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Quantitative analysis of streaming protocols for enabling Internet of Things (IoT) audio hardware

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ABSTRACT

Given that traditional music production techniques often incorporate analog audio hardware, the Internet of Things (IoT) presents a unique opportunity to maintain past production workflows. For example, it is possible to enable remote digital connectivity to rare, expensive and bespoke audio systems, as well as unique spaces for use as echo chambers. In the presented research, quantitative testing is conducted to verify the performance of audio streaming platforms. Results show that using a high-speed internet connection, it is possible to stream lossless audio with low distortion, no dropouts and around 30 ms round-trip latency. Therefore, with future integration of audio streaming and IoT control protocols, a new paradigm for remote analog hardware processing in music production could be enabled.

1 Introduction

The Internet of Things (IoT) concept facilitates the development of ubiquitous interconnected devices, where physical electronic devices anywhere are extended into the virtual world allowing, them to accept, collect, and exchange data over wired and wireless computing networks. These devices act as "physical access points to Internet services" and have been used to enable, for example, smart homes and the sharing of environmental data from remote locations [1][2]. Given that early and traditional music production techniques were largely applied with use of analog audio hardware, the IoT paradigm presents a unique opportunity to maintain past (and perhaps lost or disappearing) music production processes and workflows. With IoT connected hardware, for example, it is possible to enable remote digital connectivity to rare, expensive and bespoke audio systems, as well as unique spaces for use as reverb and echo chambers. Furthermore, the IoT paradigm allows the possibility of a 'virtually-extended music studio', where a producer may work remotely on a project whilst still accessing processors and devices that are located in their personal studio. IoT for music production could therefore revolutionize the equipment hire market, enable new forms of creative collaboration, and redefine the technical boundaries for software plugins and audio equipment design.

IoT systems are already capable of enabling remote control of analog devices (such as smart home heating and security systems), however, a key obstacle to enable the paradigm of IoT controlled audio hardware is the need for reliable and high quality audio streaming to and from remote audio processing units. A number of technologies have been experimented with in this realm, mostly for enabling remote 'jamming' (collaborative improvisational performances) by musicians in real time, via internet connected systems. The SoundWire group at the Center for Computer

Research in Music and Acoustics at Stanford University conducts research into using the internet for music production and composition, and has created the JackTrip software as a means to distribute multitrack, high quality, uncompressed audio across the internet with low latency [3]. Additional research in this area includes the LOLA low latency audio visual streaming system (Conservatorio di Musica Giuseppe Tartini in Italy) [4] and Open Sound Control (UC Berkeley Center for New Music and Audio Technology) [5]. Whilst it is currently possible to achieve near-real-time two-way streaming of audio data (with accompanying video to assist remote performances), to date this has not been widely implemented with high quality, uncompressed audio packets. In order to unlock the IoT audio paradigm, it is essential that lossless audio data can be transmitted over the public internet to remote systems with very low latency and zero data losses or dropouts.

In the presented research, quantitative testing is conducted to verify the performance of audio streaming platforms which can enable networked, lossless audio delivery to support an IoT-based professional music system. The research initially uses signal analysis to compare two streaming platforms (JackTrip and WebRTC) and incorporates mechanisms to measure audio dropouts, distortion artefacts, and latency associated with each platform. Following, the preferred platform is tested to a finer level of detail over different computing networks.

2 Audio Streaming Test and Analysis Methods

2.1 Test Procedures

Three specific audio streaming experiments are conducted. These are:

1. Comparing the performance of JackTrip and WebRTC streaming platforms.
2. Investigating the performance of lossless streaming on local area networks with wired and wireless connections.
3. Evaluating streaming performance under differing wide area network conditions.

The primary aim of the chosen streaming tests is to observe and compare discrepancies between the source and transmitted audio files, as well as identifying errors that arise as a result of the streaming process. Typical audible streaming errors include clicks, pops, buzzing sounds, or gaps of silence in the output audio file, which can be owing to a number of signal processing and data transmission issues. Two pulse-code-modulation Microsoft Wave audio files are utilised for the streaming trials [6]. These are:

a. 10 second 1 kHz sine wave
b. 30 second frequency sine sweep from 0-22.5 kHz

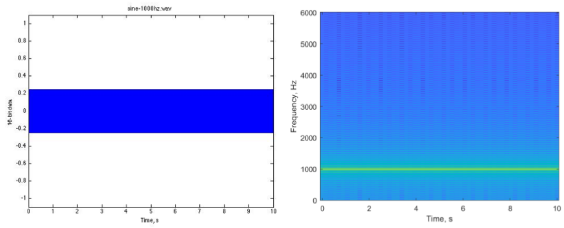

Figure 1. Sine wave source audio waveform and spectrogram.

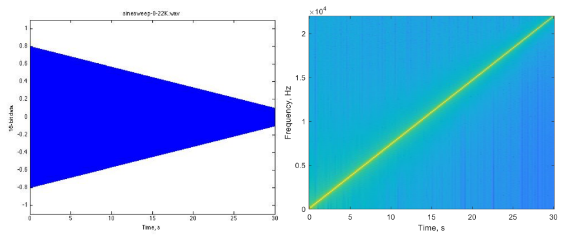

Figure 2. Sine sweep source audio waveform and spectrogram.

The 10 second 1 kHz sine wave (Figure 1) provides a consistent stream of audio at a single frequency, allowing easy observations of data drop outs or distortion to the signal that may occur as a result of streaming. The 30 second 0-22.5 kHz frequency sine sweep (Figure 2) determines if the streaming platforms accurately preserve or alter any specific range of frequencies within the audible human hearing range. Both audio files are single channel (mono) and presented at a 44.1 kHz sampling rate.

Each audio trial is conducted 5 times allowing them to be evaluated for performance consistency and repeatability. Audio is streamed from one networked computer to a secondary computer, where the transmitted audio is then recorded as a new Wave audio file, matching the settings of the source file. Audio is analysed using Matlab scripts to visualise waveform and spectrogram data as well as measure specific performance characteristics of the transmitted audio in comparison to the source. The focuses of the streaming analysis are described in detail in each case below.

2.2 Measuring Dropouts

In the context of this research, audio dropouts account for any sudden loss or fluctuation in the audio data that causes instantaneous step changes in the transmitted signal, altering its characteristics from the source sound. Dropouts can produce undesired glitches including clicks, pops, and intermittent loss of sound in the audio playback resultant from interruptions to the data packet stream [7]. Dropouts become more pronounced in real-time applications because the low-latency requirements “inhibit retransmission of lost packets,” and issues such as network link failures, routers discarding packets, packets being received out of order or delayed in delivery (jitter), and packets being disregarding by the receiver after being received too late for playback all contribute to these interruptions [8]. Some examples of audio dropouts in a 1000 Hz sinewave are shown in Figure 3.

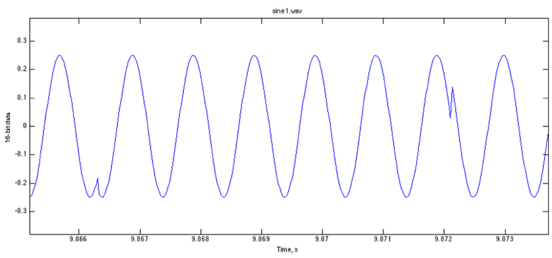

Figure 3. Example audio dropouts identified at 9.8663 seconds and 9.8721 seconds.

It is possible to count audio dropouts when using a test sinewave by evaluating the sample-to-sample difference in the received audio data. The greatest possible inter-sample difference for a 1 kHz sinewave, normalised to unity amplitude and sampled at 44.1 kHz, is approximately 0.15, which is observed at the sine wave’s maximum gradient at the point of zero crossing, as shown diagrammatically in Figure 4.

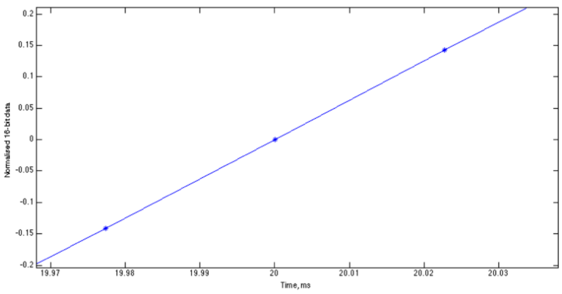

Figure 4. A 1 kHz sine wave’s maximum gradient and inter-sample difference (=0.1425) when sampled at 44.1 kHz.

The exact value for the maximum sample-to-sample difference is calculated as follows:

The gradient of a sine wave is calculated by

$$\frac{d}{dt}\sin\omega t = \omega\cos\omega t \tag{1}$$

where ω is the angular frequency (rad/s) and t is time (s). The maximum gradient is hence when $cos(\omega t) = 1$, so the maximum gradient of a sine wave is simply $\omega = 2\pi f$, where f is frequency (Hz). The sample period, P, for a signal sampled at 44.1 kHz is 1/44100 seconds, so the maximum amplitude increment per sample of a 1 kHz sinewave, is calculated as

$$2\pi f P = 2\pi * 1000 * \frac{1}{44100} = 0.1425 \tag{2}$$

It is therefore possible to count dropouts by identifying any consecutive sample value step changes in the received sine wave that exceeds 0.1425 multiplied by the amplitude of the sinewave. It is of course possible for a dropout to leave samples perfectly aligned as a matter of coincidence, and in such rare cases may be missed by the proposed dropout counting method. While more elaborate algorithms for identifying dropouts might be possible, the method proposed here is sufficiently accurate for evaluating the relative performance of network audio platforms.

2.3 Measuring Distortion Artefacts

As discussed by Moore et al. [9] and Toulson et al. [10], for example, nonlinear distortion refers to the introduction of harmonic and inharmonic frequency components that were not present in the original signal. The amount of unwanted harmonic distortion can be calculated as total harmonic distortion (THD), where harmonic frequencies are measured at integer multiples of the fundamental test frequency. THD is usually calculated as a percentage based on the ratio of the power sum (root-mean-square) of all the harmonic components to the power sum of all the harmonics plus the fundamental [11].

When evaluating a single sinusoid test signal, spectral powers which are not identified as fundamental or harmonic are classified as noise. The noise can also be quantified as a percentage of the fundamental frequency power (N), so allowing the value of THD+N to be calculated. THD+N is a much simpler quantity to measure collectively (rather than separately for THD and separately for N) for a single sinusoid test, since it essentially refers to the power of spectral components that are evident in the processed signal when the raw test signal component is removed, as discussed by Prism Sound [12], who are leading manufacturers of audio test and measurement equipment. In line with the published recommendations, THD+N is measured in this research by applying brick wall filters in the frequency domain after the signal spectra has been calculated. The filtering includes a notch filter around the 1000 Hz test frequency, with a low-cut filter implemented at 22 Hz, and a high-cut filter implemented at 22 kHz. In order to be sure of removing any side bands in the signal spectrum, the notch filter is set relatively wide to cut all frequencies between 900 Hz and 1100 Hz. The filtering profile applied for calculating THD+N is shown in Figure 5, which displays the frequency spectra of a distorted 1 kHz sine wave as an example.

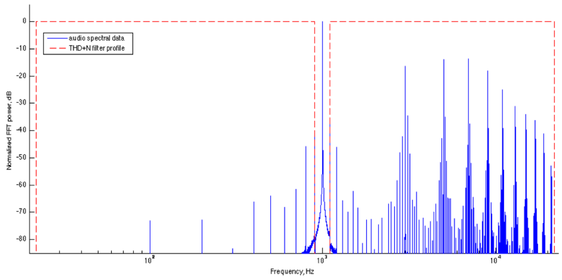

Figure 5. Example distorted 1 kHz sinewave spectrum with THD+N filter profile.

Processed audio can exhibit additional distortion and noise depending on the mechanisms and tools used in the transmission or recording. While there are no widely agreed values for acceptable THD+N ranges, it is desirable to obtain the smallest ratio possible, and for the purpose of this research, relative comparison between test results is of most value.

2.4 Measuring Latency

The official definition of latency in digital technology is "the time required online or in a network for the one-way or round-trip transfer of data between two nodes" [13]. While latency measurements up to 150 ms is deemed acceptable in traditional telephony cases, the average person begins to perceive an individual sound as two distinct sounds after 30 ms of latency [14] and some musicians can perceive the effects of latency at much lower thresholds, sometimes lower than 25 ms dependent on the style of music [15]. Particularly for live-performance and real-time audio scenarios, audio transfer relies on small buffers with no compression [15] and, due to these strict parameters, a "sudden, unexpected, increase in latency can cause a drop out in the signal at the destination" [16].

Source-to-destination latency measurements are useful for networked musical performances and online jamming sessions, but an IoT music application where audio needs to transmitted to a remote node and returned to a central location benefits from the observation of round-trip latency times. Measuring the roundtrip time of audio transmission over the network can become complicated when incorporating heterogeneous A-D (analogue to digital) and D-A (digital to analogue) processors that account for additional delays in their hardware or software. Bouillot and Cooperstock [15] propose a manual mechanism for measuring latency using a multi-channel audio editor to compare the time difference between the playback of the source file and a captured recording of the audio as it delivered to a remote node over the network and returned back to the source. Building upon this concept, round-trip latency measurements in this research are obtained by configuring the audio interface of the server computer to loopback any audio streams received from the client. As the client computer transmits audio, it simultaneously records the audio returned from the loopback server and the timing delay between the two streams determines the round-trip latency. Halving the round-trip delay time determines latency from source to destination.

Figure 6. Observing delay between the source and returned audio streams to determine round-trip network latency, using Logic Pro X software.

By setting the linear timecode (LTC) in a desired audio editing software to display in milliseconds, the round-trip latency can be determined by observing the offset of the start time of the recorded audio as compared to the initial source audio file as shown in Figure 6.

3 Comparing the Performance of JackTrip and WebRTC

JackTrip and WebRTC are both viable platforms for internet-based audio streaming applications due to their offers of high quality media distribution with low latency. JackTrip is presented as an effective tool for online jamming, allowing musicians in various remote locations to play instruments together and engage in real-time musical performances over the internet [3]. Such performances are perceived as synchronous with minimal, if any, noticeable timing differences despite large physical distances. In comparison, WebRTC is widely used for online video chat applications that offer similar benefits to Skype, allowing video and voice conversations to occur naturally and in real-time through a web browser [17]. The transfer of high quality audio with low latency is the driving appeal for both platforms, however they differ in the fact that JackTrip caters more towards music applications, which includes retaining the accurate frequency profiles of musical instrument sounds, where WebRTC employs mechanisms to optimise voice conversations, including codecs such as the iSAC and iLBC audio codecs by Global IP Solutions that are incorporated into many Voice over I.P (VoIP) applications [17].

3.1 JackTrip vs WebRTC Waveforms and Spectrogram Results

In order to conduct the streaming tests, a Mac Pro desktop computer housed at Anglia Ruskin University in Cambridge, UK was configured as a server computer to allow streaming connections from computers both internal and external to the network. A secondary MacBook Pro laptop connected by Ethernet on the same local area network (LAN) was configured as a client and connected to the server. The tests produced 5 recordings of each audio sample, resulting in 10 recordings total for both the JackTrip and WebRTC scenarios.

During these tests, particularly noteworthy observations came from the sine sweep, which showed filtering of higher frequencies in the WebRTC tests, as shown in Figures 7 and 8.

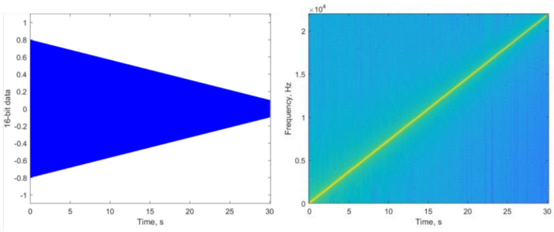

Figure 7. Example 0-22.5 kHz sine sweep LAN capture waveform and spectrogram with JackTrip.

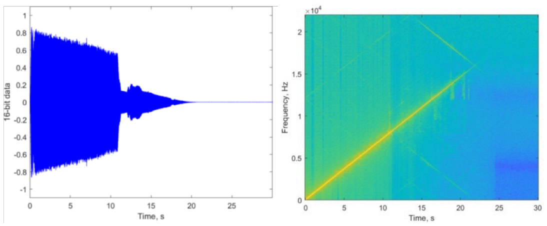

Figure 8. Example 0-22.5 kHz sine sweep LAN capture waveform and spectrogram with WebRTC.

The analysis of the JackTrip audio recordings showed that audio streamed across Jacktrip accurately maintains the characteristics of the source audio without any additional filtering or processing during transmission. The waveforms of the WebRTC captures do, however, show explicit differences from that of the source audio file and are especially prevalent in the sine sweeps, where filtering and compression is observed correlating to WebRTC's use of VoIP codecs tailored for video and voice chat scenarios.

3.2 JackTrip and WebRTC Distortion, Dropouts, and Latency on LAN

The amount of distortion, number of dropouts, and latency was evaluated for the sinewave source file transmitted using JackTrip and WebRTC over the LAN. The results are averaged over five repeats of each test, shown in Tables 1 and 2.

	Number of Dropouts					
Platform	T1	T2	T3	T4	T5	Average
JackTrip	0	0	0	0	0	0
WebRTC	0	2	0	1	0	0.6

Table 1. Dropouts for sinewave tests on LAN.

	THD+N (%)					
Platform	T1	T2	T3	T4	T5	Average
JackTrip	0.0358	0.0358	0.0369	0.0358	0.0358	0.03602
WebRTC	0.387	1.2633	0.518	0.6178	0.4668	0.65058

Table 2. Distortion measurements for sinewave tests on LAN.

Special conditions needed to be made for WebRTC latency tests as a simple loopback server could not be set up as with JackTrip. In order to retrieve latency measurements for WebRTC, a 1.5 meter audio cable was used to physically loopback the audio transmit to the server back to the client. As a result, 3 latency measurements were taken; one round-trip latency measurement for JackTrip using a loop back server, and two for both JackTrip and WebRTC using a physical loopback. Given that a sound signal propagates through a cable at approximately the speed of light [18], any additional delay through the cable is miniscule and mirrors a one-way, source-to-destination connection. Results are given in Table 3.

	Roundtrip Latency (ms)					
Platform	T1	T2	T3	T4	T5	Average
JackTrip *(loopback server)*	26	32	32	32	20.5	28.5
JackTrip *(physical loopback)*	13	8.5	16.5	17.5	19	14.9
WebRTC *(physical loopback)*	85.5	92	109.5	80.5	89.5	91.4

Table 3. Latency measurements over LAN.

While audio streaming over the LAN proved to be reliable and sufficient for real-time music delivery applications, tests showed a higher quality of performance from JackTrip compared to WebRTC. This is proven by accurate representations of JackTrip waveforms to the source audio as compared to WebRTC and lower latencies achieved by JackTrip. These initial tests were conducted over wired Ethernet connections; however, the development of wireless internet networks has made computing resources widely available for mobile applications without the need to be tethered to a specific location. The proceeding tests implemented JackTrip streaming over Wi-Fi.

An Ethernet connection provides a straight physical connection between a computing device and the internet. As a radio signal, however, Wi-Fi is subject to interference from other wireless broadcasting devices. These interferences often also result in increased latency due to competing network traffic and worse, dropouts and distortion [19]. Increased latency, dropouts and distortion can all be identified in both Table 4. and visibly in the spectrograms shown in Figure 9.

	JackTrip Wi-Fi Measurements					
Category	T1	T2	T3	T4	T5	Average
Dropouts	78	113	131	20	14	71.2
THD+N	2.7718	1.598	3.0396	2.2725	2.2855	2.39348
RT Latency	41.5	41.5	35.5	41.5	64.5	44.9

Table 4. JackTrip LAN streaming measurements over Wi-Fi.

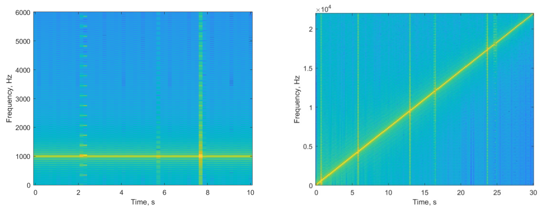

Figure 9. Sample spectrograms of 1 kHz sine wave and 0-22.5k sine sweep transmitted over Wi-Fi

4 Wide Area Network Testing

JackTrip showed capabilities to support high quality networked audio distribution and objectively outperformed WebRTC for measured and perceived audio quality. The following set of trials examine JackTrip streaming capabilities in real-world environments independent of the Local Area Network, and incorporate external computers housed outside of Anglia Ruskin University, extending distributed music applications into the Wide Area Network (WAN). Two scenarios were evaluated; JackTrip streaming over commercial or commodity networks, and JackTrip streaming over high-speed National Research and Education Networks (NRENs). Both scenarios were tested on Ethernet connections between London and Cambridge, UK with the commercial network streaming between a residential home and Anglia Ruskin and high-speed research networking streaming using the UK's JANET network at the University of Westminster, London. A 256 sample buffer size was used as determined to be most effective for the commercial networks.

Results are given below as example spectrograms (Figure 10), and tabulated dropout and noise measurements (Tables 5 and 6 respectively) for both NREN and commercial streaming tests.

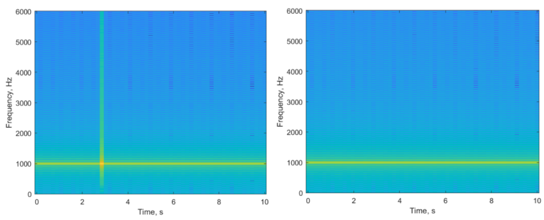

Figure 10. Sample spectrograms of 1 kHz sine wave for commercial network (left) and high-speed research network (right)

	Number of Dropouts					
Network	T1	T2	T3	T4	T5	Average
Commercial	0	1	1	0	6	1.6
NREN	0	0	0	0	0	0

Table 5. Dropouts for sinewave tests on commercial and NREN networks.

	THD+N (%)					
Buffer Size	T1	T2	T3	T4	T5	Average
Commercial	0.0358	0.8307	0.1799	0.0356	2.2168	0.65976
NREN	0.0359	0.0356	0.0357	0.0357	0.0358	0.03574

Table 6. Distortion measurements for sinewave tests on commercial and NREN networks.

Over the set of trials, the number of dropouts occurring over the commercial/commodity network did not show promising results. Streaming over an

NREN performed optimally, providing only 1 error in all of the tests and resembling the success rates of a local area network, indicating a level of consistency and reliability that is necessary for commercial applications which might utilise audio-IoT systems. Given the success of the NREN streaming in these tests, it was evaluated further to identify if the system latency could also support audio-IoT applications. As displayed in Table 7, the measured average round-trip latency of the NREN was just under the range where delays are perceived by the human ear.

NREN Roundtrip Latency (ms)					
T1	T2	T3	T4	T5	Average
30	25	36	31	25	29.4

Table 7. Latency measurements on NREN network.

Results indicate that the commercial/commodity network displayed challenges in supporting real-time audio streaming and showed dropouts in almost all trials conducted. It is possible that the lower bandwidth designations and overall network congestion on commercial networks can create more opportunities for dropped packets and errors in the data stream [20]. In the present conditions, these tests showed that the current commercial/commodity computing networks are not yet capable of supporting real-time high quality, low latency music transfer.

NRENs provide high bandwidths suitable for transferring large data sets and have demonstrated success in low latency audio streaming applications with JackTrip [3] as well as high-speed video and audio transfer with the LoLa low-latency media platform [21]. Minimal audio errors were observed with almost all but one test showing no audio dropouts. Round-trip latency was kept at a minimum to ensure unperceivable delays in transmission, and it is valuable to note that professional digital audio workstation systems (such as Pro Tools and Logic Pro) are capable of implementing delay compensation functionality for known and reliable latencies in the region of those measured. In present conditions, the research showed viable results for implementing an IoT-based music system with real-time audio transfer, utilising high-speed LAN and NREN networks.

5 Conclusions

Results show that using JackTrip on a LAN or high-speed WAN, it is possible to stream 24-bit 44.1 kHz PCM audio with low distortion, no dropouts and around 30 ms round-trip latency. The test system presented is further demonstrated, for example, to enable a remote acoustic reverb chamber to be effectively used with a conventional digital audio workstation with both remote control of the hardware effect parameters (i.e. amplification and wet/dry mix control) and round-trip streaming that is comparable to that of conventional DSP intensive music production tools, such as pitch correction and convolution reverb. As a result, the IoT paradigm is shown to be viable for future innovation when commercial internet bandwidth is improved to the capabilities of the current high-speed networks. With future integration of audio streaming and IoT control protocols, user interface development, and the effective design of customer service models, a new paradigm for remote analog hardware processing in music production could therefore be realised.

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