Estimating the Deliverable Quality of a Fully Redundant Dispersity Routing System

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Abstract—**The public Internet in its current form does not provide consistently the levels of service that real-time services such as Voice over Internet Protocol (VoIP) demand. Indeed, the scope of this gap is such that quality and reliability problems are characteristic of these services. Fully redundant dispersity routing exploiting the path diversity readily available in the Internet is one approach of mitigating these quality and reliability problems. This paper presents a model for estimating the quality that may be expected from fully redundant dispersity routing systems using paths with known packet loss and loss burstiness characteristics. That model is then applied to estimate the quality that may be expected from fully redundant dispersity routing systems of 2 – 6 paths and, for contrast, to the estimated quality that may be expected from single path systems. The insights gained by this application may be useful when selecting paths for a fully redundant dispersity routing system to satisfy some quality goal. A brief study into the accuracy of the model indicates that for two paths, 50% of the estimations are within 0.05 of the simulated Mean Opinion Score (MOS), and 98% within 0.32.**

Keywords: ***Dispersity Routing; QoS; Telephony; VoIP***

I. INTRODUCTION

While the quality of Voice over Internet Protocol (VoIP) services over the public Internet is satisfactory most of the time, quality and reliability problems are sufficiently frequent to be characteristic of these services. In some cases these problems may be so severe and persistent that users revert to traditional telephony, despite its (usually) higher cost.

Simulations using real VoIP traffic data measured in a commercial call center show that fully redundant dispersity routing can improve the quality of VoIP [1]. Using the E-model [2] to measure quality as a Mean Opinion Score (MOS) [3] objectively, these simulations establish that two paths may already increase the proportion of calls with a 'very satisfied' rating [2] from the observed 84.1% to 99.9%. Clearly, fully redundant dispersity routing has the potential to increase the quality of VoIP to be more on par with traditional telephony than VoIP is currently [1].

The main contribution of this paper is a model for estimating the quality that may be expected from a fully redundant dispersity routing system using paths with known packet loss and loss burstiness characteristics. The value of this model is that not only does it obviate the need for time-consuming simulations to arrive at a quality estimate; it also offers these estimates without needing to observe real VoIP traffic data first. Furthermore, since the model does not rely on observing actual VoIP traffic data, it is not bound by that which can be observed. Instead, estimates may be made for scenarios that have not yet been observed, such as very high loss scenarios.

Another contribution of this paper is an application of the proposed quality estimation model to the packet loss and loss burstiness characteristics measured in a commercial call center. That application estimates the quality that may be expected from fully redundant dispersity routing systems of 2 – 6 paths. Together with an estimate of the quality that may be expected from a single path system, these estimates may be of use as a planning tool. By conveying a sense of the relationships between packet loss, number of paths and user experience interpretations, they enable projections such as (1) the highest loss rate tolerable by a system expected to satisfy some quality goal with a given number of paths, and (2) the number of paths needed to meet a particular quality goal.

Path diversity has been exploited in [4] and [5] as non-redundant dispersity routing, and [6]–[9] as path switching. In contrast, this paper exploits path diversity as fully redundant dispersity routing. While the MOS is used in this paper as a quality indicator as in [6]–[9], [4] use the Noticeable Loss Rate in their study of packet dispersion on VoIP quality. This paper, unlike [4]–[9] however, uses real measured VoIP traffic data.

The remainder of this paper is structured as follows. Section II presents the background discussing alternative approaches to dispersity routing, dispersity routing itself in its various forms, and ends with a summary of the MOS as an objective method for quantifying quality. Section III then presents a model for estimating the quality that may be expected from a fully redundant dispersity routing system. Next, section IV applies the model to estimate the quality that fully redundant dispersity routing is likely to deliver, and section V describes a brief study into the accuracy of the model. Section VI concludes the paper.

II. BACKGROUND

This section begins with an introduction of dispersity routing alternatives. Next, dispersity routing is outlined, including the form used in this paper. The section concludes with a summary of the MOS as a means of quantifying quality.

A. *Forward Error Correction*

Forward Error Correction (FEC) approaches add redundant data into the data stream at the source, for use by receivers to recover any lost data, at least in part [10]. However, when delivering both the data and the redundant data along a single path, this approach increases demand on that single path. Furthermore, for real-time communication care must be taken to ensure that the redundant data is available and can be used to recover any lost data within the time constraints of the real-time communication. As loss is usually bursty [11][12], and the duration of these loss bursts may exceed these time constraints, FEC techniques may not be able to mask all failures in a real-time environment [1].

B. *Path Switching*

Path switching exploits path diversity just as dispersity routing does in order to mitigate quality and reliability problems. However, rather than actively replicating the data along multiple paths concurrently, path switching relies on maintaining backup paths, and then switching to a backup path when detecting a problem on the current path [6]–[9].

Switching may also occur preemptively when better performance is predicted on another path, in order to avoid the outage that would occur when reacting to a problem only once the problem has occurred. By switching preemptively, an outage is avoided that begins when the degradation occurs, and continues until the degradation is detected and the switch to an alternative path completes. In order to avoid switching to a path that is about to experience the same, or even worse, degradation than the current path, the ability to predict accurately which paths will give better performance over long time scales is helpful. Indeed, it is sufficiently helpful to warrant actively probing the backup paths for increased accuracy, despite probing consuming resources of its own.

C. *Dispersity Routing*

In contrast to path switching which seeks to use the best of many possible paths by switching among them, dispersity routing uses many paths in parallel. Maxemchuk [13] identifies three forms of dispersity routing.

Non-redundant dispersity routing harnesses the combined resources of multiple paths. The data to be communicated is divided among the paths such that each path is given a subset, and the set of paths collectively communicates a single instance of the data. Fully redundant dispersity routing uses multiple paths to communicate the data, with each path given a full copy of the data. Given N paths, a total of N copies of the data are sent, one instance for each of the N paths. Partially redundant dispersity routing seeks to balance the performance gains possible using non-redundant dispersity routing with the quality gains possible using fully redundant dispersity routing. This is achieved by encoding subsets of the data into blocks using techniques such as erasure codes and then sending these blocks along the set, or subset, of paths.

The last two forms seek to exploit the de-correlated failure behavior of diverse paths to mitigate failures on individual paths. For brevity, in the remainder of this paper dispersity routing refers to fully redundant dispersity routing.

TABLE I. USER SATISFACTION INTERPRETATIONS OF MOS ESTIMATES

MOS	**User Satisfaction**
4.34	Very satisfied
4.03	Satisfied
3.60	Some users dissatisfied
3.10	Many users dissatisfied
2.58	Nearly all users dissatisfied

D. *Mean Opinion Score*

The Mean Opinion Score (MOS) [3] is a measure of quality, with 1 being the lowest and 5 being the highest ranking as perceived by a human. Table I enumerates 5 user satisfaction interpretations and the lowest MOS at which these may be expected [2], associating a real-world meaning to a score. To arrive at a MOS objectively, the E-model [2] computes a MOS estimate from 21 parameters equating to observable telephone system characteristics such as latency, loss, burstiness, and codec. Of the characteristics relevant to VoIP, the most significant threat to quality at the packet level is packet loss [1][14]. Furthermore, the distribution of that packet loss affects perceived quality; packet loss that occurs in bursts is perceived to have more of a detrimental impact on quality than packet loss that occurs randomly.

III. A MODEL FOR ESTIMATING DELIVERABLE QUALITY

This section begins by describing the architecture of a dispersity routing system. The section then presents a model for estimating the quality that a dispersity routing system is most likely to deliver, given paths of known packet loss and loss burstiness characteristics.

A. *Dispersity Routing System Architecture*

Fig. 1 depicts the architecture of a dispersity routing system for communicating a stream of one or more packets. Packets entering the system (on the left) are replicated such that for a system of N paths there are exactly N instances of each packet. The system gives one packet instance to each path for delivery, each path being given exactly one instance of every packet in the stream.

Although the latency of a path is likely to differ to that of other paths, ideally a dispersity routing system uses paths with comparable latencies [1]. Paths may lose packet instances, packet instances may experience variations in latency (that is,

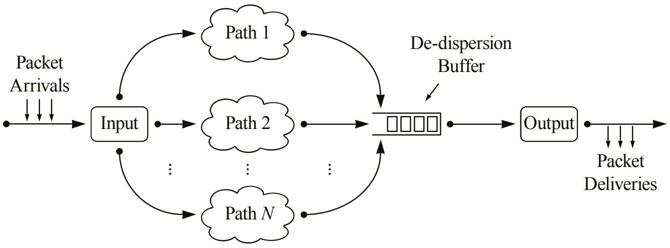

Figure 1. Dispersity routing system of N paths. Packets enter the dispersity routing system on the left, traverse all N paths concurrently, pass through a de-dispersion buffer, and leave the dispersity routing system on the right.

jitter), and packet instances may arrive out of order. Packet instances that successfully traverse a path enter a de-dispersion buffer, which discards all but the first instance to arrive of each packet and schedules delivery of the rest from the system.

The de-dispersion buffer may adopt a delay, similar to a de-jitter buffer except that any packet instances that miss the delay window are not discarded as late arrivals. This de-dispersion buffer delay compensates for differences in the path latencies causing jitter and out of order packets [1]. However, in this paper no de-dispersion buffer delay is adopted and the buffer delivers packet instances as soon as they arrive. Therefore, the latency of the dispersity routing system may be estimated simply as the minimum path latency.

B. Estimating Deliverable Quality

To estimate the deliverable quality that a dispersity routing system is most likely to deliver, the packet loss and loss burstiness characteristics are computed for that system. Together with an estimate of the latency and knowledge of the codec used, the E-model may then compute a MOS estimate from these computed characteristics.

The loss and burstiness characteristics of a path may be modeled by a Markov model such as the 4-state Markov model [14]–[16] depicted in Fig. 2. This model distinguishes between periods of high loss and periods of low loss. A high loss period, known as a *loss burst*, is not necessarily a period of total loss; some packets during a loss burst may *not* be lost. Conversely, a low loss period, known as a *gap*, is not necessarily a period of absolutely zero loss; some packets during a gap may *not* be received. The four states of the Markov model describe the four possible combinations of loss burst and gap with packet loss and receipt.

Gaps and loss bursts are distinguished by defining the minimum number of consecutively received packets in a gap, called G_{min}. Any lost packet in a gap must be separated by at least G_{min} consecutively received packets from any other lost packet to be considered a part of a gap. A loss burst is any period that is not a gap. This paper adopts 16 for G_{min}, as recommended by [16].

Let the discrete state space $\mathcal{Z} = \{1, 2, 3, 4\}$ represent the states Gap Receive, Burst Receive, Burst Loss and Gap Loss respectively. Furthermore, let the state transition matrix $\mathbf{P}$ express the state transition probabilities, such that $p_{i,j}$, the element in row i and column j, is the probability of a transition from state $i \in \mathcal{Z}$ to state $j \in \mathcal{Z}$ occurring.

$$\mathbf{P} = \begin{bmatrix} p_{1,1} & \cdots & p_{1,4} \\ \vdots & \ddots & \vdots \\ p_{4,1} & \cdots & p_{4,4} \end{bmatrix} \qquad (1)$$

Unlike the typical *right stochastic matrix* where each row vector sums to unity (that is, $\sum_{j \in \mathcal{Z}} p_{i,j} = 1 \; where \; i \in \mathcal{Z}$), here all the elements of state transition matrix $\mathbf{P}$ sum to unity (that is, $\sum_{i \in \mathcal{Z}} \sum_{j \in \mathcal{Z}} p_{i,j} = 1$). The difference is that, instead of $p_{i,j}$ being the probability of, being in state i, going to j as opposed to the other states, $p_{i,j}$ is the probability of transitioning from state i to j as opposed to all other possible state transitions.

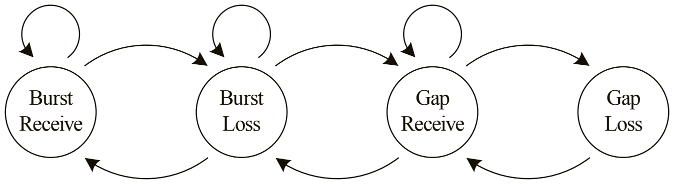

Figure 2. A 4-state Markov model considers periods of high loss as *loss bursts* and all other periods as *gaps*. Packets are lost in the *Burst Loss* and *Gap Loss*, and received in the *Gap Receive* and *Burst Receive* states.

It is clear that the probability of loss, l, is the sum of the probabilities of transitioning to the loss states Burst Loss and Gap Loss. Formally, given discrete state space $\mathcal{X} = \{3, 4\}$, that is $\mathcal{X} \subset \mathcal{Z}$, that represents the two loss states Burst Loss and Gap Loss respectively, $l = \sum_{i \in \mathcal{Z}} \sum_{j \in \mathcal{X}} p_{i,j}$. Besides the probability of receiving a packet, r, being $r = 1 - l$, r is also the sum of the probabilities of transitioning to the receive states Gap Receive and Burst Receive. For completeness, given discrete state space $\mathcal{G} = \{1, 2\}$, that is $\mathcal{G} \subset \mathcal{Z}$, that represents the receive states Gap Receive and Burst Receive respectively, $r = \sum_{i \in \mathcal{Z}} \sum_{j \in \mathcal{G}} p_{i,j}$.

The packet loss and loss burstiness characteristics of a dispersity routing system using a set of N, where $N \geq 2$, paths $\mathcal{P} = \{1, 2, \dots, N\}$ each characterized by state transition matrix $\mathbf{P}_{i \in \mathcal{P}}$, may be described by the Kronecker product of these matrices. For completeness, the packet loss and loss burstiness characteristics of a single-path (that is, non-dispersity routing) system may be characterized by the state transition matrix $\mathbf{P}_{i=1}$ of its only path. Therefore, the packet loss and loss burstiness characteristics of a system using a set of N paths, where $N \geq 1$, may be described by $\mathbf{W}$ as,

$$\mathbf{W} = \begin{cases} \mathbf{P}_{i=1}, & N = 1 \\ \bigotimes_{i=1}^{N} \mathbf{P}_i, & N \geq 2 \end{cases}. \qquad (2)$$

Let $Z = |\mathcal{Z}| = 4$ be the cardinality (that is, the number of elements) of set $\mathcal{Z}$, and $X = |\mathcal{X}| = 2$ be the cardinality of set $\mathcal{X}$. Clearly, X^N columns of $\mathbf{W}$ (that is, those representing a state transition to a loss state on all N paths) contain probabilities of simultaneous packet loss on all N paths. Therefore, the sum of these X^N columns is the probability of simultaneous packet loss on all N paths. Given matrices $\mathbf{P}_i$, where $i = \{1, 2, \dots, N\}$, let $\mathbf{p}_k^{(i)}$ be the kth column vector of the ith matrix $\mathbf{P}_i$. Furthermore, let function $f_{z_1, z_2, \dots, z_N}$ compute for $N \geq 2$ the index in $\mathbf{W}$ of the Kronecker product of the column vectors $\left\{\mathbf{p}_{z_i}^{(i)} : i = \{1, 2, \dots, N\}\right\}$. For $N = 1$, let $f_{z_1, z_2, \dots, z_N}$ equate to the identity function. Therefore, let $f_{z_1, z_2, \dots, z_N}$ be defined as

$$f_{z_1, z_2, \dots, z_N} = 1 + \sum_{i=1}^{N} (z_i - 1) \, Z^{(N-i)}. \qquad (3)$$

Formally, the set of indices of the X^N columns in $\mathbf{W}$ that represent a state transition to a loss state on all N paths then is

$$\mathcal{L} = \left\{f_{z_1,z_2,\dots,z_N} : z_i \in \mathcal{X}, i = \{1,2,\dots,N\}\right\}. \tag{4}$$

Therefore, the probability of packet loss by a system with these N paths — which is the probability of simultaneous packet loss on all N paths — is

$$P(loss) = \sum_{i=1}^{Z^N} \sum_{\mathrm{j} \in \mathcal{L}} w_{i,j}. \tag{5}$$

Given that the set of indices of the rows in $\mathbf{W}$ that do *not* represent a state transition *from* a loss state on all N paths is

$$\mathcal{R} = \{\mathrm{r} \in \{1,2,\dots,Z^N\} : \mathrm{r} \notin \mathcal{L}\}, \tag{6}$$

the probability of the system traversing from a receive state to a loss state is

$$P(burst) = \sum_{i \in \mathcal{R}} \sum_{\mathrm{j} \in \mathcal{L}} w_{i,j}. \tag{7}$$

Note that computation of $\mathbf{W}$ becomes expensive for large numbers of paths; given N paths, $\mathbf{W}$ is a $Z^N \times Z^N$ matrix. As computation of $P(loss)$ and $P(burst)$ requires only a subset of the elements in $\mathbf{W}$, computation may be simplified by computing only those elements actually needed.

The E-model characterises loss burstiness as a *burst ratio* that may be calculated using a 2-state Markov model [2] and which captures "very short-term dependencies between lost packets, i.e., consecutive losses" [15]. In the *loss* state of this model the probability of packet loss is 1 [15], as opposed to the *receive* state where the probability of packet loss is 0.

Let p be the probability of transitioning to the loss state from the receive state, computed as

$$p = \frac{P(burst)}{1 - P(loss)}. \tag{8}$$

The burst ratio for the E-model may then be calculated as

$$BurstR = \frac{P(loss)}{p}. \tag{9}$$

Having computed the packet loss probability (5), the burst ratio (9), and knowing the codec used, the E-model may then compute a MOS estimate. The default values recommended by the E-model [2] are adopted for all parameters except for those derived from the above.

IV. DELIVERABLE QUALITY OF DISPERSITY ROUTING

This section begins by describing a model that maps packet loss probabilities in the range 0 – 1 to state transition matrices of the 4-state Markov model used to model the packet loss and loss burstiness characteristics of a path. Using that loss burstiness model, this section then applies the model described in section III above to estimate the deliverable quality that may be expected from dispersity routing systems of 2 – 6 paths, and from single path (that is, non-dispersity routing) systems. A discussion of the results then concludes this section.

TABLE II. COEFFICIENTS OF LOSS TO STATE TRANSITION MAPPINGS

From	To	Coefficient 1	Coefficient 2	Coefficient 3
Gap Receive	Gap Receive	3.116E-01	-1.312E+00	9.999E-01
Gap Receive	Burst Loss	-6.974E-02	6.961E-02	1.246E-04
Gap Receive	Gap Loss	-6.772E-03	6.663E-03	1.084E-04
Burst Receive	Burst Receive	-2.072E-01	2.074E-01	-1.400E-04
Burst Receive	Burst Loss	-2.789E-02	2.791E-02	-2.175E-05
Burst Loss	Gap Receive	-6.974E-02	6.961E-02	1.246E-04
Burst Loss	Burst Receive	-2.789E-02	2.791E-02	-2.175E-05
Burst Loss	Burst Loss	1.044E-01	8.958E-01	-2.112E-04
Gap Loss	Gap Receive	-6.772E-03	6.663E-03	1.084E-04

A. Relating Packet Loss and Loss Burstiness

The loss burstiness model fits a second degree polynomial for each of the 9 state transitions possible in the 4-state Markov model used to characterize packet loss and loss burstiness in this paper. Each polynomial is a constrained linear least-squares fitting of a set of 13 624 points. A point comprises of (1) the packet loss probability (as the independent variable) observed for a real VoIP call measured in a commercial call center, and (2) the value of that polynomial's state transition probability (as the dependent variable) observed for that call.

All polynomials are constrained to pass through the state transition probability expected at packet loss probability 1. Since at that packet loss probability the only state transition possible is Burst Loss to Burst Loss, all polynomials are constrained to pass through state transition probability 0 at packet loss probability 1, except for the polynomial for Burst Loss to Burst Loss state transitions. That polynomial is constrained instead to pass through state transition probability 1 at packet loss probability 1, because the only state transition possible at 100% packet loss is from Burst Loss to Burst Loss. The coefficients thus established from the packet loss and loss burstiness characteristics measured for 13 624 VoIP calls are presented in table II for completeness. It can be shown analytically that the polynomials sum to unity.

B. Applying the Model

While the model for estimating deliverable quality accommodates paths with differing packet loss and loss burstiness characteristics, this section assumes that each path experiences the same characteristics for the sake of simplicity. Furthermore, to illustrate the significance of including loss burstiness characteristics, in this section MOS estimates are computed assuming both bursty and non-busty loss.

Let i be the number of paths, where $i \in \{1, 2, \dots, 6\}$, and let l be the packet loss probability, where $l \in \{0, 0.01, \dots, 1\}$. For each value of i, a MOS estimate assuming bursty loss and a MOS estimate assuming non-bursty loss are computed for each value of l. The MOS estimate assuming non-bursty loss is computed directly using the E-model. To compute the MOS estimate assuming bursty loss, the state transition matrix of the 4-state Markov model that models the packet loss and loss burstiness characteristics of the paths is computed first. That matrix is computed by evaluating for each of the 9 state transitions possible the second degree polynomial at loss rate l using its coefficients computed above. From the state transition matrix, the deliverable quality may then be estimated as a MOS estimate using the quality estimation model described above.

C. Discussion

Fig. 3 shows (from left to right) the model's MOS estimates for a single path system and dispersity routing systems of 2 – 6 paths. Each path experiences packet loss probabilities in the range 0 – 1. The solid curves plot MOS estimates assuming bursty packet loss, and the dashed curves plot MOS estimates assuming non-bursty packet loss. Furthermore, the horizontal dashed lines mark the minimum MOS values for the five user satisfaction experience interpretations of (from top to bottom) 'very satisfied' to 'nearly all users dissatisfied' as in table I.

The difference in Fig. 3 of the curves for bursty and non-bursty loss illustrates the significance of including burstiness when estimating the MOS. Here, estimates assuming random loss may be up to 0.6 higher than those assuming bursty loss.

For bursty loss, Fig. 3 illustrates that a user experience of 'very satisfied' may be achieved with a dispersity routing system of six paths despite each path experiencing loss probabilities up to 0.45. To deliver the same experience with two paths, those paths may not experience loss probabilities exceeding 0.09. The figure also shows that a dispersity routing system of two paths each experiencing loss probabilities up to 0.38 may deliver a user experience as low as 'nearly all users dissatisfied'. Adding one more path of the same packet loss and burstiness characteristics increases the user experience by two degrees to 'some users dissatisfied'.

Also evident in Fig. 3 is a change in the shape of the estimated MOS curves with an increasing number of paths. This change is due to the reduction in packet loss probability and loss burstiness that dispersity routing is able to achieve with additional paths.

Fig. 4 depicts the deliverable MOS estimates in terms of the user satisfaction interpretations enumerated in table I. For the interpretations of (from bottom to top) 'very satisfied' to 'nearly all users dissatisfied' as shown in table I, the highest tolerable packet loss probability is plotted for 1 – 6 paths. Diminishing returns of additional paths are visible clearly in Fig. 4 as the gains in packet loss probabilities decreasing with increasing numbers of paths.

V. ACCURACY OF THE MODEL

While the quality estimation model above computes the most likely MOS estimate for a particular dispersity routing system, being stochastic other outcomes are possible. This section considers the accuracy of the model by comparing the MOS estimated by the simulation of a dispersity routing system with an estimate by the quality estimation model of the deliverable MOS for that system. To simulate a dispersity routing system of N paths, this paper adopts the approach in [1]. However, as estimates and simulations for systems suffering no or little loss tend to agree, VoIP calls with little or no loss are avoided in this simulation. Therefore, the VoIP calls with the lowest 100 MOS estimates are selected as the pool for the simulation.

Let i be the number of paths in the range 2 – 6. For each value of i, up to 10 000 random and different combinations of i VoIP calls are selected from the pool. No call is selected more than once for a given combination. For two paths, all possible $\binom{100}{2} = 4950$ combinations are selected. The length of a combination is bounded by the shortest VoIP call adopted for one of its paths.

For each selected combination, a dispersity routing system with these i paths experiencing the packet loss and latency observed in the i real VoIP calls selected for this combination is simulated. The first packet traversing a path adopts the loss and latency observed by the first packet in the real VoIP call

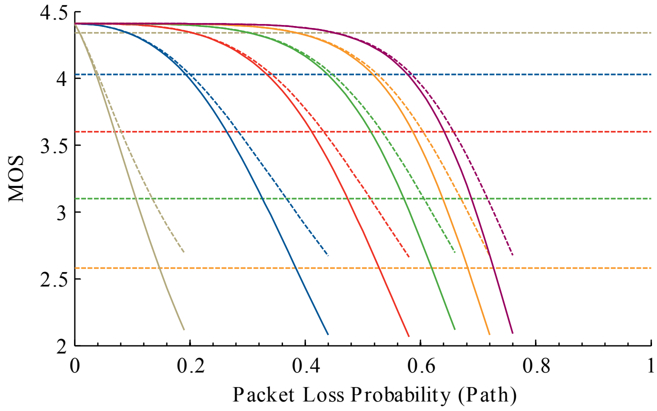

Figure 3. Deliverable MOS estimates for systems (solid curves from left to right) of 1 – 6 paths. Dashed curves show corresponding estimates assuming non-bursty loss. Horizontal lines mark minimum user satisfaction MOS.

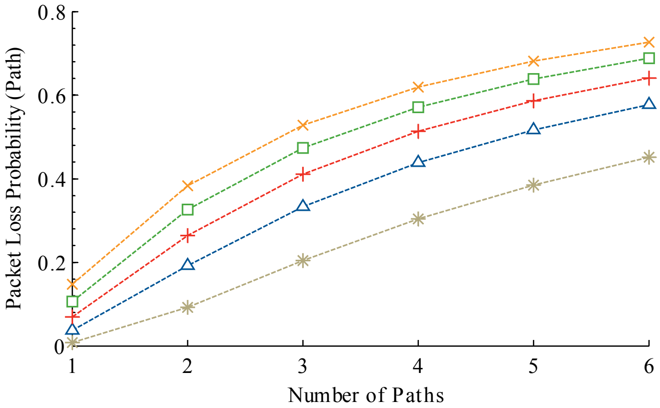

Figure 4. Maximum tolerable packet loss probabilities for the five user satisfaction interpretations of (from bottom to top) 'very satisfied' to 'nearly all users dissatisfied', for 1 – 6 paths adopting bursty loss characteristics.

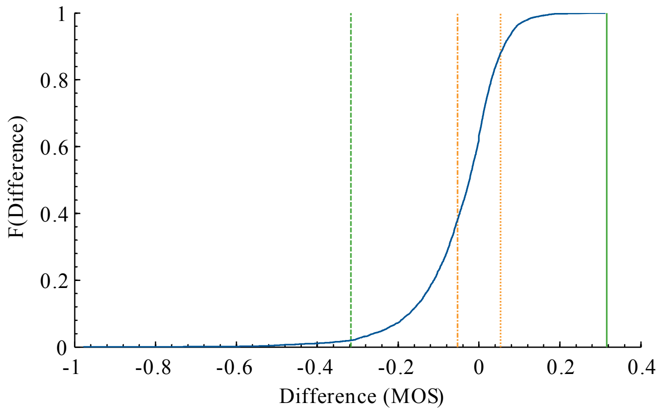

Figure 5. Cumulative distribution of differences between modeled and simulated. For each modeled a single combination is simulated. Also shown are (from inside to outside) the 50% and the 98% range.

selected for that path, the second packet traversing the path adopts those of the second packet in the real VoIP call, and so on. From the resulting output packet stream a MOS estimate is computed. In addition to obtaining a MOS estimate by simulation, an estimate from the quality estimation model is also obtained for each selected combination. The state transition probabilities for the 4-state Markov chain that describes the loss and burstiness characteristics of a path are computed easily from the VoIP call selected for that path.

Fig. 5 depicts the cumulative distribution of the differences between the simulated and the modeled MOS estimates for a system of two paths. The difference is computed as the simulated MOS estimate minus the modeled MOS estimate. Fig. 5 shows that 50% of the modeled estimates are within 0.05 of the simulated estimates, and 98% within 0.32. As the differences become smaller with increasing numbers of paths, a system of two paths represents the worst case.

This indication of accuracy is preliminary only, as for each combination a single simulation only is executed. The quality estimation model computes the most likely MOS estimate for a system using paths with those packet loss and loss burstiness characteristics. In contrast, each simulation computes a MOS estimate for the output of just one instance with those path characteristics. Other instances are possible that share the same path characteristics. Therefore, further work is needed that considers more instances of each combination.

VI. CONCLUSIONS

This paper proposes a model for computing the most likely quality deliverable from a dispersity routing system with known path characteristics. Applying the proposed quality estimation model to data extrapolated from a loss burstiness model built with data measured in a commercial call center, suggests a ‘very satisfied’ rating is achievable with six paths each experiencing loss probabilities just exceeding 0.45. The same application also helps to convey a sense of the quantities involved in dispersity routing, by establishing how much packet loss is tolerable with a given number of paths to achieve a given user satisfaction criteria. With this knowledge, the parameters of path characteristics, numbers of paths, and user satisfaction experiences may be weighed against each other in the search of a satisfactory combination.

Along with the model, a preliminary indication of accuracy is offered. While further work is needed to arrive at a stronger indication of accuracy, the indications offered show that the model is viable, and that 50% of estimates are within 0.05 and 98% of estimates are within 0.32 of the simulated MOS.

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