QoS Metrics for Cross-Layer Design and Network Planning for B3G Systems

Nuno Anastácio, Francisco Merca, Orlando Cabral, Fernando J. Velez, *Senior Member*
Instituto de Telecomunicações, DEM - University of Beira Interior
Calçada Fonte do Lameiro
6201-001 Covilhã, Portugal
nuno_rodolfo@netmadeira.com, francisco_merca@hotmail.com, ocabral@e-projects.ubi.pt , fjv@ubi.pt

Abstract— **After 3G mobile and wireless systems take-up, the research community is now directing its interest towards unified ways of looking at system design, optimization, and Quality of Service (QoS) issues to satisfy the requirements of next generation multi-service mobile and wireless IP-based networks. Their implementation requires the development of IP QoS arquitecture and mechanisms encompassing the various layer of the OSI Model to cope with the new requirements. This work identifies the requirements on latency (link radio), latency (end-to-end), bit error rate, data rates, and traffic characterisation for mobile multimedia network beyond 3G. Characterisation parameters were identified and values were suggested for their range of variation, in the context of cross-layer design.**

Keywords —Cross layer Design, QoS Metrics, Service Classes, Economic Impact, Characterization parameters.

I. Introduction

A layered architecture, like the seven-layer open systems interconnect (OSI) model [1], divides the overall networking task into layers and defines a hierarchy of services to be provided by the individual layers. The services at the layers are implemented by designing protocols for the different layers. The architecture forbids direct communication between nonadjacent layers; communication between adjacent layers is limited to procedure calls and responses.

In the framework of reference layered architecture, the designer has two choices at the time of protocol design. On the one hand, protocols can be designed by respecting the rules of the reference architecture. In a layered architecture, this would mean designing protocols such that a higher-layer protocol only makes use of the services at the lower layers and is not concerned about the details of how the service is being provided. Following the architecture also implies that protocols would not need any interfaces not presented in the reference architecture. Alternatively, protocols can be designed by violating the reference architecture, for example, by allowing direct communication between protocols at nonadjacent layers or sharing variables between layers. Such violation of a layered architecture is called cross-layer design with respect to the reference architecture.

Some examples that include a violation of a layered arquitecture include creating new interfaces between layers, redefining the layer boundaries, designing protocol at a layer based on the details of how another layer is designed, joint tuning of parameters across layers. Other violation of a a layered architecture involves giving up the luxury of designing protocols at different layers independently. Protocols designed this way impose some conditions on the processing at other layer(s) [2].

The goal of this work is to select appropriate performance metrics for the joint optimisation problem from the cross-layer design point of view for key applications in B3G (beyond 3G) wireless communication systems.

In Section II, one presents the characterisation of services according to the 3rd Generation Partnership Project (3GPP) classes: conversational, streaming, interactive, and background. Then, were chosen applications that can represent each class of service, and the most appropriate values for the parameters were defined for each service. In Section III, one identifies values for parameter which accounts the economic impact of each service and will set the ground for future definition of utility functions. In Section IV, the parameters are characterised, and their range of variation is identified. Finally, conclusions are drawn in Section V.

II. Service Classes

A. *Service Classes According to 3GPP*

In B3G, the number of services is high, and the specific service characteristics cannot be totally predicted. However, they can be included into the following four groups, defined by the ITU for IMT-2000 systems: conversational class, streaming class, interactive class, and background class.

Conversational class is specified to carry bi-directional real time traffic; so, it has the highest delay requirements. An example is circuit-switched telephony. The streaming class is specified for one-way real-time traffic where the data can be processed as a continuous stream, allowing a delay that may be neutralized by the receiving end. An example is watching a real-time video stream. The interactive class is defined for applications where the user requests data from a remote end, with low bit error rate, but certain delay is allowed. A typical example is Web Browsing. The background class is defined for applications between machines, with the weakest delay requirements. An example is e-mail. Within each service class, a range of residual bit error rate (BER) and service data unit (SDU) error ratio are supported, Table I. Another approach to model the services is via the consideration of basic components [3] (block, transaction, and streaming traffic).

TABLE I. Services Classification according to Delay, residual BER and SDU error ratio

Class	**Conversational**	**Streaming**	**Interactive**	**Background**
Delay	<= 80 ms	<= 250 ms	unspecified	unspecified
Residual BER	5×10^{-2}, 10^{-2}, 5×10^{-3}, 10^{-3}, 10^{-4}, 10^{-5} ,10^{-6}		4×10^{-3}, 10^{-5} , 6×10^{-8}	
SDU error ratio	10^{-1}, 10^{-2}, 7×10^{-3}, 10^{-3}, 10^{-4}, 10^{-5}		10^{-3}, 10^{-4} , 10^{-6}	

B. *Cross-Layer QoS performance requirements*

1) *Real-time Conversational Class*

A good example of a known use of real-time conversational class is the telephony speech (e.g. GSM), but with mobile Internet and multimedia a number of new applications will require this scheme, for example voice over IP and video conferencing tools. Real time conversation is always performed between peers (or groups) of live (human) end-users. This is the only scheme where the required characteristics are strictly given by human perception (the senses). Therefore this scheme raises the strongest and most stringent quality of service (QoS) requirements.

The real time conversation scheme is characterised by low transfer times because of the conversational nature of the scheme. The time relation (variation) between information entities of the stream shall be preserved in the same way as for real time streams. The maximum transfer delay is limited by the human perception of video and audio conversation. Therefore the limit for acceptable transfer delay is very strict, as failure to provide low enough transfer delay will result in unacceptable lack of quality. The transfer delay requirement is therefore both significantly lower and more stringent than the round trip delay of the interactive traffic case [4].

a) *Voice*

Audio transfer delay requirements depend on the level of interactivity of the end users. Under the conversational class, for voice application, the general limits for one-way end-to-end delay we have a preferred range that varies between 0 and 150 ms, and an acceptable range (with increasing degradation) that varies between 150 and 400ms. The 150ms (voice) will be the most appropriate value to use for the end to end delay. In the link radio the maximum delay is less than 20ms [5]. With codec frame length of 20ms, the one-way end-to-end delay should be less than 100ms.

There are several classes in AMR speech codec [6]. The speech parameter bits delivered by the speech encoder are rearranged according to their subjective importance before they are sent to the network. The rearranged bits are further sorted, based on their sensitivity to errors and are divided into three classes of importance: 1, 2 and 3. Class 1 is the most sensitive, and the strongest channel coding is used for class 1 bits in the air interface. The AMR speech codec can tolerate about a 1% frame error rate (FER) of class 1 bit without any deterioration of the speech quality. For class 2 and 3 bits a higher FER is allowed. The required BER for Class 1 bits is 10^{-4} while for the Class 2 is 10^{-3}, and for a higher Class a BER of 10^{-2} might also be feasible. In [7] it was considered a BER of 10^{-4}, being this one the most appropriate value.

Requirements for information loss are influenced by the fact that the human ear is tolerant to a certain amount of distortion of a speech signal. It is has been suggested in studies that acceptable performance is typically obtained with frame erasure rates (FER) up to 3 %. The human ear is highly intolerant of short-term delay variation (jitter) it is therefore paramount that this is reduced as lower level as is practical. A limit as low as 1ms is suggested as the target to be considered. In [6] the bit rate considered for the voice application varies from 4.75 to 12 kb/s, while in [7] it varies from 4 to 25 kb/s [7] being this one the range considered in this work.

b) *Video Telephony*

Video Telephony implies full-duplexing, and carries both video and audio. It is intended for use in a conversational environment. Hence, in principle the same delay requirements as for conversational voice will apply, i.e., no echo and minimal effect on conversational dynamics, with the added requirement that the audio and video must be synchronized within certain limits to provide “lip-synch” (i.e., synchronization of the speaker's lips with the words being heard by the end user). In fact, due to the long delays incurred in even the latest video codecs, it will be difficult to meet these requirements [4]. This application has a maximum delay of 50ms for link radio [5], and of 400ms (image) end-to-end, over IP [5]. It takes this time because it has to synchronize. The human eye is tolerant to some loss of information; hence some degree of packet loss is acceptable depending on the specific video coder and amount of error protection used. It is expected that the latest video codecs will provide acceptable video quality with frame erasure rates up to about 1%. One also considers a BER of 10^{-4}.The data rate for this application varies between 32 and 384 kb/s [7].

c) *Remote Gaming*

Requirements for remote gaming are obviously very dependent on the specific game but it is clear that demanding applications will require very short delays. A value of 250ms is proposed in [8] but one considers values lower than 300ms for real time action games, and less than 900ms for real time strategy games [9]. There are two kinds of values for maximum delay for remote gaming (playing chess online or play action games); other values were also found in literature (one way delay between 50ms and 100ms to satisfy the most demanding users). Remote gaming is considered to be in the conversational class but it can also be integrated in the interactive class. The proposed BER requirements varies from 10^{-6} to 10^{-7}. The data rate for this application varies between 64 and 1000 kb/s [10].

2) *Interactive Class*

This scheme applies when the end-user, that is either a machine or a human, is on-line, requesting data from remote equipment (e.g. a server). Examples of human interaction with the remote equipment are: Web Browsing and data base retrieval, server access. Examples of machines interaction with remote equipment are: polling for measurement records, and automatic data base enquiries (tele-machines). Interactive traffic is the other classical data communication scheme that, on an overall level, is characterized by the request response pattern of the end-user. At the message destination there is an entity expecting the message (response) within a certain time. Round trip delay time is therefore one of the key attributes. Besides, the content of the packets must be transparently transferred (with low bit error rate) [4].

a) *Web Browsing*

The Web Browsing application refers to retrieving and viewing the HTML component of a Web page; other components, e.g., images and audio/video clips are dealt under their separate categories. From the user point of view, the main performance factor is how fast a page appears after it has been requested. For this application one has a delay requirement between 2 to 4 s/page, although it is recommended improving it to 0.5s. The maximum delay for link radio is 300ms [11]. For this application, one has a BER of 10^{-6} [7], a FER lower than 1% [7], and a bit rate lower than 2000 kb/s [12].

b) E-mail

E-mail is generally thought to be a store and forward service which, in principle, can tolerate delays of several minutes or even hours. However, it is important to distinguish communications between the user and the local email server and server to server transfers. When the user communicates with the local e-mail server, there is an expectation that the mail will be transferred quite rapidly, although not necessarily instantaneously [4]. Under this application one has the server to server delay that can be of several minutes or hours (in [12] 5 min), while the user to local server delay varies between 2 to 4 s (the latter being the one considered in this class). E-mail has a maximum BER of 10^{-6} [7] while the bit rate is ~1500 kb/s [12].

3) Streaming Class

When the user is looking at (listening to) video (audio) the streams scheme applies. The real time data flow is always aiming at a live (human) destination. It is a one way transport. This scheme is one of the newcomers in data communication, raising a number of new requirements in both tele-communication and data communication systems. The delay variation of the end-to-end flow must be limited, to preserve the time relation (variation) between information entities of the stream. However, as the stream normally is time aligned at the receiving end (in the user equipment), the highest acceptable delay variation over the transmission media is given by the capability of the time alignment function of the application. Acceptable delay variation is thus much greater than the delay variation given by the limits of human perception [4].

a) Video

Video streaming is mainly a one-way application from server to user(s). In MPEG-4 video [6], there is a bit rate that varies between 24 to 128 kbps or higher, a end-to-end delay that varies between 150 and 400ms. In other references, other values are presented, like less than 10s [13] and 200ms [14], truly real time or not, respectively. However one considers the values between 150 and 400ms. In the link radio the maximum delay should be lower than 1s [13]. BER varies from 10^{-6} to 10^{-3}, with a significant degradation for the latter.

b) Data

FTP is a data application that can be classified as streaming service with relaxed delay tolerance, and have "0" final information loss at the receiving end. The maximum delay in link radio for this application is 500ms [7], while in end-to-end our maximum delay is less than 10s. BER is 10^{-6} [7]. The bit rate varies between 64 and 400 kb/s [7].

4) Background Class

The Background Class has no stringent requirement on delay of services. For Email the server to server delay has a wide range with median of several hours. For low priority transaction services, the delivery delay is 30 seconds.

C. Packet Error Rate

One considers the following ranges of BER supported by the network [8]:

- 10^{-3} to 10^{-7} for real time applications;
- 10^{-5} to 10^{-8} for non-real time applications.

Packet Error Rate (PER) can be calculated by using: PER = Packet_Size[bytes]·8·BER .

D. IP Multimedia Sub-system

IP Multimedia Sub-system, IMS, allows operators to provide their subscribers with multimedia services that are built on Internet applications and protocols [15]. IMS enables IP connectivity between users using the same control and charging mechanisms, Fig.1. The basic session initiation capabilities provided by Session Initiation Protocol (SIP) are used to establish peer-to-peer sessions.

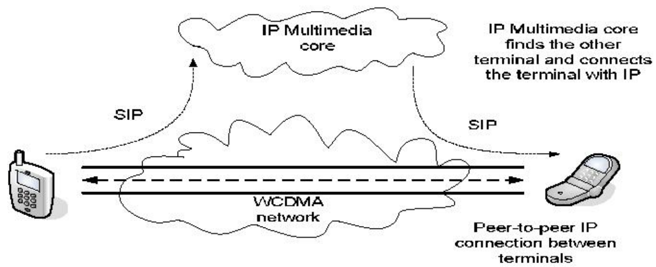

Figure 1. Basic principles of the IP Multimedia Sub-system.

IMS provides the means for network operators to maintain their role in the value chain by providing new multimedia services and predictable end user performance. The same platform can be used in both real time services, like VoIP, and non-real time services, like content sharing [16].

III. CHARACTERISATION PARAMETERS

According to [14] it is useful to distinguish characterization parameters into service, traffic, communication, service components, activity model ones, and operation environments as well. From the operation environments, in the mobile domain, terminal mobility is also of key relevance.

A. Traffic characteristics

The service characteristics are the following [17]:

Intrinsic time dependency - time-based (TB, where data blocks must be displayed consecutively at predetermined time instants), and are different from non-time-based (NTB);

Delivery requirements - real-time (RT, for immediate consumption), or non-real-time (NTR, stored for later consumption);

Burstiness - It is defined as the ratio between the peak and the average bit rates [18], several types of communication being highly 'bursty' in nature.

B. Service Characteristics

The service characteristics are the following:

Latency/end-to-end delay - Absolute delay is one of the key QoS performance parameters that must be satisfied by the broadband network [19]. In the context of service characterisation, it is the maximum transfer time (in one way) that is tolerated by the service. To provide interactive response to viewers the response time between a user action and its effect should be less than 100ms and sometimes even 50ms. By definition, latency requirements only apply to real-time applications [19], thus there is no latency requirement for non-real time ones, although one identifies the associated delay to good QoS. At the application layer, the latency requirement arises from the human response time for different real-time applications. It translates into end-to-end packet latency (absolute delay) at the network layer. Such packet latency is the sum of processing delay, packetisation delay, transmission delay, queuing delay and propagation delay.

Delay (link radio) - It is the maximum transfer time in the link radio, i.e., from the BS to the mobile, or from mobile to the BS, that is tolerated by the service.

Data rate - It is important to define the average useful data rate associated with services. Specifying the average bandwidth requirement for a 'bursty' application is a challenge, because it varies according to the duration for which the average is taken. The main difference between data rate concept and bit rate concept is that the first one is the useful information only, while the bit rate can be referred has the gross "speed" of data transmission.

Bit error rate (BER) - It is a non-dimensional variable that expresses service tolerance to uncorrected errors in the bearer service, including non-delivered information. It is calculated as the ratio between bits received with error or omitted, and the overall received bits.

Packet error rate (PER) - It is an important QoS parameter for Wireless networks. However, it is mostly derived using BER and an error independence assumption.

Frame erasure rate (FER) – It is a measure of the number of frames of data that contained errors and could not be processed. FER is usually expressed as a percentage or an exponent.

SDU Error ratio - SDU error ratio is defined as the fraction of lost or erroneous SDUs. If resources are reserved for the given UMTS bearer, the SDU error ratio does not depend on the loading conditions.

Adaptive Multi-Rate - A variable rate speech codec selected by the 3GPP for the 3G (WCDMA) evolution of cell phone system. Using the Algebraic Code Excited Linear Prediction (ACELP) compression technology, AMR provides toll quality sound at transmission rates from 4.75 to 12.2 Kbps.

Reliability - It is the probability of a device performing its purpose adequately for the period of time intended under the operating condition encountered.

Throughput - The amount of data transferred from one place to another or processed in a specified amount of time. Data transfer rates for disk drives and networks are measured in terms of throughput. Typically, throughputs are measured in kbps, Mbps or Gbps.

Jitter (delay variation) – It occurs due to different delays of the packets, which is a consequence of the bursty traffic load in network nodes. This is the measure of changes in the channel delay, compared to average delay. In circuit switched-type connections, the jitter can be thought to be insignificant; consequently the delay is a constant.

C. *Communication Characteristics*

The communications characteristics consist on burstiness, service classes, BER (bit error rate), and protocol. To support broadband applications, and based on QoS parameters, three classes of services must be supported [20]:

Best-effort delivery - It is being addressed by the ATM Forum with the ABR class of service.

Real-time (RT) delivery of time-based information - It is the CBR (Constant Bit Rate) or VBR (Variable Bit Rate) with time requirements (bounds on delay variation). This can be supported by reserving peak bandwidth for each application over a QoS-based network.

Real-time delivery of non-time based information - It is probably the most challenging class of service to support, because it is 'bursty' and has an absolute delay requirement. Such unpredictability makes it very difficult to efficiently allocate bandwidth to support this class of services.

For NRT information, only best effort delivery is necessary, and information should be stored at the receiver. Low data rates are used, except when further resources are available.

Table II presents a summary for the range of variation of cross-layer QoS metrics.

IV. ECONOMIC IMPACT

By taking service characteristics into account, i.e., intrinsic time dependency and delivery requirements [14], a resource allocation parameter can be defined based in the categories for services and applications from [11], which is a measure for the urgency of the traffic, accounts for the economic impact of each service, and are a first contribution for the definition of utility functions [21]. This is a compound way for joining the different service characteristics, in unique categories which vary here from 1 to 5 with the following correspondences:

1 - Radio bearer of NRT and background services - Best-effort delivery (NRT/NTB), e.g., E-mail;

2 - Radio bearer of NRT interactive services - Real-time delivery of non-time-based information (RT/NTB), e.g., Web Browsing and FTP;

3 - Radio bearer of almost RT service – Almost real-time delivery of time-based information (RT-NRT/TB), e.g., Video Streaming;

4 - Radio bearer of RT service - Real-time delivery of time-based information (RT/TB), e.g., Voice and Video-telephony;

5 - Radio bearer RT service of time-based information but with strict/special delivery requirements, e.g., low delay and BER, e.g., Remote Gaming.

V. CONCLUSIONS

This work presents a description of characterisation and classification parameters for B3G mobile system in the context of cross-layer design. A first assignment of parameter values for these services is identified in Table II. It shows the range of variation for all the parameters. Together with the resource allocation parameter, it is very useful for cross-layer design purposes.

TABLE II. CROSS-LAYER QOS METRICS

Services/ applications	Reliability		Latency (link radio)	Latency (end-to-end)		Bit Rate	Traffic Characterization		
	BER	***FER***	***Max Delay (ms)***	***Jitter (ms)***	***Max Delay (ms)***	***R_b[kb/s]***	***Intrinsic Time Dependency***	***Delivery Requirements***	***Burstiness***
Voice	10^{-4} [7]	< 3% [7]^3	≤ 30 [11] 20[5]	< 1 [13]	150 [7] 100 [20]	4-25[7] 4.75-12[6]	TB [7]	RT [7]	1 [14]
Web Browsing	10^{-6} [7]	< 1% [7]^3	≤ 300 [11]	N/A [13]	Few seconds[7] <4000/page[13]	< 2000 [7]	TB [7]	RT [7]	1-20 [14]
Video Streaming	10^{-4} [7] 10^{-6}-10^{-3} [6]	<1% [7]^3	< 1000 [22] 10000 [13]	-	200[7] <10000[13] 150-400[14]	32-384[7] 24-128[6]	TB/NTB [7]	RT [7]	1-5 [14]
E-Mail	10^{-6} [7]	0% [8]	-	N/A [13]	5min [12]^1 4000[14]^2	< 1500 [12]	TB [12]	NRT [12]	1-20 [12]
Video Telephony	10^{-4} [14] 10^{-6} [7]	< 1% [7]^3	150 [18] 50 [5]		200[17] <150[13] <400[13] 300 [5]	32-384 [7]	TB [7]	RT [7]	1 -5 [7]
FTP	10^{-6} [7]	0% [8]	500 [14]	N/A [13]	10000 [7] <10000 [13]	64-400[7] 384 [12]	NTB [10]	RT [10]	1-50 [7]
Remote Gaming	10^{-7}:10^{-6} [10]	0% [8]	-	N/A [13]	50[10] 100[12] <300[9]^4 250[8] <900 [9]^5	64-1000 [14]	TB/NTB [10]	RT [10]	1-30 [10]

1: server to server delay, 2: user to server delay, 3: UMTS System, N/A – Not Available, 4: RT action games, 5: RT strategy games, RT – Real Time, NRT – Non RT, TB – Time Based, NTB – Non TB.

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