The Impact of Control Information Prioritization on QoS Performance Metrics

Jasmina Baraković
BH Telecom, Join Stock Company
Sarajevo, Bosnia and Herzegovina
e-mail: jasmina.barakovic@bhtelecom.ba

Sabina Baraković
Ministry of Security of Bosnia and Herzegovina
Sarajevo, Bosnia and Herzegovina
e-mail: sabina.barakovic@msb.gov.ba

Himzo Bajrić
BH Telecom, Join Stock Company
Sarajevo, Bosnia and Herzegovina
e-mail: himzo.bajric@bhtelecom.ba

Abstract— **Real-time protocols such as Real-time Transport Protocol (RTP) and its companion protocol Real-time Transport Control Protocol (RTCP) are the solution for customers who have progressively come to desire real-time services. The characteristics of RTP and RTCP packets differ since they carry different content types, i.e., real-time data and control data, respectively. The standard approach assumes that RTP real-time data packets and RTCP control packets are classified into the same media-oriented service class. That might impact the Quality of Service (QoS) for real-time services with regard to network status. Therefore, this paper proposes an idea of QoS enhancement by efficiently transmitting user control information and establishing the proper transmission rate dynamically in dependence upon the status of a network. To accomplish this task, RTCP control packets are classified the Signaling service class, which should be given the absolute preferential treatment over all other User service classes. Results obtained with NCTUns simulator and emulator show a significant impact of control information prioritization on QoS performance metrics and indicate that this approach could be a new starting point for research activities in the future.**

Keywords-control information prioritization; QoS; RTCP; RTP

I. INTRODUCTION

Customers have progressively come to desire real-time services, and the solutions to their desires came in the form of real-time protocols. Using packet switching and the routing of Internet Protocol (IP) packets means that delays are introduced in the access and core networks [1], and hence a data acquired in real-time is transformed undesirably in the networks to non-real-time in the receiver. Real-time protocols are used to correct these delays and the degradation that is introduced during data processing and IP routing in the network. The goal of real-time protocols is to transport, control, and reassemble the information-bearing bits into real-time data for display at the receiver.

The real-time protocols include Real-time Transport Protocol (RTP) [2], which is an Internet Engineering Task Force (IETF) standard. The RTP offers data packet sequencing to enable correct ordering of packets at the receiver because packets can take different routes to their destination. It also provides identification of data source and type of payload, timing and synchronization, monitoring of delivery for diagnosing, reducing transmission problems, and sending the feedback of correct delivery and the quality of data transmission to the sending device. Furthermore, it integrates traffic sources of different types. This is used to merge heterogeneous traffic from multiple transmitting sources into a single flow. Because packets are time stamped at their source and destinations, delays encountered in transit can be estimated and unnecessary jitter can be removed to enable real-time display and enhance Quality of Service (QoS).

This protocol comes together with its companion protocol Real-time Transport Control Protocol (RTCP) [2]. Its function is to provide a means for reporting the performance of data transfer in the network. Roles of RTCP are to expose the states of the client/server to each other so that they understand and exchange the parameters of the communication, and report on the quality of communication. This protocol issues and transmits periodic control packets from participants to all other participants in a session. Two types of RTCP packets are exchanged by the RTCP: the Sender Reports (SRs) and the Receiver Reports (RRs). Fields in the report packets contain descriptions of the state of the session. Compound RTCP packets can be formed by concatenating several report packets and transmitting them as one packet. The QoS can be determined using the parameters in the reports provided by the RTCP [3]. The RTCP reports on the number of packets sent and received since the last report (throughput) and hence the number of lost packets. Because packets are time stamped, the RTCP also provides the round-trip delay, the state of the paths, and of course, the jitter associated with the packets.

These protocols, RTP and RTCP, are designed to be independent of the underlying transport and network layers. In its conventional implementation, RTP does not have a standard Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) port on which it communicates. The only standard that RTP obeys is that UDP communications are done via an even port and the next higher odd port is used for RTCP communications.

Since RTCP packets are sent using a different UDP source and destination port, it is not unlikely that the RTCP packets will receive a different treatment by the network. The standard approach requires that RTCP packets must be marked with the same DiffServ Code Point (DSCP) as the RTP packets in an effect to gain similar treatment from the network as that provided to the RTP packets. However, utilizing the same DSCP for the RTCP packets as that used for the RTP packet does not resolve all problems as follows.

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First, RTCP packets vary in size and are generally larger then RTP packets which effects their treatment by a network. Second, RTCP packets are sent at a rate as little as $1/500^{th}$ of the rate that RTP packets are sent which may also affect their treatment by the network. Third, resource reservations made to protect the RTP flows of packets are unlikely to be made to protect the RTCP flow; and if the reservation was made for the RTCP packets, it could fail, and/or be treated differently because of the vastly different traffic profiles. In summary, the QoS statistics determined by RTCP packets may be different than the actual QoS experienced by RTP packets carrying the actual media.

Accordingly, this paper proposes an idea of QoS enhancement by efficiently transmitting user control information and establishing the proper transmission rate dynamically in dependence upon the status of a network. This is attained through the classification of RTCP control packets into the Signaling service class rather than media-oriented service classes. According to our previous work [4], Signaling service class should be given the absolute preferential treatment over all other User service classes. From the RTCP perspective, it is beneficial to receive the absolute preferential treatment over RTP as it provides more accurate statistics for the measurements performed by RTCP, and thereby increase the QoS.

In the remainder of the paper, the related work on QoS evolution and signaling is discussed in Section II. The concept of control information prioritization is described in Section II. It also highlights the difference between the standard and novel approach in terms of control information prioritization. Section III considers the impact of standard and novel approach on QoS performance metrics using simulation methodology. It discusses the obtained results, together with their analysis to show that the conclusions are warranted. Section IV concludes this paper and outlines open issues for future work.

II. RELATED WORK

First studies proposing QoS frameworks for IP networks started to appear within IETF. To support QoS in IP networks, IETF proposed two frameworks. These are Integrated Services (IntServ) [5], based on connection-oriented resource reservation principle and Differentiated Services (DiffServ) [6], based on service differentiation approach. IntServ provides QoS to end hosts by reserving end-to-end resources using the Resource Reservation Protocol (RSVP) [7], when the end hosts signal their QoS needs. DiffServ obviates the need for a resource reservation protocol and offers the benefits of provisioning differentiated services. DiffServ is a starting point to guarantee QoS by providing different service classes, which are configured using specific combination of QoS mechanisms.

Although Multiprotocol Label Switching (MPLS) [8] is not considered as a QoS framework for IP networks, it provides a number of advantageous features to network operators. According to MPLS, data are transmitted along the so-called Label Switched Paths (LSP), which are established using either RSVP modification [9] or specifically developed Label Distribution Protocol (LDP) [10]. Modern QoS-aware networks such as DiffServ, MPLS or DiffServ/MPLS are specifically designed to be flexible enough to reallocate network resources in the best possible way, such that the required performance is provided using minimum amount of resources [11].

To be able to provide QoS when needed, it is necessary not only to deploy proper and effective QoS frameworks, but also to have means to signal to the network entities in charge to set-up QoS the desired level of service. In other words, it is necessary to build a framework that, interworking with a proper protocol, will signal QoS in an efficient and reliable way. The IETF Working Group Next Step in Signaling (NSIS) [12] has been chartered to address these issues, and define the framework for QoS signaling. Moreover, the Telecommunication Industry Association (TIA) has published a standard TIA-1039, QoS Signaling for IP QoS Support, which is involved in solutions to the problem of improving end-to-end QoS performance, e.g., Control for High-Throughput Adaptive Resilient Transport (CHART) system [13]. Another approach related to providing QoS under network congestion is described in [14].

In general, many QoS related projects show that there is an important background of work that aims at achieving QoS provisioning with the different tasks it involves [15]. Neither of referenced approaches discusses the possibility to enhance the QoS for real-time services by efficiently transmitting user control information. Our approach is based on an idea of prioritizing control information transmission in order to improve the QoS performance of real-time services.

III. CONTROL INFORMATION PRIORITIZATION

To accomplish the task of prioritizing control information transmission, DiffServ addresses the clear need for relatively simple and coarse methods of categorizing traffic into various service classes and applying QoS parameters to those classes. Different service classes are constructed using DSCPs, traffic conditioners, Per Hop Behaviors (PHBs), and Active Queue Management (AQM) mechanisms [16].

Though the IETF standards provided a consistent set of PHBs for services marked to specific DSCP values, they never specified which service should be marked to which DSCP value. Cisco led many confusions and disagreements over matching services with standards-defined code points in 2002 to put forward a standards-based marking recommendation in its strategic architectural QoS Baseline document [17]. Eleven service classes that could exist within the enterprise were examined and extensively profiled, then matched to their optimal RFC-defined PHBs. More than four years after Cisco put forward its QoS Baseline document, RFC-4594 was formally accepted as an informational RFC in 2006 [16]. An informational RFC is an industry-recommended best practice. This RFC puts forward twelve service classes and matches these to RFC-defined PHBs.

There are more than a few similarities between Cisco's QoS Baseline and RFC-4594, as there should be, since RFC-4594 is essentially an industry-accepted evolution of Cisco's QoS Baseline. However, there are some differences that merit attention. Cisco has completed a marking migration for Call Signaling from AF31 to CS3 (as per the original QoS Baseline). The most significant of the differences between Cisco's QoS Baseline and RFC-4594 is the RFC-4594 recommendation to mark Call Signaling to CS5. In summary, the Cisco modified version of RFC-4594 is very similar to RFC-4594, with the one exception of swapping Call Signaling marking and Broadcast Video [17].

Since RFC-4594 is to be viewed as industry best-practice recommendation, enterprises and service providers are encouraged to adopt this recommendation, with the aim of improving QoS consistency, compatibility, and interoperability. However, since it is a set of formal DiffServ QoS configuration best practices, and not a requisite standard, modifications can be made to these recommendations as required by specific needs and constraints. To meet the QoS requirements as defined in International Telecommunication Union – Telecommunication Sector (ITU-T) Recommendation Y.1541, we proposed a modification of these configuration guidelines with regard to Signaling service class [4].

The approach proposed in our previous work is based on configuring Signaling service class by using priority queuing system to give it absolute preferential treatment over all other User service classes. The priority queuing system is a combination of a set of queues and a scheduler that empties them in priority sequence [16]. When asked for a packet, the scheduler inspects the highest priority queue dedicated to Signaling service class and, if there is traffic present, returns a packet from that queue. Failing that, it inspects the next highest priority queue, and so on. A packet assigned to Signaling service class is marked with a new DSCP value, which should be requested from the Internet Assigned Numbers Authority (IANA). This DSCP value should be lower than one used to configure the Network Control service class and higher than one reserved for all User service classes defined in RFC-4594 and RFC-5865 [18].

Though this approach is signaling protocol independent, our previous work has discussed it in the context of Session Initiation Protocol (SIP) [4]. This work considers the previously proposed approach with regard to RTCP.

Since the RTCP packets simply signal information regarding the reception of the RTP packets, we propose to mark them with DSCP value associated to Signaling service class rather than media-oriented service classes. In this manner, it is possible to provide reliable and efficient transmission of user control information in order to increase the QoS by monitoring the media status, reporting on media quality, and taking any necessary corrective actions based on the media status. Thus, the main contribution of this paper lies in highlighting the fact that prioritizing control information transmission improves the QoS performance of real-time services.

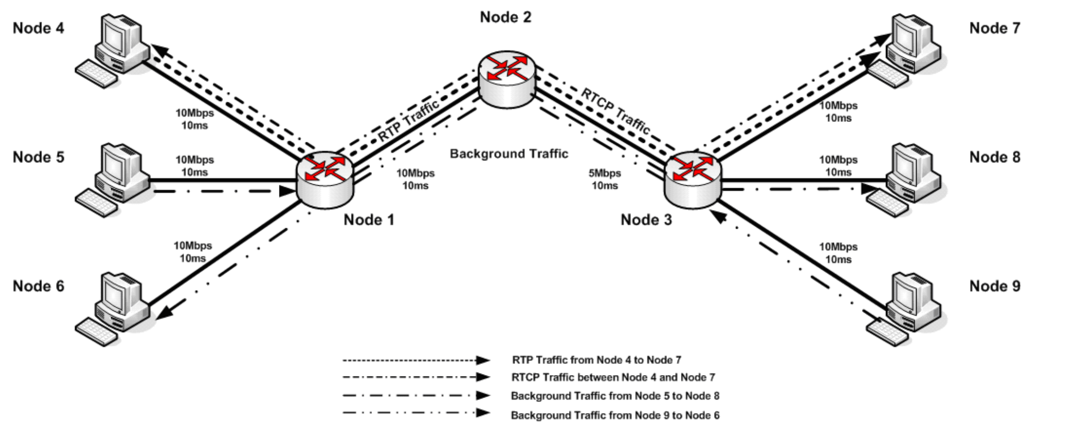
IV. Simulation Methodology

A. Simulation Setup and Environment

In order to investigate the impact of novel approach of control information prioritization on QoS performance metrics, the simulations are performed using NCTUns network simulator and emulator 6.0 [19], because it enables conduction of RTP/RTCP and DiffServ simulations.

The simulations are based on two different simulation scenarios performed on the simple network topology shown on Figure 1. The scenarios differ in control information prioritization. In Scenario 1, RTCP control packets are marked with the same DSCP as RTP packets and classified into the Multimedia Streaming service class. On the other hand, in Scenario 2, RTCP control packets are marked with higher DSCP then one used for RTP packets. The RTCP control packets are classified into Network Control class, since it is the only class in NCTUns that provides the absolute preferential treatment over all other User service classes, as Signaling service class should.

The single DiffServ domain, which constitutes the illustrated network topology, is composed of two boundary routers and one interior router. Boundary routers are responsible for classifying and conditioning traffic. Traffic classification is based on five-tuple (source IP address, destination IP address, source port number, destination port number, protocol). Several parameters are distinguished for this purpose as shown in Table I.



a. RTP (Real-time Transport Protocol); b. RTCP (Real-time Transport Control Protocol).

Figure 1. Network topology.

TABLE I. SIMULATION ENVIRONMENT DETAILS

Node Name	Node Type	Source IP[a] Address	Application	Port	Destination IP[a] Address
Node 1	Boundary Router	NA[b]	NA[b]	NA[b]	NA[b]
Node 2	Interior Router	NA[b]	NA[b]	NA[b]	NA[b]
Node 3	Boundary Router	NA[b]	NA[b]	NA[b]	NA[b]
Node 4	Sending Node	1.0.1.2	adapt_bw	5004, 5005	1.0.6.2
Node 5	Sending Node	1.0.2.2	stg	Default	1.0.7.2
Node 6	Receiving Node	1.0.3.2	rtg	Default	NA[b]
Node 7	Receiving Node	1.0.6.2	rtprecvonly	5004, 5005	NA[b]
Node 8	Receiving Node	1.0.7.2	rtg	Default	NA[b]
Node 9	Sending Node	1.0.8.2	stg	Default	1.0.3.2

a. IP (*Internet Protocol*); b. NA (*Not Applicable*).

Traffic conditioning is based on the token bucket scheme. Since traffic conditioning helps to prevent network congestion, it is set to be disabled in both simulation scenarios. Interior router is responsible for dispatching the incoming packets to the different service class queues for receiving different QoS treatments. The forwarding treatments (i.e., PHBs), which are used for the purpose of investigating the impact of control information prioritization on QoS performance metrics, are listed in Table II.

The links between network nodes are dimensioned to implement simple network configuration with one bottleneck link (Node 2 → Node 3). The configured link capacities, as well as propagation delays are shown in Figure 1. The delay of all links is 10 ms, and links capacity is 10 Mbps except link between Node 2 and Node 3. The bottleneck link capacity is 5 Mbps.

The network is loaded by two types of traffic flows that belong to different service classes. The Standard service class is intended for all background traffic that will receive normal (undifferentiated) forwarding treatment through the network. The Multimedia Streaming service class is used for transport of video traffic using RTP. The RTCP control packets use this service class in Scenario 1, and the Network Control service class in Scenario 2. The Network Control service class is primarily intended for routing and network control. Since this service class has preferential treatment over all other User service classes, it is considered as alternative to the Signaling service class in Scenario 2.

The traffic generators components are stg and rtg, respectively for sending and receiving background traffic. Using UDP greedy mode, stg application on Node 5 generates packets of 1400 bytes during 999 s, whereas stg application on Node 9 generates packets of 1024 bytes during 500 s. The RTP and RTCP traffic is generated using rtprecvonly and adapt_bw application programs, which are implemented on Node 7 and Node 4, respectively. Application rtprecvonly receives RTP and RTCP packets, sends RTCP packets, but does not send RTP packets. Application adapt_bw uses RTCP packets to report the received QoS at the receiver so that sender can dynamically adjust the sending rate of its RTP packets. The RTP packets are received on port 5004, whereas the RTCP packets are received on port 5005. The RTP is used to transport video traffic with characteristics listed in Table III.

The simulations are run for 500 simulations seconds. Due to nearly permanent characteristics of background traffic, this period can be considered sufficient. The RTP/RTCP session active time is from 5th to 500th simulation second. Start time and stop time for background traffic generator implemented on Node 5 is 150th and 350th simulation second, respectively. The activity of background traffic generator implemented on Node 9 is started at 0.5th simulation second and ended at 500th simulation second.

B. Simulation Results and Analysis

Obtained simulation results show significant impact of the novel approach on critical QoS performance metrics in comparison to standard approach, which is particularly obvious during the maximum network load. The standard approach involves marking both RTP and RTCP packets with same DSCP value (i.e., Scenario 1). The novel approach is based on marking RTCP packets with the higher DSCP value than one used for RTP packets (i.e., Scenario 2). The considered performance metrics, which are important for QoS of real-time services, include: throughput, RTP packet loss rate, cumulative number of lost RTP packets, Round-Trip Time (RTT) and jitter for RTP packets.

TABLE II. DISTINGUISHED QOS CONFIGURATION PARAMETERS

QoS[g] Configuration Setup in Boundary Router				
	Source IP[e] Address	***Source Port***	***DSCP[d]***	
			Scenario 1	***Scenario 2***
Rules	1.0.1.2	5004	AF[a]31	AF[a]31
	1.0.1.2	5005	AF[a]31	CS[b]6
	1.0.6.2	*	DF[c]	DF[c]
	1.0.8.2	*	DF[c]	DF[c]
QoS[g] Configuration Setup in Interior Router				
	PHB[f] Type	***Rate [Mbps]***	***Queue Length [packets]***	
Queues	DF[c]	7.5	10	
	AF[a]41	2	10	
	CS[b]6	0.5	10	

a. AF (*Assured Forwarding*); b. CS (*Class Service*); c. DF (*Default Forwarding*); d. DSCP (*DiffServ Code Points*); e. IP (*Internet Protocol*); f. PHB (*Per Hop Behavior*); g. QoS (*Quality of Service*).

TABLE III. VIDEO TRAFFIC CHARACTERISTICS

Encoding name	nv [20]
Sampling rate [Hz]	90000
Bits per sample	0.555555
Frame rate [fps]	30

As shown on Figure 2a, the novel approach based on classification of RTCP control packets into the Signaling service class, provides throughput in range from 5 kBps to 20 kBps during the interval of maximum network load. On the other hand, throughput obtained in the standard approach, in which RTP and RTCP packets are classified into the same media-oriented service class, does not exceed 10 kBps during the same period. Therefore, the novel approach results in higher throughput than the standard approach. This is a consequence of the control information prioritization, which makes establishing the proper transmission rate dynamically in dependence upon the status of a network.

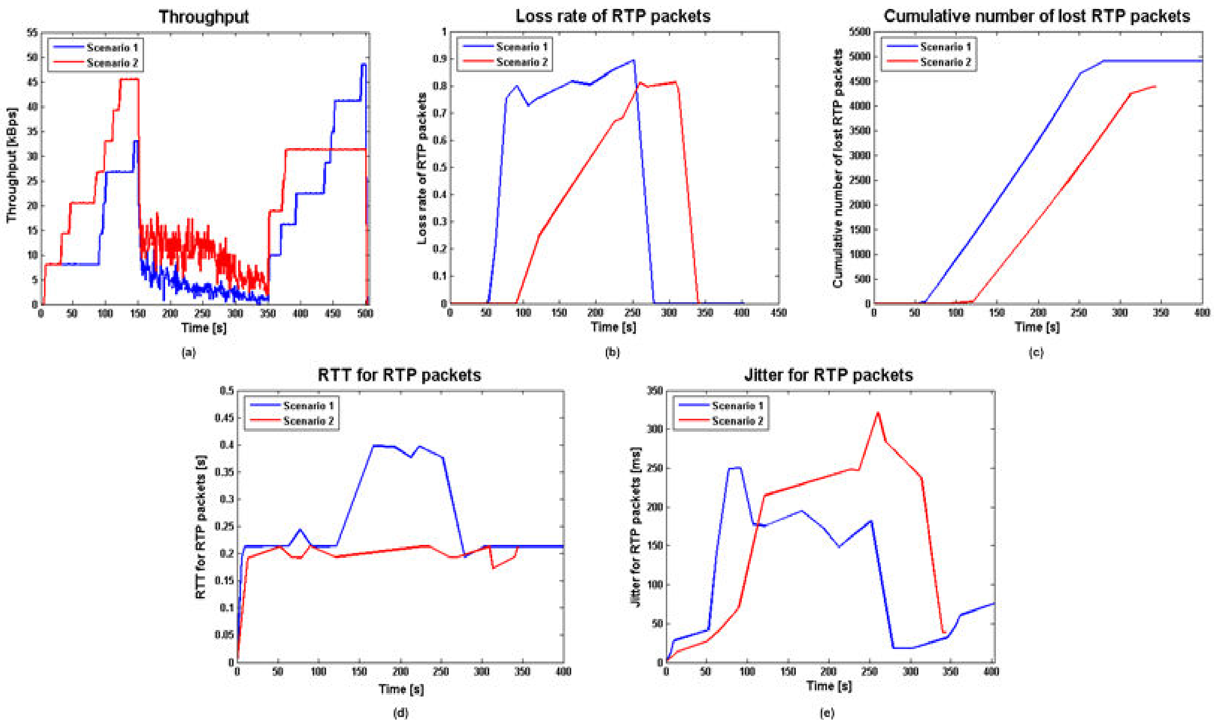
The RTP packet loss rate shown on Figure 2b implicates that the novel approach to the classification of RTCP packets performs better than standard one. In the novel approach, RTCP packets adjust the sending rate of RTP packets more dynamically than in the standard approach. Thereby, packet losses are prevented, which results in lower cumulative number of lost RTP packets (Figure 2c). Accordingly, cumulative number of lost RTP packets equals 4383 and 4892 packets in novel and standard approach, respectively.

Since the novel approach uses RTCP packets for more dynamic adjustment of the sending rate of RTP packets, it also prevents RTP packets to experience long delay caused by network status. Standard approach, on the other hand, does not adjust the sending rate as fast as the novel approach, since the QoS statistics determined by delayed RTCP packets may be different than the actual QoS experienced by RTP packets carrying the media. The lack of the standard approach in comparison with the novel one is shown on Figure 2d. It is noticed that the RTT of RTP packets in standard approach is higher than in novel one during the interval of maximum network load. Moreover, it is kept constant during the simulation, which results in lower jitter than in novel approach (Figure 2e).

V. Conclusion and Future Work

This paper proposes an idea of improving the QoS performances of real-time services by prioritizing user control information transmission and establishing the proper transmission rate dynamically in dependence upon the status of a network. To accomplish this task, the approach proposed in our previous work is used. According to that approach, Signaling service class is configured using priority queuing system to give it absolute preferential treatment over all other User service classes. Since the proposed approach is signaling protocol independent, this paper considers it with regard to RTCP. In this respect, RTCP packets are classified into the Signaling service class rather than media-oriented service classes in order to transmit the user control information reliably and efficiently.

Obtained results show the significant impact of control information prioritization on QoS performance particularly during the maximum network load. Prioritizing control information transmission provides more accurate statistics for the measurements performed by RTCP, and thereby increases the QoS.



a. RTP (Real-time Transport Protocol); b. RTT (Round-Trip Time).

Figure 2. Simulation results: (a) Throughput; (b) Loss rate of RTP packets; (c) Cumulative number of lost RTP packets; (d) RTT for RTP packets; (e) Jitter for RTP packets.

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However, the opportunity for the improvement of this approach may be identified. Encouraged by the positive results, the intention is to elaborate proposed idea and investigate it in detail. That could include the research activities on behavior of various types of multimedia real-time traffic under various network conditions. Additionally, the intention is to emulate the real-time traffic, in order to test the functions and performances of real-world devices and services, and observe how they would perform under various network conditions.

Also, since today's innovation and technology development depends on the users' satisfaction, it is necessary to investigate users' perceived quality of the real-time services with regard to this approach. Being linked to various investigation areas, this idea is hopefully going to become a new starting point for research activities in the future.

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