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Quantification of Quality of Experience for Edge-Based Applications

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Abstract

In future Internet, multi-network services correspond to a new paradigm that intelligence in network control is gradually moved to the edge of the network. As a consequence, the application itself can influence or determine the amount of consumed bandwidth. Thus the user behaviour may change dramatically. This impacts the Quality of Service (QoS) and the Quality of Experience (QoE), a subjective measure from the user perspective of the overall value of the provided service or application. A selfish user or application tries to maximize its own QoE rather than to optimize the network QoS, in contrast to a legacy altruistic user.

In this paper we present the IQX hypothesis which assumes an exponential functional relationship between QoE and QoS. This contribution is a first step towards the quantification of the QoE for edge-based applications, where an example of VoIP is taken into account. Starting from a measurement of the Skype application, we show the basic properties of selfish and altruistic user behaviour in accordance to edge-based intelligence. The QoE is quantified in terms of MOS in dependence of the packet loss of the end-to-end connection, whereby Skype's iLBC voice codec is used exemplarily. It is shown that the IQX hypothesis is verified in this application scenario. Furthermore, selfish user behaviour with replicated sending of voice datagrams is investigated with respect to the obtained QoE of a single user. In addition, the impact of this user behaviour on congestion in the network is outlined by means of simulations.

1 Introduction

In future telecommunication systems, we observe an increasing diversity of access networks and the fixed to mobile convergence (FMC) between wireline and wireless networks. This implies an increasingly heterogeneous networking environment for networked applications and services. The separation of transport services and applications or services leads to *multi-network services*, i.e., a future service has to work transparently to the underlying network infrastructure. For such multi-network services, the Internet Protocol is the smallest common denominator. Still, roaming users expect theses services to work in a satisfactory way regardless of the current access technology such as WLAN, UMTS, WiMAX, etc. Thus, a true multi-network service must be able to adapt itself to its "surroundings" to a much stronger degree than what is supported by the TCP/IP protocol suite.

Streaming multimedia applications for example face the problem that their predominant transport protocol UDP does not take any feedback from the network into account. Consequently, any quality control and adaptation has to be applied by the application itself at the edge of the network. Prominent examples of *edge-based applications* applying edge-to-edge control are peer-to-peer (P2P) applications such as eDonkey or BitTorrent, Skype VoIP, YouTube, etc. The network providers have to cope with the fact that these edge-based applications dynamically determine the amount of consumed bandwidth. In particular, applications such as Skype do their own network quality measurements and react to quality changes in order to keep their users satisfied. The edge-based intelligence is established via traffic control on application layer. Traffic engineering in future Internet has to consider this new paradigm.

The shift of the control intelligence to the edge is accompanied with the fact that the observed user behaviour changes. A user can appear altruistic or selfish. Selfish user behaviour means that the user or the application tries to maximize the user-perceived *Quality of Experience QoE* rather than to optimize the network *Quality of Service QoS*. Very often the selfish behaviour is implemented in the software downloaded by the user without his explicit notice. In contrast, altruistic users, whose behaviour is instructed by network provider traffic control protocols (like TCP) help to maximize the overall system performance in a fair manner. In the case of file-sharing platforms, an altruistic user is willing to upload data to other users, while a selfish user only wants to download without contributing to the network. For voice over IP (VoIP), altruistic users would reduce the consumed bandwidth in the case of facing congestion, while selfish users would continuously try to achieve a high goodput and QoE, no matter of consequences for other users.

User satisfaction with application and service performance in communication networks has attracted increased attention during the recent years. The notion of QoE was introduced in several white papers [1, 2, 3, 4], mostly in the context of multimedia delivery like IPTV. Besides of objective end-to-end QoS parameters, QoE focuses on subjective valuations of service delivery by the end users. It addresses *(a)* service reliability comprising service availability, accessibility, access time and continuity, and *(b)* service comfort comprising session quality, ease of use and level of support [2]. The necessity of introducing QoE can be explained on the example of VoIP. A voice user is not interested in knowing performance measures like packet loss or received throughput, but mainly in the experienced speech quality and timeliness of the connection.

There is however a lack of quantitative descriptions or exact definitions of QoE. One particular difficulty consists in matching subjective quality perception to objective, measurable QoS parameters. Subjective quality is amongst others expressed through *Mean Opinion Scores* (MOS) [5]. Links between MOS and QoS parameters exist predominately for packetised voice such as VoIP. Numerous studies have performed measurements to quantify the effect of individual impairments on the speech quality to a single MOS value for different codecs, for example G.729 [6], GSM-FR [7], iLBC used by Skype [8], or a comparison of some codecs [9]. Additionally, the E-model [10] and extensions [11] exist that assess the combined effects of different influence factors on the voice quality. In [12], the logarithmic function is selected as generic function for mapping the QoE from a single parameter because of its mathematical characteristics.

This work, in contrast, motivates a fundamental relationship between the QoE and quality impairment factors such as packet loss and related jitter. An e**x**ponential solution is derived for the **I**nterdependency of **Q**oE and **Q**oS hypothesis, referred to as IQX . This contribution is a first step towards the quantification of the QoE for edge-based applications, where an example of VoIP is taken into account.

The rest of this paper is organized as follows. Section 2 introduces multi-network services and the emerging of edge-based intelligence. Starting from a measurement of the Skype application, we show the basic properties of selfish and altruistic user behaviour due to edge-based intelligence in Section 3. This is realized among others by an adaptive bandwidth control triggered by QoE. Section 4 starts with the quantification of the QoE of a VoIP application. We discuss the IQX hypothesis and the exponential functional relationship between QoE and QoS. It is exemplarily verified in Section 4.1 in terms of MOS depending on the packet loss of the end-to-end connection, whereby the iLBC codec as used by Skype is taken. We assume that the selfish users of the VoIP application utilize replication of voice datagrams to maximize their QoE, while the altruistic users change to a codec with a lower quality to consume less bandwidth. As a result, the benefit of the replication is investigated from a single user's point of view in Section 4.2. The impact of this selfish user behaviour on the network congestion is briefly illustrated in Section 4.3. Finally, Section 5 summarizes this paper.

2 Edge-Based Intelligence and Quality of Experience

From traffic engineering viewpoint, the shift of intelligence to the edge is accompanied by a number of changes:

- Change of user behaviour and traffic profile: edge-based services (like Skype) perform QoS measurements itself and adapt the traffic process according to the perceived QoS (packet blocking probability or jitter). The traffic change of those applications could be quite selfish, i.e. it tries to maximize its own QoE no matter of the network overload condition.

- Change from Multi-service Networks to Multi-Networks Services: An edge-based application could use many networks with different technologies in parallel, raising the question which network has to maintain which portion of the agreed QoS. From this perspective, the QoE will be the major criterion for the subscriber of a service.

- Higher Dynamic of Network Topology: an edge-based application is often controlled by an overlay network, which can change rapidly in size and structure as new nodes can leave or join the overlay network in an distributed manner.

Multi-network services will be often customer originated services. Together with the edge-based intelligence, the change of bandwidth demand and consumption is observed which only depends on the user behaviour and the used software of that service. The bandwidth demand is no longer under control of the network provider. A good example for this paradigm change is illustrated by the huge amount of traffic for P2P file-sharing [13] compared to web traffic.

However, the multi-network service has to maintain a certain QoE for each user. As a consequence, the edge-based application is responsible

(a) to evaluate the QoE at the end user's site and

(b) to react properly on the performance degradation, i.e., that the application adapts itself to the current network situation to maintain the QoE.

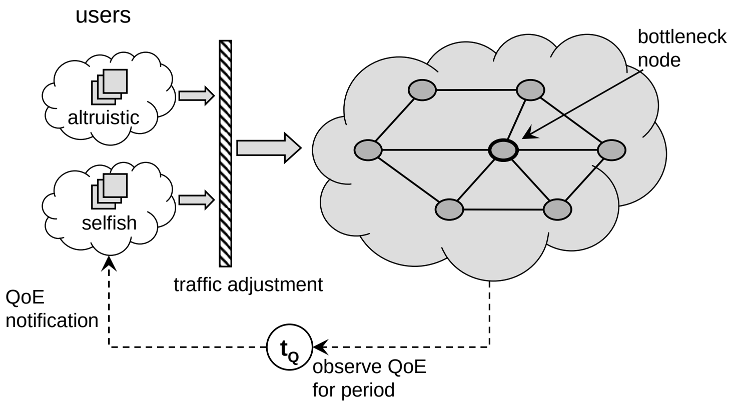


Figure 1: Quality assessment mechanisms for QoE

Figure 1 illustrates the QoE control scheme of such a multi-network service. Users are connected to each other via the corresponding access technologies. The QoE is assessed during a period t_Q of time. Accordingly, the altruistic users and the selfish users react on feedback obtained from measurements. In this paper we observe the Skype VoIP service in more detail as an example for a service with edge-based intelligence. This example shows the change in user behaviour and bandwidth demand and discusses the QoE adaptation scheme, i.e. the way Skype reacts to keep the QoE.

3 Measurement of Skype VoIP

Skype is a proprietary VoIP application which is based on P2P technology. It offers rapid access to a large base of users, seamless service operation across different types of networks (wireline and wireless) with an acceptable voice quality [8], and a distributed and cost-efficient operation of a new service. The voice quality of the Skype service is achieved by using appropriate voice codecs, such as iSAC and iLBC [14], and by adapting the sender traffic rate according to the current packet loss and jitter of the end-to-end connection. The latter one is referred to as *QoE adaptation* in the following.

This QoE adaptation can be illustrated by a measurement study presented [15]. The general measurement setup is the following: Skype user A sends audio data to Skype user B. We used an English spoken text without noise of length 51 seconds, a sample rate of 8 kHz, encoded with 16 bits per sample which is a standard audio file for evaluating VoIP and available at [16]. The wav-file is played in a loop with a pause of 9 seconds in between using the Winamp audio player on machine A. The output of Winamp is used as input for Skype (instead of a microphone). On sender A and receiver B, Windows XP is the OS, Skype 2.0.0.81 (February, 2006) is installed and a packet trace is captured with TCPDump on each machine. In order to emulate various network conditions on the link between machine A and machine B, we use the Nistnet software [17]. Nistnet is installed on a separate machine with three network interfaces and operates as gateway for A and B and to the Internet, cf. Figure 2. With this measurement setup, both Skype user A and B have access to the Internet (which is required for using this service), while packet loss is only emulated on the direct connection from A to B. Here, Skype encodes audio with the iSAC codec due to the used hardware. If the power of the machines is below 600 MHz, Skype

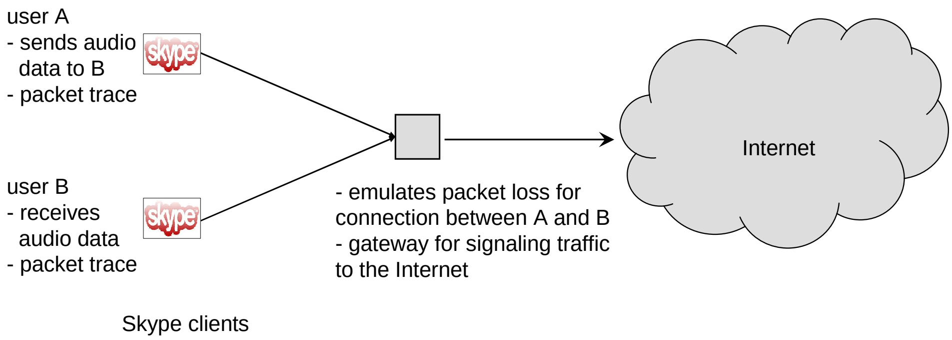


Figure 2: Measurement setup for a Skype call

will use the iLBC codec.

Figure 3 shows the reaction of the Skype software on packet loss. Every 30 ms, a packet is sent from user A to user B (with a measured standard deviation of 6.65 ms). The measured packet loss ratio on the right y-axis denotes how many packet got lost, whereby we used the average for a window size of 6 s. On the left y-axis, the average size of the voice packets on application layer is plotted in bit. Again, we used a window size of 6 s corresponding to 200 voice packets. First the Skype call is established between user A and B and we start with no packet loss. The size of a packet varies between 90 bit and 190 bit with a measured average of 150 bit. It has to be noted that the oscillations of the packet size derive from the measurement setup. During the pause interval, Skype sends still packets, but only with a size of 50 Bit.

After 5 minutes the packet loss probability is increased about 5% every two minutes, until the packet loss probability reaches 30%. The time interval of two minutes was chosen to ensure that Skype reacts to changes. We have found out that Skype needs about one minute to change e.g. a voice codec. As we can see in Figure 3, Skype reacts on the experienced QoE degradation in terms of packet loss by increasing the packet size, whereas still every 30 ms a packet is sent. The size mainly ranges between 240 bit and 320 bit with an average of 280 bit. In contrast to before, the packet size is nearly doubled. This means that Skype sends now redundant information within every voice packets while experiencing packet loss in order to maintain the QoE.

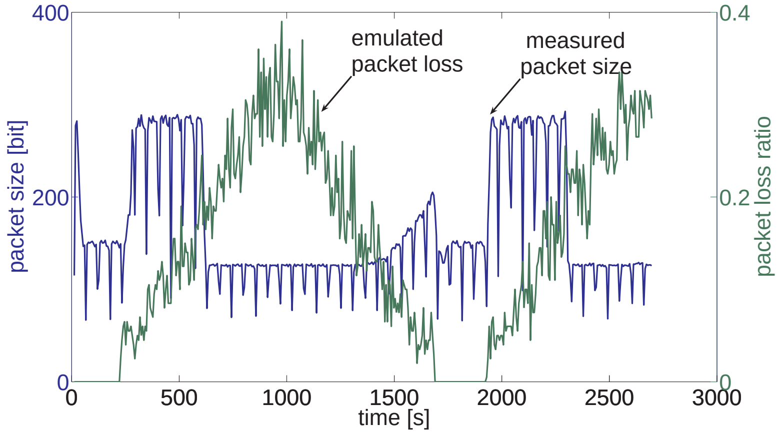


Figure 3: Measurement of Skype's QoE adaptation on changes in the end-to-end link

However, as a certain threshold is exceeded (here: about 20% packet loss), the packet size is decreased again and with 125 bit on average smaller than in the beginning. This indicates a change in the used voice codec. As soon as the packet loss probability is decreased again and falls below a certain threshold, the sender rate is again adapted by changing the packet size. In [15], we have also shown that Skype even does rerouting on application layer if the packet loss or the round trip time on the direct end-to-end connection is too high.

This measurement points out that edge-based applications try in fact to keep the QoE above an acceptable threshold. In the case of Skype, this is done by adapting the amount of consumed bandwidth. If the receiver's application detects packet loss, it instructs the sender to increase the bandwidth. For a VoIP call, this is easily possible, since the connection is full duplex and the connection from user B to user A is used to send the feedback information. Here, a change of the bandwidth consumption and the user behaviour is observed. A user – or to be more precise, the application – behaves selfish to get the maximum QoE, irrespective of the network overload condition. This observation was the starting point for this study aiming at the estimation of the QoE.

4 Quantitative Observation of QoE

In this section we focus on a fundamental relationship between the QoE and quality impairment factors, like packet loss or jitter. As an analytical solution of the relationship between QoE and loss, we formulate the IQX hypothesis (exponential interdependency of QoE and QoS) in Section 4.1. A first verification of this hypothesis is done using real measurement of the iLBC codec. Regarding the single user's point of view, the benefit of replicating voice datagrams is analytically derived with respect to the QoE in Section 4.2. The costs for this achievement are a higher amount of consumed bandwidth and the risk of worsening potential network congestion. In Section 4.3, the impact of selfish and altruistic behaviour on the network itself is discussed.

4.1 The IQX Hypothesis for Quality of Experience

We use as example in the following the *Internet low bitrate codec iLBC* [18], which is a free speech codec for VoIP and is designed for narrow band speech. Two basic frame lengths are supported: (a) 304 bit each 20 ms, yielding 15.2 kbps, and (b) 400 bit each 30 ms, yielding 13.3 kbps, respectively. The latter is used in Skype when the CPU of the used machines is below 600 MHz [8].

We performed a measurement series in which the iLBC codec (b) is explicitly used. However, with a probability p_{loss} a packet gets lost on its way from user A to user B. We vary the packet loss probability from 0% to 90% in steps of 0.9%. The audio data as described in Section 2 is used as input speech file. At the receiver side, the audio stream is piped into an audio wav-file. Each experiment is repeated ten times, i.e. 1010 measurements were conducted.

In order to express the QoE of the VoIP call, the *Mean Opinion Score MOS* [5] is used. Therefore, the audio file sent is compared with the received wav-file using the Perceptual Evaluation of Speech Quality (PESQ) method described in ITU-T P.862 [19]. The resulting PESQ value can be mapped into a subjective MOS value according to ITU-T Recommendation ITU-T P.862.1 [20]. Figure 4 shows the obtained MOS values in dependence of the packet loss probability

p_{loss} for the conducted experiments. The MOS can take the following values: (1) bad; (2) poor; (3) fair; (4) good; (5) excellent. Obviously, the higher the packet loss probability, the lower the MOS value is.

In general, the QoE is a function of n influence factors $I_j, 1 \leq j \leq n$:

$$QoE = \Phi(I_1, I_2, \cdots, I_n) \ . \tag{1}$$

However, in this contribution we focus on one factor indicating the QoS, the packet loss probability p_{loss}, in order to motivate the fundamental relationship between the QoE and an impairment factor corresponding to the QoS. Hence, the idea is to derive the functional relationship $QoE = f(p_{loss})$. In general, the subjective sensibility of the QoE is the more sensitive, the higher this experienced quality is. If the QoE is very high, a small disruption will decrease strongly the QoE, also stated in [12]. On the other hand, if the QoE is already low, a further disturbance is not perceived significantly. This relationship can be motivated when we compare with a restaurant quality of experience. If we dined in a five-star restaurant, a single spot on the clean white table cloth strongly disturbs the atmosphere. The same incident appears much less severe in a beer tavern.

On this background, we assume that the change of QoE depends on the current level of QoE – the expectation level – given the same amount of change of the QoS value. Mathematically, this relationship can be expressed in the following way. The performance degradation of the QoE due to packet loss is $\frac{\partial QoE}{\partial p_{loss}}$. Assuming a linear dependence on the QoE level, we arrive at the following differential equation:

$$\frac{\partial QoE}{\partial p_{loss}} = -\tilde{\beta} \cdot (QoE - \gamma) \ . \tag{2}$$

The solution for this equation is easily found as an exponential function, which expresses the basic relation of the IQX hypothesis:

$$QoE = \alpha \cdot e^{-\beta \cdot p_{loss}} + \gamma \ . \tag{3}$$

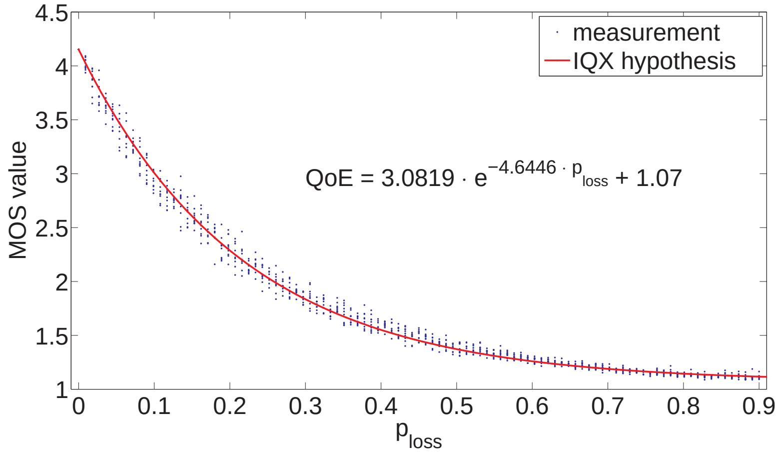


Figure 4: Exponential estimation of QoE in dependence of packet loss probability p_{loss}

For $p_{loss} \to 1$, the QoE in terms of MOS approaches its minimum of 1 from above. From the measured data, we obtain the following fit for iLBC voice codec (400 bits each 30 ms), following the IQX hypothesis:

$$QoE = 3.0829 \cdot e^{-4.6446 \cdot p_{loss}} + 1.07 \; . \tag{4}$$

It has to be noted that the packet loss is only one impairment factor indicating the QoS. For a general quantification of the QoE, additional factors like jitter have to be considered according to Eq. (1), which will be part of future work. Nevertheless, Eq. (4) will be used in the following section to derive analytically the impact of replication of voice datagrams on the QoE.

4.2 Impact of Replication of Voice Datagrams on QoE

Based on the experiences with Skype, we propose as one possibility the replication of voice datagrams to overcome a QoE degradation due to packet loss. Again, we consider the iLBC voice codec, as introduced in Section 4.1. This means that every $\Delta t = 30$ ms, a voice datagram of size $s_{voice} = 400$ bits is sent. A *replication degree* R means that the voice datagram is additionally sent in the following $R-1$ packets. As a consequence, a packet contains now R voice datagrams with a total packet size of $s_{packet} = s_{header} + R \cdot s_{voice}$. The variable s_{header} denotes the overhead for each packet caused by TCP and IP headers (20 Byte + 20 Byte) and on link layer (e.g. 14 Byte for Ethernet). Hence, the required bandwidth is a linear function in R: $C_{req} = \frac{s_{header} + R \cdot s_{voice}}{\Delta t}$. The gain of this bandwidth consumption is the reduction of the effective voice datagram loss probability $1 - p_{voice}$. For a given packet loss probability p_{loss} and a replication degree R, a voice datagram only gets lost if all R consecutive packets containing this voice datagram get lost. Thus, it holds

$$p_{voice} = 1 - p_{loss}^{R} \; . \tag{5}$$

The effect of the voice datagram replication can be seen in Figure 5 for a replication degree of $R = 1, \cdots, 6$. On the x-axis the packet loss probability p_{loss} is denoted. The QoE on the y-axis is computed according to Eq. (4) whereby the voice datagram probability in Eq. (5) is used. For

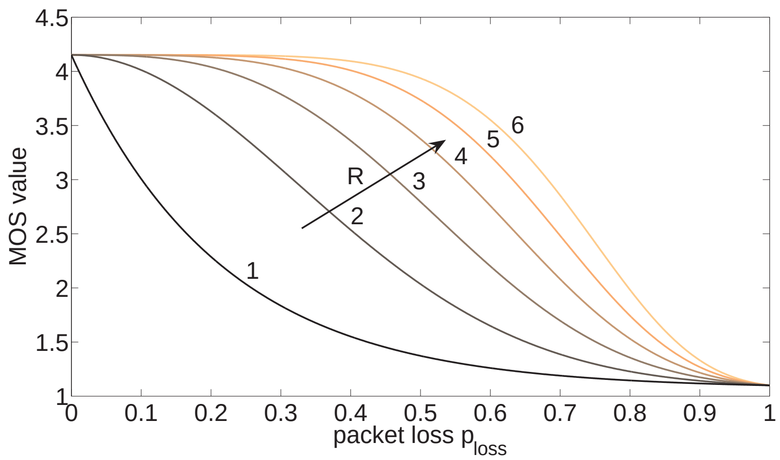


Figure 5: QoE in dependence on the replication degree (w/o jitter)

$p_{loss} = 0.2$, the QoE is only 2.29 for $R = 1$. A replication degree of $R = 2$ and $R = 3$ leads to a QoE of 3.63 and 4.04, respectively. This means the QoE could be improved from a poor quality to a good quality. A further increase of the replication degree only yields to a small gain as compared to the growth of the required bandwidth C_{req}.

Besides the increased bandwidth consumption, the replication also causes some jitter, as the voice datagrams are not received every $\Delta t = 30$ ms, but maybe in one of the $R-1$ following packets. Next, we compute the probability $\tilde{y}(i)$ that a voice datagram is successfully transmitted in the i-th try, used to quantify the jitter.

$$\tilde{y}(i) = p_{loss}^{i-1} \cdot (1 - p_{loss}) \tag{6}$$

The probability that a voice packet is received follows as

$$p_{voice} = \sum_{i=1}^{R} \tilde{y}(i) = (1 - p_{loss}) + p_{loss}(1 - p_{loss}) + \cdots + p_{loss}^{R-1}(1 - p_{loss}), \tag{7}$$

which agrees with Eq. (5). The number Y of trials which is required to successfully transmit a voice datagram is a conditional random variable. It follows a shifted geometric distribution and is defined for $1 \leq i \leq R$:

$$Y \sim \frac{\mathrm{GEOM}_1(p_{loss})}{p_{voice}} \quad \text{with} \quad y(i) = \frac{\tilde{y}(i)}{p_{voice}} = \frac{p_{loss}^{i-1} \cdot (1 - p_{loss})}{1 - p_{loss}^{R}}. \tag{8}$$

We define the jitter σ to be the standard deviation $\sqrt{\mathrm{Var}[t_{rcvd}]}$ of the interarrival time of received packets, normalized by the average time Δt between any two sent packets, $\sigma = \sqrt{\mathrm{Var}[t_{rcvd}]}/\Delta t$. For the sake of simplicity, we assume a deterministic inter packet sent time Δt and a deterministic delay $t_{s \to r}$ from the sender to the receiver. Then, the jitter can – after some algebraic

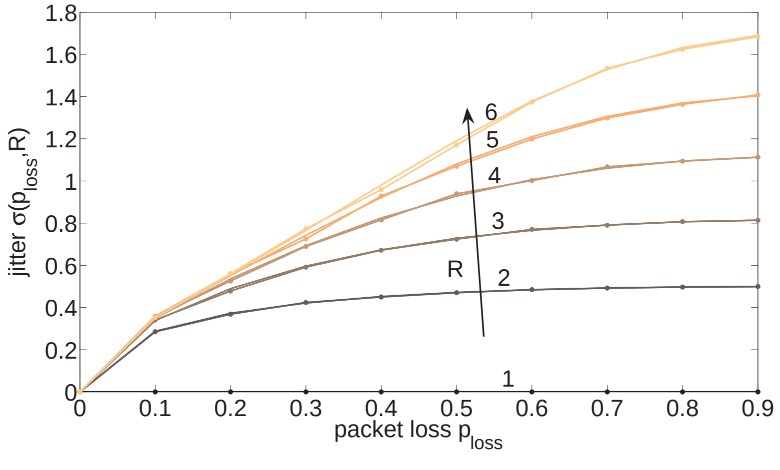


Figure 6: Increase of jitter due to replication of voice datagrams

transformations – be expressed as

$$\begin{aligned}\sigma &= \frac{\sqrt{\mathrm{E}[t_{rcvd}^2] - \mathrm{E}[t_{rcvd}]^2}}{\Delta t} = \frac{\sqrt{\mathrm{E}[(Y\Delta t)^2] - \mathrm{E}[Y\Delta t]^2}}{\Delta t} = \sqrt{\mathrm{E}[Y^2] - \mathrm{E}[Y]^2} \\ &= \sqrt{\frac{p_{loss}}{(p_{loss}-1)^2} - \frac{{p_{loss}}^R \cdot R^2}{({p_{loss}}^R - 1)^2}} \,. \end{aligned} \tag{9}$$

Figure 6 shows the jitter σ for a replication degrees $1 \leq R \leq 6$ in dependence of the packet loss probability p_{loss}. Eq. (9) is an exact formula, which we also validated by implementing a simulation. The solid lines correspond to the analytical calculation of the jitter, while the solid lines with the dots as marker show the simulation results. Both curves agree and the confidence intervals are too small to be visible.

The cost of the voice datagram replication – beside the increased bandwidth consumption – is an increased jitter. But jitter also impacts the QoE and is of course one impairment factor in Eq. (1). As a result, a maximal degree R_{max} of replication exists and a further increase does not improve the QoE anymore. ITU-T G.114 recommends a latency of the end-to-end delay of 150 ms, referred to as toll quality, and a maximum tolerable latency of 400 ms. According to the end-to-end delay $t_{s\to r}$ and the inter packet sent time $\Delta t = 30$ ms, the following inequation has to hold

$$R \cdot \Delta t + t_{s\to r} < t_{max} \tag{10}$$

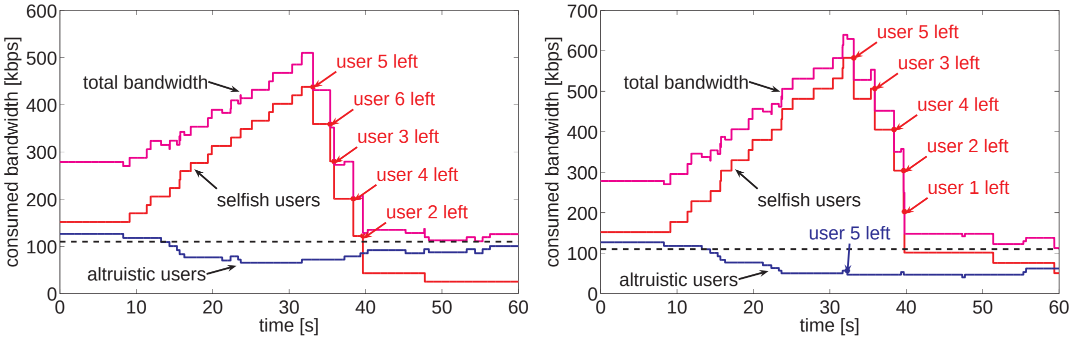
for a maximum allowed latency t_{max}. For example with $t_{max} = 200$ ms and $t_{s\to r} = 10$ ms, the maximum replication degree is limited by $R_{max} \leq 6$.

4.3 Network's Perspective for Edge-Based QoE Management

From the single user's point of view, the replication of voice data overcomes the degradation of packet loss and enables to keep a certain QoE. The cost for this achievement is a higher amount of consumed bandwidth. However, if the packet loss is caused by congestion in the network, this additionally required bandwidth worsens the network situation. We consider selfish and altruistic users which react on the perceived QoE. A single user measures the QoE during a period t_Q, the so called *QoE assessment period*. After each period t_Q, the user reacts on the obtained QoE value and adjusts the amount of consumed bandwidth, as illustrated in Figure 1. If the QoE is too low over some time, the user drops the call.

On one hand, the pure selfish user only looks on its own QoE which it tries to maximize by adjusting the throughput. This can be achieved *a)* by increasing the packet size by the replication degree R or *b)* by increasing the frequency of sending packets to $\frac{R}{\Delta t}$. On the other hand the altruistic user tries to minimize congestion in the network, i.e. the packet loss probability, in order to get a good QoE. Therefore, she uses a low-quality voice codec if packet loss, i.e. congestion, is detected.

In Figure 7, the consumed bandwidth over time of all altruistic and selfish users is considered in a congested system in which a bottleneck node of 110 kbps has to carry the traffic from six selfish and five altruistic users. While the altruistic users reduce their packet size, the selfish



(a) Increasing the packet size

(b) Increasing frequency of sending packets

Figure 7: Voice datagram replication to achieve a bit rate of sent audio information

users increase the throughput. As a consequence, packets get dropped, the QoE decreases, and the users give up after some time.

In practice, however, we do not observe or at least expect that the selfish users will lose. First of all, an edge-based application would react more sensitive than discussed in this section. An important point is how the QoE is monitored and what are the optimal thresholds to react. In addition, there is different traffic traveling through the bottleneck. TCP traffic, e.g., will be pushed away by UDP traffic. In that case, the entire system behaviour will be changed. These aspects will be considered in future work.

5 Conclusions

Multi-network services with edge-based intelligence, like P2P file-sharing or the Skype VoIP service, impose a new control paradigm on future Internet: They adapt the amount of consumed bandwidth to reach different goals. A selfish behaviour tries to keep the Quality of Experience (QoE) of a single user above a certain threshold. Skype, for instance, repeats voice samples in view of end-to-end-perceived loss, which increases the consumed bandwidth. Altruistic behaviour, on the other side, would reduce the bandwidth consumption in such a case in order to release the pressure on the network and thus to optimize the overall network performance.

In order to study such behaviour, we first focus on the quantification of the QoE for edge-based applications as a function of network Quality of Service (QoS), where an example of VoIP is taken into account. The QoE is quantified in terms of MOS in dependence of the packet loss of the end-to-end connection, whereby the iLBC voice codec is used exemplarily. The IQX hypothesis (interdependency of QoE and QoS) is proposed and verified for packet loss as a QoS indicator. IQX assumes an exponential functional relationship between QoE and QoS: $QoE = \alpha \cdot e^{-\beta \cdot p_{loss}} + \gamma$.

The impact of the bandwidth adaptation on the QoE of a single user is then quantified. We consider a selfish user which replicates voice datagrams to overcome packet loss. The gain of this increased bandwidth consumption is the reduction of the effective voice datagram loss probability. The cost of the replication – beside the increased bandwidth consumption – is an increased jitter. The jitter also impacts the QoE. As a result, a maximal degree of replication can be derived up to which an increase of the QoE can be achieved. However, if the packet loss is caused by congestion in the network, this additionally required bandwidth worsens the network situation. Thus, we illustrated the impact of selfish and altruistic behaviour on the network itself by means of simulations. Summarizing, the emergence of edge-based applications and the resulting user behaviour open a new scientific field with a lot of challenges to be solved.

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