

The Impact of Network-Based Parameters on Gamer Experience

Dorel Picovici

Institute of Technology Carlow, Ireland

David Denieffe

Institute of Technology Carlow, Ireland

Brian Carrig

Institute of Technology Carlow, Ireland

INTRODUCTION

Most of the existent games consist of multiple players, connected over a network, collaborating and competing in a virtual world. In this world, each player typically controls a single virtual entity. Communication between players can be achieved by sharing entity state information, such as positioning using synchronisation messages. These messages are periodically transmitted across the connecting network, and update the remote state of the virtual entity, which is the state replicated on other players' computers. If the games are using the Internet as a connecting network, latency, jitter, and packet loss, can have a significant impact upon the service experienced by application user (end-user). More specifically, latency or delay can be introduced by various types of delays, such as propagation, serialisation, and queuing delays.

Jitter is the variation in latency experienced by consecutive packets. Not all of the packets in a given flow will take the same path through the network. The time taken to traverse different routes is likely to vary due to factors such as their different physical distance, the number of hops, or physical link properties. On lower bandwidth links, which have a greater serialization delay, variation in packet lengths can introduce jitter. The size distribution and arrival patterns of other traffic flows on shared links may influence the queuing delay experienced by packets in one particular flow and is, in itself, a source of jitter. There are a number of different points on the network where packets may be lost. At the physical layer, all links experience some rate of data corruption, known as the bit error rate (BER). This may be caused by a high signal-to-noise ratio (SNR) during digital to analog conversion processes, which causes erroneous encoding or decoding of data, or it may be caused by faulty hardware. Forward error correction (FEC) is sometimes used at the link layer to recover from one- or two-bit errors. A cyclic redundancy check (CRC) may be applied to detect whether or not errors are included in the frame. Occasion-

ally, transient congestion, with the subsequent queuing of packets, is so severe that it causes the routing queue buffer to overflow. When this occurs, newly arriving packets will be dropped until there is sufficient space in the buffer to place new packets. On other occasions, dynamic routing changes or route flapping may result in a temporarily incomplete network path, which causes losses.

BACKGROUND

The current Internet model provides only a single level of service, known as best effort (BE) delivery of data. In this model, the network will attempt to route traffic as quickly as possible to its destination, but provides no guarantees that those packets will traverse the same path across the network, arrive in the same order, or even arrive at all. Streaming and interactive applications (often referred to as *nonelastic* applications) require upper bounds to be placed on delay and jitter to facilitate smooth delivery. Packet loss and corruption must be also bounded to ensure adequate subjective quality, as it may not be possible to retransmit packets within the required time frame. How the required level of service might best be provided to the multitude of applications in use on the Internet has been the subject of extensive study for over a decade. When discussing the motivation for quality of service (QoS) in communications networks, it is important to define the term "quality of service." The International Telecommunication Union (ITU) Telecommunication Standardization Sector (ITU-T) describes the term "quality of service" as "The collective effect of service performance which determines the degree of satisfaction of a user of the service" (ITU-T, 1994). When this definition is used, it can be clearly seen that QoS is something that can only be correctly determined by the user of a service, as it relates to the user's expectation of service quality.

The quality of a voice/video telephony call, or a multi-player game can be assessed using objective and subjective

methods. For voice/video, objective methods allow subjective quality to be predicted on the basis of psychoacoustic modeling (Hollier & Cosier, 1996). Similar applications are very much in their infancy for gaming. Subjective methods of assessment involve playing sample stimuli to a target audience in order to gather opinion data. The primary difficulty with subjective assessment is that of eliciting useful objective information when a user expresses their satisfaction or otherwise with a service. Quantifying what is meant by “good” is difficult without contextual information about the user’s previous experiences and context. This points to the inherent complexity of measuring subjective quality and the difficulty involved in trying to obtain uniform answers from a diverse population of users. On the other hand, objective methodology, though often preferred because it is easier to measure and gather data for, is not without problems.

Subjective Measures of Mean Opinion Score (MOS)

Assessment measures that are based on ratings by human listeners are called subjective measures. When used for telecommunications systems, these tests seek to quantify the range of opinions that listeners express when they hear speech transmission of systems that are under test. Properly designed subjective tests provide the most accurate way of assessing speech quality. However, the results of subjective tests are influenced by the conditions of the tests, and great care must be taken of a number of factors in order to obtain reliable and reproducible results.

Although subjective assessment of speech quality requires substantial efforts, it is indispensable as a reference for evaluating the performance of objective speech quality

Table 1. Listening-quality scale

Quality of speech	Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

Table 2. Listening-effort scale

Effort required to understand the meaning of sentence	Score
Complete relaxation possible; no effort required	5
Attention necessary; no appreciable effort required	4
Moderate effort required	3
Considerable effort required	2
No meaning understood with any feasible effort	1

Table 3. Listening-effort scale

Loudness preference	Score
Much louder than preferred	5
Louder than preferred	4
Preferred	3
Quieter than preferred	2
Much quieter than preferred	1

6 more pages are available in the full version of this document, which may be purchased using the "Add to Cart" button on the publisher's webpage:
www.igi-global.com/chapter/impact-network-based-parameters-gamer/13826

Related Content

Designing Model-Based Intelligent Dialogue Systems

Dina Goren-Bar (2001). *Information Modeling in the New Millennium* (pp. 268-284).

www.irma-international.org/chapter/designing-model-based-intelligent-dialogue/23004/

Facing the Challenges of Multi-Channel Publishing in a Newspaper Company

Airi Salminen and Kirsi Hakaniemi (2007). *Journal of Cases on Information Technology* (pp. 54-72).

www.irma-international.org/article/facing-challenges-multi-channel-publishing/3194/

GPS: A Turn by Turn Case-in-Point

Jeff Robbins (2011). *Teaching Cases Collection* (pp. 1-18).

www.irma-international.org/article/gps-turn-turn-case-point/54463/

Framing Political, Personal Expression on the Web

Matthew W. Wilson (2009). *Encyclopedia of Information Science and Technology, Second Edition* (pp. 1580-1585).

www.irma-international.org/chapter/framing-political-personal-expression-web/13788/

Risk Planning and Management

Daniel M. Brandon (2006). *Project Management for Modern Information Systems* (pp. 157-182).

www.irma-international.org/chapter/risk-planning-management/28182/