Audio packets from the mixer device were delivered to the core network over the Ethernet interface in the downlink direction. Audio packets were processed in the core network and then delivered to the base station; this processing involved about 150 µs delay.

In the 5G base station the packet scheduler determined timing for the radio transmission for each packet. Although both the audio system and the 5G system were synchronized with GPS timing, audio packets were scheduled 100 µs later than they arrived at the scheduler so that small timing fluctuations in the system could be compensated for, and audio system and 5G frame timing aligned.

To transmit data over the air with a low bit error rate, scheduled data packets were encoded with 5G L1 channel coding, which added 155 μs processing delay to the data transmission. The actual transmission of the data over the air interface based on two symbols took only 71 μs. However, the radio hardware processing delay added 107 μs to the air interface delay giving a total of 178 μs.

The received data from the air interface was then decoded (157 μs) and forwarded to the audio system through Ethernet (1 μs). In addition, jitter was introduced by the Linux test environment and other components and was compensated for via a buffer on the audio receiver.

The process for the uplink direction was like the downlink – the UE device received audio packets from Ethernet and transmitted to the audio system over the 5G radio. Delay components for uplink and downlink were the same, except the delay for channel coding - the encoding delay in the UE was 214 μs and the decoding delay in the base station was 304 μs.

The audio receiver took end timestamps and extracted the start timestamps on reception of an audio IP packet. Received samples were fed into a jitter buffer to ensure consistent processing. Finally, samples were decoded back to 24 bit / sample and were then available for further processing. Decoding was driven by the media clock. A proprietary error correction algorithm was used to conceal missing audio samples if the jitter buffer was unable to provide a sample. Error concealment was limited with regard to the lengths and frequency of errors to maintain the highest audio quality in professional audio applications.

Figure 5 shows on the left side the different latencies which occurred versus the absolute measurement time. As mentioned before, the basis for these measurements was the clock synchronization between the audio devices. The use of PTP over a dedicated Ethernet connection allowed hardware timestamps with a relative clock deviation smaller than 20 ns, allowing high-precision measurements.

The range of measured values varies between about 800 μs and 3.4 ms, where most of the values are in a range between about 800 μs and 1.8 ms. The distribution of the measured values is shown on the right side of Figure 5. Here, it is clearly visible that a transmission latency of approximately 800 μs was achieved for most packets.

Due to the isochronous nature of audio signals, the overall audio latency is not defined by the minimum occurring latency but by its jitter or maximum. Accepting no errors in the transmission, the transmission latency from this use case test would be about 3.4 ms.

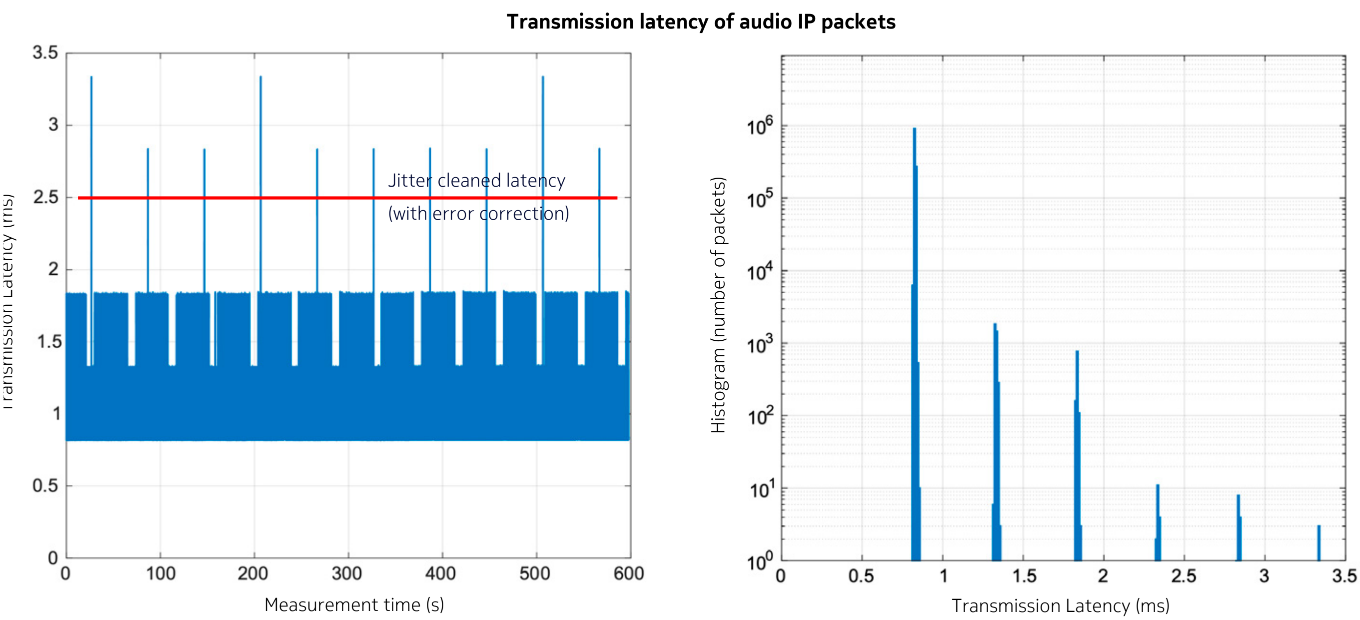
In the test system, the core network and SDR UE implementations were based on the Linux platform. Since Linux is not a hard real-time system it contributed to increased jitter in the end-to-end delay path. Higher jitter results in a more spread delay histogram in general and furthermore, a small amount of data (less than 1%) has latency increased to greater than 2 ms.

The most significant jitter in the system was introduced by the simulated core network where the jitter value could reach 2 ms. The packets delayed with variable jitter delay are scheduled to 5G for radio transmission with 500 μs granularity. While the jitter of most packets was managed by the postponed scheduling, some packets involved latency that was longer than the 5G transmission granularity and was scheduled to the next slot. This leads to sharp peaks aligned to 500 μs slot timing in the transmission latency histogram (Figure 5, right side). Therefore, latencies in the system are worst case and can be improved in commercial implementations with real-time operating systems.

Depending on the occurring error distribution and frequency, the concealment method used in this testbed allowed the jitter buffer size to be reduced to achieve a constant one-way transmission latency of 2.5 ms, resulting in a packet error rate of 10^{-5} when counting all packets slower than 2.5 ms as errors. In future, the jitter buffer could be reduced further if large latency spikes can be reduced.

Nonetheless, 2.5 ms of one-way transmission latency is an outstanding value. As described above (see Figure 1), the overall application latency is composed of two times the transmission latency plus AD, DA conversion and time for audio processing / mixing. This shows that approximately 7 ms of application (two-way) latency was achieved with 5G URLLC as a wireless high-quality link, coming very close to the specified 4 ms.

Figure 5. Transmission latency vs. measurement time (left), distribution of transmission latency (right)



Future outlook

Ecosystem

Chipsets and devices

The availability of URLLC systems for PMSE will depend on the availability of 3GPP Rel.16 based URLLC and TSN capable chipsets and devices. It is assumed that release 16 URLLC/TSN chipsets will be generic chipsets used by numerous applications and verticals including professional audio.

Standards development

While 3GPP releases 15 and 16 have implemented key 5G features to support Industrial Internet of Things (IIoT) and professional audio applications, the work continues in release 17. A specific study of enhancements tailored to video, imaging and audio for professional applications (VIAPA) is under consideration[6], and includes the support of scheduling based on a Survival Time in which consecutive packet errors are controlled.

Furthermore, when devices become available with advanced Ethernet functionalities specified in release 16, 5G time synchronization and jitter control to a sub-microsecond level will become viable. A subsequent development will be to optimize where de-jitter and buffering functions take place, e.g. in 5G rather than the professional audio applications, for best overall performance.

Deployment aspects

5G URLLC aims to support end-to-end latencies of lower than 1 ms, which would fully meet the needs of PMSE. In terms of practical deployment, enabling such low latency requires the 5G core functionality and any application processing to be performed locally in proximity to the 5G base station, to avoid introducing further latency into the chain. Deployment options for 5G-enabled PMSE should be explored further in this regard, e.g. including portable dedicated network deployments.

While 5G also supports a wealth of spectrum bands and spectrum sharing options for potential use by PMSE it should be noted that certain bands, e.g. FDD spectrum below 1 GHz, or high (mmWave) spectrum bands using high SCS values will support the lowest latency deployments. An additional consideration is the potential need to synchronize 5G PMSE spectrum use with other users of adjacent spectrum; especially TDD spectrum. Accordingly, further study into optimal 5G spectrum use by PMSE is recommended.

Conclusion

Interpretation of the results and how they lead to a solution

This paper analyzes the requirements of audio PMSE and discusses the potential of integrating audio PMSE with 5G URLLC by evaluating the network's latency performance when transferring isochronous audio data. It shows that 5G is capable of meeting the strict latency requirements of audio PMSE. Latency is one of the four key requirements of the professional audio use case; the others being reliability, spectral efficiency and synchronicity.

Furthermore, this paper discusses potential solutions for further latency optimization.

Next steps/future work

To continue the exploration into how capable 5G is to fully support professional audio, the logical next steps would include the replication of this proof of concept in a real-world environment, also including the evaluation of additional KPIs. In due course such testing should also embrace 3GPP Release 16 and Release 17 compliant equipment, following completion of the respective standards. In addition to technical feasibility, the appropriate deployment and commercial / business models will need to be fully considered, in order that 5G can be successfully positioned for use in audio PMSE.

Abbreviations

AD	Analog to Digital
DA	Digital to Analog
DCI	Downlink Control Information
eMBB	Enhanced mobile broadband
ENG	Electronic News Gathering
FDD	Frequency Division Duplex
gNB	5G base transceiver station
GPS	Global Positioning System
IEM	In-Ear Monitoring
IIoT	Industrial Internet of Things
IMT	International Mobile Telecommunications
IP	Internet Protocol
ISM	Industrial, Scientific and Medical (radio band)
ITU	International Telecommunication Union
ITU-R	International Telecommunication Union for Radio
KPI	Key Performance Indicator
MCS	Modulation and Coding Set
mMTC	Massive Machine Type Communication
NR	New Radio
PDCCH	Physical Downlink Control Channel
PDSCH	Physical Downlink Shared Channel
PER	Packet Error Ratio
PMSE	Programme Making and Special Events
PPS	Pulse Per Second
PTP	Precision Time Protocol
PUCCH	Physical Uplink Control Channel
PUSCH	Physical Uplink Shared Channel
QoS	Quality of Service
RF	Radio Frequency
SCS	Sub Carrier Spacing
SDR	Software Defined Radio
SPS	Semi-Persistent Scheduling
TDD	Time Division Duplex
TSN	Time Sensitive Networking
TTI	Transmit Time Interval
UE	User Equipment
URLLC	Ultra-Reliable Low-Latency Communications
VIAPA	Video, Imaging and Audio for Professional Applications

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About Sennheiser

Shaping the future of audio and creating unique sound experiences for customers – this aim unites Sennheiser employees and partners worldwide. Founded in 1945, Sennheiser is one of the world’s leading manufacturers of headphones, loudspeakers, microphones and wireless transmission systems. Since 2013, Sennheiser has been managed by Daniel Sennheiser and Dr. Andreas Sennheiser, the third generation of the family to run the company.

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