TELECOLLABORATION – A CASE STUDY FOR PERFORMANCE ANALYSIS OF VOIP SYSTEMS

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ABSTRACT
The evolving nature of complex traffic issues of IP (Internet Protocol)-based Telecollaboration (TC) business system technology requires an intuitive understanding for the participants not just to make sense of design issues involved but also to provide an insight of guaranteed run time performance. The aim of this study is to introduce new theoretical underpinnings for modeling of VoIP (Voice over IP) applications, which may have impact on conventional wisdom of TC in context of often contradictive aspects of user Quality of Experience (QoE). In this paper, we also focus on the performance evaluation and traffic control factors as they present an objective engineering challenge for development of analytic traffic models. Close attention is paid to performance issues such as packet loss probability, packet delay and jitter apart. This research paper concludes on highlighting the importance of mathematically formal dimensioning and evaluation of IP networks for an improved user perception of QoE and keeping a balance between traditional approaches and new theories of unconventional thinking in this emerging field.

1. INTRODUCTION
To win on the TC technology transition and prepare for tomorrow, the nature of CeNTIE’s (Centre for Networking Technologies for the Information Economy) TC (Telecollaboration) project at UTS is mainly to understand, analyse and guide the participants of TC to the real world by providing them opportunities to reap the collaborative benefits in an efficient manner. It was our aim to establish and assess the boundaries of concept formulation and early popularization of this transition through R & D of extension, enhancement and exploration of issues involved that are cognitive and non-cognitive in nature. This includes conducting experimentation for an early adoption and implementation of the TC technology apart of establishing its technical advantages. The project aims to provide a good foundation on the information and knowledge highway, where substantial investments are made in research through supporting software with ability to understand the underpinning issues of TC for steering through this rough terrain and navigate through areas of uncertainty. Whereas, our partners, vendors and focus groups provide links between technology development and its utilization to reach the end-user community during TC technology transfer and diffusion stages. It is of equal importance to make sure that uncertainty issues associated with implementation of the technology must be resolved during the diffusion phase which could then be accomplished through demonstrations and pilot trails. This work is part of the project deliverables that deal with those uncertainty issues that relate to the creation of indices for QoS, collaboration, usability and learning (V.Mahadevan, 2004). Recognition of importance of development of sound technical principles and provision of implementation support for collaborative practices are also required to ensure that the uncertainty level associated with these practices is reduced to a point where adoption can take place. In an innovative application such as TC, the proposed solutions for uncertainty issues would resolve mysteries of user experience not just to make sense of drastically different and unfamiliar facets of TC teletraffic issues but to better understand the reflective spectrum of reactions in the realms of TC. In this paper we first emphasize
the importance of rigorous physical examination and explanation of the mathematical concepts for a VoIP application through proposed analytical models in a networking environment. Finally, this paper concludes with notes on the importance of making effective use of the available subjective results of VoIP that can assist in pointing towards yet new research avenues explored by TC research communities that are not just evolutionary but revolutionary in nature.

2. THE DOMAINS, MODELS AND REALMS OF VOICE OVER IP FOR TC

![Figure 1. Domains, Models and Realms of VoIP for Telecollaboration](image)

Figure 1. Domains, Models and Realms of VoIP for Telecollaboration

Figure 1, demonstrates how the two well-known models such as Poisson and self-similar can be used to describe the Internet traffic in the measurement domain. Each of them has own advantages in the realm of analytical modeling. It is also shown that both the Poisson and self-similar models can co-exist to give a more accurate scenario. In a low congestion link, Long Range Dependence (LRD) characteristics are observed and as load increases, the model is pushed to Poisson. Similarly, as load decreases, the model will be pushed to self-similarity. In a Poisson model, bursty traffic appears to be bursty in nature in fine scale. But in coarse scale, this traffic appears to be smoothed out and looks like a random noise. Whereas, in self-similar model, scaling of bursty traffic has never been an issue. This is because it has similar characteristics on any scale that gives a more precise result due to LRD in the network traffic. Recent research on packet traces varying across a wide range of link speeds and connection loads reveals that on links with high speeds, towards the core of the Internet that is carrying traffic made up of large numbers of connections, the traffic can be close to Poisson and independent with the absence of the burstiness (Jin Cao, 2003). Considering the reason that Poisson model does not scale the bursty traffic properly, it was intentionally decided by authors to take into account this model to describe the network traffic for the purpose of this work. The small sub-sets of usability heuristics (as shown in Figure 1) recommended by Billy Vaughn Koen (2003) are taken under consideration for superficial treatment of complexity issues of TC. Yet, in the realm of mathematics, they are identified as arithmetic, mathematics, deduction, causal and behavioral analysis. As the packet loss, packet delay and jitter are the major sources of speech impairment in VoIP applications, we recommend using an ON/OFF source model. This is a two-state (active/silent) model in which the packet arrival rate is fixed (R) and the sojourn times (period spent in a state) are negatively and exponentially distributed from a Poisson process. We are able to calculate the mean duration in the active state as Ton and mean duration in the silent state as Toff using the formulas (J. M Pitts and J. A. Schormans, 2000). The other important factor that impacts on packet loss includes codec types (such as G.711, G.729A, and G.722 etc). When packet losses are grouped together (large packet bursts), codecs are found to be unable to hide the packet loss from the user. This prompts the users to know what the percentage of lost packets is. To compute the performance analysis,
we use M/D/1 heavy-traffic approximation for a basic IP fixed size packet queuing model. This is used for
demonstrating the basic queuing relationships involved, by assuming the buffer capacity as finite with FIFO
(First-in-first-out) discipline apart of estimating the actual packet loss probability. The codec selection and
choice of the compression scheme used depends on selection of parameters that are important for a specific
measurement purpose. Similarly, the jitter buffer size must strike a fine and subtle balance between delay and
quality. For example, if the jitter buffer is too small, network perturbations such as loss and jitter will cause
audible effects in the received voice and if the jitter buffer is too large, voice quality will be fine. In case of
multiple ON/OFF sources, we can assume the buffer-less approach for calculating the burst-scale loss factor.
Packet loss can also be caused by overloaded links, excessive collisions on a LAN, physical media errors and
other factors. It is also equally important to note that the different codecs have different input bit rates and
changing the voice payload overhead causes change in their voice bit rate and their relevant network
bandwidth requirements. The packet loss probability can be calculated by using parameters resulting in
increased load that include source bit rate, number of voice source, ON-OFF periods and packet overheads.
Our results demonstrate that the packet loss using G711, G729, and G723 codecs with single source input on
the same link with the same bandwidth and high bit rate source causes high load in the network. On the other
hand, when increasing the number of calls in G729 (to 20 calls) and G723 (to 25 calls), it shows that the
packet loss probability distribution in all three codecs are nearly of the same value. Increasing Ton period
(from 1 sec to 4 sec) of input source can result in increasing link bandwidth and packet loss probability. The
link load can be calculated from the formula: Link utilization = (arrival rate x activity factor) x service time
(J. M Pitts and J. A. Schormans, 2000).

Figure 2. Loss Probability Vs Link Bandwidth
(When Ton is above 5 sec)

Figure 3. Loss Probability Vs Link Bandwidth
(When Packet Overhead Increases)

The Figure 2 depicts the scenario where increasing Ton above 5 sec will increase the activity factor and
then the maximum arrival rate will become the source bit rate for after which the packet loss probability
remains the same. However, when the packet overhead increases the source bit rate and bandwidth
requirements also increase therefore causing the packet loss probability to increase as shown in Figure 3.
Heavy-traffic approximation of packet loss probability for the M/D/1 queuing model can also be
overestimated at high link utilization and underestimated when the link utilization is low for other discrete
distributions. For changing the Toff (during those silent periods in which voice sources will not send packets)
we apply the Voice activity detection (VAD). Here, we use an excess-rate M/D/1 analysis with an
assumption of use of deterministic server to demonstrate ON-OFF behavior of the model. The measured
outcome of this can be viewed as a series of overlapping packet flows by comparing with multiple loads (for
different codecs). This is demonstrated in Figure 4, where the packet loss probability using G729 and G723
voice codecs are of nearly the same values for the same link bandwidth using multiple input sources. Hence,
the impact of Toff period while studying the queuing behavior plays a major role in the design and
performance process of the network. In which case, we use different queuing models needed for the
performance evaluation. This includes calculation of the probability distribution of the queuing contents
(such as packets), delay, and the loss probabilities of the packets, the busy and idle periods of the system, etc.
Our future research work will include demonstration of use of variety of analytical techniques that will assist
us in understanding teletraffic issues involved in various sub systems of application performance assessment
in a more intuitive way (for models such self-similar etc). Sharing of virtual buffer space (independently
treated) and server capacity (appropriate to the multiplexing scenario and traffic sources used) becomes
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critical. This is due to the fact that different traffic classes/aggregates are needed for different performance requirements, which are then to be guaranteed by best effort IP network with the packet delay and packet loss. In this case, calculation of the decay rate for each virtual buffer is required not just to meet the performance requirements but to assess buffer space partitioning. Sharing of buffer space could be done for both a single output port and across multiple output ports. The packet loss probabilities for each class of traffic could be consistently expressed in terms of the desired decay rate, $d_r$, scaling factor, $S$, and buffer space, $X$. This needs to be carried out by the network operator by configuring virtual buffer partitions under software control. As part of the shared buffer analysis to cope with multi service requirements the packet loss probability is plotted against buffer capacity per output port, this is demonstrated in Figure 5. Here the packet loss probability decreases by adding more buffer ports (for different codecs). Finally, we use a token bucket configuration for traffic conditioning of aggregate IP flows for the multiple ON-OFF source models by using excess rate analysis as earlier demonstrated. The results of the relationships between different packet loss probabilities and token bucket size (for different codecs) by changing the token bucket rate allocation are shown in Figure 6. It also shows that by increasing the token bucket rate allocation and token bucket size the packet loss probabilities will decrease. The Figure 7, depicts the relationship between waiting time and link bandwidth for different codecs with different bit rates.

Figure 4. Loss Probability Vs Link Bandwidth (For Multiple Input Sources)  
Figure 5. Loss Probability Vs Buffer Capacity (Per Output Port)

In the same figure different delays are shown for different codecs. Then we use, ITU-T G.107 standard "E-model" formulas (as shown in Figure 1) to calculate delays (that includes delay estimate from Real Time Control Protocol (RTCP) packet, coding and packetization delay, delay introduce by Jitter buffer and decoder, send side’s and receiver side’s access delay etc). For calculation of impairments delayed after voice signal transmission called Id is nothing but the approximations to the delays such as taker echo, listener echo and long delays. The Figure 8 demonstrates the relationship between Id values based on the waiting time of the queuing model that we use for different codecs. It is important that the effects of equipment (i.e. codecs,
etc) parameter values (i.e. Ie, etc) for different codecs needs to be considered as well as its impact on percentage packet loss as shown in Figure 9. It is worth noting here, that the larger the Ie, the more severe will be the impairment. Parameters such as Is (the simultaneous impairment factor that accounts for impairments caused by non-optimum side tone and quantizing distortion) and A (advantage or correction factor that adjusts perceived quality based on user expectations) are of equal importance. Combining Id and Ie values, will give the R factor a value (from 0 to 100) by taking into account the different codecs that we use in our endpoints as shown in Figure 10. Then R values can be mapped into an equivalent Mean Opinion Score (MOS) as an estimate of effective voice quality over time, a numerical measure of the quality of human speech, or "customer experience" metric. MOS is basically used to manage networks and set Service Level Agreements (SLAs) from a meaningful customer quality perspective. Figure 11 shows the relationship between link utilization and MOS for different codecs.

3. QUALITY OF EXPERIENCE OF VOICE OVER IP IN THE REALM OF UNCERTAINTY FORCES

Lack of reliability of voice quality measurement capability based on Mean Opinion Score (MOS), would undermine the overall customer perception of voice quality of value-added TC business system. This is seen as mandatory in terms of ensuring acceptable communications in this emerging collaborative paradigm. The proposed use of mathematically tractable quantitative method in this study, combined with the qualitative method would enable us not just to observe the scaling properties of measured network traffic but to ensure that the dominant perceived voice quality of users that is not sacrificed for the end-user QoE (Quality of Experience). As shown in Figure 1, ISO 9241 standard on usability that defines the quality of efficiency, effectiveness and satisfaction (Frokjaer et al, 2000) has been taken for consideration. Efficiency is described...
as the relationship that exists between the accuracy and completeness with which users achieve certain goals and the resources expended in achieving them. However, efficiency is related to the technology as well as human resources that are based on time and effort. Effectiveness is described as the accuracy and completeness with which users achieve certain goals. It is also shown in Figure 1 that the important measure of effectiveness can be obtained through the business and collaboration outcomes through QoB and Qoc metrics (V.Mahadevan, 2004). On the other hand, the technology effectiveness also becomes crucial for the overall satisfaction such as MOS which then can be broken down into user satisfaction based upon the business, collaboration and technology outcomes and the processes which users follow in order to achieve the outcomes. For which, satisfaction becomes the “users’ comfort with positive attitudes towards the use of the system such as VoIP. The important sub-set of usability heuristic that are needed in the realm of uncertainty forces are identified as the position of certainty, logic and truths, progress, physical reality, consciousness and perception and arguments. These subsets of usability heuristics play a crucial role when considering TC with its own problems in philosophy and cognitive engineering. This is not just to achieve functionally unified system responses but also for the conceptual understanding of objective reality for the participants. This needs the knowledge representation for the accumulation of all uncertainty forces involved in the context. Measurement and mapping of voice quality enables businesses to achieve an optimal mix of voice and data without damaging VoIP service quality. Thus it is expected that this study of usability heuristics of VoIP would enable the participants in a TC business system that are experiencing a widespread acceptable, borderline or unacceptable voice quality to carry out the required collaborative tasks successfully.

4. CONCLUSION

The proposed measurement of packet loss, delay and jitter and mapping with non-intrusive MOS which would assist the TC participants with the better understanding of technological, business, social and human sub-systems’ acceptability issues involved. This in turn will have impact on business and social sub-systems’ acceptability for overall system’s acceptability for early adoption of value-added TC business systems. In this study, it was our intention, to demonstrate the Poisson model based approach combined with intuitive ideas of understanding the relationship between the Internet traffic-performance and the QoS requirements from the user point of view. The main aim was to manage the complexity of interpreting the results obtained. Our future work will include description of network traffic through self-similar models and investigate the co-existence that appear between the Poisson and self similar models as they both have their own individual advantages. This would further assist in understanding the relationship between the network invariants and varying nature of TC application by exploring the internal dynamics of the Internet traffic. It is important to note that modeling properties should address the research questions being investigated and support understanding of the model’s parameter settings.

REFERENCES