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In-Depth Latency and Reliability Analysis of a Networked Music Performance over Public 5G Infrastructure

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ABSTRACT

Networked Music Performances (NMP) are of increasing importance for musical enthusiasts, amateurs and professionals exploring new solutions and opportunities to rehearse or perform together at geographically distant locations. The use of public cellular connectivity to access wide area networks for such purposes can provide unique flexibility in planning NMP setups. Generally, latency and reliability are two key parameters for the end-to-end transmission of audio information through networks in this context. The stringent requirements of NMPs with respect to those parameters pose a major challenge for the current fourth generation (4G) of cellular technology. It is expected that the new and upcoming fifth generation (5G) of cellular technology will deliver significant Key Performance Indicator (KPI) improvements, and thus could be a promising enabler for musical interaction over distant locations. Currently, first public deployments of cellular 5G are being rolled out. This work presents an in-depth latency and reliability analysis of a distributed performance conducted over public 5G infrastructure in Finland. Measurement results suggest that Quality-of-Service (QoS) is needed in order to enable NMP over cellular 5G for many users and devices in a consistent, plannable and also flexible way.

1. Introduction

The increasing availability of public wide area network (WAN) infrastructure, whether wired or wireless, enables IP connectivity between almost any location worldwide for all kinds of purposes. One emerging field of application for such networks is collaborative interaction between musicians over distant locations. Latency and reliability are key parameters when transferring audio information over networks in such NMP scenarios. The network latency determines the acoustic delay among interacting artists and is of major importance for the musical outcome and the experience of participants [1]. Reliability of network transmissions is equally important. Lost audio information can either result in impairment of audio quality e.g., due to imperfect concealment or result in other disadvantages related to mitigation methods such as increased latency caused by retransmissions. End-to-end network latency and reliability are not only determined by the properties of the WAN but also by the end user's local wired or wireless access technology e.g., Wi-Fi, fiber or Digital Subscriber Line (DSL). The availability of this local infrastructure is often a key factor in the feasibility and planning of NMPs. Due to mobile use cellular technology has the potential to connect audio devices and services anywhere and anytime, enabling new interactions between musicians, audiences and musicians, and even between audiences [2].

The current generation of cellular technology (4G) is designed with KPIs of 100 ms one-way latencies (end-device to Internet-service) and 1 % packet loss, targeting general Internet-connectivity and some more forgiving real-time applications such as VoIP telephony [3], [4]. 5G is targeted to significantly improve the performance of cellular connectivity. It is expected that 5G might even deliver KPIs that can support some QoS driven applications.

The feasibility of NMPs with cellular technology is subject to ongoing research efforts. A central topic in this context is the evaluation of cellular performance with respect to NMP requirements.

2. Network Requirements of NMPs

The latency requirement in NMPs is subject to many studies (e.g., [5]–[7]). A maximum mouth-to-ear delay of 30 ms is suggested to allow synchronized and immersive interaction between musicians [1].

In contrast, reliability in NMPs is rarely considered in literature. One reason for that is the wide use of academic or leased-line networks with very high reliability in previous NMP research [1], [8]. Another reason could be the unclear definition of the term reliability in NMP contexts. While in other packet-based networking applications, reliability is often considered as the percentage of packets that are successfully transmitted from end-to-end, this statistical packet error ratio (PER) might not accurately reflect the requirements of live music streaming applications where the distribution of lost audio information over time can have significant impact on the subjectively perceived audio quality. E.g., 1 % of packet loss can describe a single burst of 600 ms of lost audio within one minute, or 10 equally distant 60 ms audio losses in a minute. Depending on the audio content and available concealment methods, a single 600 ms burst will more likely affect the perceived audio quality than ten 60 ms audio errors. The relation between packet loss, the distribution of packet loss over time, and perceived audio quality is not trivial and part of ongoing research efforts e.g., [9]. In general, this can be measured with subjective and objective tests, while objective measurements are typically preferred due to the expensive and time-consuming nature of subjective tests.

Perceptual Evaluation of Audio Quality (PEAQ) is an International Telecommunication Union (ITU) standard to measure perceived audio quality considering psychoacoustic effects. It is often used for the evaluation of audio codecs [10]. While PEAQ was used in some works to evaluate algorithms for packet loss concealment [11], other works have shown that PEAQ might not be suitable to evaluate the impact of network packet loss on audio quality, as it is not designed to reflect the specific properties of such transmission systems [9]. In [12] the average absolute error between original and impaired audio is used to measure loss in audio quality due to packet loss. [13] derived a NMP reliability requirement from a 3GPP study about the wireless link of professional microphone equipment on a live stage, which is based on a PER with assumed uniform error distribution [14].

To the best of our knowledge there is no widely accepted method for the objective evaluation of network-based packet loss in NMP scenarios. Consequently, no consensus about a reliability requirement that considers realistic packet error bursts can be found in literature.

Still, for practical purposes, an estimated PER requirement based on available information from similar applications can be useful to evaluate obtained measurement results. In general, NMP scenarios are not limited to specific proficiency levels of participating musicians or performance goals. Networked setups can include private rehearsals of music enthusiasts, master class learning situations with focus on musical timing, live concerts with local or remote audience, and remote production of musical recordings with emphasis on highest audio quality. Therefore, we foresee a wide

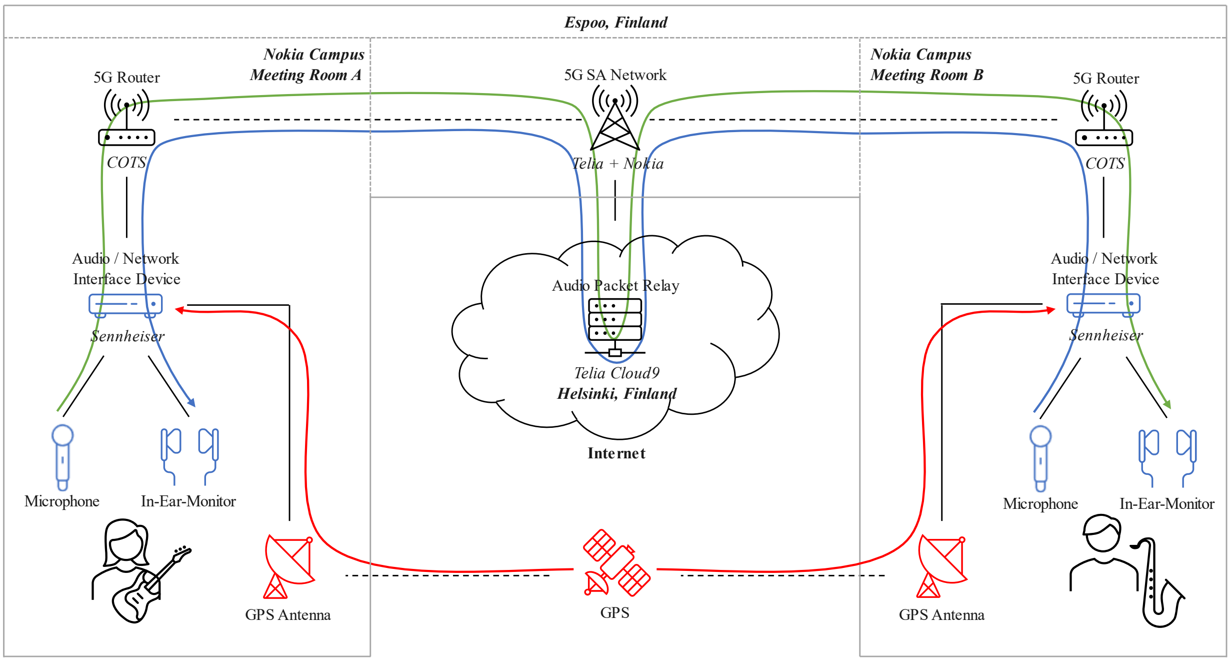

Figure 1: Overview network music performance system over public 5G infrastructure

range for potential reliability requirements in NMPs depending on individual musical goals.

The ITU recommends a packet loss ratio of 10^{-3} as a general IP network performance target to ensure best support for voice applications [15]. Forementioned 3GPP study recommends a PER of at least 10^{-6} for the transport of audio in high professional live performance scenarios [14]. Based on the assumption that NMPs need to deliver more complex audio content than speech, we suggest a potentially realistic PER requirement in the range of 10^{-4} to 10^{-6}.

3. Experimental Setup

In this work we created a NMP setup using public cellular 5G SA network infrastructure. The NMP system was setup on the Nokia campus in Espoo, Finland where two adjacent meeting rooms were acoustically connected through a 5G network. Figure 1 gives an overview on the overall setup.

Audio Setup

The audio setup of the NMP system consisted of a single microphone and an in-ear-monitoring (IEM) device in each meeting room. Each pair of microphone and IEM was connected to an audio / network interface device per room with analogue audio cabling. The interface devices translate analogue audio signals into IP-packets for network transport and vice versa. The devices are FPGA-based and allow deterministic processing for high-precision packet pacing and timestamping, as well as logging of received IP-packet latencies, jitter, packet loss, and packet reordering. A detailed description of the audio network interface devices can be found in [8], [16] and [17], where they were already used for high-precision measurements of different wired and wireless IP-networks.

In this setup, the audio devices were individually connected to commercial off-the-shelf (COTS) 5G routers with Ethernet cabling. One channel of compressed audio was transmitted through the 5G network. For transmission we selected a packet size of 5 ms as a reasonable trade-off between IP-network overhead and sample collection delay. Based on an audio sample rate of 48 kHz, each IP-packet contained 240 audio samples. The use of a proprietary codec otherwise used in professional wireless solutions for compression resulted in a data rate of approximately 0.5 Mbit/s in each direction. To optimize for latency, we used User Datagram Protocol (UDP) for transport and did not include any retransmission-scheme on application layer. The audio interface devices in both meeting rooms used the same GPS-based time for sample clocks, packet pacing and timestamping.

5G System

The COTS 5G routers (ZTE HyperBox MC801A) were connected to a Telia 5G base station as shown in Figure 1. The used 5G base station is a part of Telia commercial network operated in 3.5 GHz (3GPP band n78) frequency band. The used 5G routers were connected to a pre-commercial standalone (SA) mode. Although there were no other SA subscribers in the network, the base station offered commercialized non-standalone (NSA) 5G to subscribers in the same frequency band resources. As there was no kind of QoS-scheme in place for our SA subscription, our setup competed in a typical best-effort way with the other NSA enhanced Mobile Broadband (eMBB) applications. The used radio was a massive multiple-input and multiple-output (mMIMO) based remote radio head (RRH). The serving RRH was located on the top of the right-hand side building in Figure 2. The red triangle shows antenna direction. The two 5G routers in the adjacent meeting rooms were placed next to windows on the second floor of the building shown with red dot in Figure 2. Although the distance and angle between RRH and UEs was not ideal for the MIMO-beam, a very stable and good quality communications could be ensured through line of sight. The distance between UEs and the RRH was about 50 meters. The measured signal was good (Reference Signal Received Power (RSRP) -65 dBm).

Today, direct peer-to-peer communication is typically prohibited in mobile networks due to security reasons. A common technique to overcome such limitations is the use of a relay server on the Internet to which all mobile peers initially open an outgoing connection. The physical location of such relay servers in relation to peer devices and the point of network transition between mobile network and Internet needs to be carefully selected to keep added latency as small as possible. In this setup we deployed the relay server in Telia Cloud9 computing cloud service. The data center where Telia Cloud9 runs is located in Helsinki, Finland with only about 20 km distance from the radio site. As shown in Figure 1, audio packets traveled between audio devices and Telia Cloud9 over 5G and public Internet. The network transition between 5G user plane function (UPF) and public Internet is located in the south of Finland. In our setup the theoretically added delay from the detour can be assumed to be less than 1 ms. In other setups, where it might not be possible to place the relay server so close to the 5G routers and the network transition point, delay from additional wired transport might be significant.

4. Measurement Results

For the purpose of evaluating the connection between both endpoints over the 5G system we made long-term and high-precision latency, jitter, and packet loss measurements in order to obtain a quantitative basis for the evaluation of the state-of-the-art of current mobile technology in the context of NMP. Additionally, we invited two musicians to use the setup for a small rehearsal session to also

Figure 2: Layout and distance between remote radio head and 5G routers (from Google Maps)

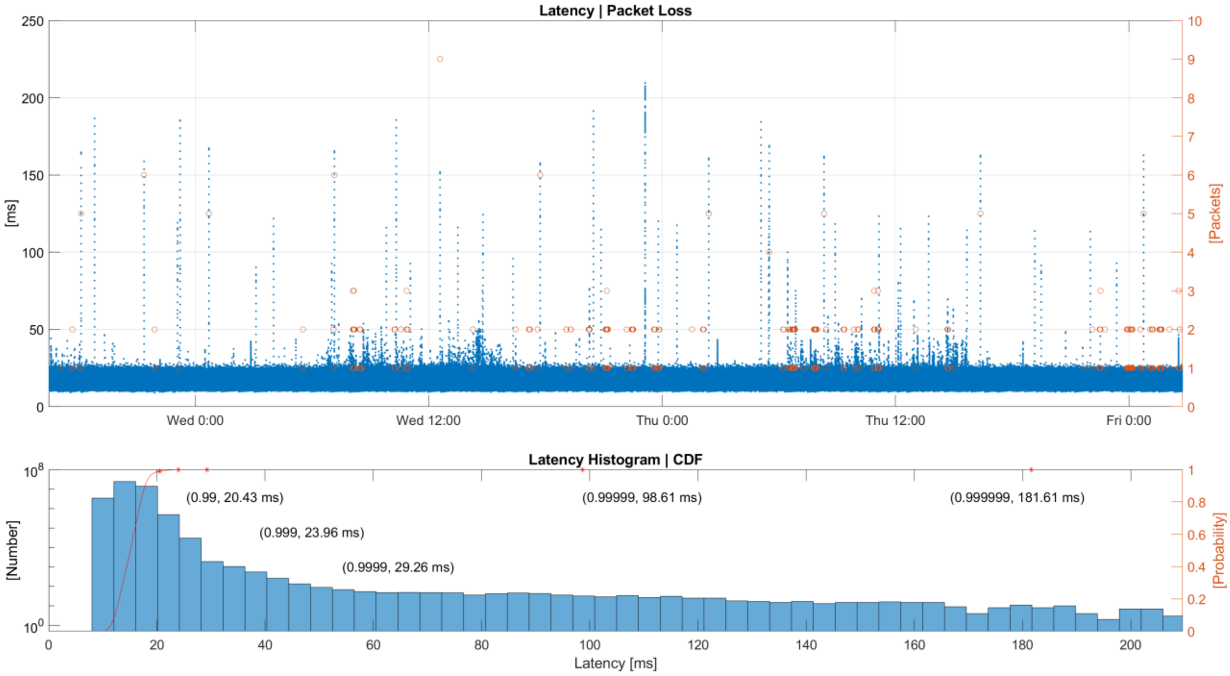

Figure 3: One-way latency over time (top, left axis), consecutive lost packets over time (top, right axis), latency histogram (bottom, left axis), and the latency CDF (bottom, right axis) of ~58.3 hours of audio over public 5G

collect subjective feedback.

1. Quantitative Evaluation

For our experiments we continuously transmitted audio for 3500 minutes (58.3 hours) from one endpoint to the other, and vice versa, while measuring the performance of the IP connection by means of latency, jitter, packet loss, and packet reordering of every IP-packet we sent. We started the experiment on a Tuesday evening (local time) and ended on a Friday just after midnight of the same week.

Overview

Figure 3 shows one-way latency over time (top, left axis), consecutive lost packets over time (top, right axis), latency histogram (bottom, left axis) and the latency cumulative distribution function (CDF) (bottom, right axis) of every audio IP-packet sent from one audio device to the other over the 5G network (about 42 million packets in total). Overall, the fastest packets show a latency of about 8.88 ms, the slowest are at 209.62 ms, and the mean latency over more than two days is 15.09 ms. 99 % of the packets are at a delay of about 20 ms. During the measurement time about 21000 packets are lost in total, which equals to a PER of about 0.05 %. No out-of-order packets were received.

Over the course of the measurement time some latency spikes of 100 ms to 200 ms stand out in contrast to the general visual baseline below 50 ms. However, at the time of finalizing this document, the cause for these peaks is not identified. By testing different architecture setups, we were able to exclude some components as source of the spikes (audio devices, 5G routers, audio relay service, 5G Radio Access Network, 5G core) while others (Configuration and parameters of 5G Radio Access Network and 5G core, Firewalls) could not be ruled out. We assume these latency spikes could be optimized by extensive investigation as they seem to be caused by unwanted behavior of a component in

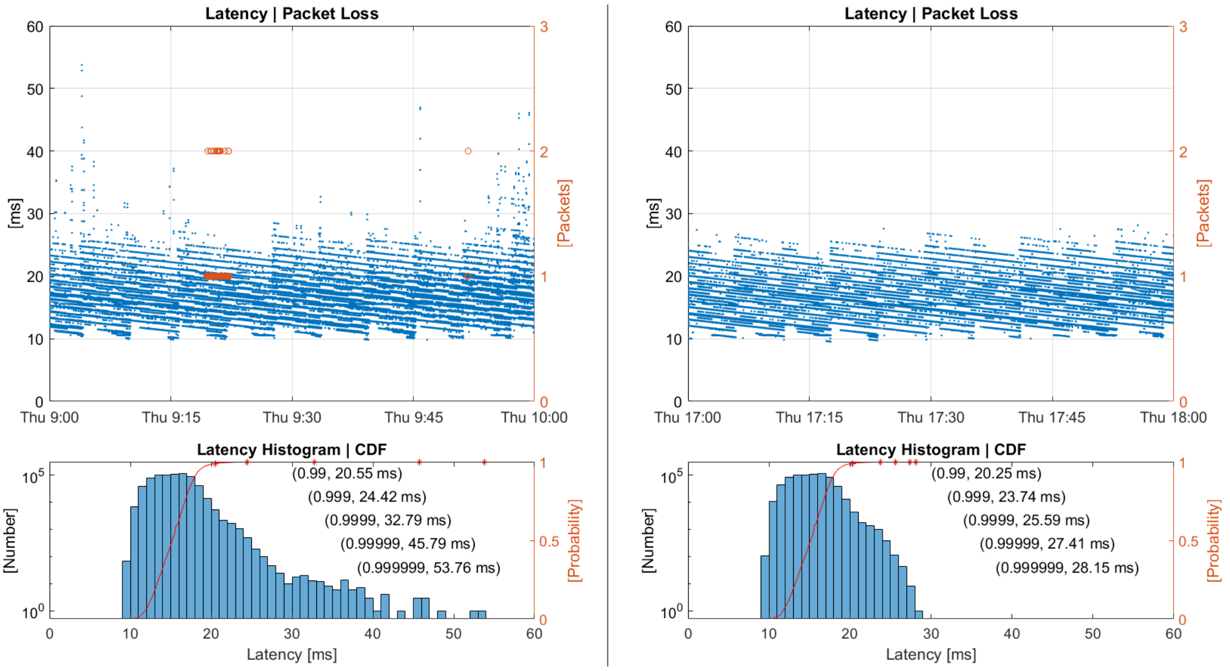

Figure 4: Exemplary network KPI comparison of two different times of day

the network and not by a systematic behavior by one of the deployed technologies.

From the depiction in Figure 3 (top) it is evident that latency jitter and packet loss significantly fluctuate over time. While jitter seems to generally be higher during daytime, more precisely during typical working hours (“nine-to-five”), packet loss seems to be uncorrelated to specific times of the day. Still, we observed phases were no packet loss occurred for several hours, while at other times the packet loss appeared in consistent bursts over longer periods. Packet loss also seems not to be correlated to jitter. Although, it is difficult to draw general conclusions due to the overall very low packet loss, the seemingly independence of jitter and packet loss could indicate that both have different causes.

The complexity and heterogeneity of the transport network used in our NMP trial and the fact that we only obtained end-to-end measurements makes it challenging to identify exact reasons for observed behavior. Assuming everything is working as intended, jitter and packet loss in wired networks typically only occur due to congestion, as bit corruption and link errors are negligible [18]. Here packet loss is correlated to jitter as it is a consequence of load-buffer-overflows. In wireless networks jitter and packet loss can have additional reasons, such as mobility, interference, handovers, and signal fading [18]. The static nature of our setup indicates that mobility, handovers, and signal fading most likely only played a minor role.

Different Times of Day

Figure 4 illustrates the fluctuation of one-way latency and packet loss over the day by comparing two exemplary sections of each one-hour length. The left side of Figure 4 shows latency over time (top, left axis), packet loss over time (top, right axis), latency histogram (bottom, left axis) and CDF (bottom, right axis) Thursday between 9 AM and 10 AM. The right side shows the same day between 5 PM and 6 PM.

Both periods show a very similar mean latency of about 16 ms. During the morning period several latency-spikes up to about 60 ms can be seen. Additionally, a large burst of packet loss occurred after 9:15 of about 5 minutes of length. In contrast the afternoon time shows relatively constant latency without noteworthy spikes, and no packet loss at all. Comparison of the CDFs in Figure 4 illustrate the performance difference. While the latency statistics for the lower end of 99 % of all packets are almost identical, the numbers for the upper end of 99.9999 % differs by a factor of approximately 2, reflecting the difference in latency spikes.

Dimensioning of Jitter Buffers for NMP

In NMPs it is desired to have a constant end-to-end streaming latency on application layer to ensure sufficient audio quality and stable tempo. This is typically achieved using jitter buffers to compensate the varying transmission latencies of individual packets. The dimensioning of such a buffer is not trivial and needs to be carefully considered, since its size trades off latency and audio-quality.

One practical approach for dimensioning a jitter buffer is to perform measurements like we presented in this paper to understand the network performance over time. From such insight an operating point based on the possible trade-off between latency and robustness can be selected. Since this approach is based on observing past behavior of the network, and assuming the future will perform in a similar way, it includes inherent insecurity, especially in best-effort wide area networks with many users. Thus, this approach works generally better if measurement data is acquired over a long period of time, and if the measurement period is most similar in terms of expected behavior of other network users compared to the time of operation.

The streaming latency after configuration of the jitter buffer size is constant and consists of the amount of audio that is packed into one IP packet (5 ms in this case) and the jitter-compensated transmission latency of the packets. The latter consists of the latency of the fastest packet plus the jitter buffer size. For the same network conditions different streaming latencies can be realized by selecting a different jitter buffer size.

In addition to actual lost packets in the network, packets that arrive too late for the selected jitter buffer size have also to be considered lost for the playback. Thus, the packet error ratio for a streaming application consists of the network packet error ratio plus the ratio of late packets. A jitter buffer with a size of 10 ms will handle packets which are maximum 10 ms slower than the fastest packet. All slower packets arrive too late and are therefore lost.

To dimension the jitter buffer, we considered the impact of the resulting streaming latencies on the reliability of the stream. Figure 5 gives the ratio of lost audio packets (lost & late network packets) for different assumed streaming latencies that are calculated from the measured packet latencies in our trial. Technically these configurations only differ in the size of the jitter buffer.

For all given assumed streaming latencies the ratio of lost audio packets is above the range of 10^{-4} to 10^{-6} specified in section 2. The recommended maximum streaming latency of 30 ms would result in a too low reliability (audio PER of 0.091481 %)

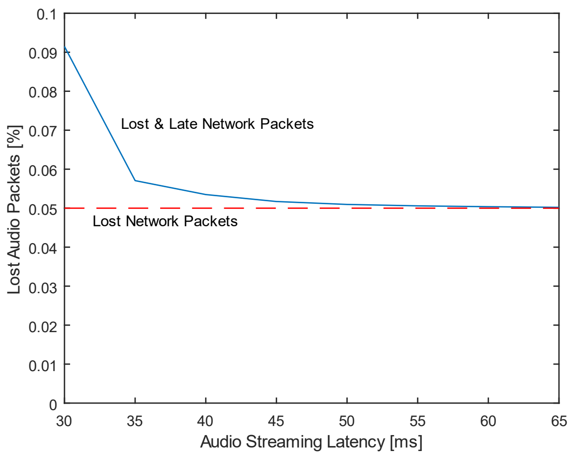

Figure 5: Trade-off between audio streaming latency and lost audio packets (lost & late network packets)

and therefore we investigated how much the jitter buffer needed to be increased to achieve a sufficient reliability. Increasing the assumed streaming latency from 30 ms to 40 ms by increasing the jitter buffer the reliability of the stream increased by a factor of about two. For higher assumed streaming latencies this ratio is dominated by actual lost network packets (0.05%) and not late packets. Therefore, a further increase of the buffer size would result in only minor improvement of the reliability. If the target is to optimize for the reliability, our measurements suggest selecting a jitter buffer size with a resulting stream latency of at least 50 ms for our particular setup.

The typical approach to deal with the remaining lost audio is to use a packet loss concealment (PLC) algorithm. For many PLC algorithms not only the ratio of lost audio packets (Figure 5) but also the length of the consecutive lost audio is important [19]. Figure 6 shows the calculated length of consecutive lost audio for different assumed audio streaming latencies. The maximum length of consecutive lost audio and the overall amount of lost

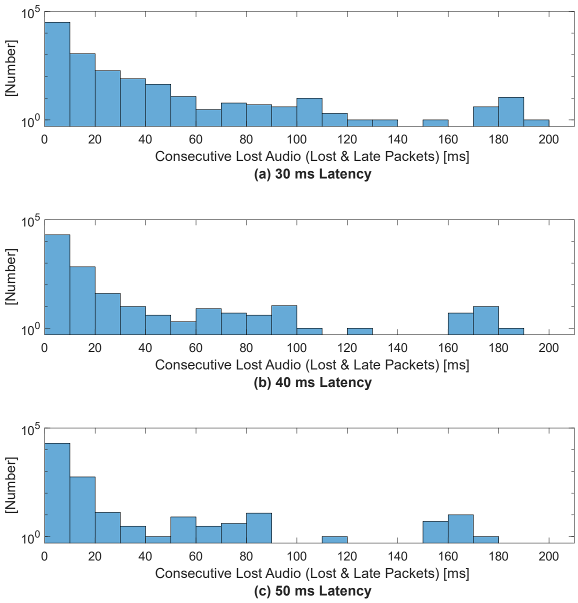

Figure 6: Logarithmic histogram of consecutive lost audio (lost & late network packets) for different assumed audio streaming latencies

audio (Figure 5) is decreased if a higher streaming latency is assumed for the application. Higher streaming latency is a result of a larger jitter buffer and therefore results in less packet discards because they are late.

Figure 6 (a) to (c) also show an accumulation of consecutive lost audio between 150 ms and 200 ms. This is caused by the unexplained and unusual latency spikes (see section 4). In general, most of the consecutive audio loss bursts are relatively short. This might indicate that PLC algorithms could be able to recover some of these gaps in the audio signal, depending on the audio content and the time between audio loss bursts.

2. Qualitative Evaluation

For a qualitative evaluation, we asked two musicians to use our NMP system for a rehearsal session. A duet of a saxophonist and a guitarist performed two jazz pieces (Autumn Leaves by Joseph Kosma and Summertime by George Gershwin) for about 1 hour from 6 PM to 7 PM on a Tuesday (Note: this was a different Tuesday, than the one partially captured in Figure 3). Since we knew of the partially non-predictable network performance, we selected the size of jitter buffers so that an application end-to-end latency of about 55 ms was achieved. The latency budget included about 50 ms of network latency / jitter, 5 ms of packet size for the capturing of audio samples and a delay smaller than 1 ms for processing and analogue-digital conversion.

We were aware that this latency is higher than what is recommended generally in literature. We assumed that the musicians will still be able to rehearse to some extend since it is known that musicians are able to trade-off quality- of-experience and latency with mitigation-strategies such as "leader-follower" [1].

The subjective feedback we collected from the two musicians after the rehearsal session indicated that the NMP experience was generally very favorable. The artists enjoyed performing together over 5G. Nevertheless, in some time periods during the rehearsal the musicians reported audible dropouts. Whether due to packet loss or latency jitter is unknown.

Note, that the musicians might have been biased in their subjective opinion as they are employees of Nokia's.

5. Discussion and Conclusions

Our long-term measurement with a NMP system over public 5G infrastructure showed that the network performance varied to a considerable extent depending on time of day. We assume that the cause for that is the use of non-exclusive wireless and wired network resources based on best-effort scheduling schemes. We observed a correlation of degraded network performance with typical working hours over multiple days. This might indicate that the main reason for temporary increased jitter in our setup is to be found in the **local** 5G cell or network. If the cause for degraded network performance would correlate to competing users in the public **wide area** networks we used, the increased jitter would have followed typical traffic pattern in such networks with peaks at around 8 PM [20].

We have seen that the network performance during some periods was well within the requirements of NMPs. From our measurements it is also evident that in other periods the network performance did not meet the general requirements due to the influence of other competing network users. It can be assumed that the negative impact will increase in the future since today only a minority (~10 % in 2022 [21]) of all cellular subscribers are using 5G, although the share might be higher in Nokia's campus area than international averages. Nevertheless, the musicians in our exemplary NMP were able to perform with each other over the 5G infrastructure for the time period of an hour.

From this we conclude: in order to enable NMP over cellular networks for many users / devices in a consistent, plannable and also flexible way, ad-hoc end-to-end QoS is needed. For that, the relay-service also needs to be located within the QoS domain, or peer-to-peer communication is required, to avoid the use of best-effort-based networks such as today's public Internet. The 3GPP 5G architecture provides the concepts needed to potentially support this vision ("slicing", "multi-access edge computing"). Deployments and commercialization of these new features are expected to be seen in the near future.

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