The Role of QoS in WebRTC and IMS-based IPTV Services

Michael Maruschke
Hochschule fuer Telekommunikation Leipzig (HfTL)
University of Applied Sciences,
Leipzig, Germany
Email: maruschke@hftl.de

Jens Zimmermann
Fixed Mobile Engineering Deutschland
Deutsche Telekom Technik GmbH,
Darmstadt, Germany
Email: jzimmermann@telekom.de

Kay Haensge
Telekom Innovation Laboratories
Deutsche Telekom AG
Berlin, Germany
Email:kay.haensge@telekom.de

Tilmann Bach
Technische Planung und Rollout
Deutsche Telekom Technik GmbH,
Berlin, Germany
Email: tilmann.bach@telekom.de

***Abstract*—This paper describes the considerable role of Quality of Service (QoS) for Web Real-Time Communication (WebRTC) clients connected with an IP Multimedia Subsystem (IMS)-based IP Television (IPTV) infrastructure. To raise the quality of experience for IPTV customers, the article focuses on the merging of the technical capabilities arising from both the IMS-based telecommunication networks including IPTV specific components and the WebRTC clients. The ongoing WebRTC standardization process as well as the state of the art WebRTC-QoS trends are considered. To enrich typical IPTV services with appropriate network QoS characteristics a scheme has been developed. The author's concept presents a proposal of an architecture featuring an integrated QoS functionality for WebRTC in conjunction with IPTV services. With our new approach, a WebRTC user inside a 4G mobile network can benefit from the integrated end-to-end quality for real-time IPTV services like Live TV. Composed of several open-source-based testbed solutions, a first prototype has been developed illustrating the QoS initiation procedures primarily.**

***Keywords*-QoS; WebRTC; IMS-based IPTV; EPS;**

I. INTRODUCTION

In times of ever-growing bandwidth needs by Internet users, applications and tightened network resources on side of the network infrastructure providers the importance of QoS mechanics rises heavily. Technologies enabling QoS needs to get deployed more and more corresponding to the communication context (e.g., for conversational voice: delay sensitive, for file transfer: packet loss sensitive). This especially embraces those applications that are not affected by QoS reservations a network provider manages thus far, the so called Over The Top (OTT) services. This includes all the currently established WebRTC services. If the WebRTC client requests for QoS ensured network resources while starting a new communication session, the telecommunications network can provide adequate resources. The advantages are obvious: both end-users and network providers can benefit from such an approach. For the end-users it is possible to experience a high quality even in OTT applications like WebRTC services and for the network provider new business cases are revealed when it is possible to sell the QoS features (inherited from Evolved Packet System (EPS) and fixed line Next Generation Network (NGN) networks) to OTT applications and end-users.

The provisioning of bandwidth many times over the needful proportion (the so called "‘overprovisioning"’) is going to get increasingly unsuitable. This applies to all kinds of IPTV services, especially if they deliver their content with High Definition (HD) or Ultra-High-Definition resolutions. The realisation of live TV with 4K display resolution would increase the end-to-end data rate enormously. Taking those trends into account, it seems reasonable for network infrastructure operators to provide the bandwidth in a more effective manner by using network QoS techniques.

This journal paper discusses the QoS as an important characteristic for real-time-based telecommunication services in general and for the particular field of IMS-based IPTV services. On the basis of the various QoS requirements for different IPTV services (like Live TV, Audio or Video on Demand) a new QoS resource class mapping is developed by the authors. Based on [1], we propose an architectural concept to enrich an established WebRTC session accessing an IMS-based IPTV network infrastructure with QoS features. Particularly, for an WebRTC end-user, which has access to a 4G mobile network, a new end-to-end QoS control mechanism has been developed and verified. Consuming real-time TV services like Live TV using a Web browser on a mobile device, the user benefits from a QoS concept that combines the well established QoS technology from the 4G mobile network with our new WebRTC QoS enrichment. While the authors paper [1] proposed the principle combination of WebRTC with IMS-based IP-TV services, this journal contribution is focussed on the enrichment of QoS for an WebRTC client consuming TV services with real-time characteristic. In contrast to this, our other publication [2] is aimed to establish QoS only for an conversational voice call. The authors' QoS extension principle is new and neither proposed by established international standardization bodies nor solved in a practical manner up to now. To verify our proposal, a first testbed has been implemented. Particularly, the QoS activation procedures have been tested.

The present journal paper is structured as follows: Specifying the used terminology like RTC and QoS, Section II offers a survey of the current status of QoS in context of WebRTC, of real-time communication in fixed and mobile telecommunication networks and of IPTV services. Based on an IPTV-QoS parameter mapping approach, Section III describes the authors new concept to integrate QoS dynamically in the WebRTC client accessing IMS-based IPTV architecture. (4G mobile network). The new architecture and their specifics are considered and the proof of concept incorporating an 4G mobile network is presented. In Section IV, a conclusion summarises the achieved results of this contribution and gives an perspective.

II. Status Quo

A. Real Time Communication and QoS

Real Time Communication (RTC) is generally characterised by the so called "Real-Time" condition. That implies that the value of the communication depends significantly upon the time at which the data is arriving at the data sink [3]. The throughput time of the data (flow rate) across the network delays the delivery of the data packets to the recipient. The tolerable delay time or latency depends on the type of the desired communication (e.g., conversational audio and video or real-time gaming). Besides the delay time, two other characteristic performance aspects can significantly influence the quality of the real-time communication. First, the circumstance that the transmitted data packets are routed through the network passing an unequal number of network elements, which results in variable arrival times at the recipients side. In packet-based networks, the varying delay of the transferred data is called packet delay variation or simply Delay Variation (DV) [4]. Sometimes, the different packet delivery times are also named as jitter [5]. Furthermore, if data packets reach the incoming data buffer on the receiver side too late, they will be discarded and consequently counted as lost packets.

Basically, QoS encompasses both the service categorization and the overall performance of the network communication for each service category. The International Telecommunication Union - Telecommunication Sector (ITU-T) describes QoS as the unity of all characteristics of a telecommunication service which are necessary to satisfy the service user. This includes the defined and the implied requirements for the complete customer satisfaction. Mean Opinion Score (MOS) methods are often used to indicate the measured or detected service quality. Initially developed for a subjective quality evaluation, the MOS method with it's rating score from 1 (bad) to 5 (excellent) is more and more used for objective QoS measurement methods.

Appropriate QoS characteristics like End-to-End-Delay of the transmitted packets, the Packet Delay Variation and the size of the tolerable Packet Loss are required to satisfy the telecommunication customer's expectations. The end-to-end quality of any telecommunication service depends on the performance of all involved components; the technical end-user system as well as each relevant network entity (see Figure 1). Hence, the QoS one-way-delay parameter is influenced by all components. For speech transmission, an one-way-delay (also known as 'mouth-to-ear-delay') of 150 ms is experienced as very good and all delays above 400 ms are considered to be unacceptable for the consumer [6].

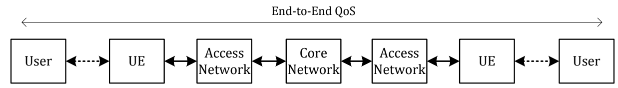

Figure 1: Schematic contributions to end-to-end QoS [7].

Focusing especially on IP-based networks, several QoS requirements are reasonably for a successful and effective operation of RTC. Therefore, the ITU-T also presents a generic QoS classification for public IP-based networks and the related network performance parameters IP Transfer Delay (IPTD), IP Delay Variation (IPDV), IP Packet Loss Ratio (IPLR) and IP Packet Error Ratio (IPER) (see also table 1 and table 3 of [8]). From a network topology point of view, this recommendation addresses all relevant network parts (Access Network; Aggregation Networks, Core Networks) but not the end-user or home-network side. Table I illustrates the eight defined IP network QoS classes and their corresponding QoS network parameters. Note that the parameter IPER is contributing insignificantly to the overall packet loss and therefore is not shown in the Table I. The value 'U' stands for 'unspecified'.

Furthermore, a guidance on usage of those eight QoS classes is given as follows, outlining corresponding communication examples:

- QoS class 0, for real-time, jitter sensitive and high interactive applications like Voice over IP (VoIP) and Video conferences;
- QoS class 1, for real-time, jitter sensitive and high interactive applications like VoIP and Video conferences, but with less constrained delay requirements;
- QoS class 2, for transaction data, highly interactive (signaling traffic);
- QoS class 3, for transaction data, interactive;
- QoS class 4, for low loss only applications like short transactions, bulk data, non real-time buffered video streaming;
- QoS class 5, for all other traditional applications of default IP network without any QoS demands
- QoS classes 6 and 7 have a provisional character and are designed for applications similar to applications in QoS class 0 or 1, but with more strictly demands for the packet loss rate.

B. QoS in IMS-based Telecommunication Networks

IMS, as an architectural framework for supporting IP multimedia services, was originally designed by the 3rd Generation Partnership Project (3GPP) for mobile core networks and successive expanded for fixed-line-based core networks [9]. It addresses multiple IP multimedia applications like speech and video communication, shared online whiteboards, telepresence conferences and multicast services. To provide these in a flexible and appropriate manner, telecommunication network operators differentiate their services for the customer regarding to the QoS characteristics [9]. Basically, QoS should be negotiable for

TABLE I: IP network QoS class definitions and network performance parameters by ITU-T Rec. Y.1541.

Network Performance Parameter	QoS Classes							
	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5	Class 6	Class 7
IP Transfer Delay	100 ms	400 ms	100 ms	400 ms	1 s	U	100 ms	400 ms
IP Delay Variation (Jitter)	50 ms	50 ms	U	U	U	U	50 ms	50 ms
IP Packet Loss Rate	0,1 %	0,1%	0,1%	0,1%	0,1%	U	0,001%	0,001%

IP multimedia sessions and their individual media components (like audio or video) both at the time before establishing a connection as well as during the established connection. QoS related concepts (QoS signaling, QoS resource reservation and allocation, etc.) belongs to this framework inherently due to the fact that IMS describes one technological concept to realise a NGN. Next to other capabilities like mobility of packet-based telecommunication networks, the QoS is a key feature in a NGN as defined by the ITU-T [10].

Mobile networks

QoS has always been considered in the standardization process of mobile networks. Therefore, in Universal Mobile Telecommunication System (UMTS) networks (3G) four QoS classes has been introduced. They are named as follows:

- Conversational Class,
- Streaming Class,
- Interactive Class, and
- Background Class [11].

These classes are also entitling the possible use cases for each of the four types. Conversational is used for real-time audio and video communication. Streaming can be used to stream audio or video data towards the User Equipment (UE) by having a small buffer and non-critical real-time constraints. Interactive is used for general user data transfer such as web browsing and application information exchanges. The last and lowest prioritised class is background. It is used for non time-critical applications, like email polling.

With the specification of EPS and fourth generation (4G) mobile networks, the 3GPP has developed a new, only packet-based core network domain (also known as Evolved Packet Core (EPC)) for both conversational voice/video communication as well as other packet or IP-based applications like public Internet communication. For this modern all-IP network, the 3GPP also defines a new QoS concept involving all relevant EPS network components like the UE, the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and the EPC. For enabling QoS inside the EPS a so called EPS Bearer is used to fulfill all requirements of the media delivery. Figure 2, depicts this EPS Bearer, which is correlated with the overall Bearer-based End-to-End (E2E) QoS concept of the 3GPP network infrastructure. The detailed description can be found in [12]. QoS classes are composed of a subset of standardized characteristics, they describe the packet forwarding treatment in 4G networks. For the QoS class determination the so called QoS Class Identifier (QCI) was introduced. Based on typically applications, various QoS groups separated in nine QCI values are defined in [13].

Since the 3GPP standardization institution provides their own QoS treatments, a mapping is always required. Therefore, Table II shows different QoS classes originated in 3GPP Release 99 networks (UMTS-3G) and EPS-based 4G networks, in relation to the main relevant QoS parameters Delay and Packet Loss. It should be clarified that in Table II with "Prio" the packet scheduling priority is meant and Packet Error Loss Rate (PELR) stands for an upper bound for the rate of non-congestion-related packet loss. While for the first four QoS classes (QCI 1 to 4) a Guaranteed Bitrate (GBR) is allocated, the following classes (QCI 5 to 9) are supported with Non-GBR characteristics. It is obvious, that real-time applications like audio and video conversation demands more strict packet delay requirements than traditional Internet traffic like file sharing, which is more critical regarding the packet loss characteristic. While for telecommunication carriers the guaranteed delivery of IMS Signaling is relatively important, the customer needs the stricter QoS resources for voice, video and interactive real-time gaming (see also QCI value 3 or 7). It is noticeable that for video streaming services (both live streaming and buffered streaming) exists several QoS relevant QCI values. Thus, live streaming services are typified by QCI value 2 and 7 with similar QoS characteristics (packet delay and packet error loss rate), but with different priority and bitrate guaranties. To handle QoS for buffered streaming services, a couple of QoS classes (QCI values 4, 6, 8 and 9) is specified. The assigned QoS relevant parameters packet delay and packet error loss rate are identical once again meanwhile the priority and the bitrate guarantee differs. Note that QCI value 9 is typically used for the default bearer of a UE for non privileged subscribers.

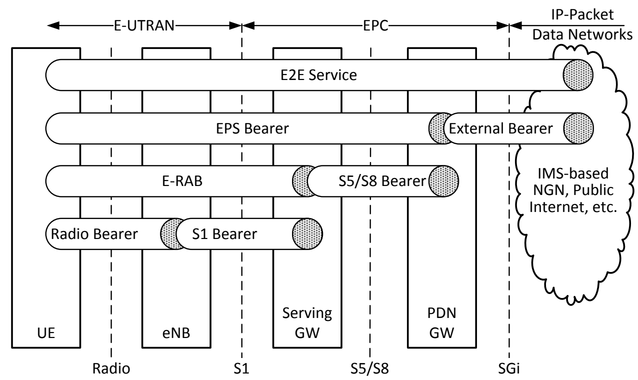

Figure 2: 3GPP overall QoS concept, based on [12].

Fixed line networks

QoS in fixed-line-based IMS networks differs relating to the used IP Connectivity Access Network (IP-CAN). While the 3GGP-based mobile networks use their typical access networks like UMTS Terrestrial Radio Access Network (UTRAN) or E-UTRAN, the fixed-line-based IMS networks provide the telecommunication services to the residential customer via any Digital Subscriber Line (xDSL) technique. Besides the fact that the access transmission technology differs, the

TABLE II: QoS class mapping between 3G and 4G mobile networks, based on [13].

UMTS Traffic Class (3G)	QCI (4G)	Prio	Bitrate Guarantee	Packet Delay Budget	Packet Error Loss Rate	Example Services
Conversational	1	2	GBR	100 ms	1 %	Conversational Voice
Conversational	2	4	GBR	150 ms	0,1 %	Conversational Video (Live Streaming)
Conversational	3	3	GBR	50 ms	0,1 %	Real Time Gaming
Streaming	4	5	GBR	300 ms	0,0001 %	Non-Conversational Video (Buffered Streaming)
Interactive	5	1	Non-GBR	100 ms	0,0001 %	IMS Signaling
Interactive	6	6	Non-GBR	300 ms	0,0001 %	Video (Buffered Streaming), TCP-based (e.g., www, chat, ftp, p2p file sharing, progressive video)
Interactive	7	7	Non-GBR	100 ms	0,1 %	Voice, Video (Live Streaming), Interactive Gaming
Interactive	8	8	Non-GBR	300 ms	0,0001 %	Video (Buffered Streaming), TCP-based (e.g., www, chat, ftp, p2p file sharing, progressive video)
Backround	9	9	Non-GBR	300 ms	0,0001 %	Video (Buffered Streaming), TCP-based (e.g., www, chat, ftp, p2p file sharing, progressive video)

Customer-Premises Equipment (CPE) (any terminal and associated equipment located at a subscriber's premises and connected with a carrier's telecommunication channel) is distinct from the connected network infrastructure. Therefore, mobile network customers are utilizing mobile devices like cellulars, smartphones or tablet PCs. In contrast to this, in fixed line networks the residential customer uses a home-network where various end devices are connected to each other and accessed on the CPE, which can be build by an Integrated Access Device (IAD) for IMS-based NGNs.

In fixed access line NGNs, the QoS management functions are supported by a Resource Admission Control Subsystem (RACS), which is responsible for elements of the policing control including resource reservation and admission control in the access and aggregation networks [14]. Any multimedia services like VoIP or IPTV can request particular QoS parameters, such as data throughput, latency, jitter and packet loss from the transport network side. Then, the RACS is responsible to manage this QoS requests by evaluating this in the context of predefined policy rules, and performing the reservation and allocation of adequate QoS resources through all affected transport network elements.

To ensure QoS aware NGN service delivery in fixed access networks, the RACS specification distinguishes between two abstract QoS architecture principles [14]:

- *guaranteed QoS:* traffic delivery service with absolute demand on some or all of the QoS parameters, such as throughput, latency, jitter and packet loss, and
- *relative QoS:* traffic delivery service without absolute demand on some or all of the QoS parameters.

In contrast to this, the support of QoS unaware ("Best Effort") networks as well as the support of networks that have statically provisioned QoS differentiation does not require any RACS functionality. To determine the various QoS classes the DiffServ classification is applicable [15].

Sometimes, for statically QoS support, the QoS marking inside the IAD-based Home-Network on CPE side is carried out by utilizing the Differentiated Services Code Points (DSCP) classification mechanism for each media flow.

C. QoS in IMS-based IPTV services

In principle, Video Quality is defined as the indicator, which evaluates the quality of the video stream delivered to the user [16]. This indicator describes the perception of the end-user in term of the video quality. Typically, objective perceptual video quality measurements models are utilizing a reference-based approach, like the ITU-T recommendation J.247 [17] describes it. Taking the end-user context by consuming Live TV or Video on Demand (VoD) into account, is it not applicable to operate with a video quality assessment method, which is functional with a full reference. The current issue is that there is no standardized approach for an objective video quality measurement model that does not need any reference signals. Therefore, in context of end-user quality survey of IPTV services the following indicators are proposed to characterize the quality of IPTV services [16]:

- Channel Availability (indicates the availability of Live TV/VoD channels proportional to the attempted channels),
- "Black Screen" Occurrences (e.g., effected by a major loss of video packets during a long period of time),
- Blockiness Occurrences (produced by a low-quality video compression when too few bits are present; it is perceptible by the contrast of color),
- Frozen Picture Occurrences (some picture appearing as stopped/frozen from time to time),
- Lip Desynchronization Occurrences (the synchronization of audio and video stream is not well),
- Zapping Delay (time, which is needed to be switch from one TV/VoD channel to another),
- Transmission Delay (indicates the delay to transmit the audio/video signal from the delivery point to the end-user's TV device; important for some cases like football matches),
- and others.

While reference [16] is focused on the context of end-user quality characterization and their indicators, it is obvious that the network-based QoS parameter Transfer Delay (TD), DV and Packet Loss Rate (PLR) have also high relevance.

As already introduced in [1], IMS-based IPTV services can be differed among each other regarding their kind of service or feature, based on [18]. For instance, linear live Television (TV) or real-time VoD will demand other network performance characteristics then the Electronic Program Guide (EPG) feature. While the linear live TV service demands more stringent QoS performance characteristics like low transfer delay, low packet delay variation and minimal packet loss, the IPTV content control feature EPG accepts less tightened QoS performance. The specification [18] also describes a general approach for dynamic QoS resource modification between Standard Definition (SD)-TV and HD-TV.

To fulfill an adequate QoS support for all typical IPTV services, a high-level guideline for the use of traffic management is given by [19]. Therein, a potential ITU-T Y.1541 performance class mapping for typical IPTV service applications is provided and visible in Table III. In general, the IPTV service category *Streaming* demands most of the QoS resources, because it is "live"-communication with strict real-time prerequisites. Besides, the rubric *Download* includes video content consuming service like near VoD. It becomes also known as progressive video download principle using by consuming Youtube-Videos.

TABLE III: Potential mapping of IPTV services to ITU-T Y.1541 QoS classes [19].

IPTVservice	Service applications	ITU-T Y.1541 QoS class							
		5	4	3	2	1	0	7	6
Streaming	Live TVcontent						x		x
	Video content					x		x	
	Audio content					x			
	Content control				x				
	Live speech						x		
	Live low- resolution video content						x		
Download	Video content		x						
	Data	x							
Upload	Video content		x						
Message exchange	Interactive			x					
	Non- interactive	x							
Middleware/ application	Portal			x					
	Payment transactions			x					

D. WebRTC and QoS

WebRTC introduction

A main focus of the upcoming WebRTC technology is to integrate real-time communication into standard web browsers without the need of any additional browser plug-in or software. The World Wide Web Consortium (W3C) defines an Application Programming Interface (API) for web developers [20]. It empowers the browser to capture video and audio inputs of the client's device. While the W3C is responsible for the web developer API, the Internet Engineering Task Force (IETF) standardizes all corresponding protocols in an active working group named "'Real-Time Communication in WEB-browsers - RTCweb" [21].

WebRTC has been specified to use secure transport of the Real-Time Transport Protocol (RTP) packets with Secure Real-Time Transport Protocol (SRTP) [22] including the mandatory Datagram Transport Layer Security (DTLS) encryption protocol [23] used for key negotiation [24]. For solving Network Address Translation (NAT) problems, WebRTC also provides Session Traversal Utilities for NAT (STUN) [25], Traversal Using Relays around NAT (TURN) and Interactive Connectivity Establishment (ICE) [26] capabilities. WebRTC requires Session Description Protocol (SDP) for the negotiation of the session properties and uses the whole SDP's Offer/Answer-Model.

The generic architecture of a WebRTC client is described by [27] and illustrated in Figure 3. The components can be described as follows:

- Webserver, which provides the web application to load and includes a server with which the client connects to handle all signaling;
- Browser, a generic web browser;
- Web application, application source code executed by the web browser;
- Browser RTC Function, WebRTC component in the web browser with voice, video and transport engines;
- Signaling Path, which is not specified but is needed to transfer the SDP information between the WebRTC Client and any other signaling endpoint (alternative bypassing the Webserver);
- Media Path, which transports the payload;
- STUN Connection, which is a component to bypass NAT restrictions.

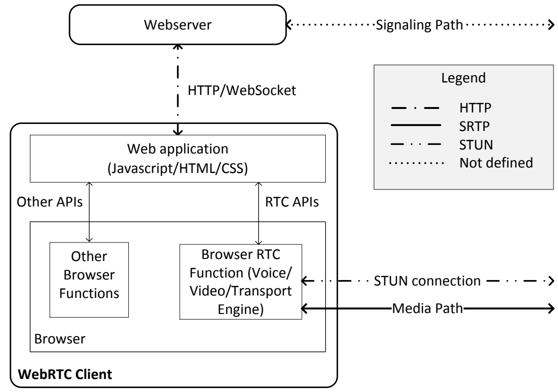

Figure 3: WebRTC client based on [27].

For a WebRTC client to successfully run, it is necessary to use a capable web browser. That means the browser has to implement the Browser RTC Function. Currently, web browsers like Google Chrome, Mozilla Firefox and Opera provide this component by default. Therefore, all devices that are able to run one of these browsers have the ability to use web-based real-time communication. This includes all, desktop and tablet computers, laptops and smartphones. At the moment, there are restrictions in some operation systems like Apple's iOS or Microsoft's Windows Phone.

In this subsection, we intend to find out if QoS is a topic for the developing WebRTC technology and their standardization bodies. Typical use cases for WebRTC and their requirements are described in [28]. Based on a simple video communication service, some use cases are presented, involving voice or video or data communication respectively combinations of those. For instance, to realize a "Multiparty on-line game with voice communication" quick updates of the game state are required, and they have higher priority than the voice [28]. Generally, the browsers should be able to render good audio and video quality for an adequate and acceptable jitter and packet loss values and must support a time synchronized audio and video playback function. If a WebRTC Client is accessed behind a residential router that supports any kind of data traffic prioritization, the user should be able to take advantage of this QoS support, provided by the network side. Summarizing, from an application layer point of view the WebRTC use cases do not define some exact and comparable values for the network related QoS parameter like delay, jitter or packet loss.

DSCP-Marking

An IETF-Draft proposes a QoS mechanism at the WebRTC client side using DSCP [29]. This document provides DSCP values for browsers to use for various classes of traffic. It proposes how WebRTC applications can mark data packets for a packet prioritization. It assumes that residential or wireless networks support traffic preferential treatment, based on DSCP. For all other cases including cellular mobile-based network access, this suggestion is not appropriate. However, if the real-time packet is transmitted towards a QoS capable core network domain, a QoS class mapping is needed. Besides, client side marking of IP packets is also a topic for the admission control processes. The issue is, if the marking is allowed and the network enforces the requested QoS parameters, other (unwanted) applications may also request prioritized forwarding. This situation may lead to overall high prioritized traffic with no benefit for the actual intended application. Therefore, the authorization of client-side requested QoS needs to be clarified. This draft does not cover any mapping processes for QoS management inside an 3GPP-based mobile network Release 8 or inside a fixed-line-based network, which does not use DSCP marking mechanism.

QoS concepts for WebRTC accessed to a mobile EPS network

Currently, few concepts exist which propose the enrichment of QoS for an WebRTC Client accessing a mobile EPS network. The 3GPP specification (Section Annex U of [9]) proposes a new architecture for including WebRTC Clients into an IMS-based EPC network. Hence, it describes only, that QoS support can be provided. However, any definite method or mechanism for this is missing. To fill this gap, an authors proposal for a QoS support method for WebRTC users, which are connected to an EPS-based IMS network infrastructure is documented in [2]. To fulfill this QoS support, the 3GPP-based architecture (taken from Section Annex U of [9]) became enhanced with the following entities named

- *WebRTC Client with QoS Awareness*,
- *WebRTC QoS Signaling Function (WQSF)*, and
- *eP-CSCF*, which is a modified eP-CSCF*.

This proposed WebRTC QoS Architecture is depicted in Figure 4. The concept and its added entities will be described briefly. Provided in reference [2], the WebRTC communication is enriched with the capabilities to request QoS resources via signaling all used RTC data flows to the WQSF, which adapts the information and starts a signaling session with the Policy and Charging Rules Function (PCRF) of the underlying core access network. This concept reuses the standardized EPS architecture as well as the new proposed integration of WebRTC clients into the IMS network [9]. The WebRTC Client acts as an UE that provides QoS awareness. It allows requesting QoS characteristics during the active conversation phase. With the help of periodical and event-based transmissions of the QoS relevant information (used media type like audio or video, and IP flow information like IP addresses and port numbers) of each established media stream, it is possible to use the EPS related QoS mechanisms dynamically. This means that the web application can start with one media type (e.g., audio) and add another media type (e.g., video) during the conversation phase. All involved components (e.g., end device, network entities) have to support such a dynamic mechanism to enforce the changed QoS resources.

For further understanding of the QoS control mechanisms in the described architecture, Figure 5 depicts the general QoS initiation procedure between the WebRTC Client and the EPS-based network architecture.

The Web Application starts with the QoS control sending relevant application layer QoS requirements, such as flow and media type information, towards the Application Function (AF). After a possible negotiation and session information signaling,

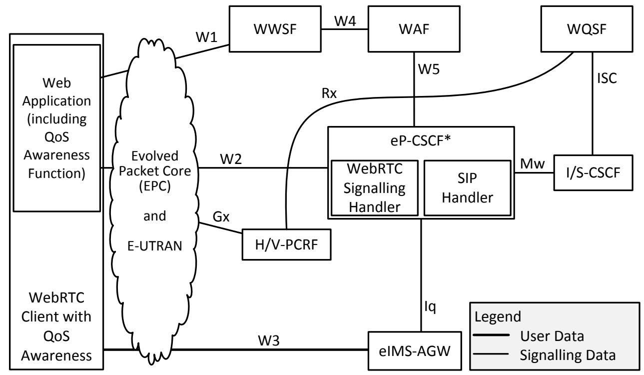

Figure 4: WebRTC QoS architecture for an EPS network [2].

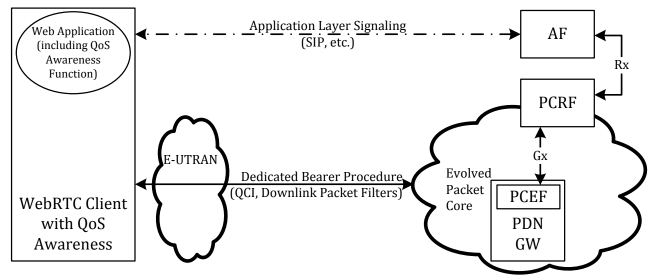

Figure 5: Principle of QoS initation.

the AF is aware of all QoS request parameters and will start the Policy and Charging Control (PCC) procedures by sending the mapped parameters into Diameter Attribute-Value-Pairs (AVP)s via the Rx interface towards the PCRF. The application can rely on the AFs endpoint to request QoS for its application flows. The PCRF will authorize the request and will start the resource allocation via the Gx interface to the Policy Control Enforcement Function (PCEF).

More details including the used QoS signaling and enforcement procedures, the evolved protocols and interfaces respectively message sequence charts are presented in [2].

E. Summary of Status Quo

As we have illustrated in this section, the role of QoS in telecommunication networks is heterogeneous. Depending on the specific scope of the standardization bodies 3GPP and ITU-T, the QoS relevant metrics and parameters differ among themselves. For instance, utilizing the network performance parameters IPTD, IPDV (Jitter) and IPLR the ITU-T recommendation Y.1541 [8] is clearly focused on the IP network layer (Open System Interconnection/International Organization for Standardization (ISO/OSI)-Layer 3 [30]). Regarding the area of applicability of the affected network segments all involved core networks and access networks are intended, but not the UE. Otherwise, the 3GPP defines in their Technical Specification TS 23.203 [13] a QoS parameter called QCI, which is closely associated with packet forwarding treatment characteristics like guaranty of bitrate, packet scheduling priority, packet delay budget and packet error loss rate. The scope of this standardized QoS parameter QCI is focused on the EPS Bearer lever (ISO/OSI Layer 2) between the mobile UE side, the mobile access network, the mobile core network (e.g., EPC), but not the IP backbone network, which is located behind the mobile core network. Concluding this, it means that a QoS mapping between heterogeneous telecommunication network ecosystems, which are supporting RTC like conversational voice or live TV is not simple. The challenge is to realize end-to-end QoS over such various networks transparently and perceivable for the end-user. From this perspective, QoS should be focused on a *horizontal QoS level* (see also Figure 1). Otherwise, the QoS metrics differ, depending on the functional layer (e.g., the ISO/OSI protocol layers). It is obvious that the overall QoS result depends on close interworking of all involved protocol layers. For instance, in an UE the QoS E2E service can not be realized successful, if all lower QoS layers (EPS Bearer, E-UTRAN Radio Access Bearer (E-RAB) and Radio Bearer) are not assuring their own QoS support (see also Figure 2). This QoS perspective is also called *vertical QoS level*. An overall QoS concept needs to be comply both, the *horizontal* and the *vertical QoS level*.

Independent of the above described QoS taxonomy and metrics, from the end-users perspective the expectations on a communication application are highly relevant. The perceived QoS by the users strongly depends on the performance of the network. However, it is measured by the opinion of the users. A typical subjective metric to measure this QoS performance by the consumers is commonly known as a MOS method. Though various users are consuming the same content (e.g., a Youtube Video) it will result in subjective and different perception and quality ratings. As an example, one user may rate the perceived quality as good, while another user will rate the same communication application as not acceptable.

Regarding IPTV, the standardization bodies define different IPTV Services and for each they assign different objective network QoS classes (see also Table III). For that, in Section III we propose an IPTV Service mapping into the relevant network QoS classification.

III. Concept

A. QoS Enrichment for IMS-based IPTV

The authors concept of the enrichment of QoS for typical IPTV services is based on specifications provided by the standardization bodies ITU-T and European Telecommunications Standards Institute (ETSI) [19] [18]. The most common services and features described are Live/Linear TV or Video, respectively Audio on Demand as streaming application and also Download applications like near video on demand (e.g., Youtube video downloading) or the feature EPG as type of data download.

Various IPTV services with its particular real-time communication characteristics demand different QoS resources. Therein, the IPTV services are related to network QoS classes, defined by 3GPP and ITU-T [13] [8].

We propose an assignment mechanism, which can be used for IPTV services to correlated to their appropriate network QoS classes. This mapping approach is depicted in Table IV.

In this table, a mapping is performed between the IPTV Service Applications, taken from ITU-T Y.1920 with the category '*Example Services*' of EPS QoS classes in 4G networks (see also Table II).

As already described in the summary of Section II, the differences between the two network QoS class approaches (3GPP versus ITU-T) were indicated. Therefore, those consolidation make sense.

As a result on the basis of the correlated IPTV services (see also Table IV), a correlation of concrete services/features to the adequate network QoS classes given by 3GPP and ITU-T is shown. Accordingly, and for instance, the Live/Linear TV service is associated with the EPS QoS class parameter QCI value 2, priority value 3 and GBR, as well as the ITU-T Y.1541 classes 0 or 6. This also encompasses the linked network QoS parameter delay, packet loss and delay variation. Taking into account that the EPS QoS classes and their corresponding parameters Delay Budget and PELR are focused on the network segments UE - Access Network - EPC, on the ISO/OSI protocol layer 2. In contrast to this, the ITU-T QoS classes are aimed at IP core and IP access network parts of the IP layer. Therefore, both the vertical as well as the horizontal QoS levels differ.

As depicted in Table III, the preferred ITU-T QoS classes for video content are class 1 and class 7. Based on this information the mapping to the QCI classes leads to QCI class 4, 6 or 8. Each of these QCI classes share the same values for the technical parameters delay and packet loss. The QCI class 4 is used as favourite, because it has a higher priority as the other two classes and it uses a GBR. For audio content the ITU-T class 1 should be used. The mapping to the QCI approach is similar to video content. As result the QCI class 4 is used for audio content, too.

Another IPTV service, described in [1], is the content control. For this service, the Table III gives information about the use of ITU-T class 2. The mapping of the technical parameters lead to the QCI class 5. Otherwise, compared to real-time audio or video, the content control has not such high demand for the bit rate. But the content control should not be affected by congestion, so a higher priority is more preferable. Therefore, the QCI class 5 is a valid approach.

B. Consolidating EPS and IMS-based IPTV Architectures for QoS support

The proposed architecture is based on the presented IMS-based IPTV architecture from [1], depicted in Figure 6, and the analyzed concept for WebRTC accessing to an mobile EPS network [2]. The principal components are marginally accommodated. The innovation in this approach is the enrichment of QoS for the IMS-based IPTV environment with WebRTC. The consolidated architecture of the proposed concept is depicted in Figure 7. Components and interfaces of the architectures are described in the following section.

Components and interfaces, which result from the approach from [1], are:

- Components:
    - Webserver
    - WebRTC client
    - Signaling GW (SGW)
    - Core IMS
    - SDF
    - SSF
    - SCF
    - MCF
    - MDF (modified for WebRTC)

TABLE IV: Mapping of IPTV services towards 3GPP/ITU-T QoS classes.

IPTV Service Application (ITU-T Y.1920)		IMS-based IPTV Services (ETSI TS 182027)	EPS QoS Classes (3GPP TS.23.203)			QoS Classes (ITU-T Y.1541)
			QCI	Priority	GBR / Non-GBR	
Streaming	Live TV content	Linear / Broadcast TV	2	3	GBR	0; 6
	Video content	VoD, Network PVR, Time-Shift TV	4	5	GBR	1; 7
	Audio content	AoD	4	5	GBR	1
	Content control	Content control	5	1	Non-GBR	2
Download	Video content	Push VoD, Near VoD	6; 8; 9	6; 8; 9	Non-GBR	4
	Data	EPG	9	9	Non-GBR	5
Upload	Video content	Interactive TV	6; 8; 9	6; 8; 9	Non-GBR	4

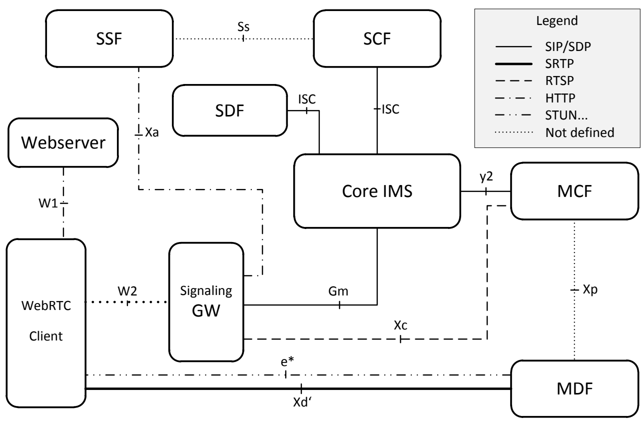

Figure 6: Architecture for WebRTC clients connected with an IMS-based IPTV ecosystem [1].

- Interfaces:
  - W1
  - W2
  - e*
  - Xp
  - Xd (modified to Xd')
  - Gm
  - Xa
  - Ss
  - ISC
  - y2

The web server is needed to provide the WebRTC application sources. The WebRTC application is executed in a WebRTC capable browser. The architecture of a WebRTC client is depicted in Figure 3. The application provides signaling functions for the communication with the core network via the inserted Signaling Gateway (SGW). Therefore, to make IMS-based IPTV services accessible to WebRTC clients, Generic IPTV Capabilities described in [18] are supported. As described in [1], the SGW implements some of these generic capabilities.

This gateway function converts session control messages coming from the WebRTC client side into Session Initiation Protocol (SIP) messages for the IMS core network side and vice versa. The SGW generates and forwards SIP messages towards the IMS core network and acts in place of the WebRTC client as a SIP capable signaling endpoint. As shown in Figure 6 the SGW also converts the session control messages from the WebRTC client into Hypertext Transfer Protocol (HTTP) and Real-Time Streaming Protocol (RTSP).

The core IMS is formed by the components, which are specified in [9]. This bulk of components in the core IMS implement several services, such as registration, provisioning, routing, accounting, billing, etc.. For more details of the components and their interworking and special function see [9].

The architecture includes also several IPTV functional components, which are standardized in [18]. The Service Discovery Function (SDF) in Figure 7 provides information about available services and related SSFs. The Service Selection Function (SSF) provides information that contains the metadata of the available content. The Service Control Function (SCF), is a SIP Application Server (AS) and the reference point for IMS UEs to start and control the IPTV sessions. The Media Control Function (MCF) controls the media transport of the MDF and receives instructions of the SCF and the UE. Also the selection of the right MDF is part of the MCF. The selection is made by several information, for example on codec information or geographical location. After a successful selection, the MCF transmits important session description information to the MDF. The Media Delivery Function (MDF) contains the media data and transmits them to the UE. In the architecture, the MDF is a modified MDF for WebRTC clients, which is proposed in [1]. It provides all necessary functions to establish a session with a WebRTC client. It supports STUN, ICE, DTLS and SRTP functionality. Also, a streaming engine with WebRTC capable codecs are implemented. These components are described more

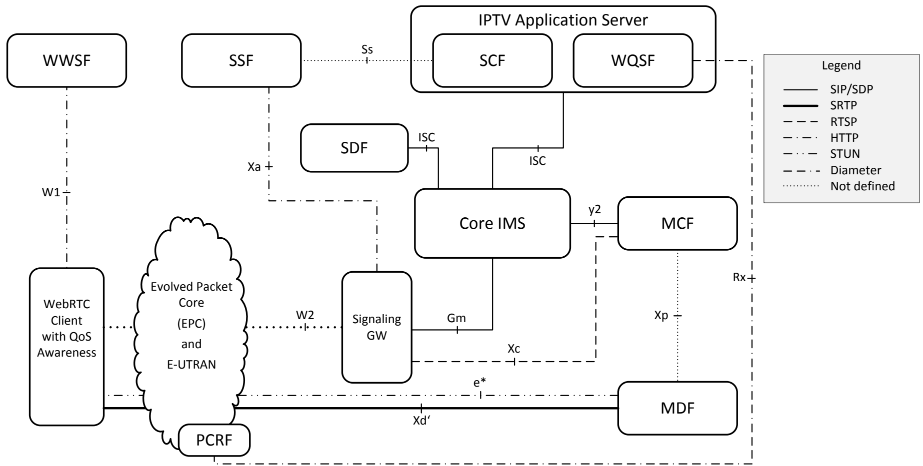

Figure 7: Consolidated architecture for IMS-based IPTV for QoS support.

in detail in [1].

The interfaces, shown in Figure 6, differ from the standardized architecture. The interfaces for the service interconnection are relocated from the user side to the SGW, which resides in the Telco's network infrastructure. The functionality and the used protocols of the Gm, Xc and Xa interfaces, located between the added SGW and the service or the core functions, are still conform to their specification [31]. The Ss, ISC and y2 interfaces remain unaffected. In addition to these interfaces above, some are modified or added and differ from the specification. These are described in more detail below.

The formerly undefined Xp interface between the MCF and the MDF is extended in the range of functions, respectively through the extension of the MDF.

The W1 interface is a reference point between the WebRTC client and the Webserver. It is used to download the application's source file using HTTP. Via this interface, the user receives the latest WebRTC web application.

The W2 interface is located between the WebRTC client and the SGW. The used protocol for this reference point is not defined and could be design by the developer. Therefore, the developer can choose from several state-of-the-art client-server protocols. The meaning of protocol messages regarding this interface must cover the sense of the transfered protocol messages from the interfaces Gm, Xa and Xc.

The Xd' interface between the WebRTC client and the MDF is responsible for media delivery using SRTP [32]. The original Xd interface only supports RTP/RTCP or HTTP for media delivery, so the modified interface for WebRTC interconnection is named Xd'. This modification results from the mandatory use of a secure connection in WebRTC [22].

The e* interface is a second reference point between the WebRTC client and the MDF. This interface is added in the concept of [1]. It is used for STUN connectivity checks between the WebRTC client side and the MDF. STUN is a mandatory to implement feature of WebRTC because the most WebRTC clients are behind NAT firewalls.

To enrich the whole environment with QoS, some components and interfaces have been added or modified. The resulting architecture is depicted in Figure 7 and its components and interface are described as follows:

- Components:
  - WWSF (added)
  - WebRTC client with QoS Awareness (modified)
  - IPTV Application Server combining SCF with WQSF (added)
  - EPC (added)
- Interfaces:
  - W2 (modified)
  - Rx (added)

In contrast to Figure 6, the WebRTC Web Server Function (WWSF) is depicted and represents the Webserver of the original architecture. The WWSF provides the actual WebRTC web application sources. This function may also be used for web-based authentication and authorization.

The WebRTC application is modified to support QoS. Therefore, to make QoS accessible to the WebRTC client, a special QoS awareness function is integrated. This integrated awareness function retrieves periodical and event-based flow information of the established and used media channels.

The IPTV Application Server acts as SIP AS combining the SCF with the WQSF, both functions are acting together as a single endpoint against the core IMS. The SCF part handles the IPTV specific messages and provides functions to control the session in a way which is described in [18]. The internal WQSF processes the WebRTC related flow information into Diameter-based QoS request messages and hence acts as an AF against the EPS network. More details of this WQSF are described in Figure 4 and in [2].

As access and control network we considered the EPC connected with the E-UTRAN. Together, they both build the overall EPS. To achieve QoS for the customers, the PCC concept, involving the PCRF and PCEF (as well as other internal components), is fully integrated into the EPS infrastructure.

The interface W2 is extended with specific QoS information messages. The functionality to support messages from the Gm, Xa and Xc remains unaffected.

A newly added interface is the Rx interface. This reference point is located between the IPTV AS, here the AF of the

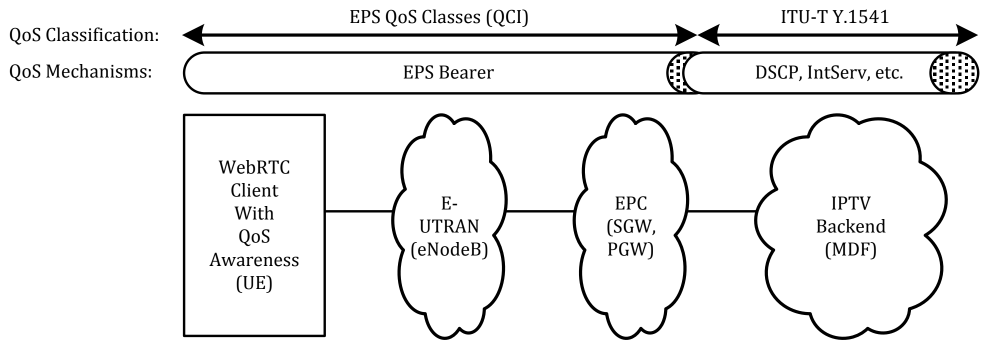

Figure 8: QoS Enforcement on the User Data Plane.

WQSF part of the AS, and a PCRF of the access network. The utilized protocol at this reference point is Diameter and it is used to transmit QoS related messages. The specific Diameter messages are described in [2] in more detail.

Figure 8 depicts the E2E chain for QoS enforcement between the UE and the MDF. In this chain two QoS classification schemes are applied, which exploits the mapping as proposed in Table IV. On the mobile network side, the user traffic treatment is categorized through the EPS QCI. To fulfill the QCI requirements, the user traffic is treated in an adequate EPS Bearer. On the IPTV Backend side, the user traffic treatment is categorized based on the recommendation of ITU-T Y.1541. For this, multiple QoS enforcement mechanisms can be used (e.g., DSCP, IntServ, MPLS). To achieve the expected quality on the consumer side, a mapping of the different QoS mechanisms is also necessary.

C. Proof of Concept (EPS and QoS with IMS IPTV over WebRTC)

Testbed components

To verify the functionality and the usability of the proposed concept, a testbed is prepared. With this testbed, the QoS initiation procedures for the content on demand use case (audio and video) is implemented and tested. However the actual enforcement of the QoS parameters on the user data plane is still outstanding.

For testing the concept the Google Chrome browser in version 38, which supports WebRTC, is used. The basis of this testbed is formed by an open-source IMS core network implementation originating from Fraunhofer FOKUS institute and now available on reference [33].

E-UTRAN functionality is provided by a LTE Femto Cell prototype with integrated eNodeB functionality by the company ip.access [34]. As mobile core network, an OpenEPC Rel. 3 testbed, initially developed from Fraunhofer FOKUS institute, is used [35].

The WWSF is an Apache HTTP Web Server, which provides the web application. The WebRTC client is implemented by using HTML5 and JavaScript. Based on this the Graphical User Interface (GUI) of the client is a responsive web site design using the jQuery mobile framework. This framework makes web sites accessible to all smart phone, tablet and desktop devices. The clients source code, based on JavaScript, utilizes the WebRTC API.

The SGW is written in C# and designed to handle several WebRTC Client sessions simultaneously. The prototyped SGW provides the main functionality for the interaction with the Gm and the Xa interface. The SGW firstly appears in [1] and is more sophisticated to cover the QoS support. Based on the sipsorcery project, an enhanced SIP protocol stack supporting IMS specific extensions is implemented [36].

Also, the IMS-based IPTV components are prototyped and inherited from [1]. All prototyped IPTV components, written in Java, are based on the technical specification [18]. The IPTV AS, containing the SCF and the WQSF, accepts the SIP requests via the JAIN-SIP stack. The SCF part handles all relevant SIP IPTV messages. The WQSF adapts the SIP requests containing the QoS relevant information into Diameter requests, based on JavaDiameterPeer library included in the OpenIMSCore project. The implementation of the MDF parses the session information, passed by the MCF. Within the MDF, the information are distributed to different engine functionalities. Open-source frameworks are used for

- The ICE agent with the STUN functionalities (icedjava) [37]
- The DTLS key exchange (BouncyCastle) [38],
- The SRTP implementation (srtplight) [39],
- And the streaming server (FFmpeg) [40].

Message Sequence

A sequence for Video on Demand over WebRTC with additional QoS reservation is depicted in Figure 9. The sequences can be described as follows:

Sequence 1) depicts the service discovery whereas the WebRTC client maps the service discovery messages into feasible messages to the Signaling Gateway via W2 and vice versa and whereas the Signaling Gateway maps the service discovery messages into feasible messages to the SDF via Gm and ISC and vice versa.

Sequence 2) depicts the service selection whereas the WebRTC client maps the service selection messages into feasible messages to the Signaling Gateway via W2 and vice

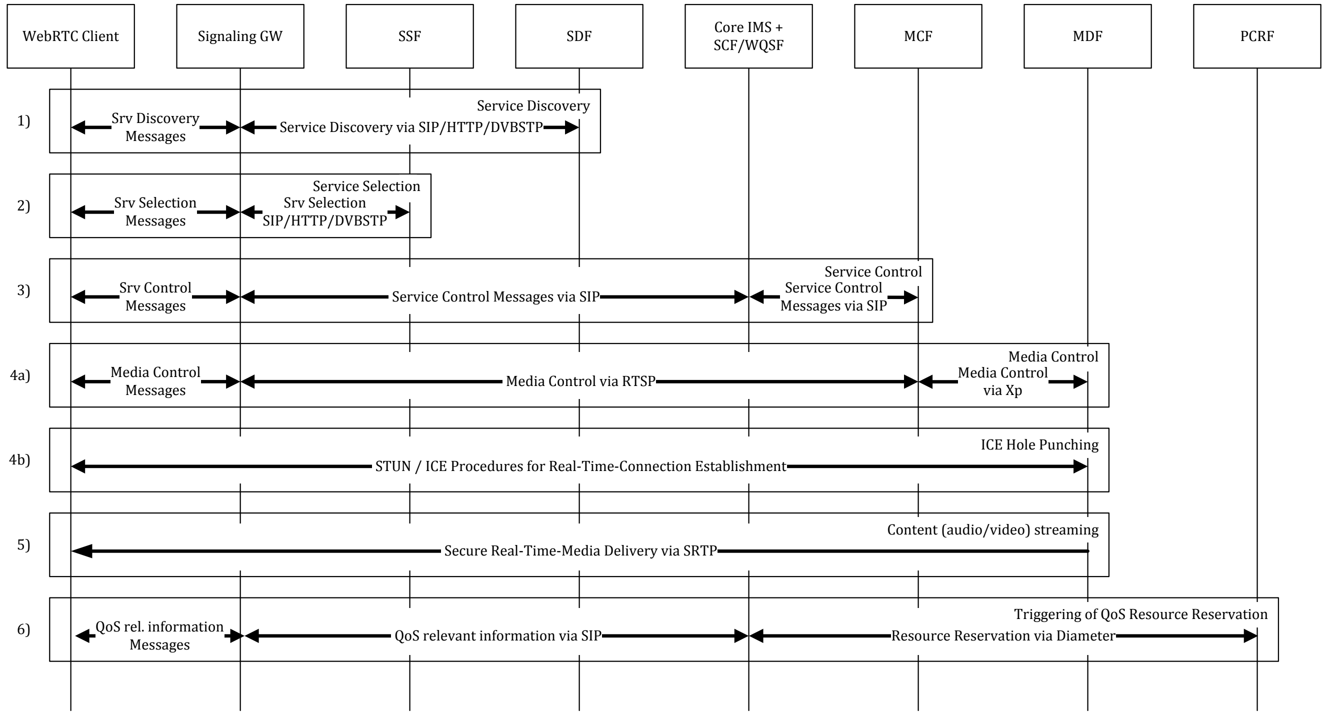

Figure 9: Sequence chart for WebRTC-based VoD with QoS resource reservation.

versa and whereas the Signaling Gateway maps the service selection messages into feasible messages to the SSF and via Xa vice versa. In the messages from the SSF, the client receives the service identifier to start the service.

Sequence 3) depicts the service control whereas the WebRTC client maps the service control messages into feasible messages containing the service identifier to the Signaling Gateway via W2 and vice versa and whereas the Signaling Gateway maps the service control messages into feasible messages to the IMS core via Gm and vice versa. With smart SIP routing the core transmits the messages to the SCF via IMS Service Control (ISC) by triggering on the service identifier. The SCF initiate the service delivery with session control messages via y2. Following session control message are transmitted from the user side to the SCF and vice versa.

Sequence 4a) depicts the media control whereas the WebRTC client maps the media control messages into feasible messages to the Signaling Gateway via W2 and vice versa and whereas the Signaling Gateway maps the media control messages into feasible messages to the MCF 16 via Xc and vice versa. Sequence 4a) further depicts the media control whereas the MCF maps the media control messages into feasible messages to the MDF via Xp and vice versa. The concurrent running sequence 4b) depicts the ICE/STUN procedures to establish a real-time path between the WebRTC Client and the MDF via e*.

Sequence 5) depicts the secured real-time streaming of audio/video between the WebRTC client and the MDF via Xd.

Sequence 6) depicts the QoS control process with signaling of QoS relevant information from the WebRTC client towards the targeted WQSF. The QoS relevant information is then used to request QoS related resources on the EPS network. This Sequence 6 is further described in the following subsection.

Triggering of QoS Resource Reservation

For triggering the QoS resource reservation, Figure 10 depicts a detailed message flow chart where the QoS relevant information is forwarded from the WebRTC client towards the QoS related network entities.

The UE's running WebRTC application retrieves all relevant information from the browser's internal *Browser RTC* function of the active WebRTC PeerConnection and extracts all connectivity pairs in which the user data is transmitted, received, or both. The application stores this connectivity information (including local and remote IP addresses as well as their transported media type) and prepares it into an overall JavaScript Object Notation (JSON) document, displayed in Listing 1. For each media flow, an array element will be added into the JSON document. As an overall QoS parameter, the document contains a field named *serviceType*, which incorporates the name of the used IPTV service. With that, all involved network entities can adapt the specific characteristics taken from Table IV for the IPTV application flow handling. In the example, the *serviceType* is set to *streaming_video_vod*. Analogue to this, it is also possible to use other IPTV services based on Table IV.

The presented JSON document (see Listing 1) will be relayed from the WebRTC client side into the SIP-based IMS environment. The Signaling Gateway enforces the conversion of the QoS relevant information received from the WebRTC proprietary signaling channel into standardized SIP messages (including the JSON document as SIP body).

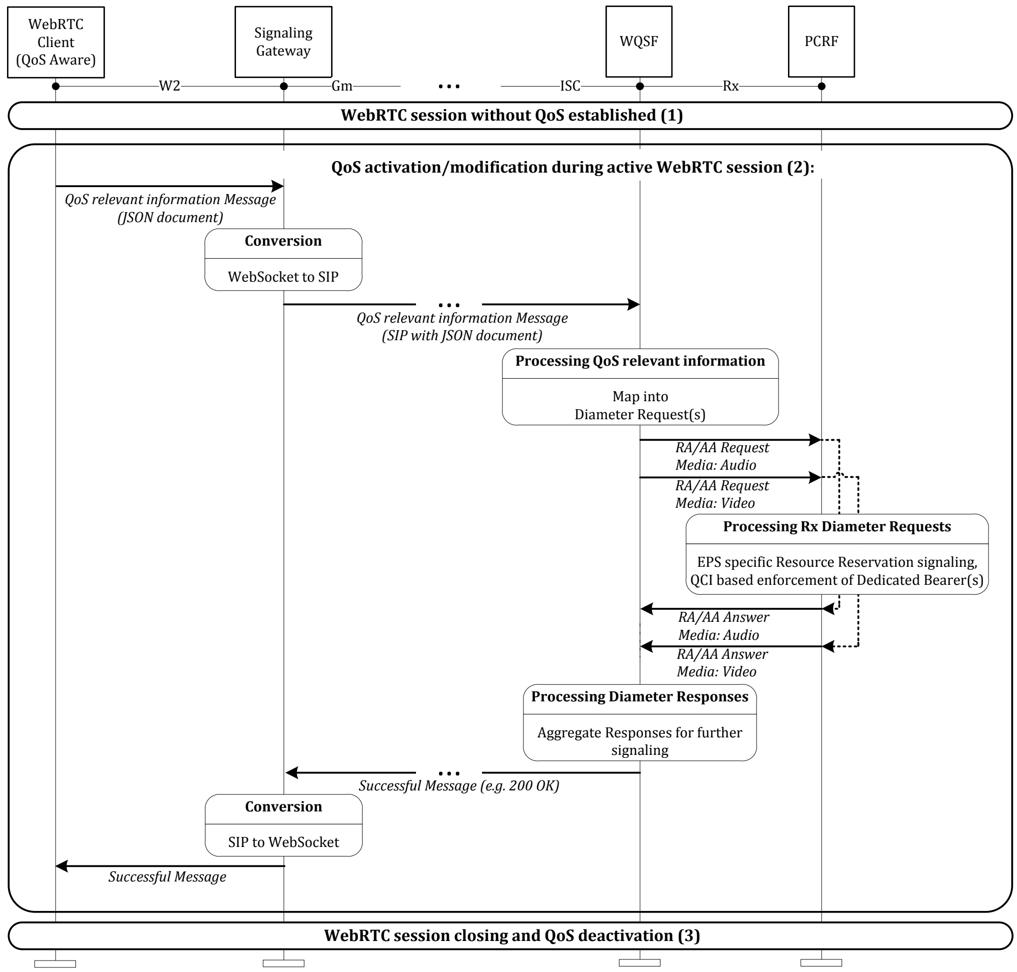

Figure 10: Sequence for QoS activation during a WebRTC session.

Listing 1: JSON document including QoS relevant information, provided by the WebRTC client.


```
{
 "cmd": "statisticInformationrequest",
 "attributes": {
  "sessionId": "406abf0d12174....",
  "localIdentity": "alice@domain.test",
  "messageType": "update",
  "remoteIdentity": "tvservice@domain.test",
  "serviceType": "streaming_video_vod"
  "flows": [
   {
   "remoteAddress": "192.168.7.4:34961",
   "localAddress": "192.168.13.63:56878",
   "mediaType": "audio"
   },
   {
   "remoteAddress": "192.168.7.4:40546",
   "localAddress": "192.168.13.63:56878",
   "mediaType": "video"
   }
  ]
 }
}
```


The WQSF performs the processing of the incoming JSON document. According to the content of the objects in the JSON document the WQSF generates related Diameter-based AVPs, as shown in Listing 2. For instance, the *AF-Application-Identifier* contains the intended *serviceType*: *streaming_video_vod*. Furthermore, the request also comprises the *Flow-Description* for the given video media flow.

Listing 2: Mapping of VoD with media type video into a Diameter Request.

```
Command Code: 265 AA-R (Request)
ApplicationId: 3GPP Rx (16777236)
...
AVP: Subscription-Id(443)
  AVP: Subscription-Id-Type(450)
    val=END_USER_SIP_URI (2)
  AVP: Subscription-Id-Data(444)
    val=sip:alice@domain.test
AVP: Media-Component-Description(517) vnd=TGPP
  AVP: Media-Type(520) vnd=TGPP val=VIDEO (1)
  AVP: Media-Sub-Component(519) vnd=TGPP
    AVP: Flow-Number(509) val=1
      AVP: Flow-Description(507) val=
        PERMIT OUT udp from 192.168.13.63 56878
```

```
        to 192.168.7.4 40546
      AVP: Flow-Description(507) val=
        PERMIT IN udp from 192.168.7.4 40546
        to 192.168.13.63 56878
...
AVP: AF-Application-Identifier(504) vnd=TGPP
  val=73747265616d696e675f766964656f5f766f64
  // Hex ASCII for: "streaming_video_vod"
```

Sequence 1 in Figure 10 shows the WebRTC session establishment phase. All procedures for this establishment such as the exchange of security keys, candidates for addressing issues etc. are covered in here but will not be described further. The first sequence is finished at this point. The establishment of the QoS characteristics will be processed in Sequence 2.

The second sequence depicts the actual QoS resource reservation interaction of the WebRTC session as well as the information sending process towards the WQSF. The WebRTC QoS Awareness Function processes QoS relevant session information of each media channel and forwards this as a JSON document towards the SGW. The SGW performs a conversion of the WebSocket message into a SIP request including the JSON document. This request will be forwarded through the core components towards the WQSF which maps this QoS relevant information into adequate Diameter requests. For each flow and media type (audio and video), described in the *flows*-array, the WQSF generates respective Re-Auth- or Authorise-Authenticate-*Request* (RAR/AAR) messages based on the JSON document [41]. The PCRF receives the Diameter messages and executes an resource reservation signaling based on the EPS QoS Class principles [13]. The proposed mapping is based on Table IV. Based on the given *AF-Application-Identifier* and the described *Media-Type* (audio and video), the QCI-based establishment and enforcement with Dedicated Bearers towards the PCEF inside the EPS core network. After processing the incoming Diameter Requests, the PCRF responses with Diameter-Answers towards the WQSF respectively. Inside the WQSF, the answers are stored and aggregated, related to that base JSON document. If all requests are answered successfully, the WQSF replies with a *successful message* (i.e., SIP Response 200 OK) along the initial way through the core network components. Finally, the SGW performs a conversion of the SIP response into WebSocket message.

The third sequence depicts the session closing procedures and has an analogue behavior. Initiated by the WebRTC client side, the session closing and QoS deactivation process follows same pattern as Sequence 2.

IV. Conclusion

The authors showed how QoS management could work for a WebRTC client which is connected to a mobile 4G network. In particular, we proposed a novel concept, which includes the following:

- An adequate QoS class mapping (see also Table IV), which can be implemented into all involved network elements (4G mobile and IPTV core network);
- An aggregated network architecture (see also Figure 7);
- A detailed QoS control concept, which is already prototyped and tested in practice (see also Figure 10);
- A concept for E2E QoS enforcement on the user data plane.

Considering an use case for live TV, respectively VoD, it is pointed out how WebRTC clients could benefit from a resource reservation rather than having no QoS allocation. For IPTV services like Near VoD (e.g., YouTube watching) a QoS support as proposed in Table IV should be helpful for mobile 4G end-users. We see a high potential to combine our WebRTC-based QoS concept with the VoD services utilizing recent HTML5 technologies. The technical details for a successful consolidation are for further study.

The following aspects are relevant within this journal contribution and are still outstanding at the moment:

- Implementation of the QoS relevant enforcement components in the User Plane and their performance testing;
- Analyzing the subjective end user expectations correlating to the different IPTV services;
- Involving other IPTV quality assessment methods (instrumental and perceptual).

The authors are positive about the increasing relevance of end-to-end QoS in heterogeneous networks in the near future.

Furthermore, several fifth generation (5G) white-papers forecast the heavily growing need for QoS resources for the next years [42]. With the identified new upcoming technologies therein, such as 3D audio, 3D video and ultra-high-definition formats and codecs, the demand for lower latencies and higher per-user data rates will increase tremendously. In that process, it is important to detect the necessary QoS demands for each contexts individually. Not until then, it is possible to deal fairly on the finite network resources and serve all users with the best experience for their individual real-time applications.

QoS should be made accessible, not only statical by network providers pre-configured parameters but also dynamically allocatable through the users applications regarding his current state of communication. The author's proposed concept, providing an interface for the user applications to request for a QoS reservation, could be a solution for the future.

Acknowledgment

The Telekom Innovation Laboratories (T-Labs) and the Hochschule fuer Telekommunikation Leipzig (HfTL) are actively cooperating since 2011. Both are working together on common topics in the area of fixed-/mobile converged network infrastructure including IMS-based services. They are participating in ongoing projects relating WebRTC with Telco Assets like QoS and interoperability. This paper also presents some of the results and acquired experience arising from [1] and [2].

References

[1] T. Bach, M. Maruschke, J. Zimmermann, K. Haensge, and M. Baumgart, "Combination of IMS-based IPTV services with WebRTC," ICCGI 2014, The 9th International Multi-Conference on Computing in the Global Information Technology, IARIA, June 2014, pp. 140–145.

[2] K. Haensge and M. Maruschke, "QoS-based WebRTC access to an EPS network infrastructure," in 2015 18th International Conference on Intelligence in Next Generation Networks (ICIN 2015), Paris, France, Feb. 2015, pp. 9 – 15.

[3] C.Aras, J.Kurose, D.Reeves, and H.Schulzrinne., "Real-time communication in packet-switched networks," Proceedings of the IEEE, Jan 1994, pp. 122 – 139.

[4] ITU-T, "Internet protocol aspects Quality of service and network performance;Internet protocol data communication service IP packet transfer and availability performance parameters," International Telecommunication Union (Telecommunication Standardization Sector),

REC Y.1540, Nov. 2007. [Online]. Available: http://www.itu.int/rec/T-REC-Y.1540-200711-S/en

[5] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," RFC 3550 (Standard), Internet Engineering Task Force, Jul. 2003, updated by RFCs 5506, 5761, 6051, 6222. [Online]. Available: http://www.ietf.org/rfc/rfc3550.txt

[6] ITU-T, "TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS, International telephone connections and circuits General Recommendations on the transmission quality for an entire international telephone connection-One-way transmission time," International Telecommunication Union (Telecommunication Standardization Sector), REC G.114, May 2003. [Online]. Available: http://www.itu.int/rec/T-REC-G.114-200305-I/en

[7] ——, "Quality of telecommunication services: concepts, models, objectives and dependability planning Terms and definitions related to the quality of telecommunication services; Definitions of terms related to quality of service," International Telecommunication Union (Telecommunication Standardization Sector), REC E.800, Sep. 2008. [Online]. Available: http://www.itu.int/rec/T-REC-E.800-200809-I/en

[8] ——, "Internet protocol aspects Quality of service and network performance;Network performance objectives for IP-based services," International Telecommunication Union (Telecommunication Standardization Sector), REC Y.1541, Dec. 2011. [Online]. Available: http://www.itu.int/rec/T-REC-Y.1541-201112-I/en

[9] 3GPP, "IP Multimedia Subsystem (IMS); Stage 2," 3rd Generation Partnership Project (3GPP), TS 23.228 v11.4.0, Mar. 2012. [Online]. Available: http://ftp.3gpp.org/specs/html-info/23228.htm

[10] ITU-T, "Next Generation Networks Frameworks and functional architecture models;General overview of NGN," International Telecommunication Union (Telecommunication Standardization Sector), REC Y.2001, Dec. 2004. [Online]. Available: http://www.itu.int/rec/T-REC-Y.2001/en

[11] 3GPP, "Quality of Service (QoS) concept and architecture," 3rd Generation Partnership Project (3GPP)(Release8), TS 23.107 v8.2.0, Sep. 2011. [Online]. Available: http://www.3gpp.org/DynaReport/23107.htm

[12] ——, "E-UTRA and E-UTRAN overall description," 3rd Generation Partnership Project (3GPP), TS 36.300 v8.12.0, Mar. 2010. [Online]. Available: http://ftp.3gpp.org/Specs/html-info/36300.htm

[13] ——, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Policy and charging control architecture (Release 8)," 3rd Generation Partnership Project (3GPP), TS 23.203 v8.14.0, Jun. 2012. [Online]. Available: http://www.3gpp.org /DynaReport/23203.htm

[14] ETSI, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control Sub-System (RACS): Functional Architecture," European Telecommunications Standards Institute (ETSI), ES 282003 v3.5.1, Apr. 2011. [Online]. Available: http://pda.etsi.org/pda/queryform.asp

[15] K. Nichols, S. Blake, F. Baker, and D. Black, "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers," RFC 2474 (Proposed Standard), Internet Engineering Task Force, Dec. 1998, updated by RFCs 3168, 3260. [Online]. Available: http://www.ietf.org/rfc/rfc2474.txt

[16] ETSI, "Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part4," European Telecommunications Standards Institute (ETSI), ES 202765-4 v1.2.1, May 2014. [Online]. Available: http://pda.etsi.org/pda/queryform.asp

[17] ITU-T, "Objective perceptual multimedia video quality measurement in the presence of a full reference," International Telecommunication Union (Telecommunication Standardization Sector), REC J.247, Aug. 2008. [Online]. Available: http://www.itu.int/rec/T-REC-J.247-200808-I/en

[18] ETSI, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IPTV Architecture; IPTV functions supported by the IMS subsystem," European Telecommunications Standards Institute (ETSI), TS 182027 v3.5.1, Mar. 2011, Available: http://www.etsi.org/deliver/etsi_ts/182000_182099/182027/03.05.01_60/ts_182027v030501p.pdf [retrieved: May, 2015].

[19] ITU-T, "IPTV over NGN Guidelines for the use of traffic management mechanisms in support of IPTV services," International Telecommunication Union (Telecommunication Standardization Sector), REC Y.1920, Jul. 2012. [Online]. Available: http://www.itu.int/rec/T-REC-Y.1920-201207-I

[20] C. Jennings, A. Narayanan, D. Burnett, and A. Bergkvist, "WebRTC 1.0: Real-time communication between browsers," W3C, W3C Working Draft, Feb. 2015, http://www.w3.org/TR/2015/WD-webrtc-20150210/.

[21] IETF. Rtcweb status pages. [Online]. Available: http://tools.ietf.org/wg /rtcweb/ [retrieved: May, 2015]

[22] C. Perkins, M. Westerlund, and J. Ott, "Web Real-Time Communication (WebRTC): Media Transport and Use of RTP draft-ietf-rtcweb-rtp-usage-11," Internet-Draft, Internet Engineering Task Force, Dec. 2013, Available: http://tools.ietf.org/id/draft-ietf-rtcweb-rtp-usage-23.txt [retrieved: May, 2015].

[23] E. Rescorla and N. Modadugu, "Datagram Transport Layer Security Version 1.2," RFC 6347 (Proposed Standard), Internet Engineering Task Force, Jan. 2012, Available: http://www.ietf.org/rfc/rfc6347.txt [retrieved: May, 2015].

[24] E. Rescorla, "WebRTC Security Architecture draft-ietf-rtcweb-security-arch-07," Internet-Draft, Internet Engineering Task Force, Jul. 2013, Available: http://tools.ietf.org/id/draft-ietf-rtcweb-security-arch-11.txt [retrieved: May, 2015].

[25] J. Rosenberg, R. Mahy, P. Matthews, and D. Wing, "Session Traversal Utilities for NAT (STUN)," RFC 5389 (Proposed Standard), Internet Engineering Task Force, Oct. 2008, Available: http://www.ietf.org/rfc/rf c5389.txt [retrieved: May, 2015].

[26] J. Rosenberg, "Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal for Offer/Answer Protocols," RFC 5245 (Proposed Standard), Internet Engineering Task Force, Apr. 2010, updated by RFC 6336. Available: http://www.ietf.org /rfc/rfc5245.txt [retrieved: May, 2015].

[27] H. Alvestrand, "Overview: Real Time Protocols for Brower-based Applications draft-ietf-rtcweb-overview-08," Internet-Draft, Internet Engineering Task Force, Sep. 2013, Available: http://tools.ietf.org/id/draft-ietf-rtcweb-overview-13.txt [retrieved: May, 2015].

[28] "Web Real-Time Communication Use Cases and Requirements," RFC 7478 (Informational), Internet Engineering Task Force, Mar. 2015. [Online]. Available: http://www.ietf.org/rfc/rfc7478.txt

[29] S. Dhesikan, C. Jennings, D. Druta, P. Jones, and J. Polk, "DSCP and other packet markings for WebRTC QoS: draft-ietf-tsvwg-rtcweb-qos-03," Internet-Draft, Internet Engineering Task Force, Nov. 2014, Available: https://tools.ietf.org/html/draft-ietf-tsvwg-rtcweb-qos-03 [retrieved: May, 2015].

[30] International Organization for Standardization, "Information technology - Open Systems Interconnection - Basic Reference Model," International Organization for Standardization, REC ISO/IEC 7498-1:1994, Nov. 1994. [Online]. Available: http://www.iso.org

[31] ETSI, "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based IPTV stage 3 specification," European Telecommunications Standards Institute (ETSI), TS 183063 v3.5.2, Mar. 2011, Available: http://www.etsi.org/deliver/etsi_ts/183000_183099/183063/03.05.02_60/ts_183063v030502p.pdf [retrieved: May, 2015].

[32] M. Baugher, D. McGrew, M. Naslund, E. Carrara, and K. Norrman, "The Secure Real-time Transport Protocol (SRTP)," RFC 3711 (Proposed Standard), Internet Engineering Task Force, Mar. 2004, updated by RFC 5506. Available: http://www.ietf.org/rfc/rfc3711.txt [retrieved: May, 2015].

[33] Openimscore.org. Opensourceims. [Online]. Available: http://www.openimscore.org/ [retrieved: May, 2015]

[34] ip.access Ltd. ip.access: leaders in 2g, 3g and 4g end-to-end small cell solutions. [Online]. Available: http://www.ipaccess.com/ [retrieved: May, 2015]

[35] OpenEPC. Openepc.com. [Online]. Available: http://www.openepc.com/ [retrieved: May, 2015]

[36] T. Bach. sipsorcery-fork. [Online]. Available: https://github.com/hftl-ims-research/sipsorcery-fork [retrieved: May, 2015]

[37] inspired social. Open source ICE implementation. [Online]. Available: http://code.google.com/p/inspired-social/source/browse/trunk/StunServer/net/mc_cubed/icedjava?spec=svn20&r=20#icedjava [retrieved: May, 2015]

[38] Legion of the Bouncy Castle Inc. Bouncy Castlee Crypto API. [Online]. Available: http://www.bouncycastle.org [retrieved: May, 2015]

[39] steely gint. set of classes implementing a simple (S)RTP stack. [Online]. Available: https://github.com/steely-glint/srtplight [retrieved: May, 2015]

[40] FFmpeg: a open source cross-platform solution to record, convert and stream audio and video. [Online]. Available: http://www.ffmpeg.org/index.html [retrieved: May, 2015]

[41] "Diameter Network Access Server Application," RFC 7155 (Proposed Standard), Internet Engineering Task Force, Apr. 2014. [Online]. Available: http://www.ietf.org/rfc/rfc7155.txt

[42] 4GAmericas. 4G Americas' Recommendations on 5g Requirements and Solutions. [Online]. Available: http://www.4gamericas.org/en/resources/white-papers/ [retrieved: May, 2015]