Paper

Quality Expectations of Mobile Subscribers

Przemysław Gilski and Jacek Stefański

Department of Radio Communication Systems and Networks, Gdańsk University of Technology, Gdańsk, Poland

Abstract—Mobile systems, by nature, have finite resources. Radio spectrum is limited, expensive and shared between many users and services. Mobile broadband networks must support multiple applications of voice, video and data on a single IP-based infrastructure. These converged services each have unique traffic holding and quality requirements. A positive user experience must be obtained through efficient partitioning of the available wireless network resources. The 3rd Generation Partnership Project (3GPP) has developed a comprehensive Quality of Service (QoS) parameter to address this problem. The regular control of service quality is critical for operators to ensure user Quality of Experience (QoE), establish new business models and monetize services. It enables operators to employ fair-use resource policies and maintain network performance during peak traffic times. Wireless mobile communication is tending towards an integrated system of Internet and telecommunication technologies, where mobile users move freely anytime and everywhere. They desire to communicate with any device using the best service available. In this paper QoS management issues in mobile communication are described. The authors present an insight into subscriber behavior and related factors that affect the QoE of mobile data services.

Keywords—cellular networks, mobile communication, quality of experience, quality of service, wireless communication.

1. Introduction

Due to the networks evolution from Circuit Switched (CS) to Packet Switched (PS) technologies, telecommunication services have experienced a huge increase in transmission capabilities, e.g., medium, bandwidth, throughput. It also helped new services to emerge, including Voice over IP (VoIP) telephony and multimedia streaming.

Currently, users require only an Internet Protocol (IP) access connection, either via a Wireless Local Area Network (WLAN) hotspot or a cellular connection. This IP communication trend requires an appropriate QoS, in order to fulfill the user expectations.

As the number of users in both telecommunication and Internet networks grows, it becomes clear that real-time services are becoming more difficult to implement due to erratic delay and packet loss.

Nowadays, mobile broadband networks carry multiple services that share radio access and core network resources. In addition to best-effort services, wireless networks must support delay-sensitive real-time services. Each service has different QoS requirements in terms of packet delay tolerance, acceptable packet loss rates and required minimum bit rates.

2. Quality of Service Background

QoS can be defined as a set of predefined technical specifications necessary to achieve the required service functionality. Each user specifies his requirements, so that the network can adjust its bandwidth, making use of different QoS schemes in order to satisfy the request. This can be an important factor when comparing services offered by different vendors or providers. When both price and feature are similar, quality becomes the key differentiator.

The degradation of QoS can be caused by a number of factors, including [1]:

- congestion (caused by traffic overflow – bottleneck effect),
- delays (caused by network equipment),
- distance or retransmission of lost packets,
- shared communication channels (collisions and large delays are common),
- limited bandwidth (poor capacity management).

2.1. QoS in Mobile Telecommunication

As mobile networks evolve to high-speed IP-based infrastructures, the wireless industry is ensuring high-quality services by developing QoS and policy-management techniques in addition to adding network capacity.

Mobile telecommunication is a type of communication used for transmitting voice or data over long distance. It consists of services such as: wireless telephony, satellite communication, WLAN and other 802.1x networks, IP-routed networks including the Internet, etc. However, the current global Internet is a best-effort service.

This service does not guarantee anything, even delivering a packet from one point to another within a single network. The destination node does not know the delivery speed or time. While delivering an e-mail message, delay is not a problem. But when it comes to real-time services like VoIP calls, if the delay becomes too large or too many packets are lost, the service quality becomes unacceptable [2], [3].

2.2. QoS Parameters

In order to keep track whether the contracted QoS are being met, the parameters must be monitored and resources should be reallocated in response to system anomalies. If a change of state happens and the resource management cannot make resource adjustments to compensate it, the application can either adapt to the new level of QoS or degrade to a reduced service level. The measurement of QoS is based on parameters including: delay, jitter, packet loss, throughput and many other, depending on the application and management scheme [4].

- Delay (latency) – a parameter related to communication. Since end points are most often distant, the transfer of information will consume time to reach the other side. Can be measured either one-way (from source to destination node) or round-trip (from source to destination and back to source node). The round-trip delay is used more frequently in the form of the ping command. It only sends a response back when it receives a packet without processing it. The final result is the minimum delay time possible for sending a packet from source to destination in the tested link.
- Jitter – a delay variation introduced by the transmission of multiple packets over the network. Can seriously affect the quality of audio-video streaming. In order to compensate it, all collected packets should be hold until the last (slowest) packet arrives on time and then rearranged to be played in the correct order. Jitter buffers are clearly visible when using audio-video streaming websites.
- Packet loss – occurs when one or more packets transported across the network fail to reach their destination. Some packets may fail to arrive when the buffer is already full. The loss of packets can be caused by other factors, e.g., signal degradation, high network load or defect in network elements. Wireless networks are more vulnerable to packet loss due to interference caused by other systems, multipath fading, multiple obstacles, etc.
- Packet error rate – the number of incorrectly received packets due to corrupted bits, often expressed as a percentage.
- Throughput – the amount of data that can be processed in a fixed time space, usually measured in bits per second. Throughput is a good way of measuring capacity of a communication link, regardless of connection type. However, it may not reflect the real user experience.
- Reliability – the availability of a connection, describes the ability of a system or component to function under stated conditions for a specified time period.

2.3. QoS Class Indicator

The Quality Class Indicator (QCI) specifies the treatment of IP packets received on a specific bearer. The *bearer* is a basic traffic separation element that enables differential treatment for traffic with different QoS requirements. It provides a logical transmission path between the User Equipment (UE) and Packet Data Network Gateway (PDN-GW). Packet forwarding of traffic traversing a bearer is handled by each functional node, e.g. eNodeB in Long Term Evolution (LTE).
The 3GPP has defined a series of standardized QCI types summarized in Table 1 [5].

Table 1
3GPP standardized QCI characteristics

QCI	Packet delay budget [ms]	Packet error loss rate	Exemplary service
1	100	10^{-2}	Conversational voice
2	150	10^{-3}	Conversational video (live streaming)
3	50	10^{-3}	Real-time gaming
4	300	10^{-6}	Non-conventional video (buffered streaming)
5	100	10^{-6}	IMS signaling
6	300	10^{-6}	Video (buffered streaming), TCP-based (e.g. www, e-mail, chat)
7	100	10^{-3}	Voice, video (live streaming), TCP-based (e.g. www, e-mail, chat)
8–9	300	10^{-6}	Video (buffered streaming), TCP-based (e.g. www, e-mail, chat)

where: QCI 1–4 are Guaranted Bit Rate (GBR) and QCI 5–9 Non-GBR resource type; IMS – IP Multimedia Subsystem, TCP – Transmission Control Protocol.

For first deployment, the majority of operators will likely start with three basic service classes: voice, control signaling and best-effort data, whereas in the future premium services such as high-quality video transmission.

3. Quality of Experience

Subscribers expect their mobile devices provide high-quality connectivity and performance at all time. Any interruption in data services is as critical as an interruption in voice. However, while voice services have a standardized measurement of quality called Mean Opinion Score (MOS), there is no equivalent for mobile data.
Mobile data services encompass a wide variety of content types and usage patterns, including e-mail, audio-video

MOST WIEDZY Downloaded from mostwiedzy.pl

streaming, application downloading or online gaming, all with different characteristics. Depending on the service being used, mobile subscribers have varying quality expectations for mobile data performance and usability.
When subscribers consume content, their QoE is not determined strictly by the speed achieved via wireless technologies. They make subjective assessments based on a combination of factors as: speed, smoothness, latency. Operators know, the better the experience, the longer and more frequently subscribers will consume content.

3.1. Defining Subscriber QoE

Mobile operators do not have unlimited technical resources and capital. The radio spectrum is finite and even if operators increase capacity, bandwidth-hungry applications such as Peer-to-Peer (P2P) services and video streaming will eventually consume any excess capacity. Table 2 demonstrates how subscriber QoE expectation varies by service type [6].

Table 2
Comparison of QoE expectations and performance requirements by service type

Service	QoE expectation	Performance attributes
Internet	Low (best-effort)	• Variable bandwidth consumption, • Latency and loss tolerant
Business services	High (critical data)	• High bandwidth consumption, • Highly sensitive to latency, • High security
P2P	Low (best-effort)	• Very high bandwidth consumption, • Latency and loss tolerant
Voice	High (low latency and jitter)	• Low bandwidth (21–320 kb/s per call), • One-way latency (< 150 ms), • One-way jitter (< 30 ms)
Video	High (low jitter and packet loss)	• Very high bandwidth consumption, • Very sensitive to packet loss
Interactive gaming	High (low packet loss)	• Variable bandwidth consumption, • One-way latency (< 150 ms), • One way jitter (< 30 ms)

There is a significant distinction between real-time services such as video conversation or voice and best-effort services like Internet browsing. Real-time services must reserve a minimum amount of guaranteed bandwidth and are more sensitive to packet loss and latency. Subscriber QoE is based on a number of factors such as:

– mobile application responsiveness,

– time required to download a Web page,

– stalling in a video,

– video content resolution.

Figure 1 describes top Android and iOS applications [7], [8].

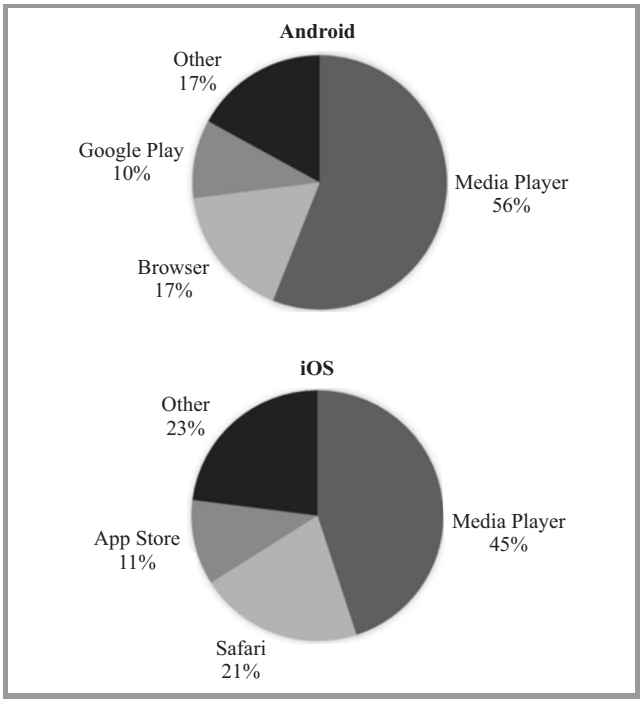

Fig. 1. Top Android and iOS applications.

According to presented data, about a half of mobile data is associated with Media Player software. The dedicated application stores and browsers account more than a quarter, whereas other application including Facebook, eBay and Instagram generate about 20% of mobile data volume.
In case of mobile Web page downloads across multiple mobile operators worldwide results show that over 50% of Web pages take more than 8 s to load and that 20% of Web pages take 20 s or more (Fig. 2) [9].
Depending on network conditions and the time of a day, mobile videos stall between 5–35% of the time. In some cases, stalling can lead some subscribers to abandon their sessions, causing frustration and loss of interest. It is visible, that conventional traffic management solutions do not work well in this case. Video is based on a variable bit rate. Its peak rates can exceed the shaped bandwidth of traditional traffic management solutions, leading to clips, stalling and eventually a poor experience. Figure 3 shows the change in global share of mobile video volume by format between 2010 and 2013 [8].
As shown, in 2010 90% of mobile video data was associated with the FLV format. Currently, the most popular video format is MP4, closely associated with smartphones, representing 67% of the global mobile video volume.

MOST WIEDZY Downloaded from mostwiedzy.pl

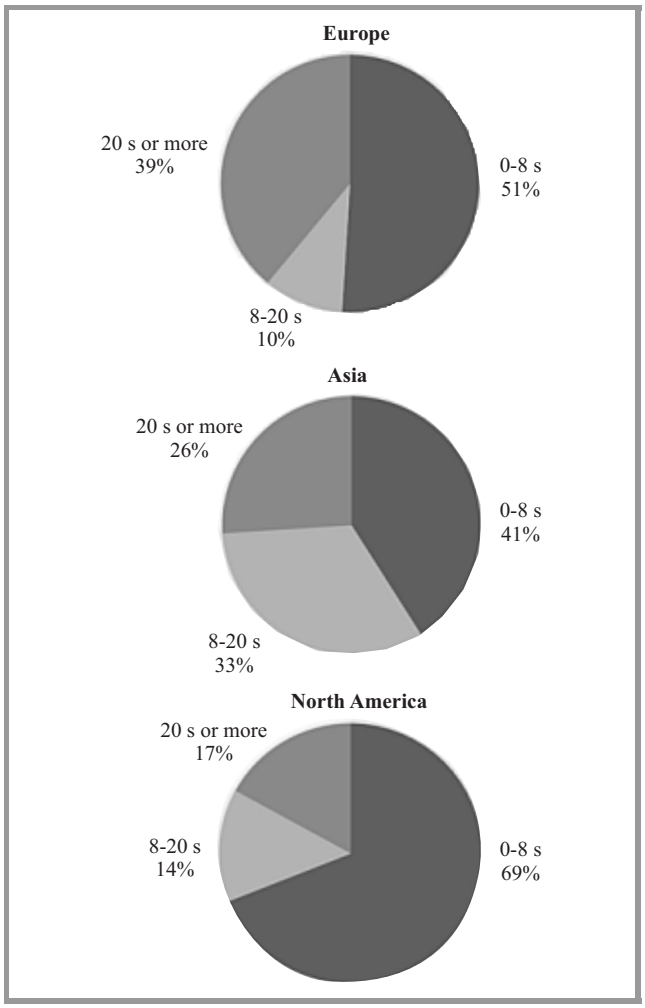

Fig. 2. Web page download times.

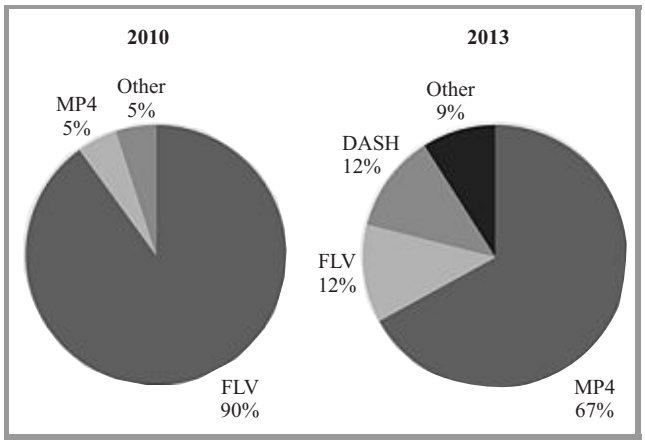

Fig. 3. Global share of mobile video volume by format.

Higher-resolution videos drive a disproportionate percentage of overall wireless network traffic, as shown in Fig. 4 [9].

Wireless networks that support this kind of videos deliver a better visual QoE to their subscribers. However, these multimedia must be effectively optimized to ensure that the overall subscriber QoE is not negative, considering screen size, resolution or connection speed.

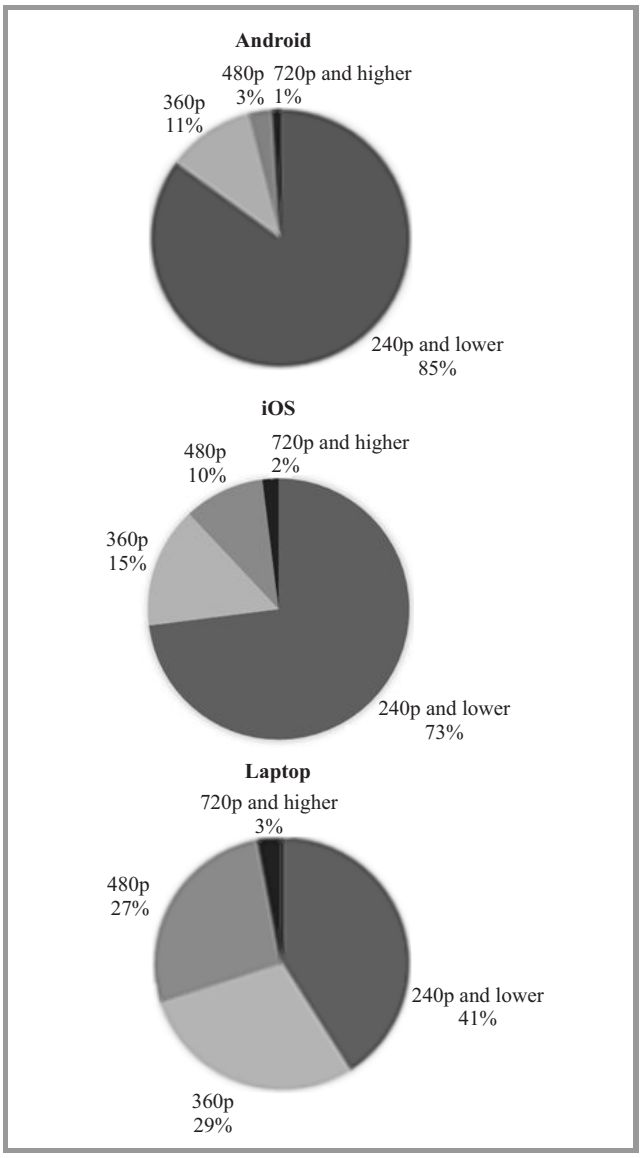

Fig. 4. Video request by resolution.

4. Summary

The widespread and availability of mobile smart devices will fuel the rapid growth in subscribers and sheer data volume. Operators world wide are racing to add new services and more powerful devices. They are making substantial investments to upgrade their networks capacity and performance.

If data continues to grow, operators will be forced to smarter manage the traffic. The economic realities and physical limitations of available spectrum prevent operators from simply adding more and more network capacity. Operators must plan today for future evolution of the network, which means working with vendors that have a solid roadmap for QoS and policy mechanisms in their products.

MOST WIEDZY Downloaded from mostwiedzy.pl

Quality plays a major role in wireless networks. Further traffic management and optimization technologies could allow network operators as well as service providers and vendors to improve subscriber QoS and QoE. Network efficiency could be optimized through application detection combined with adaptive traffic management in order to dynamically adjust to network conditions in real-time. As a result, it could help to boost mobile data usage, attract new customers, and raise satisfaction.

References

[1] M. Koivisto and A. Urbaczewski, "The relationship between quality of service perceived and delivered in mobile internet communication", *Inform. Syst. and e-Business Manag.*, vol. 2, pp. 309–323, 2004.

[2] J. A. Garcia-Macias, F. Rousseau, G. Berger-Sabbatel, L. Toumi, and A. Duda, "Quality of Service and mobility for the wireless internet", *Wirel. Netw.*, vol. 9, pp. 341–352, 2003.

[3] "Quality of Service (QoS) Concept and Architecture", ETSI TS 123 107 V11.0.0, 2012.

[4] F. Farid, S. Shahrestani, and C. Raun, "Quality of Service concerns in wireless and cellular networks", *Commun. of the IBIMA*, vol. 2013, pp. 1–5, 2013.

[5] 3GPP TS 23.203 V8.9.0, "Policy and Charging Control Architecture", 2010.

[6] "Quality of Service (QoS) and Policy Management in Mobile Data Networks", Ixia, Dec. 2013.

[7] "Mobile Analytics Report", Citrix, June 2013.

[8] "Mobile Analytics Report", Citrix, Nov. 2013.

[9] "Quality of Experience for Mobile Data Networks", Citrix, 2013.

Przemysław Gilski received his B.Sc. and M.Sc. degrees in Telecommunications Engineering from Gdańsk University of Technology (GUT), Poland, in 2012 and 2013 respectively. Currently he is a Ph.D. student at the Department of Radio Communication Systems and Networks (DRCSN), GUT. His research and development interests include digital video and audio broadcasting systems, software-defined radio technology, location services and radio navigation systems, as well as quality measurements in mobile networks.

E-mail: pgilski@eti.pg.gda.pl
Faculty of Electronics, Telecommunications
and Informatics
Department of Radio Communication Systems
and Networks
Gdańsk University of Technology
Gabriela Narutowicza st11/12
80-233 Gdańsk, Poland

Jacek Stefański received his M.Sc., Ph.D. and D.Sc. degrees in Telecommunications Engineering from Gdańsk University of Technology (GUT), Poland, in 1993, 2000 and 2012, respectively. From 1993 to 2000 he worked as an assistant professor at the Department of Radio Communication Systems and Networks (DRCSN), GUT. Since 2001 he has been working as an associate professor at the DRCSN. His research and development interests include analysis, simulation, design and measurements of cellular, wireless and trunked radio systems, techniques of digital modulation, channel coding, signal spreading, radio signal reception, measurement of radio wave propagation, field strength prediction, software radio design, location services, ad-hoc sensor networks, radio monitoring systems and radio navigation systems. He is the author and co-author of more than 200 papers. He is a member of the Electromagnetic Compatibility Section of the Electronics and Telecommunications Committee, Polish Academy of Science and the Institute of Electrical and Electronics Engineers organization.

E-mail: jstef@eti.pg.gda.pl
Faculty of Electronics, Telecommunications
and Informatics
Department of Radio Communication Systems
and Networks
Gdańsk University of Technology
Gabriela Narutowicza st11/12
80-233 Gdańsk, Poland

MOST WIEDZY Downloaded from mostwiedzy.pl