LTE ‘Real World’ Performance Study

Broadband and Voice over LTE (VoLTE) Quality Analysis:

TeliaSonera, Turku, Finland

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Executive Summary

In January 2011 Epitiro announced its commitment to test commercially available installations of 4G LTE services for the purpose of understanding the actual performance delivered to subscribers.

In March 2011 Epitiro partnered with Finnish consultancy group *Genre Mobile* to conduct indicative research into TeliaSonera’s LTE rollout in Finland. Around-the-clock measurement of broadband performance and Voice over LTE call quality was conducted using Epitiro’s 4G ‘Consumer experience’ test probes that automatically connect to the internet as a typical consumer. Testing was completed over a 5 day period with over 20,000 data points being collected. A 3G test probe also measured services to enable direct comparison to 4G LTE services.

The results of the testing clearly showed 4G LTE to perform significantly better than 3G services. Broadband download speeds of up to 48.8Mbps were recorded with an average of 36.1Mbps. 3G download speeds were 4.1Mbps on average. Average latency for 3G was 117ms, while 4G services were five times faster at an average of 23ms.

Analysis of international Voice over LTE calls, initiated on the Finnish LTE network and terminating on the BT’s UK 21CN PSTN network, showed that average call quality, according to the ITU-T PESQ MOS analysis, was excellent – with less than 0.3% of calls being of inferior quality.

Based on observations, TeliaSonera’s LTE rollout is equal to high-speed fixed line service quality seen in leading countries. Further, the low latency times measured indicate that the LTE service in Finland would be capable of handling demanding web-based applications including VoIP, video streaming and HD IPTV.

Being a new service, the subscriber base is minimal and not at the same levels as 3G. Thus the 4G network is not currently subject to the effects of contention from simultaneous demand. In-motion resting was not completed due to the relatively small area in which 4G is currently available so in-motion performance on trains, in cars etc. is not yet understood. Epitiro will remain interested to understand how performance varies for in-motion work and as popularity of 4G services increases.

The Potential of LTE

In recent years the market demand has grown considerably for mobile broadband with consumers and businesses now enjoying the full benefits of emailing and web browsing on the move. However, limitations in speed and high latency (compared to fixed line broadband) has made 3G services unsuitable for time-sensitive applications such as VoIP, video streaming and on-line game play.

The fourth generation (4G) mobile communications protocol theoretically can reach speeds in excess of 100 Mbps with minimal latency thus potentially becoming an access technology capable of handling all applications from basic email to bandwidth-demanding HD video.

As affordable higher-speed 3G+ (HSPA) services have been the driver behind the increase in smartphones, 4G is expected to drive a new wave of devices and applications that can benefit from high-speed, high-quality mobile broadband.

The easier expansion of services geographically is also a driver behind 4G. Wireless communications has long been envisaged as a solution for providing rural communities and developing nations with high-speed broadband service. With its accessibility via mobile phones and USB modems has the potential to become the preferred ‘last mile’ link for high-speed broadband access in the future.

LTE in Finland

In November 2010 TeliaSonera announced the availability of Finland’s first 4G LTE service in Turku and Helsinki, forecasting typical speeds between 20Mbps and 80Mbpsand operating on the 2.6GHz frequency band.

At the time of the announced 4G service TeliaSonera had not decided its suppliers for the national 4G LTE network rollout with Ericsson and Nokia Siemens Networks (NSN) infrastructure being used in Turku and Helsinki, respectively. The 4G modems that TeliaSonera launched in Finland are Samsung USB modems supporting 4G, 3G and 2G network connectivity.

Scope of Testing and Methodology

The aim of the project was to undertake initial findings of LTE performance compared to available 3G services in context of both broadband and IP voice quality (Voice over LTE).

From March 21st to March 25th 2011, Epitiro's AT400 network measurement probes conducted testing at 30 minute intervals. Each test consisted of Epitiro's automated mobile broadband test probes connecting to both 4G and 3G services and collecting a series of key performance indicators through active test sequences. Testing was conducted from environmentally controlled locations in Turku, Finland where strong radio signals were available. *Genre Mobile* provided the in-country logistics.

The test probes were dedicated solely to testing the LTE and 3G services and were remotely supported and monitored by Epitiro NOC staff based in the UK. Each test probe executed the same set of test scripts with the sole difference between them being the USB modem type (3G / 4G).

Pre-test engineering analysis indicated that the data caps of 30GB per month for LTE and 20GB per month for 3G would not be exceeded.

Data was analysed using Epitiro's ISP-I Quality of Service Analysis solution. ISP-I is designed to analyse large volumes of voice, video and broadband data from both fixed and mobile networks.

Significant statistical analysis was undertaken to ensure the findings were representative of true Subscriber experience. The data presented represents a 95% confidence level around the mean. That is, Epitiro is confident that should the same survey be undertaken, it is 95% certain the results would be the same.

Applications and Key Performance Indicators

The project test scripts were configured to collect key performance indicators on metrics that affect popular applications such as web browsing, VoIP telephony, video streaming, file downloading and online game play.

Web Browsing

The web browsing quality of experience is generally associated with the time it takes to locate and download a web page within a web browser application. The speed or bandwidth of a customer’s connection is one a factor. In this research Epitiro examined underlying aspects such as the cached and non-cached HTTP download speeds and DNS Server resolution times. We also recorded average web page download times for an understanding of actual consumer Quality of Experience.

File Download

Whether it is email attachments, MP3 songs or programme updates, internet services are frequently used by consumers to download large data files. We conducted multi-threaded TCP Throughput tests to fully understand uplink and downlink capacity.

Voice over LTE (VoLTE)

An increasingly popular application for IP-based networks is VoIP telephony including free (Skype), commercial services (SIP) and femtocell-based mobile calls via consumer broadband (xDSL and Cable) services. Regardless of the increasing volumes of data traffic it is important for operators to deliver a good voice experience. LTE is the first 3GPP radio access technology (RAT) that does not support circuit switched voice calls, in that it does not have dedicated channels for circuit-switched (CS) telephony. LTE instead relies on the end-to-end IP connection from the smartphone to the core network and beyond.

With LTE we are migrating from a world of circuit switched certainty for voice quality, to a world of uncertainty with VoIP where the network needs to provide "end to end" QoS to deliver acceptable voice quality.

Overall voice quality was analysed as well as call setup time, call success ratio and underlying metrics that affect telephony.

IPTV/On-Line Media

Applications such as streaming of internet music, on-line radio stations and video media (BBC iPlayer) are increasingly becoming popular with end users. Again, packet loss is one essential metric this that was captured as a key performance indicator of potential customer experience.

Gaming

Popular interactive gaming on the web – Xbox, Sony Playstation – allows end users around the globe to compete but relies on the internet to be responsive to player commands. Round trip latency time was measured using ICMP Ping tests.

Technical KPIs

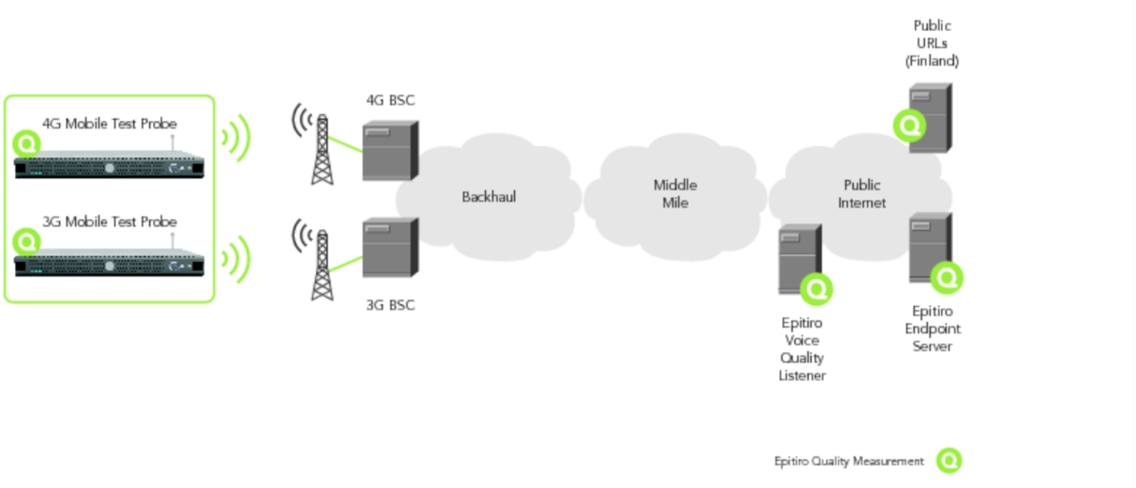
The technical KPIs collected for broadband performance include download speed, upload speed, DNS resolution time, web browsing speeds, web page load times, network jitter, network packet loss and latency. Voice quality analysis was completed using PESQ, the ITU standard for measuring a ‘mean opinion score’ based on comparison of transmitted and received audio quality.

Considerations

The purpose of this project was to capture indicative results of LTE performance and as such the scope of testing completed in this project was limited in some areas.

- Data was captured over 5 days which does not allow us to show trends in performance as additional LTE subscribers partake in the service.
- In order to ensure future reports on LTE performance in Europe are comparable, download and upload speeds were measured to popular European peering points in Amsterdam (AMSIX) and the UK (Telehouse). Speeds measured to endpoints within Finland might show services to be faster.
- Voice quality was measured via IP connectivity only. Interoperability testing was not conducted between IP and the UK's 21CN PSTN network.

Epitiro Test Configuration

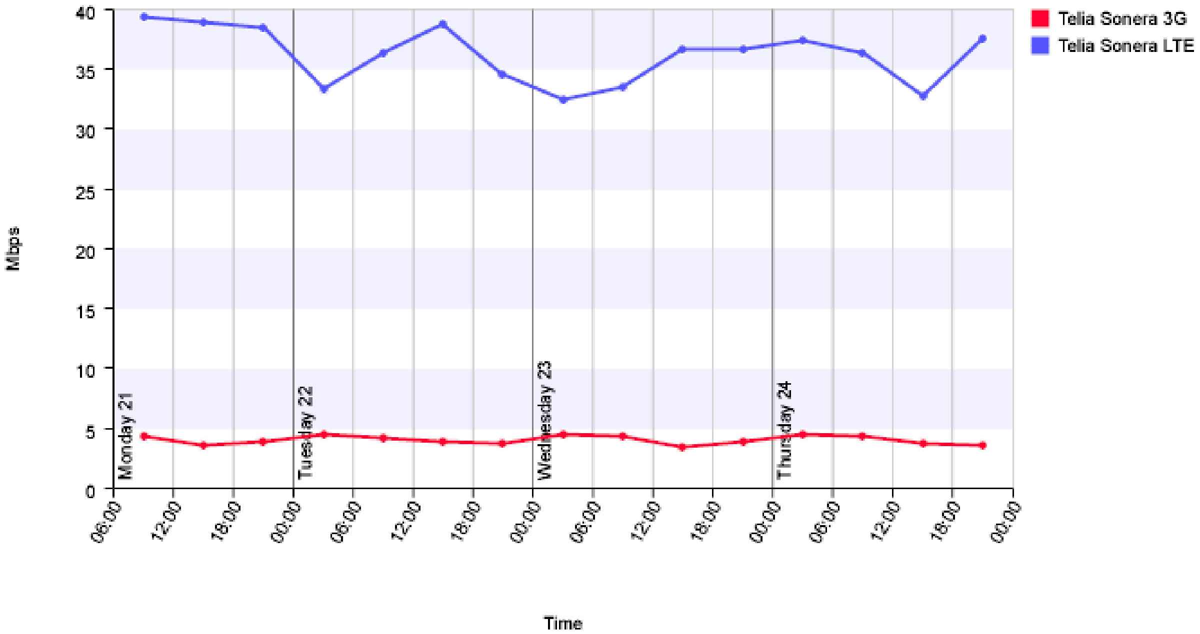


LTE Speed Test Results

Particular attention was given to the analysis of speed as TeliaSonera indicated that subscribers could expect download speeds between 20Mbps and 80Mbps.

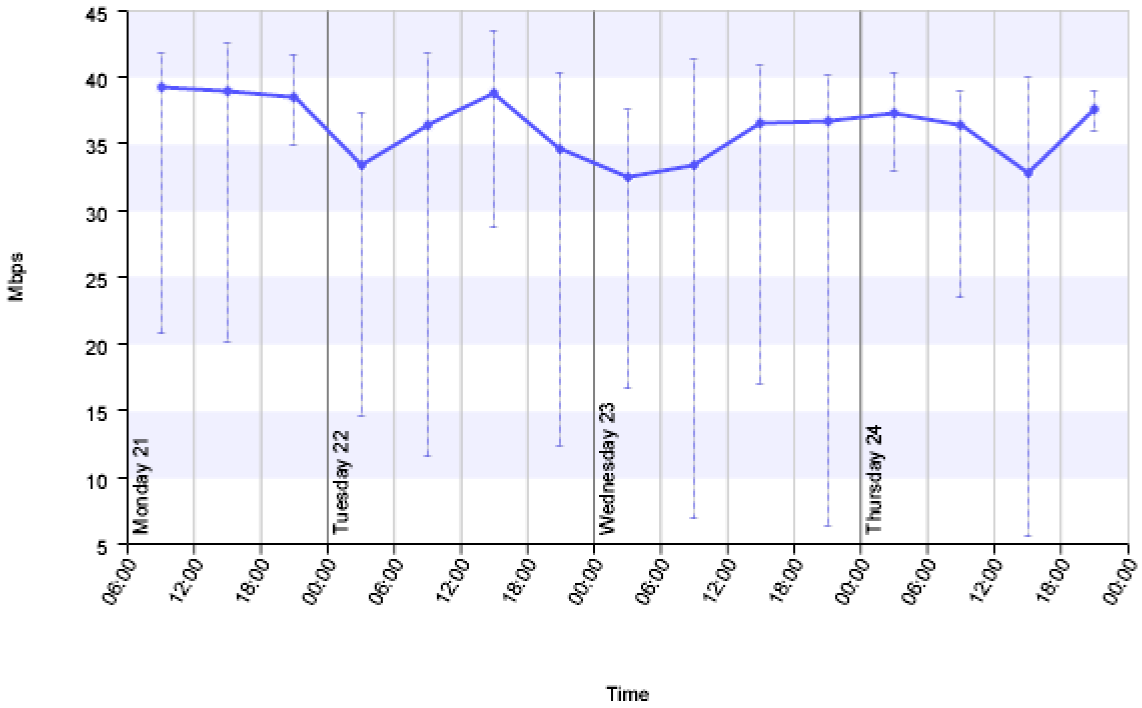
The 3G download speed measured was an average of 4.1 Mbps as shown in the Fig 1. LTE download speed was an average of 36.1 Mbps – almost 10 times faster than the average 3G download speed.

Fig 1: 4G and 3G Average Download Speeds



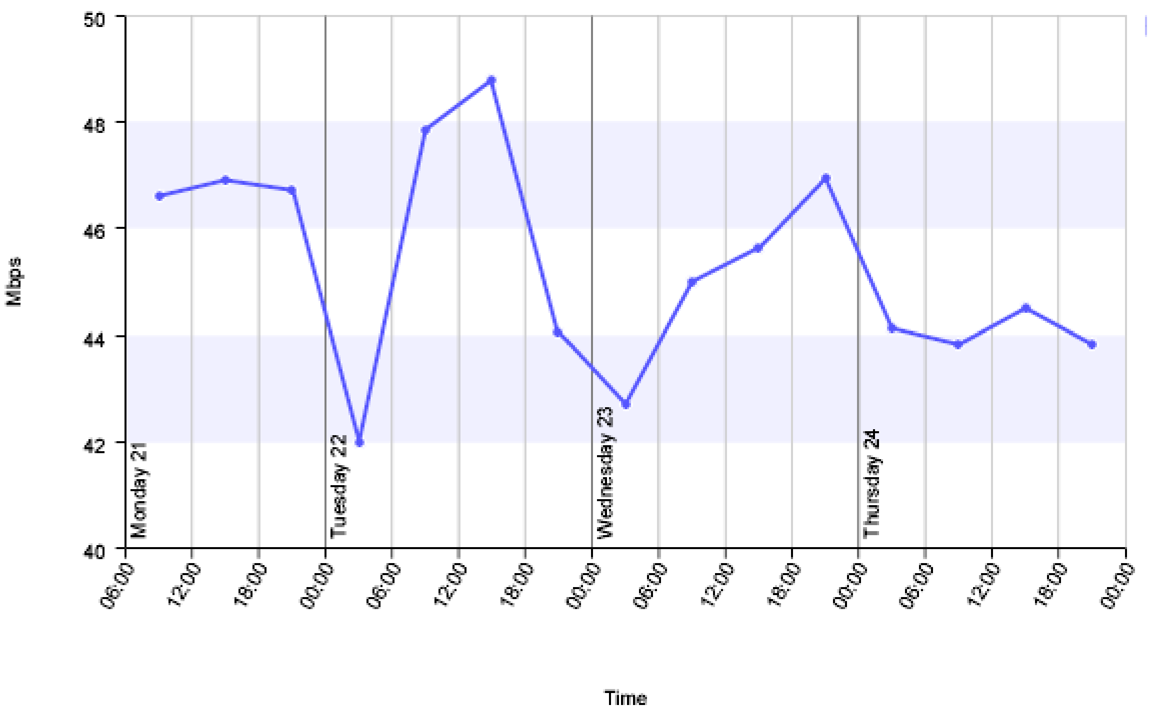
A more detailed look at both services investigated the consistency of speeds. Consistency is necessary for real-time communications such as VoIP or video. Fig 2 shows the average speed along with the peak and minimum measurements for each time period. Speed measurements over TeliaSonera's 4G network varied with measurements dropping as low as 5.6Mbps. However, the overall average 4G speeds were 30Mbps or higher throughout the entire test period.

Fig 2: 4G Average, Min and Max Download Speeds



Absolute peak speeds across the period tested are shown in Fig 3 with peaks per period ranging from 42Mbps to 49Mbps.

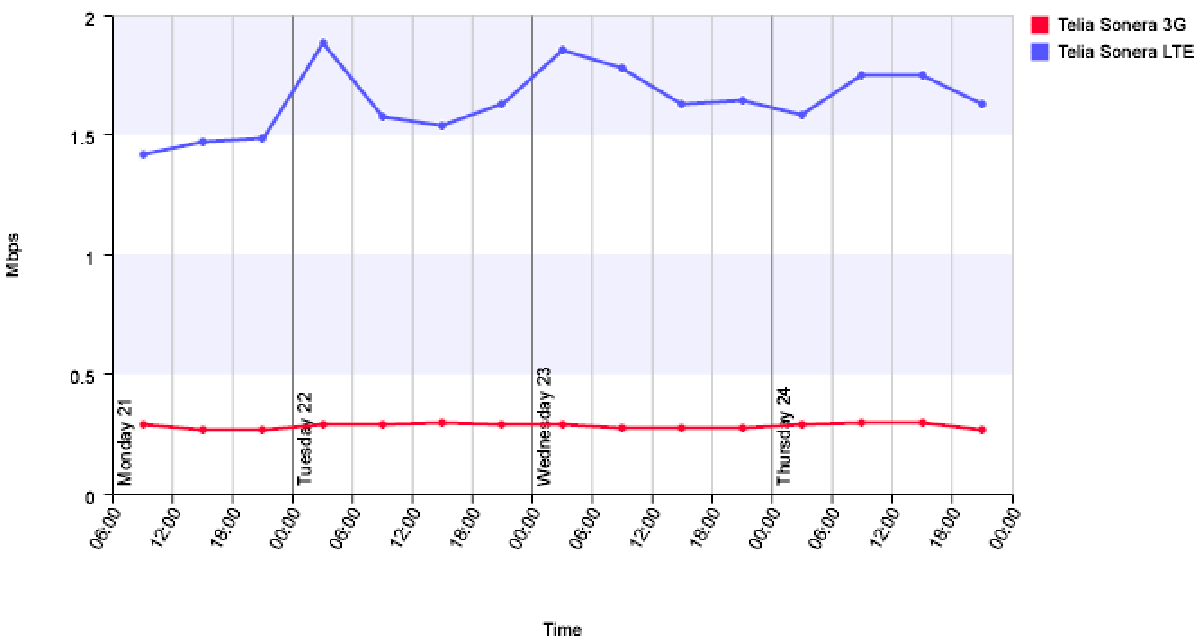
Fig 3: 4G Peak Download Speeds



Upload speed is often overlooked in broadband communications as traditionally the bandwidth requirement has been limited, needing only to accommodate small-sized IP signalling instructions or the sending of emails and attachments. Fig 4 shows the 3G speed averages 0.29Mbps which is adequate for performing these tasks.

However, applications such as VoIP and video-conferencing require significantly more speed. LTE upload speeds averaged 1.7Mbps which meets Skype's recommendations for Basic video calling (0.3Mbps), HD video calling (1.5Mbps) and Group video calling (0.5Mbps).

Fig 4: 4G and 3G Average Upload Speeds



Summary

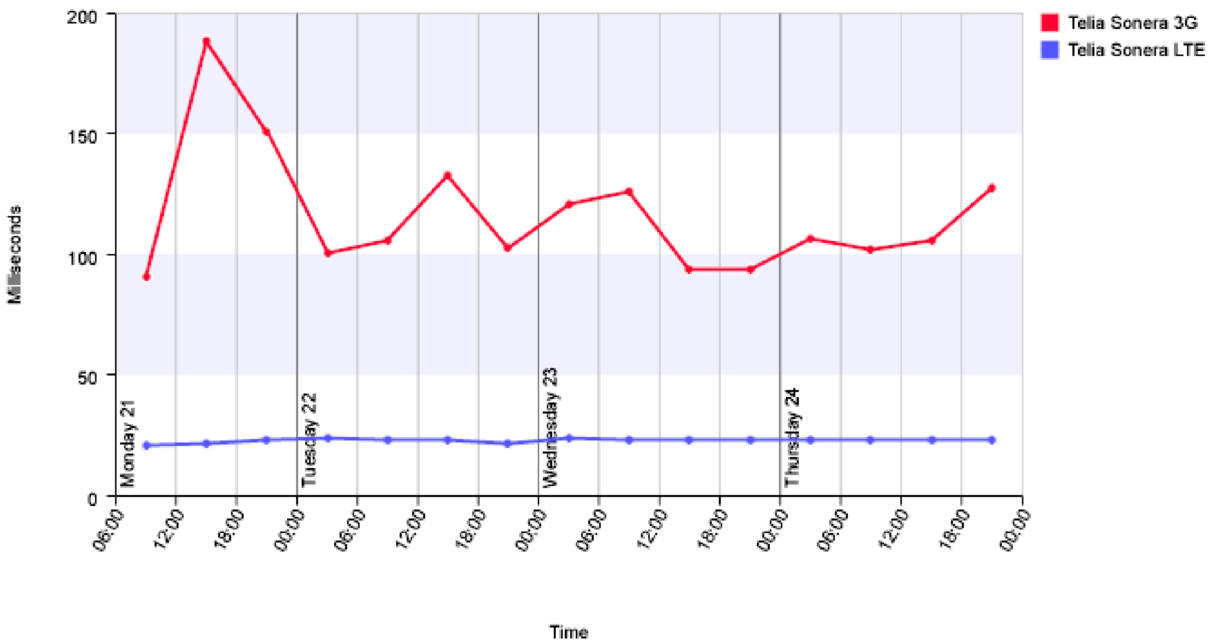
With average download speeds of 36.1Mbps and peak speeds of 48.8 Mbps the performance of LTE is confirmed to be much faster than 3G. These speeds also meet TeliaSonera's forecast of providing services between 20 – 80Mbps on average, though a few times speeds measured were below the 20Mbps minimum. Still, it's clear that the capability of LTE to provide high speeds in a commercial consumer deployment is real.

Network Latency Test Results

Network Latency is the time it takes for a network to respond, and is the chief issue that prevents 3G mobile networks from providing reliable Quality of Experience for real-time applications such as VoIP telephony and on-line game play. Streaming video can also be affected where latency times exceed the capability of jitter buffers. Higher delay times, combined with other network degradations, conspire to render experience unsatisfactory.

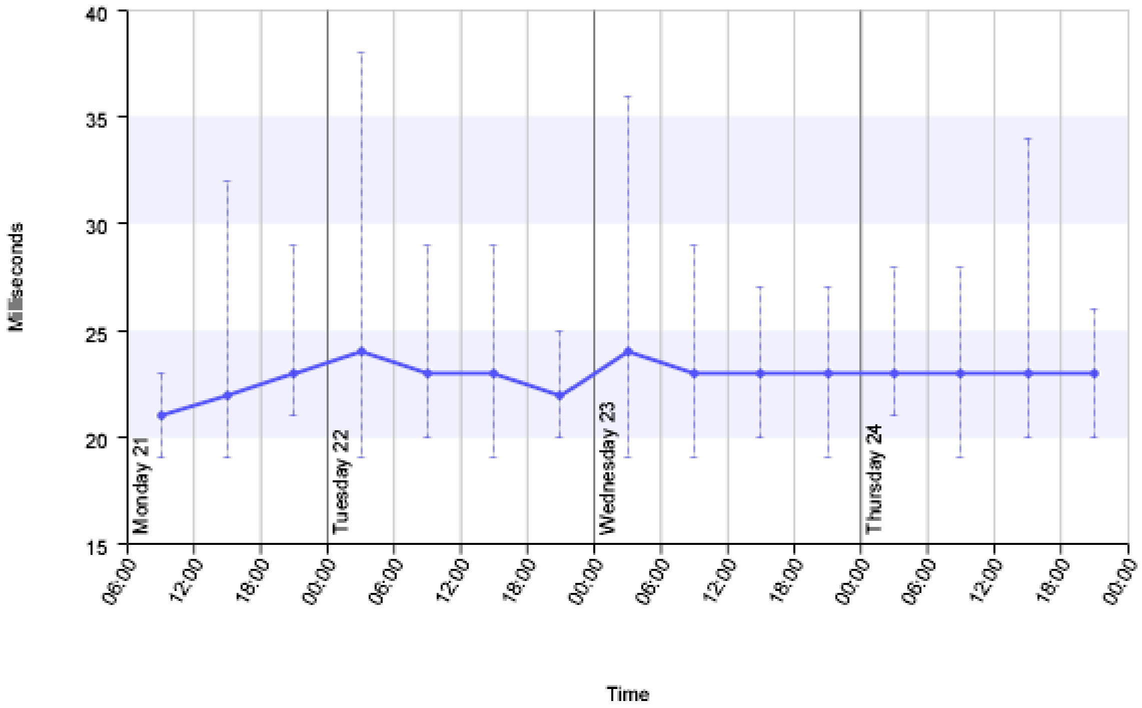
Fixed line latency measurements are typically about 50ms in leading countries. Latency for 3G networks is typically 2 to 3 times slower, thus affecting real-time communications. As shown in Fig 5, 4G latency times were steady at 23ms, while 3G latency varied between 98ms and up to 189ms.

Fig 5: 3G and 4G Average Network Latency Time



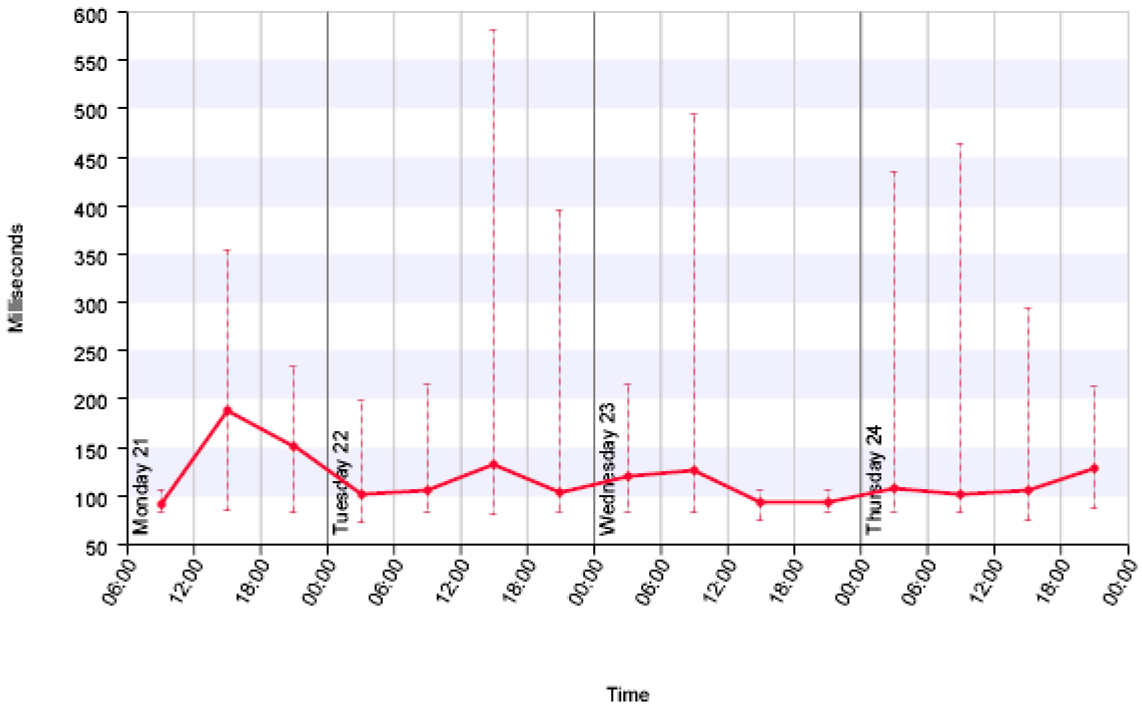
Further examination shows that 4G latency performance was very stable with a maximum recorded time of 38msec, well within the requirements for responsive network applications as shown in Fig 6.

Fig 6: 4G Peak, Average and Minimum Network Latency Time



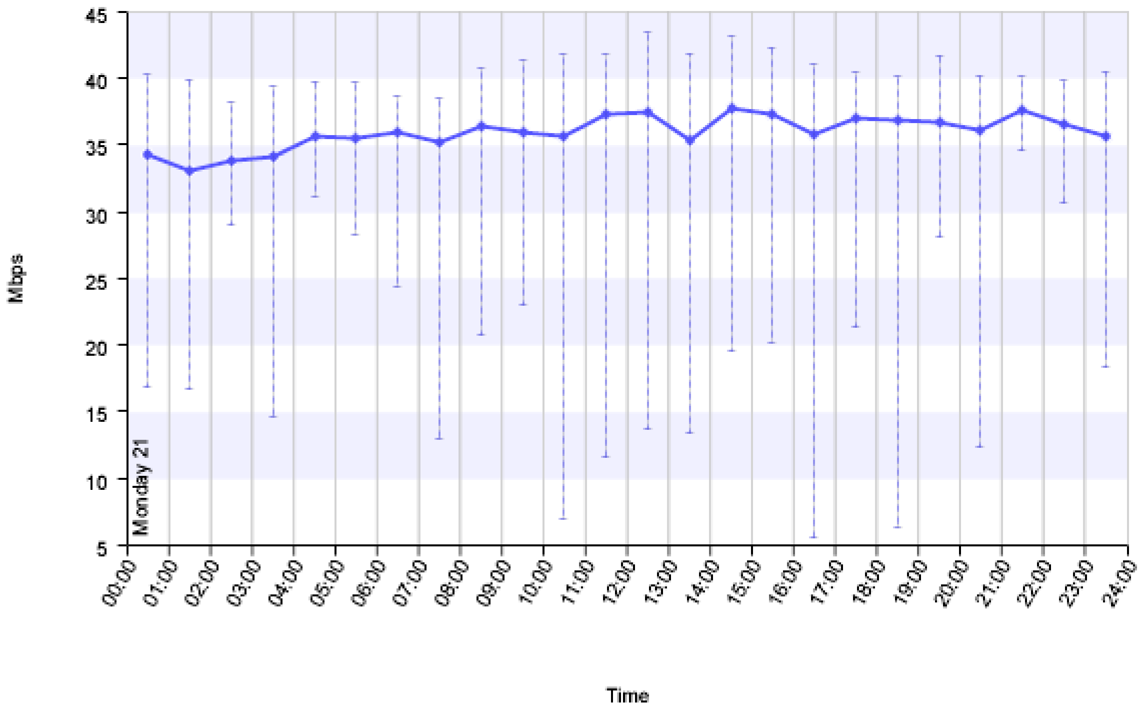
As seen in Fig 7 latency times for 3G were often erratic, showing extremes into the hundreds of milliseconds. This would provide a poor experience for on-line gaming and VoIP telephony.

Fig 7: 3G Peak, Average and Minimum Network Latency Time



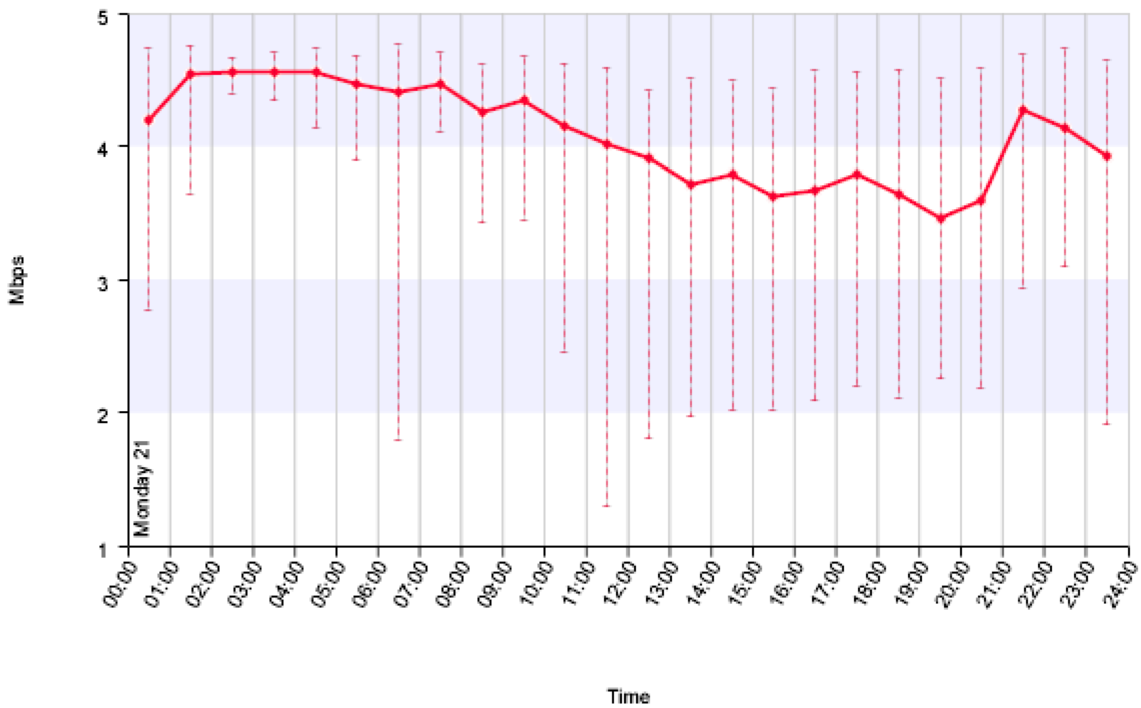
Speeds for LTE were also analysed on a 24 hr basis to understand behaviour by time of day. In Fig 8 there is no evidence of LTE speeds dropping during peak periods of the day.

Fig 8: 4G Peak, Average and Minimum Network Latency Time by Hour of Day



Conversely, 3G speeds varied by time of day as seen in the Fig 9. Speeds declined at about 08:00 a.m. until about 9 p.m. (21:00) which could indicate typical contention for resources.

Fig 9: 3G Peak, Average and Minimum Network Latency Time by Hour of Day



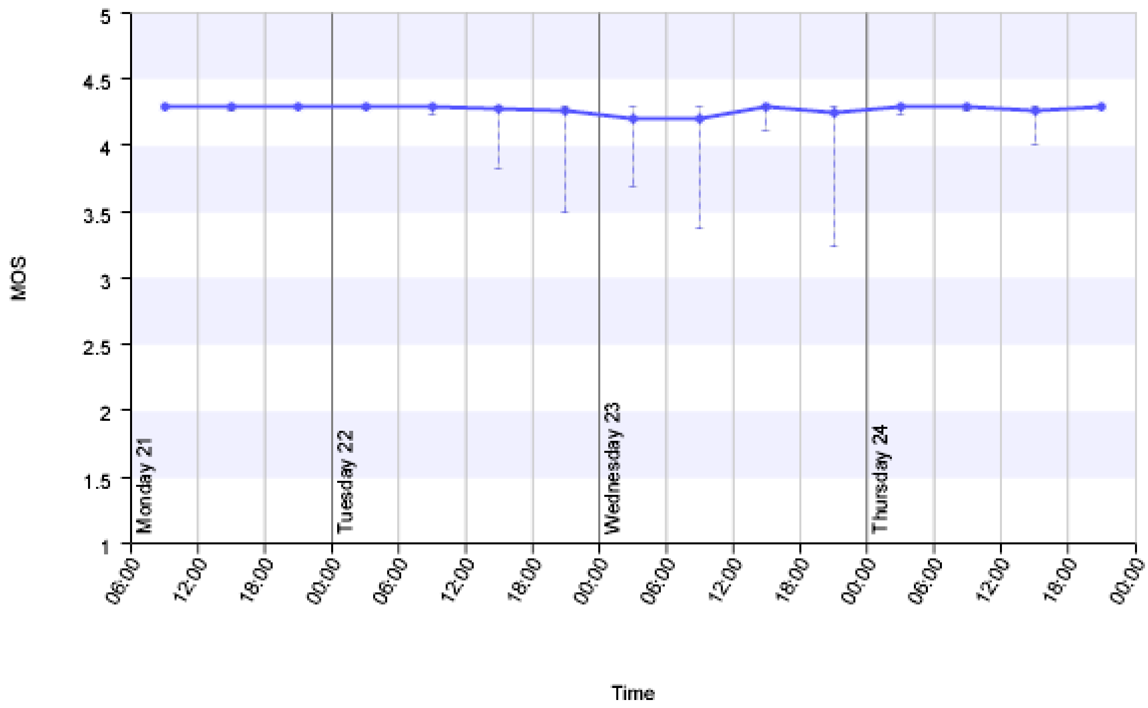
Voice over LTE (VoLTE) Quality Test Results

In November 2000, SIP was accepted as a 3GPP signalling protocol and permanent element of the IP Multimedia Subsystem (IMS) architecture for IP-based streaming multimedia services in cellular systems. SIP is the control protocol for IMS networks and given the industry-supported VoLTE initiative, LTE smartphones and tablets will have an integrated SIP client in order to make calls and communicate in new ways not previously considered. SIP was used to control all the Voice sessions contained in this report.

Voice over LTE (VoLTE) test results were calculated using the ITU-T P.862 PESQ model which transmits a reference voice signal in both directions and measures the degraded output voice quality received at the far end. The results are then scored and presented as a 'mean opinion score' or MOS which represents a subjective assessment of the received transmission. Voice quality is plotted on a scale of 1 to 5 where 1 would be perceived as 'bad' quality and 'excellent' would be a score of 4 or higher.

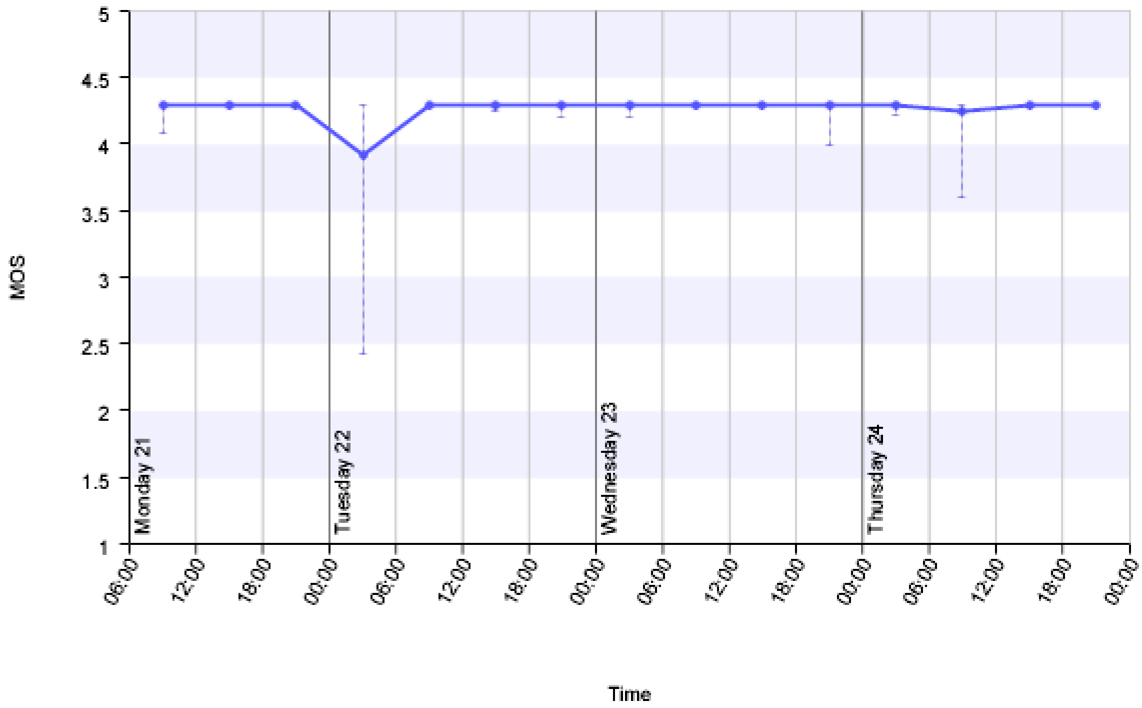
The downstream results for LTE are shown in Fig 10. As seen by these results the voice quality of these calls was 'excellent' on average, with only a few calls measuring lower. Even so, these calls measured 3.2 or higher which would be subjectively viewed as 'good'.

Fig 10: 4G Peak, Average and Minimum Voice MOS (International calls to London) - Downstream



Upstream results in Fig 11 were very similar though one test measured a low MOS score of 2.3. In practice, a MOS score this low would likely result in the parties hanging up and re-attempting the call.

Fig 11: 4G Peak, Average and Minimum MOS (International Calls to London) – Upstream



VoIP or Voice over LTE (VoLTE) sessions comprise two types of network traffic – the signalling messages required to setup and manage calls between users, and the encoded voice conversations. The call setup and management protocols, in this case using Session Initiation Protocols (SIP), use minimal bandwidth and do not have stringent latency requirements. The real challenge is to satisfy the bandwidth demands of the encoded conversation between callers.

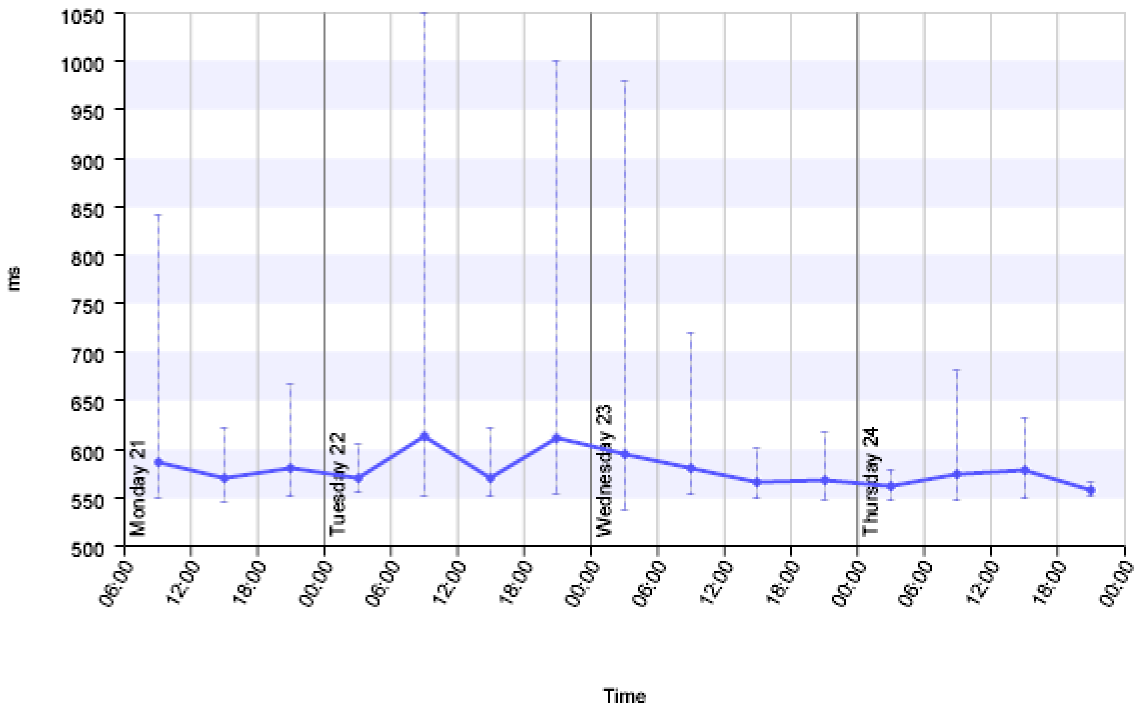
Each side of the call requires a continuous amount of bandwidth for the duration of the call. As an example, the G.711 CODEC uses the same encoding as used on PSTN/POTS - at a bit rate of 64 kbps and is generally limited to 160 bytes (20ms of voice) since larger payloads would increase the encoding latency and cause perceptible audio delay.

Call Set-up Times

Call Setup Time is the time required from the initial dialling of digits to actually establishment of a voice connection. Today, Subscribers are generally accustomed to fast call setup times and will expect to get similar or better performance in any new network.

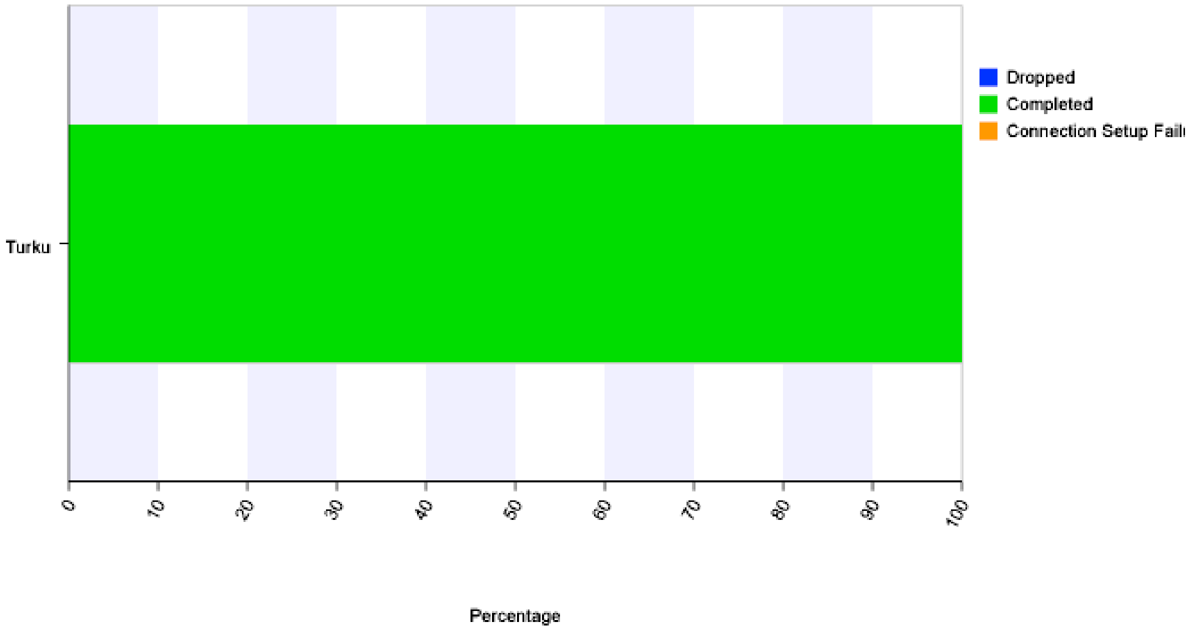
The Call Setup Times recorded using the 4G LTE services were approximately 0.6 seconds which is extremely fast, especially considering the call was originating in Finland and terminating in the UK. If such call international set-up times can be maintained then the outlook for subscribers is very good.

Fig 12: Call Set-up times



The Call Success Ratio, defined as the ratio of successful connects to dial attempts, is also of importance to the service providers. Fig 13 shows that all of the calls were successfully completed and not one of the calls made during the project lifecycle was dropped.

Fig 13: Call Success Ratio



Voice Latency

Voice Latency is the time delay incurred in speech by the telephony system from the moment that the speaker utters a word until the listener actually hears the word. This is known as "mouth-to-ear" latency or the "one-way" latency. Excessive voice latency can cause uncomfortable delays and lead to 'talker overlap' in two-way phone conversations.

Voice latency comprises in-country network latency (measured at 23ms via LTE in Finland), latency through media gateways and other infrastructure en route and jitter buffer processing time. The three types of latency or delay inherent in an IP-based telephony networks are:

- Propagation delay
- Serialization delay
- Handling delay

Propagation delay is caused by the length a signal must travel via light in fibre or electrical impulse in copper-based networks. Handling delay—also called processing delay—defines many different causes of delay (voice encoding, compression and switching) and is caused by devices that forward the frame through the network. Handling delays can impact traditional phone networks, but these delays are a larger issue in IP-based environments.

A packet-based network experiences delay for other reasons. Two of these are the time necessary to move the actual packet to the output queue and queuing delay. When packets are held in a queue (typically because of congestion), the result is queuing delay.

Callers notice the effects of round-trip delay when it exceeds 250ms, and thus the one-way latency budget for voice calls should be in the region of 150ms. The voice latency measured via TeliaSonera's 4G LTE SIP calls to and from the UK was approximately 165ms on average, with peak measurements near 200ms. Certainly these are more than acceptable measurements given that the calls were international.

Fig 14: Downstream Voice Latency

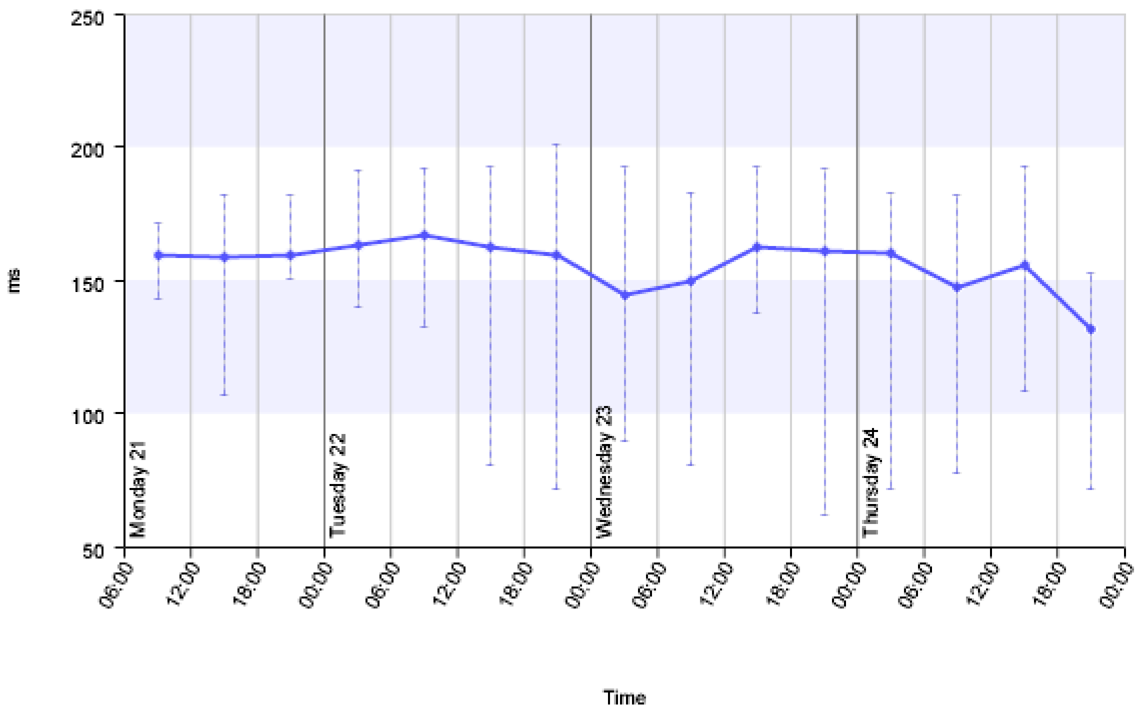
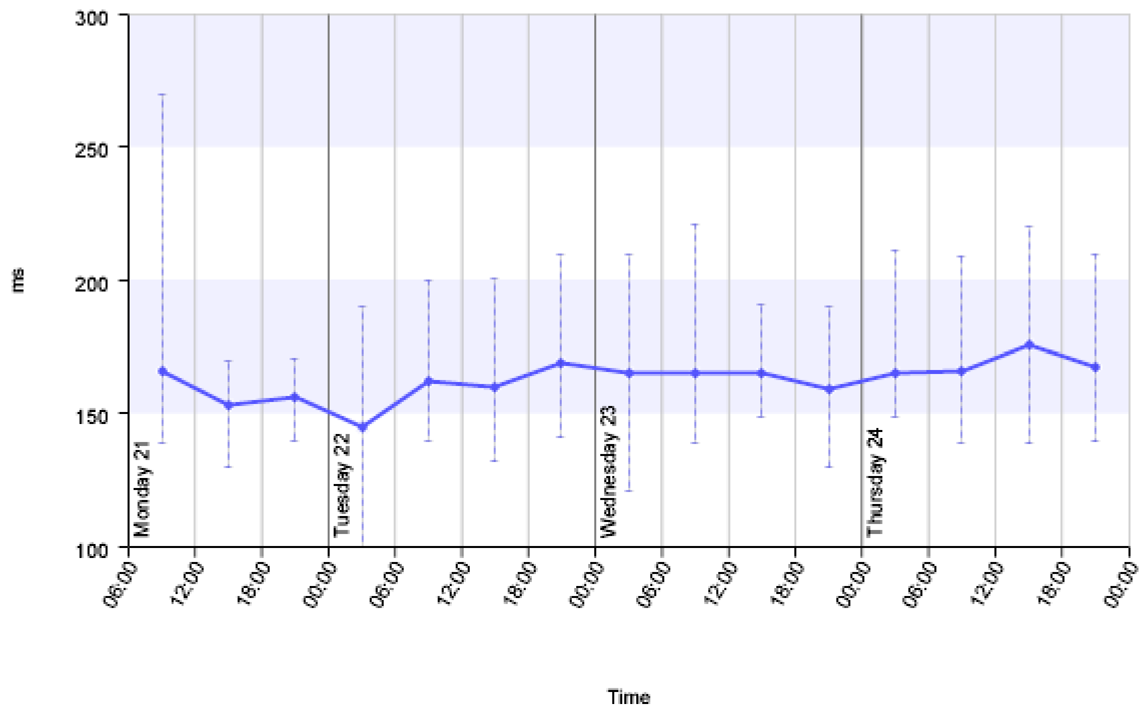


Fig 14: Upstream Voice Latency



Packet Loss

Packet Loss can be caused by many different reasons; overloaded links, excessive collisions, physical media errors and others. Voice CODECs take into account the possibility of packet loss, especially since RTP data is transferred over the unreliable UDP layer.

Packet loss starts to be subscriber-affecting when the percentage of the lost packets exceeds a certain threshold (in the region of 4% of the packets), or when the losses are grouped together in bursts – which tends to be the case by their nature. In those situations, even the best CODECs will be unable to hide the effects from the user - resulting in degraded voice quality. For these reasons, it is important to know both the percentage of lost packets, burst behaviour and whether these losses are specifically on the Uplink or Downlink side of the network service.

In Figures 16 and 17 the average and peak packet loss (burst) is well under the 5% threshold with the exception of a measurement on Tuesday March 22nd where upstream packet loss exceeded 9% - if this was a real conversation it would have resulted in garbled audio for the listener. For all other calls the packet loss was comfortably under 0.5%.

Fig 16: Downstream Packet Loss

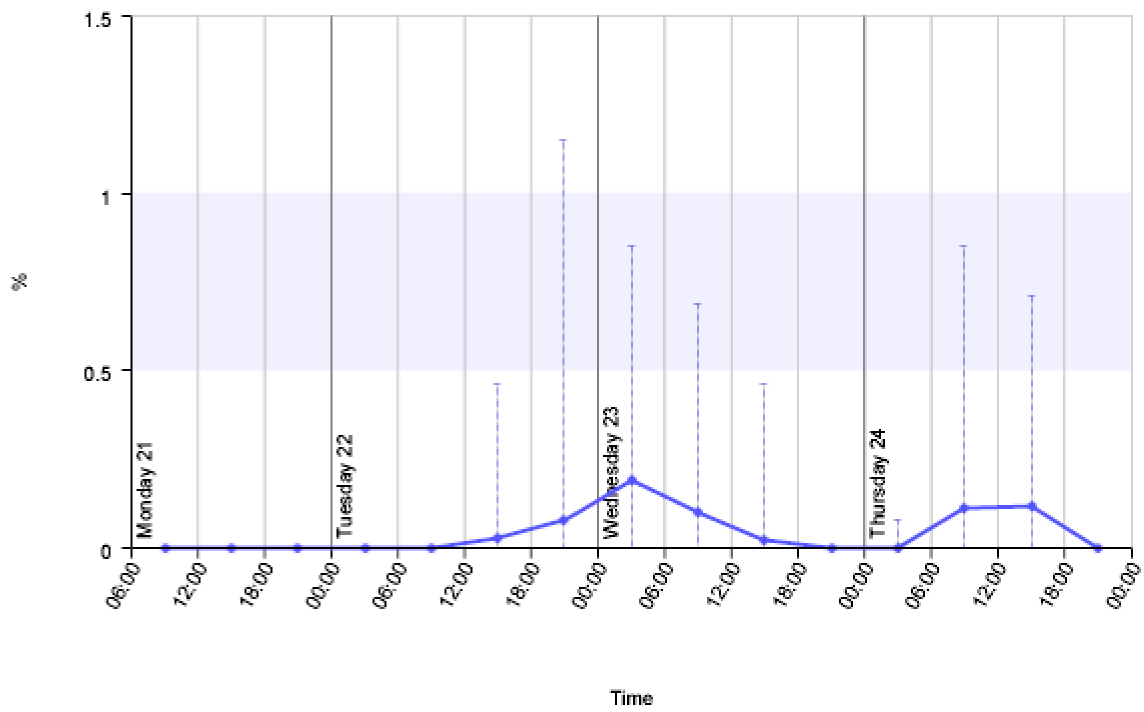
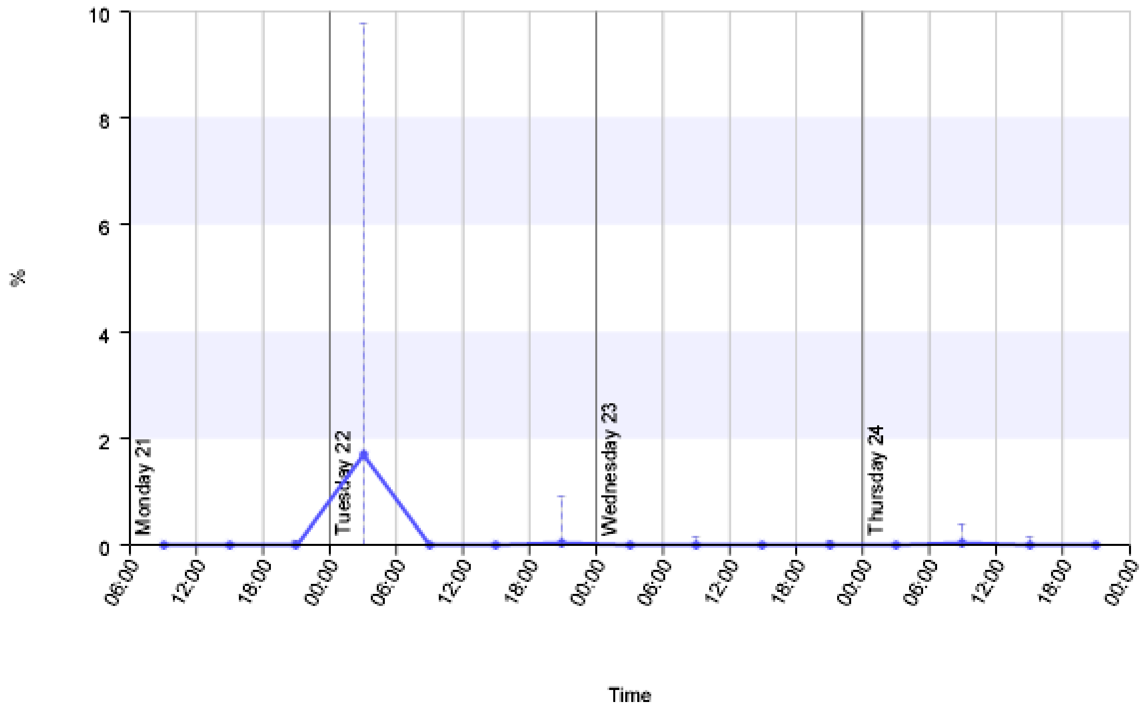
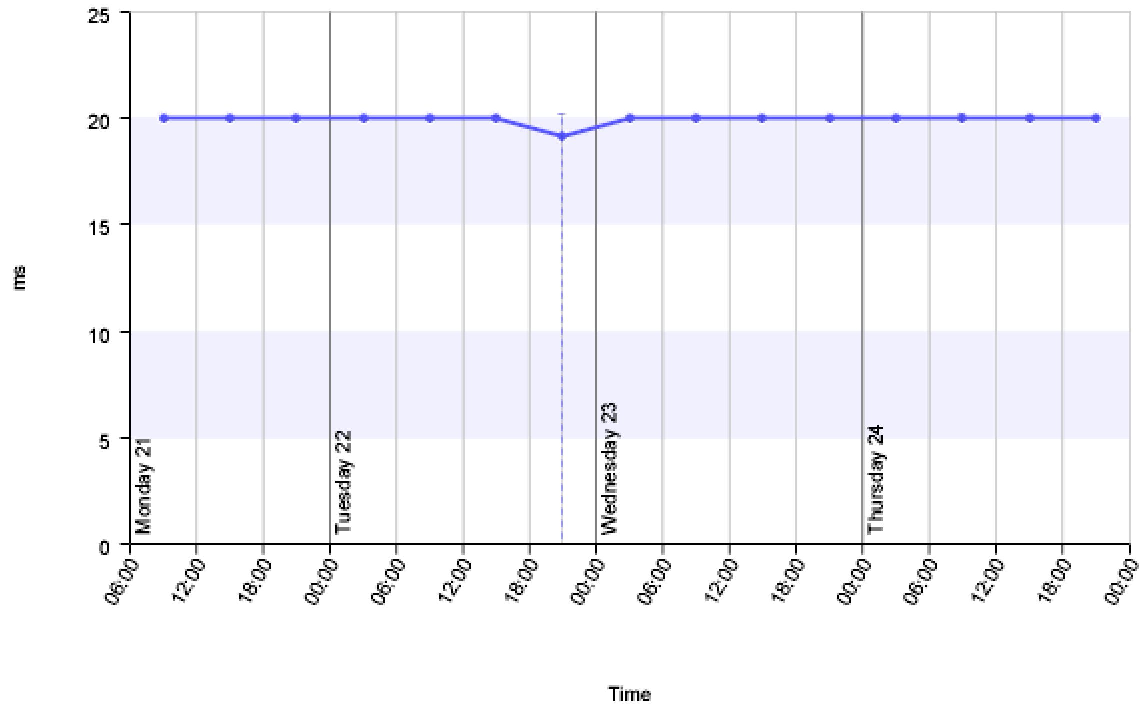


Fig 17: Upstream Packet Loss



VoIP Packet Delay Variation (PDV)

PDV is the difference in end-to-end delay between selected packets in a flow with any lost packets being ignored (RFC 3393). The effect is incorrectly referred to as jitter. The delay is specified from the start of the packet being transmitted at the source to the end of the packet being received at the destination. The term PDV is defined in ITU-T Recommendation Y.1540, *Internet protocol data communication service - IP packet transfer and availability performance parameters, section 6.2*.



Conclusions

TeliaSonera's LTE network in Turku Finland is delivering the promise of high-speed mobile communications with an average speed of 36.1Mbps, and a peak of 48.8Mbps.

High speed combined with the low latency observed shows that this rollout may be able to handle on-line game play, VoIP and HD video.

Voice quality was deemed 'excellent' for the most part with the exception of a couple of calls where the quality was merely 'good'. Only one single call had voice quality in the upstream direction that would be deemed 'poor' and likely result in the parties hanging up. However, this was just 1 call of the 334 calls made.

Of further interest to the industry will be an examination of how the service performs as subscriber numbers increase. Additional subscribers could cause congestion at the point of access or in the backhaul.

In-motion measurements will also be of interest to see how real-time applications perform with challenges unique to those situations, including cell handoff.

This initial testing indicates that LTE, as a commercially available service, is capable of providing subscribers with speeds of up to 10 times faster than 3G with latency that is equal to high quality fixed line services. Given the data Epitiro collected, LTE in Finland should be able to handle the most demanding of applications.

ipQ - Epitiro’s Mobile Performance Analysis Solution

ipQ™ is a scalable end user device-based broadband measurement solution that finally allows network operators to ‘see’ IP service quality as experienced by fixed and mobile customers. Using software probes that download to smartphones and PC’s, ipQ™ provides realtime insight into the performance of services from the Subscriber perspective.

For mobile networks, both radio (coverage) and IP performance is measured from Android-based smartphones or PC’s equipped with dongles. No other solution available today matches this ability to understand all aspects of mobile broadband.

Fixed network operators can readily see ‘past the CPE’ and understand how services are experienced in the home to PC’s that are wired or connected via WiFi. Used for network management or solving a single subscriber’s fault, ipQ™ puts quality of experience data at your fingertips.

ipQ™ easily scales to provide national coverage and information about service quality that is out of reach with conventional testing and measurement methods. Software test apps are downloaded over the web to potentially thousands of subscriber ‘test points’ to quickly create a substantial database of your network - and your competitors. Powerful geo-spatial maps, charts and graphs can be generated in the on-line dashboard or data can be exported.

About Genre Mobile

Genre Mobile Ltd provides insight, expertise, and tools for more profitable mobile data networks.

Genre Mobile Ltd helps Operators to understand mobile data behaviour to improve customer experience and optimize mobile network utilization.

With Genre Mobile Ltd tools, the operator can analyze usage, segment customers and simulate offering changes. This insight helps to select optimal offering characteristics, and to implement the changes automatically to network.

http://www.genremobile.com

About Epitiro

Epitiro is recognised as the global leader in fixed and mobile broadband benchmarking, providing subscriber experience insight to ISPs, MNOs, media providers, multinational corporations and government regulators.

Clients such as Ofcom, Vodafone, Orange, Virgin Media, Telecom New Zealand, Telefonica O2, Tiscali, Singapore IDA, Saudi Telecom Company, Bahrain Telecom Regulatory Authority and many others benefit from Epitiro's coverage of fixed and wireless broadband performance.

Founded in 2001, Epitiro is headquartered in the UK.

http://www.epitiro.com

Annex A - Technical Perspective

With the emergence of LTE, mobile network operators are beginning to deploy VoIP-based services and there are challenges to ensure acceptable voice quality. Given that a large portion of mobile operator revenue is from voice services, call quality in LTE is of considerable importance.

The combined effect of end-to-end delay and packet loss can reduce customer satisfaction and calls will be terminated early triggering a loss in revenue. More significantly, customers who have significant on-going poor experiences with call quality are likely to churn – dropping a service in favour of a competitor's offering.

Sources of Delay and Packet Loss

In wired broadband networks, packet loss typically comes from congestion. Routers along a call path will drop packets when their buffers become full or when packets arrive too fast to be dealt with effectively. With giga-rate (Gb) wired connections, this hardly happens these days – so much so that low delay and packet loss characterise wired networks

Communication on the IP network is inherently less reliable in contrast to the circuit-switched public telephone network, as it does not provide a network-based mechanism to ensure that data packets are not lost, or delivered in sequential order. It is a best-effort network without fundamental Quality of Service (QoS) guarantees. Therefore, VoIP implementations may face problems mitigating latency and jitter.

By default, IP routers handle traffic on a first-come, first-served basis. Routers on high volume traffic links may introduce latency that exceeds permissible thresholds for VoIP. Fixed delays cannot be controlled, as they are caused by the physical distance the packets travel; however, latency can be minimized by marking voice packets as being delay-sensitive with methods such as Differentiated Services (DiffServ), MPLS and MPLS-TP.

Traditional Wireless networks

In wireless networks the challenges are different. Here, the radio path between the base station (eNodeB in LTE networks) and the UE is the primary cause of delay and packet loss. If wireless networks are to compete with wired networks, then the radio link must be made as robust as possible.

LTE Networks

LTE is an all-IP (Internet Protocol) mobile broadband service wherein the transmission of real-time data such as voice and video is carried entirely over packet-switched connections. In LTE, the mechanism for providing end-to-end Quality of Service (QoS) is based on two parameters. First, for each User Equipment (UE) a Layer 2 Packet Delay Budget (L2PDB) is defined for each session. This ensures that delay sensitive packets arrive at their destination on time.

Annex B – Voice / VoLTE Testing Parameters

The SIP protocol is an Application Layer protocol designed to be independent of the underlying transport layer; it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP). It is a text-based protocol, incorporating many elements of the HTT/Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP).

Testing for voice quality was executed using both G.729 and G.711 codecs. SIP was used to control all the voice sessions in this study.

Jitter Buffer Management Considerations

The minimum performance requirements for jitter buffer management of voice media, as described in 3GPP TS 26.114. were met. Voice packet flows can typically tolerate delays on the order of 100 ms and packet losses of up to 1 % (percent), since state-of-the-art AMR voice codecs perform well up to these error rates.

Data Transport

The Epitiro LTE test devices use RTP over UDP as described in IETF RFC 3550 and IETF RFC 768, respectively, to transport voice. The devices both used the standard port numbers for sending and receiving RTP packets. This facilitates interworking with fixed/broadband and PSTN/Mobile networks.

Bearer considerations

The dedicated bearer for IMS SIP signalling was not utilised. Instead, the default bearer was used when UE creates a connection to the PDN which is used for standard data transmission, as defined in 3GPP specifications. On that basis, it is important to note that the results obtained during this project are likely to be lower than calls with a dedicated bearer for IMS-based VoLTE services.

Test Configurations

Metric	Test Endpoints	Parameters
Download and Upload Speed	Endpoint - London Endpoint - Amsterdam	Protocol: TCP Test Threads Down: 2 Test Threads Up: 1
HTTP	http://www.iltalehti.fi/etusivu/ http://www.iltasanomat.fi/ http://yle.fi/http://www.hs.fi/ http://www.mtv3.fi/	Timeout: 60000ms Cached: Y Uncached: Y Images: Y
Ping	www.iltalehti.fi www.mtv3.fi Endpoint - London Endpoint - Amsterdam	N/A
DNS	www.iltalehti.fi www.iltasanomat.fi yle.fi www.hs.fi www.mtv3.fi	N/A
PESQ Analysis	Epitiro Listener: UK	Codec: G.711 a-law , G.729

Table of Results

Metric	3G	4G	Number of Tests
Avg Download Speed	4.1 Mbps	36.1 Mbps	334
Peak (best) DL Speed	5.2	48.8	334
Avg Upload Speed	0.3 Mbps	1.7 Mbps	334
Avg Latency	117 ms	23 ms	334
Peak (worst) Latency	500 ms+	38 ms	334
Downstream avg MOS	N/A	4.3	668
Downstream Peak (best) MOS	N/A	4.3	668
Downstream Worst MOS	N/A	3.2	668
Upstream avg MOS	N/A	4.3	668
Upstream Peak (best) MOS	N/A	4.3	668
Upstream Worst MOS	N/A	2.3	668
Call Setup Time avg	N/A	600 ms	668
Call Completion Ratio	N/A	100%	668
Voice Latency DS avg	N/A	165 ms	668
Voice Latency US avg	N/A	165 ms	668
Packet Loss DS avg	N/A	0.2%	668
Packet Loss US avg	N/A	0.1%	668

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20-103-1006.001