PRELOADING TIME OF MULTIMEDIA STREAM AS KEY PERFORMANCE INDICATOR FOR CELLULAR NETWORKS

Radko Krkoš

Doctoral Degree Programme (2), FEEC BUT

E-mail: krkos@phd.feec.vutbr.cz

Supervised by: Vít Novotný

E-mail: novotnyv@feec.vutbr.cz

Abstract: The aim of this paper is a proposal of preloading time as a key performance indicator for assessment of user perceived performance of streamed multimedia playback in cellular networks. The general formula is presented and special cases are explored. Implications of this parameter are considered and later analysed. A Q coefficient of capacity allocation optimality is introduced.

Keywords: Q coefficient, streaming capacity allocation, buffer size dimensioning

1. INTRODUCTION

Cellular networks are ubiquitous today and therefore, they are commonly used by customers for consuming information. According to [1], 4G networks are such a technological step, that they enable users to switch from static content to multimedia, if permitted by monthly allowance plan. Streamed multimedia is easier to use and more effective for users, but also much more bandwidth intensive. The change in customer behaviour presents a significant caveat for mobile operators, because they need an apparatus to measure, express and assess the performance of this service with increasing importance.

An established way for assessing performance in cellular networks is using a set of well-known Key Performance Indicators (KPIs). These KPIs are defined directly by 3GPP, the initiative that specifies cellular network standards. An issue with the KPIs, as defined in [2] and [3] is that they are too low level to be either: a) understandable by customers; b) useful for performance assessment of a specific service; but they can be used as input variables for higher level KPIs.

2. MULTIMEDIA STREAMING

When dealing with multimedia streaming in contrast to text and pictures download, it is required to look at an extended set of parameters that affect the user experience. In addition to available bandwidth, also the source bit rate, playing time, transmission delay and jitter are of great importance.

2.1. AVAILABLE BANDWIDTH

Available bandwidth represents the amount of data that can be transmitted per time quantum and directly impacts the time that users have to wait for a particular service; it is commonly measured in bits per second. In cellular networks, available bandwidth is highly dependent on many factors, such as local network load, number of cells in an area and position in the serving cell. As such, it can take values from a wide range.

2.2. SOURCE BIT RATE

The source bit rate is the amount of data required to represent one second of encoded source stream, it is measured in bits per second. Source bit rate is entirely different for two types of multimedia streams that exist; audio and video. While for audio streaming, bit rates as low as 128kbps are used, for video streaming, especially with the advent of Full HD mobile terminals, the bit rates are an order of magnitude higher [4].

2.3. PLAYING TIME

Playing time, length of stream in seconds of real time consumption, is for most viewed videos due to the viewing environment and use case short. According to [1], it is under 5 minutes, from [5], it can be seen that average length of video linked from news or blog sites is 252 seconds.

2.4. DELAY AND JITTER

Delay is the time between the packets of stream leaving the source and arriving at the destination, it is measured in seconds, and jitter is the variation of delay for individual packets, also measured in seconds. A significant delay and jitter are prevalent in all packet switched networks and 4G cellular networks are no exception. Delay and jitter have to be part of computation, but only as an additive component. Statistical analysis of delay and jitter in cellular networks was presented in [6].

3. ANALYSIS

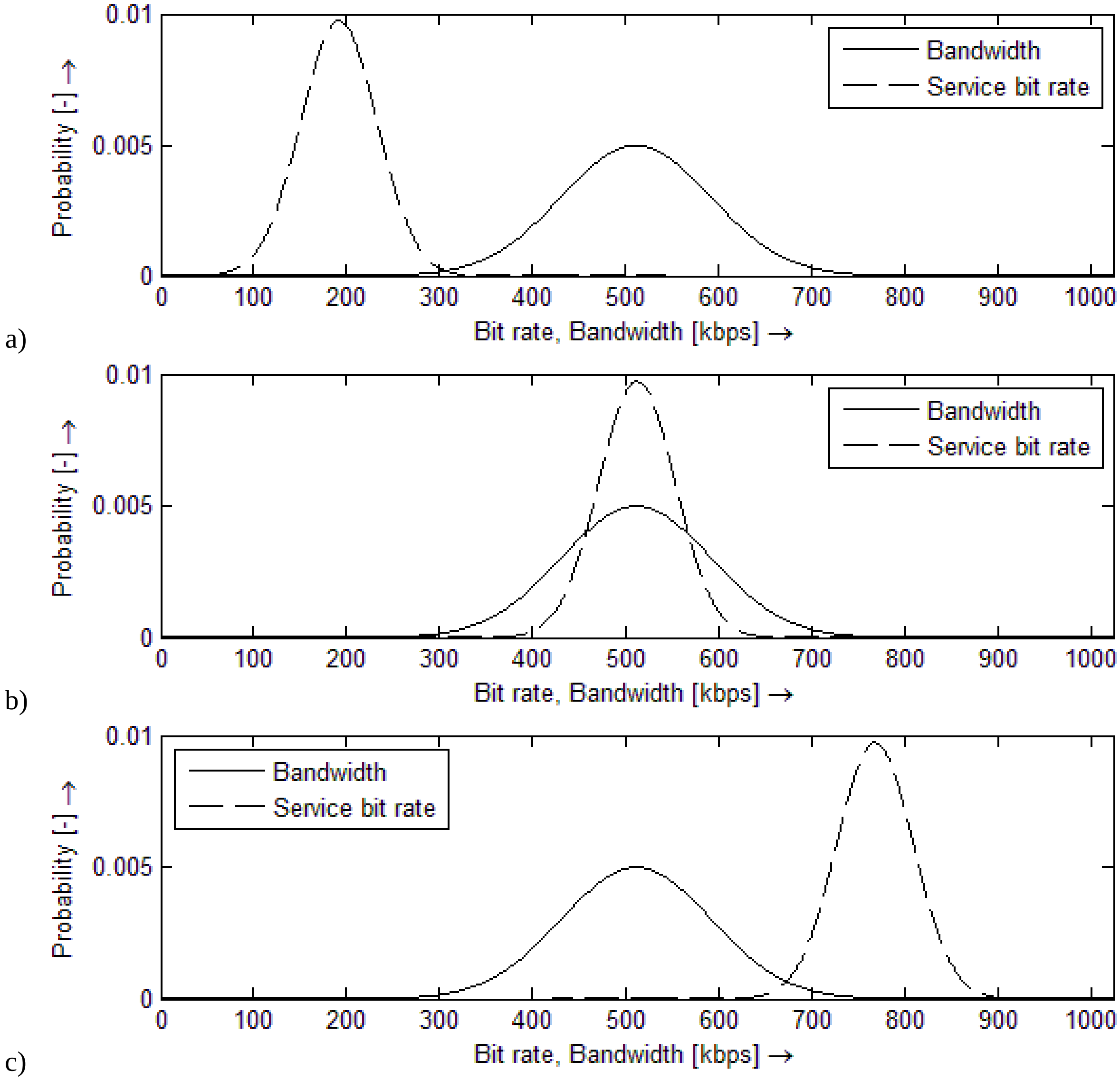

Figure 1: Example bit rate and bandwidth distribution with bandwidth mean of 512kbps;
a) Bit rate mean = 192kbps; b) Bit rate mean = 512kbps; c) Bit rate mean = 768kbps;

Multimedia streaming parameters with the biggest influence on preloading time are the bandwidth and bit rate. Figure 1 displays three possible situations of bit rate and bandwidth. In section a, bit rate mean is lower than that of bandwidth, in b, they are equal and in c the bit rate mean is higher. The y axis shows the probability of particular bit rate (source or bandwidth). The sum of probabilities of all bit rates is 100%, so the maximum probability of one individual bit rate from the distribution is less than 1%.

3.1. GENERAL EQUATION

The multimedia stream preload time depends on buffer size and available bandwidth with addition of network transport delay, as displayed in equation (1):

$$preload = \frac{BUFFER\_SIZE}{f(BANDWIDTH)} + f(DELAY) \text{ [s]} \tag{1}$$

The buffer size is dimensioned directly with source bit rate and the sum of playing time and jitter, as displayed in equation (2).

$$BUFFER\_SIZE = \left(f(BIT\_RATE) - f(BANDWIDTH)\right) \cdot \left(f(PLAYING\_TIME) + f(JITTER)\right) \text{ [B]} \tag{2}$$

When the equations (1) and (2) are combined, the result is a general formula for multimedia stream preload time, equation (3). It is worth noting that according to equation (3), the preload time is a point in a 5-dimensional space.

$$preload = \frac{f(BIT\_RATE) \cdot \left(f(PLAYING\_TIME) + f(JITTER)\right)}{f(BANDWIDTH)} + f(DELAY) \text{ [s]} \tag{3}$$

3.2. SPECIAL CASES

Equation (3) describes the general case of multimedia stream preload time. Two special cases show interesting properties. The first one is when bit rate is sufficiently smaller than available bandwidth (best case), which is illustrated in figure 1a. In this case, the effects of available bandwidth, source bit rate and playing time become negligible and therefore preload time can be computed according to equation (4).

$$preload \approx DELAY + JITTER \text{ [s]} \tag{4}$$

Second special case is when bit rate is sufficiently larger than available bandwidth (worst case), as can be seen in figure 1c. In this case, the equation (3) is appropriate for preload time computation, but the effect of delay and jitter is very small and so a simpler equation, (5), can be used.

$$preload \approx \frac{f(BIT\_RATE) \cdot f(PLAYING\_TIME)}{f(BANDWIDTH)} \text{ [s]} \tag{5}$$

3.3. THE Q COEFFICIENT

From the analysis of the worst case scenario in section 3.2, bit rate and available bandwidth are the most important components, with playing time being only a linear factor affecting the preload time if bit rate and available bandwidth are approximately the same. Therefore, if the playing time is omitted from equation (5) and functions are transformed to average value, an equation (6) for a coefficient Q is obtained, which can be used for buffer size dimensioning bandwidth allocation planning.

$$Q = \frac{BIT\_RATE}{BANDWIDTH} \text{ [-]} \tag{6}$$

From equations (3), (4), (5) and (6), it is possible to construct a graph of preload time as a function of coefficient *Q*, which is shown in figure 2.

As displayed in figure 2, the preload time starts at a non-zero value, equal to the sum of delay and jitter. Then the preload time slightly increases until *Q* coefficient becomes 1, where it takes steeper tendency. This increase is due to intersection of distribution functions of bit rate and available bandwidth. If the standard deviation of either is lower than that shown in figure 1, for example when constant bit rate source is streamed, then the actual preload time is less probability based and more deterministic. In the last phase, the increase of playing time is linear with the value of *Q* coefficient, which reflects the proportionality of available bandwidth to source bit rate. Preload time normalization, as used in figure 2, is performed by division of preload time by playing time, to make it independent on source playing time.

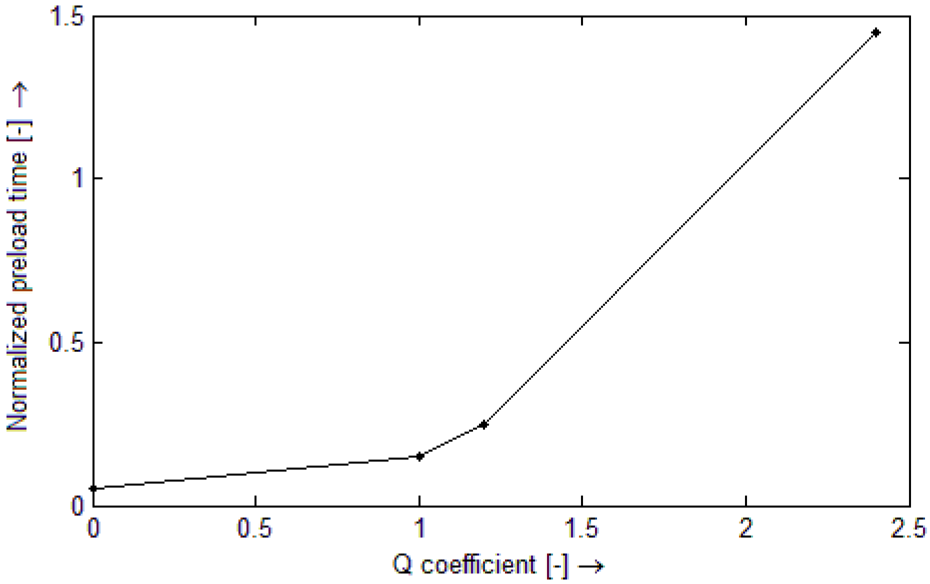

Figure 2: Estimation of preload time as a function of Q coefficient

4. PRELOADING TIME OF MULTIMEDIA STREAM

The equation (3) is too general to be of use as a Key Performance Indicator. It is possible to simplify it by moving the result from a 5-dimensional space to a scalar value. This can be done by a choice of a proper function. One of these functions is average value. If average value was used as a function in eq. (3) and instead of playing time, buffer time was used, an equation (7) is devised.

$$preload = \frac{avg(BIT\_RATE)\cdot\left(avg(BUFFER\_TIME)+avg(JITTER)\right)}{avg(BANDWIDTH)} + avg(DELAY)\ \ [\mathrm{s}] \qquad (7)$$

Equation (7) is much easier to use in practice, but it implies the fact, that distribution functions and standard deviations of all variables are the same. This may not be always true, but is a suitable simplification, which in practice does not yield a large error.

Proposed equation (7) can be used to compute the preloading time required for uninterrupted playback of streamed multimedia of specified bit rate and playing time, when average available bandwidth is known. The calculated preload time is in seconds. The preloading time can be directly used to compare the waiting time for service in different cellular networks or in different locations in the same cellular network, or to map time dependency of multimedia streaming service quality.

5. IMPLICATIONS

The coefficient Q, introduced in section 3.3, acquires values from three intervals: $Q \to 0$ for the best case, $Q \approx 1$ for general usage pattern and $Q >> 1$ for the worst case. Both extreme cases lead to large buffer sizes. When source bit rate is significantly higher than bandwidth, almost whole content has to be preloaded for playback to be uninterrupted. When bit rate is much lower than available bandwidth, then three situations are possible. First, the content is received at maximum speed and cached locally, which again leads to large buffer size, or the buffer size is limited and preloading is stopped on its fulfilment, which leads to bursty traffic pattern. The third option is traffic shaping to limit content streaming bandwidth to the value of average bit rate.

Traffic shaping effectively leads back to the general usage pattern value of $Q \approx 1$. This value yields the best results overall. Both extreme cases require large buffer sizes, what means a higher probability of unnecessary transmission when user cancels the playback after a preloading period. This causes higher resource usage and added cost for operator with no benefit perceived by the user.

6. CONCLUSION AND FUTURE WORK

Preloading time, a Key Performance Indicator for assessment of user perceived performance of streamed multimedia playback in cellular networks was presented. This indicator extends the set of KPIs defined by 3GPP in [2], [3] and is specifically targeted for assessment of a single service – multimedia streaming, which is gaining importance, as users change their behaviour because of the enhanced service offered by 4G cellular networks.

The future work will consist of measurement and determination of distribution parameters of all input parameters of devised general formula, specifically available bandwidth, source bit rate and playing time, but also delay and jitter. This might lead to a more accurate version of proposed equation for preloading time calculation. Also, the proposed equation can be used for dimensioning the buffer size in terminal equipment, which is beneficial, because devices from the target group of mobile terminal equipment are still resource constrained.

ACKNOWLEDGEMENT

This research work is supported by projects SIX CZ.1.05/2.1.00/03.0072, EU ECOP CZ.1.07/2.2.00/28.0062, and MPO FR-TI4/696, Czech Republic.

REFERENCES

[1] WALLS, Karla: How mobile usage changes when customers get 4G speeds. In: CIQUAL: Customer Insight [online]. 2012-11-05. Available from: http://www.ciqual.com/blog/how-mobile-usage-changes-when-customers-get-4g-speeds/

[2] 3GPP TS 32.450. Key Performance Indicators (KPI) for Evolved Universal Terrestrial Radio Access Network (E-UTRAN): Definitions. Release 11.0.0. 2012-09-26. Available from: http://www.3gpp.org/ftp/Specs/html-info/32450.htm

[3] 3GPP TS 32.455. Key Performance Indicators (KPI) for the Evolved Packet Core (EPC): Definitions. Release 11.0.0. 2012-09-26. Available from: http://www.3gpp.org/ftp/Specs/html-info/32455.htm

[4] VATOLIN, Dmitriy, KULIKOV Dmitriy, PARSHIN Alexander, ARSAEV Marat and VORONOV Alexander: MPEG-4 AVC/H.264 Video Codecs Comparison. In: MSU Video Codec Comparison [online]. Moscow: Graphics & Media Lab Video Group, 2012 [cit. 2013-03-03]. Free edition. Available from: http://www.compression.ru/video/codec_comparison/h264_2012/mpeg4_avc_h264_video_codecs_comparison.pdf

[5] Inside YouTube Videos. SYSOMOS INC. Business intelligence for Social Media [online]. 2010. Available from: http://www.sysomos.com/reports/youtube/

[6] GRAJA, Hubert, PERRY, Philip and MURPHY, John: A statistical analysis of IP packet delay and jitter in cellular networks. 5 pages. Available from: http://pel.ucd.ie/files/statistical%20analysis.pdf