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A Comprehensive Analytical Model for Video over IP in Telecollaboration Business System Environments

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Abstract - **This paper proposes the emerging Telecollaboration (TC) business system framework supported by Video over IP applications. Such environment allows for the measurement of user experience. Initially, a simple but comprehensive analytical model of Video over IP for the development of Quality of Service (QoS) indices is demonstrated. This is followed by a presentation of the model consisting of several illustrative units that provide an integrated context for estimation and mapping of the quality and quantity measures of Video over IP in the ubiquitous Internet environment. The Pitts's software methodology is applied, addressing the second-order network performance statistics in the context of jitter (i.e., delay and packet-loss variation). Finally, at the network layer, the behavior of source, queuing, multi-service requirement mechanisms and a set of workflows, connections and aggregates are measured to justify the quality inherent usability aspects of video quality. It is expected that the proposed modeling approach overcomes the degradation of video quality, which is seen as a fundamental problem that may occur due to the multi-user packet switched network. Our results suggest that the experimentally validated analytical modeling of Video over IP is of considerable value in promoting the TC initiatives.**

Keywords: Telecollaboration, Internet Protocol (IP), Video-over-IP, Quality of Service (QoS), and Quality of Perception (QoP)

1. INTRODUCTION

The idea of academic knowledge on Telecollaboration (TC) as an abstract form had a new impetus from the use of technologies. It uses the metaphor of technology advancements to describe the closely coupled collaboration entities that could be implemented and tested [25]. However, with the unprecedented growth of multi-service best-effort IP network, the TC business system environments are experiencing a growing demand from users with adequate network QoS ([8], [9]). Additionally, sound traffic engineering still remains the crucial issue for the performance review of the inherent usability aspects of the audio and video quality of the applications involved ([2], [10], [22]). The existing literature reports numerous inconsistent network-level analytical models, which may be due to different experimental designs; independent and dependent network traffic invariants ([2], [5], [26]). Therefore, an appropriate network-level analytical model and a concomitant simulation environment are expected to enable the technology-intensive Video-over IP application to deal with the application-level activities of an emerging TC business system ([10], [11], [12]). These activities are often seen as specific types of quantitative predictions of the system [10]. Therefore, it becomes critical to carry out these activities while manipulating the moderator variables that could produce a valid and reliable research database of the Network-level comprehensive Analytical Model. Video-over IP is generally forecast to be a substantial portion of the future IP network. We realize that only little was done on traffic engineering for Video-over IP ([2], [10], [11], [26]). This is partly due to the fact that a traffic-engineering model of Video-over IP does not provide sufficient bandwidth provisioning to ensure adequate network QoS. However, this becomes crucial when broadband digital networks are emerging. Therefore, the real design challenge is to engineer the network traffic that carries a mix of Video-over IP and other IP traffic flows. To counter this trend, a Network-level Analytical Model is required for Video-over IP to see the impact of quality and quantity measures of the inherent usability aspects of the underlying technology of an emerging TC business system ([2], [10], [11], [19]). Figure 1 depicts the Network-level comprehensive Analytical Model of the Video-over IP, which has two well-known distributions (such as Geometric and Pareto).

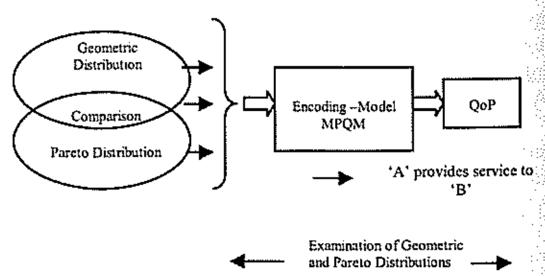

Fig. 1. A Comprehensive Analytical Model of Video-over IP

It is also depicted in Figure 1 that the comparison of both the Geometric and Pareto distributions would lead to a more accurate packetisation scenarios. Therefore, an examination of the basic premises of the comparison of both the Geometric and Pareto distributions of Video-over IP would significantly influence the application level activities of the system using a Moving Picture Quality Metric (MPQM) encoding model, as demonstrated in Figure 1 [24]. It is seen that traffic that emanates from a user (whether bursts or connections) be usually described by a source in the first stage. This is followed by the queues for describing the relevant queuing behaviour of IP packets at the second stage and multi-service requirement mechanisms (such as buffer sharing and partitioning, packet discard, queue-scheduling) for differentiated performance analysis at the third stage of the best-effort IP network ([9], [10], [11], [19], [21]). Finally a set of flows, connections and aggregates (with mechanisms such as admission control, policing, dimensioning and configuration) are depicted in the fourth stage to provide end-to-end QoS guarantee. It is expected that the order of demonstration would assist the users to understand the inherent usability aspects related to the network traffic dynamics of an emerging TC business system. The overall result is the creation of an indexed qualitative dataset by applying Pitt's methodology with a flow order that would allow more effective comparisons of the results of the network traffic dynamics of the system ([2], [6], [11], [13], [19]).

2. GEOMETRIC AND PARETO DISTRIBUTION IN VIDEO-OVER IP

Initially, Geometric- and Pareto-distribution-based source models are used individually. These alternative ways of presenting information about an arrival process is based on counting the number of arrivals in a defined time interval. For example, in discrete time, geometric inter-arrival times form a Bernoulli process, whereas, the Pareto distribution represents one of the distribution classes [19]. The geometrically distributed inter-arrival times form a Bernoulli process (with a group (batch) of packet arrivals instead of a single packet arrival) ([1, [19]). An M/G/Infinity queue is also used to match the queuing behaviour with the Geometric and Pareto distributions of the source models. Next, the input process of a discrete-time infinite server system fed by a discrete-time Poisson process of rate λ (customers/slot) and with generic service time σ is taken under consideration ([17], [19], [24]). The formulas for the probability that the queue size is greater than the buffer size for Geometric- and Pareto-distribution-based source models (Equations i and ii respectively) using M/G/Infinity queuing are given as follows ([19], [21], [22], [24]).

$$P_{\lambda}[q_{\infty} > b] \approx \frac{\rho^2}{2}(1-\rho)(1+\gamma)^2\gamma^{2(1-\rho)b} + \rho^3 \exp\left(-2\frac{1-\gamma}{1+\gamma}(1-\rho)b\right) \quad ..(i)$$

Where ρ the utilisation, γ the session duration and buffer size $b=0,1,\ldots$.

$$P_{\lambda}[q_{\infty} > b] \approx E_{\alpha-1}\left(-\frac{(\alpha-1)E[\sigma]}{\Gamma(2-\alpha)}\frac{1-\rho}{\rho^{\alpha}}b^{\alpha-1}\right) \quad ..(ii)$$

Where, R – Rate of arrivals, E[on] – Mean number of arrivals in ON state, C – Service rate, E[off] – mean number of time units in OFF state and α = activity factor of voice source. The Geometric distribution measurements of the probability that the queue size is greater than the buffer size involve the utilisation values along with the corresponding arrival rate of packets [22]. The buffer size used for Geometric distribution measurements is 100 and the service time is 0.8 ([19], [21], [24]). It is important that the value of arrival rate of packets is calculated by simply dividing the utilisation value by the service time. Figure 2 depicts the results of plotting the tail probability for an M/G/Infinity queue with Geometric and Pareto distribution. The utilisation values taken are 0.1, 0.4, 0.6 and 0.8. It is demonstrated in Figure 3 that for an utilisation value of 0.8 with a session duration (or service time) of 0.8 and buffer size of 100, the probability that the queue is found to be greater than the buffer size is approximately 6 x 10-3. However, the linear decrease of the probability suggests an exponential decay of the queue length distribution ([21], [22], [24]). Although the geometric distribution seems to be sufficient in representing traffic with single arrivals that are less than 100, Figure 3 depicts the model's inability to represent more than 100 arrivals. However, the probability is that the queue size is found to be greater when buffer size drops dramatically. This could be a problem when the traffic is highly variable and the traffic arrival comes in batches. Another issue with this distribution is its inability to represent traffic with long sojourn times.

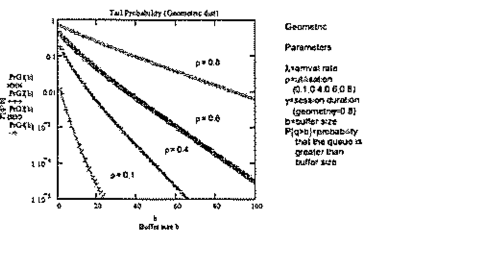

Fig. 2. Geometric Distribution (b = 100 and γ = 0.8)

Figure 3 depicts that the model becomes impractical when the service-time value approaches or is equal to 1. For this reason, a Pareto distribution model is chosen to represent the highly variable traffic, that is non-negligible probabilities for large batch sizes, or long sojourn times ([19], [22], & [24]). The Pareto distribution measurements of the probability that the queue size is greater than the buffer size involve the utilisation values along with the corresponding arrival rate of packets. The buffer sizes used for Pareto distribution measurements are 100 and 1000 and the service times are 1.5 and 1.7 ([19], [21], [22], & [24]).

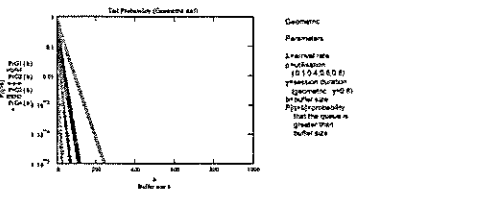

Fig 3. Geometric Distribution (b = 1000 and γ = 0.8)

Figure 4 depicts the results of plotting the tail probability for an M/G/Infinity queue with Pareto distributed source. The log–log scale plot uses utilisation values of 0.2, 0.5 and 0.8, with a buffer size, b of 100. It is important to know that the service time is also termed as session duration, and the value of μ used is very much greater than 1 ([19], [21], [22], [24]).

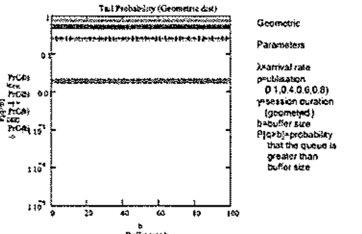

Fig 4. Geometric Distribution (b = 100 and γ = 1)

Figure 5 shows the results of plotting the tail probability for an M/G/Infinity queue with Pareto-distributed source. The log–log scale plot uses utilisation values of 0.2, 0.5 and 0.8, with a buffer size, b of 1000 and service time of 1.7. It is apparent that Pareto distribution is more suitable for representing highly variable traffics. It is also seen from Figure 6 that the heavier μ=1.5 induces larger tail probabilities than μ =1.7, whilst keeping the utilisation values the same ([1], [19], [22], [23], [24]). Next, by using the buffer analysis in packet-based networks the evaluation of probabilities associated with information loss and delay becomes very convenient. Hence, the state probability is concentrated on packet arrivals. The formulas for the buffer analysis of an individual queue depict the auto-convolution for N buffers sharing the buffer space, the estimated loss probability, and the state probability for an individual queue ([1], [19], [21], [22], [24]). Initially Auto-convolution of geometric distribution is given by a negative binomial is given as

$$P_N(k) = {}^{k+N-1}C_{N-1}\times(d_r)^k\times(1-d_r)^N \quad \text{....(iii)}$$

Where k means the size of the combined queues in that shared buffer. Next, the Probability that the shared buffer overflows is given as

$$Q_N(k-1) = 1-\sum_{j=0}^{k-1}P_N(j) = \sum_{j=k}^{\infty}P_N(j) \quad \text{....(iv)}$$

Where, R – Rate of arrivals, E [on] – Mean number of arrivals in ON state, C – Service rate, E [off] – mean number of time units in OFF state and α = activity factor of voice source ([19], [21]). Next, a geometric approximation could be applied to the tail of the negative binomial (i.e taking the geometric parameter, q from the ratio of successive queue state probabilities) is given as

$$Q_N(k-1) = \sum_{j=k}^{\infty}P_N(j) \approx \sum_{j=k}^{\infty}P_N(k)\times q^{j-k} \quad \text{....(v)}$$

$$q = \frac{P_N(k+1)}{P_N(k)} = \frac{{}^{k+N}C_{N-1}\times(d_r)^{k+1}\times(1-d_r)^N}{{}^{k+N-1}C_{N-1}\times(d_r)^k\times(1-d_r)^N} \quad \text{...(vi)}$$

Where k means the size of the combined queues in that shared buffer.

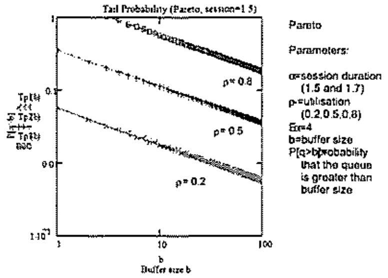

Fig 5. Pareto Distribution (b = 100 and μ = 1.5)

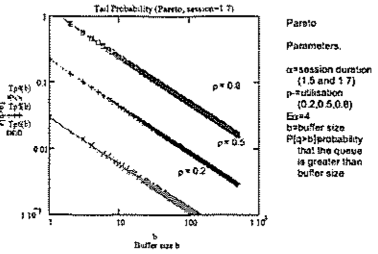

Fig 6. Pareto Distribution (b = 1000 and μ = 1.7)

It is important to know that the focus is a geometrically distributed inter-arrival time, because a closed-form solution is still obtainable ([1], [19], [22], [24]). However, it is still highly recommended that a Pareto distribution be applied in order to represent traffic that exhibits self-similarity and takes into account long-range dependence, as seen earlier ([14], [15], [16], [19]). Therefore, for the calculation of the packet-loss probability and the delay distribution of real-time packets a new aggregated model is taken under consideration. A two-state Markov chain used in this model supports the error process at the packet level, and Automatic Repeat Request (ARQ) for error correction. It is important to remember that there are actually two types of closed-form solutions used in this model ([6], [7]). The first type involves arrival rate α, while the second type involves a load parameter λ. A finite buffer seems to be sufficient, since the packets exceeding the delay limit are discarded and there is at most one arrival at each slot ([1], [10], [19], [22]). An interesting property of these solutions is the interchangeable correlations of error and arrival without affecting the packet-loss probability. Hence, when arrival correlation dominates, the approximate formula for packet-loss probability degenerates to one without correlated errors ([1], [7], [10], [19], [22]). The shared buffer measurements list the numerical results for convolution of 8 buffers with a very minor discrepancy in the comparison of exact convolution and negative binomial expression using a decay rate value of 0.78997. The numbers of buffers used are

2, 4 and 8 respectively ([1], [7], [19], [21], [22]). It is important that having separate buffers per output port is less efficient compared to a number of output ports sharing the buffer space ([1], [7], [10], [19], [22]). Figure 7 depicts the relationship of state probability against the total buffer capacity.

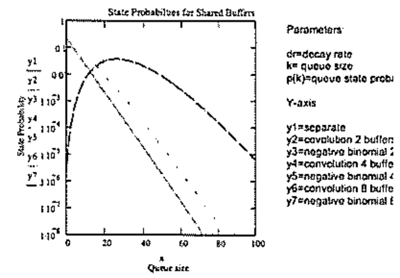

Fig 7. State Probability for Shared Buffers

Figure 8 depicts the relationship of overflow probability against the total buffer capacity. From Figures 8 and 9, it could be observed that by using a decay rate of 0.78997, for a queue size of 80, the state probability is found to be 1.4 x 10^{-4}, and overflow probability is found to be 8.3 x 10^{-4}.

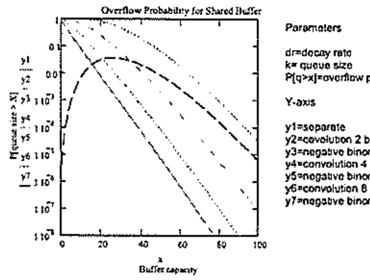

Fig 8. Overflow Probability for Shared Buffers

Figure 9 depicts the relationship of packet-loss probability against delay, with arrival rate (α) of 0.9 and probability from failure state to success state of 0.1. Because the error probability is assumed to be small and errors occur in bursts (due to high correlation), it is expected that long error-free periods interleave with relatively short error periods. In a long error-free period state, all packet transmissions are considered successful and consequently the queue lengths would likely decrease to zero before the next error burst arrives ([1], [7], [10], & [19]).

However, the load in the system should not be very high. For situations with a very high load, the queue length may not be able to reach zero (due to very few empty slots) before the next error burst occurs [7]. For this reason, the packet loss could be significouldtly larger. However, in an error period state, packet discard occurs if the delay threshold is exceeded. The chance of packet dropping depends on the queue length at the beginning of the error period and the value of the delay threshold ([6], [7], [19]).

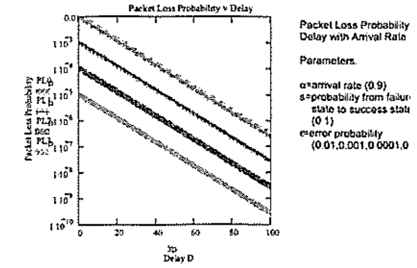

Fig 9. Packet-loss Probability vs. Delay (α = 0.9)

Figure 10 depicts that as delay increases, the packet-loss probability decreases exponentially.

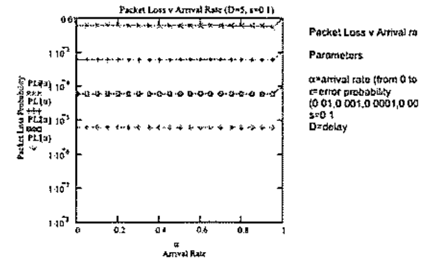

Fig 10. Packet-loss Probability vs. Arrival Rate (D = 5)

So, for a delay threshold of 50, the packet-loss probability is approximately 10^{-4}, with an error probability state of 0.01, as demonstrated in Figure 11. The measurements for packet-loss probability with varying delay thresholds and error probability. For a delay threshold of 40 with error probability of 0.01, the packet-loss probability is approximately 1.3 x 10^{-4}, whereas for a delay threshold of 40 with error probability of 0.001, the packet-loss probability is 1.5 x 10^{-5}. Similarly, the measurements for packet-loss probability under different error states with varying delay thresholds. Here, the constant values of arrival rate α and s are taken ([1], [7], [10], [19], [21], & [22]).

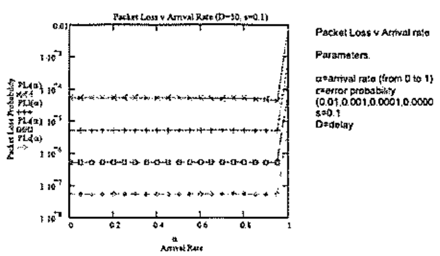

Fig 11. Packet-loss Probability vs. Arrival Rate (D = 50)

Figure 11 demonstrates the relationship of packet-loss probability against arrival rate for D value 50, whereas Figure 12 depicts the near-linear relationship between packet-loss probability and error probability when α is 0.9. It is seen from Figures 12 and 13 that a straight line with slope of zero is generated, except when the arrival rate α approaches 1. This means that the packet-loss probability is almost independent of the arrival rate. But when the arrival rate approaches 1, the packet-loss probability increases sharply until it reaches the error probability with an arrival rate of 1. However, at these arrival rates, nearly every visit to the failure state would result in a packet loss, and hence the error probability is the same as the packet-loss probability ([6], [7], [10], [19], [22]).

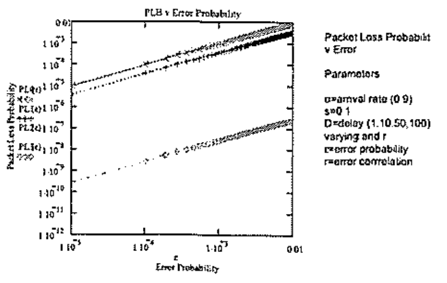

Fig 12. Packet-loss vs. Error Probability (α = 0.9)

It is also observed from Figures 12 and 13 that the packet-loss probability is much larger when the delay threshold, D (tolerable delay), is not sufficiently large when the errors are highly correlated. The measurements for packet-loss probability with varying delay thresholds D and error correlations r using the second closed-form solution, which uses load as a parameter. It is important that λ and arrival correlations v are fixed ([7], [10], [19], [22]). Figure 13 depicts the relationship of packet-loss probability against delay, with fixed load value λ 0.9 and error probability of 0.01. Since λ and ε are fixed, the correlations of errors and arrivals are determined by the values of r and v. The values of the error correlation r used are: 0.0001, 0.001, 0.003 and 0.005. v parameter on the other hand, which describes the arrival correlation, is held constant at a value of 0.002. As with the previous results, it could be observed that increasing the tolerable delay (threshold value) decreases the packet-loss probability exponentially. With a moderate delay and error probability, the packet-loss probability is acceptably low when the load is high and the correlations are moderate [1] ([7], [10], [19], [22]). Figure 14 depicts the relationship of packet-loss probability against delay, using a load of 0.8 with equal values of r and v, which means they are perfectly correlated. As seen below, the lines are almost parallel to the horizontal axis when they are equal to or smaller than 0.01. It is also demonstrated from Figure 15 that, for a moderate delay and error probability, the packet loss probably is significantly higher when the load is high and there is a low value of correlation. The packet-loss probability obtained even higher values for correlation r = v = 0.001 [5] [6] [19].

The measurements for packet-loss probability with varying delay thresholds D, arrival correlations v and error correlations r, using the second closed-form solution, which uses load as a parameter. It is important that λ and error probabilities are fixed. Similarly, the measurements for packet-loss probability, with varying delay thresholds D and error probabilitiesλ, arrival correlations v and error correlations r, are fixed ([7], [10], [19], [20], [22]).

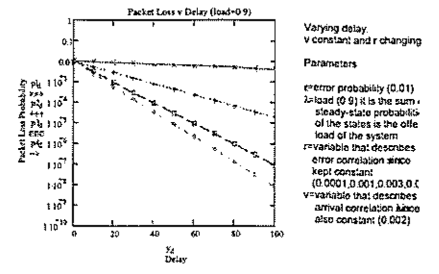

Fig 13. Packet-loss Probability vs. Delay with load (λ = 0.9)

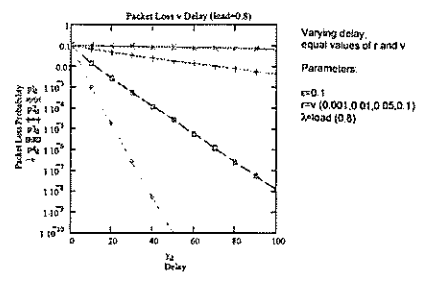

Fig 14. Packet-loss Probability vs. Delay (λ = 0.8)

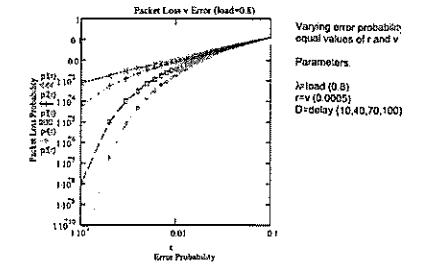

Fig 15. Packet-loss Probability vs. Delay (λ = 0.8)

Figure 15 depicts the packet-loss probability against error probability with fixed load of 0.8 and varying delay thresholds [20]. Similar to the previous graphs, the arrival and error correlations are perfectly correlated. Log scale is used for both axes to depict the results of small error probability more clearly. As the Figure 15 shows, it is apparent that, as error probability decreases, the packet-loss probability decreases rapidly. The rate of decrease depends on the delay threshold, that is, the larger the delay, the higher the decrease rate [20]. Finally, an encoding model MPQM is used as a quality-assessment tool, focusing more on the user perception of video quality. In the presented model, the relationship between the user's-perceived quality and average bit rate is described as the effect of packet loss on the user.

Verscheure's, Moving Pictures Quality Metric (MPQM) block diagram, and the quality scale are used for subjective testing for video quality and the measurements for the proposed encoding model [24]. Figure 16 demonstrates the relationship between Packet Loss Ratio (PLR) and the quality rating of MPQM [24]. It is seen in Figure 16 that, on a semi-logarithmic scale and for a provided quantiser factor (MQUANT), initially the video (encoding) quality remains constant with the PLR [24]. However, beyond a certain PLR, the perceptual quality seems to be dropping quickly. It is also seen that the higher the MQUANT value, the higher the PLR value, after which the quality drops.

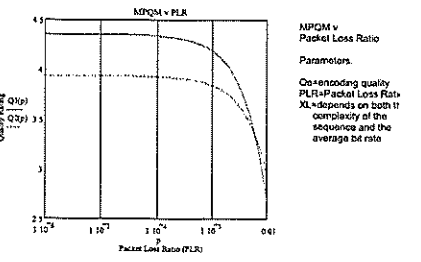

Fig 16. MPQM vs. Packet-loss Ratio

Therefore, the indexed qualitative data set for Video-over IP could be developed. Initially this may depict the approximation of the probability that the queue is greater than the buffer size with regard to utilisation and arrival rate (or session duration). By knowing this probability, a good estimate of the buffer size is obtained, which would be very useful for analysing the performance of sharing buffers. This is followed by delay parameter, a major factor in determining the packet-loss probability set to the threshold (tolerable delay) to the largest acceptable value [20]. This indexed dataset may also depict the error probability, which has a significant effect in the packet-loss probability. Finally, the packet-loss probability may be fine-tuned by changing the margin bandwidth reservation if the tolerable delay is not too small ([1], [7], [10], [20], [21], & [22]). However, the bandwidth parameter is not considered at this point. Further investigation of bandwidth parameter for various reasons would add value to the data set. The mapping process based on the packet-loss probability estimated follows this. The quantiser factor and the bit rate also play a key role in tuning the quality rating. It is important to realise that increasing the coder bit rate does not necessarily enhance the image quality. As a matter of fact, quality degradation may even be produced when bit rate exceeds a certain threshold. Hence, when considering video transmission over lossy networks, the bandwidth being consumed increases the encoding bit rate above a certain threshold. This may be due to the saturation of quality [24]. The purpose of the indexed qualitative data set obtained for Network-level Comprehensive Analytical Model is to generalise network QoS characteristics statistically in accordance with specified guidelines ([10] & [11]). It is expected that the data set within the context of inexpensive, reliable and accessible TC environments would be sufficient to tackle the network dimensioning problem, provided there are a number of possible known and unknown QoS parameters [19]. It is proven that the comparative analysis of the results, through the diagnostic and prognostic nature of the data set, provides a concurrent and retrospective review of preventive technology interventions of Internet capabilities [3]. The synthesis realisations are also compatible with the true underlying network process, which would assist the users to provide solutions for the inherent usability aspects of the system. Application-level activities in particular could enhance the Quality Assurance (QA) for the users, besides assisting them with resource allocation during the system pre-deployment period.

One of the untested assumptions of current collaborative practices in TC business system environments involves the challenge of dealing with the necessary business and knowledge-driven qualitative data along with the expected results of the proposed analytical model of Video over IP. This is a field of research that has yet to produce a practice-oriented consensus on how these results could support high-level usability awareness among the users of TC business systems to engender successful collaborative businesses [8]. Some results might even support user interpretations in the distinct phase of the knowledge-based visualisation and the contextual characteristics of their occurrence [9]. The decomposing of small chunks of the proposed comprehensive analytical modeling concepts could foster the measurement of user experience within the TC community to make their collaboration more productive.

3. SUMMARY

It is expected that by applying indexing of the qualitative data set of Video over IP technique, potential failures at application layer could be reduced prior to the video delivery. However, the accuracy of measurement of network QoS and mapping into their equivalent QoP largely depends on the multiple field surveys and laboratory validation tests that constitute a part of prediction process of the synthesized data set. In conclusion, the status of the proposed analytical model directly depends on the complexity of the system under investigation as well as on the availability of full statistical knowledge. Hence, the measure of user experience in particular would allow the users to construct usability patterns for the TC business system environments under investigation. This includes generating a synthesised data set as an important stage in the simulation process.

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Rough Number Assisted Decision-Making under Uncertainty in Quality Function Deployment

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Abstract — **Decision-making in the front-end phases of product development involves much vague design information such as customers' expectations and designers' assessments. Quality function deployment (QFD) is an effective means to process such vague design knowledge in order to develop quality products that can satisfy customers' requirements. Very often, QFD uses fuzzy numbers to represent and analyse the vague and subjective perceptions of customers and designers. However, besides the torment in determining appropriate fuzzy membership functions, the rapid enlargement of resultant fuzzy boundary intervals is another big problem in developing QFD, which will result in low discerniblility of design objectives. In view of this, this work proposes a novel concept of rough number to represent and analyse the vague design knowledge. A comparative case study has shown that the proposed rough number concept not only can save the effort in determining fuzzy membership functions, but also can suppress the enlargement of boundary intervals in developing QFD.**

Keywords - vagueness and uncertainty; rough sets; rough numbers; decision-making; product development; quality function deployment

I. INTRODUCTION

Product development is a complex business process involving extensive decision-making activities which usually have to deal with both quantitative and qualitative data/information. During a product development process, especially at the front-end phases of product development, an effective and efficient decision support system can play a crucial role because as much as 70-80 percent of a product's overall costs are determined at the early design phases. Better and more informed early design decisions are therefore crucial to lowering overall product development time and cost. However, the decision-making at the early phases of product development usually involves much vague and uncertain design information such as customers' linguistic expectations and designers' subjective assessments, which cannot be effectively utilised in the decision support system. In such a circumstance, a key challenge in the early phases of product development is to describe and quantify such vague and uncertain design information in the right form of data effectively and efficiently.

On the other hand, customer-oriented product development has become vital for manufacturing companies to maintain competitive in today's global marketplace. As a result, many useful tools have been developed to assist product development so that designers can better understand customer expectations and subsequently fulfil their needs by incorporating appropriate technical measures throughout the product development process. In this respect, quality function deployment (QFD) establishes itself as one of the most useful tools to translate the vague design information about customers' requirements and designers' assessments into

Foreword

The Eighth International Conference on Intelligent Technologies (InTech'07), to be held in Sydney from 12 to 14 December 2007, is hosted by University of Technology Sydney (UTS) and the ARC Centre of Excellence for Autonomous Systems (CAS). It is the first time this annual conference takes place in Australia.

Previous InTech conferences have been the venue for internationally-recognised researchers in applied mathematics and artificial intelligence producing excellent work in the areas of Fuzzy Logic, Knowledge-Based Systems, Neural Networks, Learning/Adaptive Systems, Data Mining, Uncertainty Processing and Soft Computing, Intelligent Data Analysis, Control and Decision Science.

InTech'07 merges its tradition in laying mathematical foundations of intelligent technologies with the conference theme on "*Intelligent Technologies in Robotics and Automation*", reflecting an increasing interest in real-world applications. The conference's technical programme covers also recent advances in information technologies, business and finance management, biomedical engineering, control and power engineering, sensing, vision and image processing.

The International Programme Committee and invited reviewers were working very hard to return all paper reviews in time, with each paper being peer-reviewed at least by two specialists in the relevant area. For this, we are indebted to all colleagues involved. Out of over 70 paper submissions and interest registrations, the Committee has accepted 40 regular papers for presentation and 3 poster papers. In addition to ten regular sessions, InTech'07 features also four plenary talks, two invited papers, a Gala dinner speech, and an expert panel discussion.

The generous sponsorship obtained from the UTS Faculty of Engineering and the UTS node of the Centre for Autonomous Systems is gratefully acknowledged. We would like to sincerely thank our colleagues from University of Texas at El Paso, USA, and from the InTech founding institution, Assumption University, Thailand, for their invaluable cooperation. Many thanks go also to staff and students at the host institution, who have helped and supported the conference in one way or another. Lastly, we would like to express our heartfelt thanks to all the authors, participants and esteemed guests for their solid contributions and making for this unique event.

We hope InTech'07 will be successful, fruitful to all of you, and would like to wish you a joyful stay in Sydney.

Assoc. Prof. Q. P. Ha
InTech'07 General Chair
UTS-CAS, Nov. 2007

Panel Discussion

The International Conference on Intelligent Technologies (InTech) has traditionally had a strong focus on the development of mathematical foundations for Intelligent Technologies. Recent InTech conferences have been the venue for internationally-recognised researchers in applied mathematics and artificial intelligence producing excellent work in the areas of Fuzzy Logic, Knowledge-Based Systems, Neural Networks, Learning/Adaptive Systems/Data Mining, Uncertainty Processing and Soft Computing, Intelligent Data Analysis, Control and Decision Science.

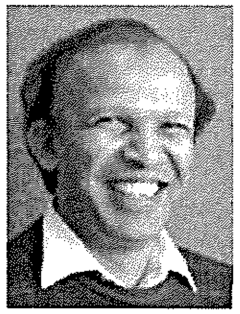
InTech'07 is complementing these with intelligent technology developments for applications in information technologies, business and finance management, biomedical engineering, control and power engineering, sensing, image processing and the conference mainstream of robotics/mechatronics and automation. Towards productive amalgamation of sciences and applications in real world deployment of intelligent technologies, a Panel Discussion will be organised on the theme:

Intelligent Technologies: Bridging the Gap between Sciences and Applications

Facilitator:

Professor Gamini Dissanayake, University of Technology Sydney, Australia
Centre for Autonomous Systems, UTS Node Director

Biography:



Gamini Dissanayake is the James N Kirby Professor of Mechanical and Mechatronic Engineering at University of Technology, Sydney (UTS). He has expertise in a broad range of topics in robotics including; robot localisation, mapping and simultaneous localization and mapping (SLAM) using sensors such as laser, radar, vision and inertial measurement units; terrain mapping; multi-robot coordination for SLAM, target tracking and probabilistic search; motion planning for single and multiple robot manipulators, legged robots, and cranes; and application of robotic systems in urban search and rescue. He leads the UTS node of the ARC Centre of Excellence for Autonomous Systems. He graduated in Mechanical/Production Engineering from the University of Peradeniya, Sri Lanka in 1977. He received his M.Sc. in Machine Tool Technology and Ph.D. in Mechanical Engineering (Robotics) from the University of Birmingham, England in 1981 and 1985 respectively.

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