DEVELOPMENT OF A BURST MODEL FOR SPEECH TRANSMISSION WITH CELL LOSSES IN AN ATM NETWORK

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ABSTRACT

Speech quality, with cell losses in an ATM network, has been evaluated previously using random cell loss rates. However, it has been shown that cell loss behavior for speech transmission in ATM is more appropriately described by the burst model due to the burstiness properties of the aggregate voice cell arrival process.

A mathematical model, called the burst model for cell loss behavior in an ATM network, has been developed. The burst model predicts cell loss behavior very accurately as link capacity increases. According to the burst model, it has been calculated that SNRseg decreases by 1.5dB at the link load of 98% for the STS-3C link.

I. INTRODUCTION

Recently there has been a large amount of interest in the area of an integrated multimedia network (voice, video, data) communication such as Broadband ISDN (BISDN). Asynchronous Transfer Mode (ATM) is the switching and multiplexing technique associated with BISDN. An ATM network loses cells by network congestion or by transmission error. Here, our concern is cell losses by network congestion.

The evaluation of speech quality in ATM with cell losses has been performed previously using random cell loss rates[1]. However, the assumption of random cell losses in ATM is not realistic because once the queue is full, it remains full for a certain period, during which time many cells may be lost, resulting in consecutive cell losses. Analysis of information loss in packet voice systems has indicated that cell losses are more likely to occur in bursts rather than at random[2][3]. Therefore, speech quality with cell losses should be evaluated with a proper model which should include the burst cell loss characteristics of an ATM network.

Two models are discussed in Section II. Simulation results for the validation of the burst model and the discussion of the effects of cell losses on speech quality are provided in Section III. Finally, Section IV contains conclusions.

II. ATM SIMULATION MODELS

A. RANDOM MODEL

The random model assumes that cell losses in ATM are distributed randomly. Note the random model is not affected by network parameters such as network load and link capacity which is a situation very different for the burst model.

This model generates cell losses with the desired cell loss rate using a random number generator. This is a straightforward approach because the characteristics of an ATM network are not directly involved in the computation of random cell losses. For example, one can use the random model for the evaluation of cell recovery techniques.

However, a more realistic model should be used for the evaluation of speech quality with cell losses in ATM because the random model does not take into consideration the behavior of voice traffic in an ATM network, which is burst like in its behavior[2][3].

B. BURST MODEL

The ATM cell size is 53 octets (5 octet header and 48 octet payload). Four octets out of the 48 octet payload field are assumed to be used for an adaptation layer header[3]. Therefore, the voice information field is 44 octets in length.

In an ATM network, digital speech interpolation (DSI) is used for voice compression. DSI is an approach that is used to remove silences from the speech signal, that is, voice cells are generated only during talkspurts and not during silences.

A voice source is "active" when the talker is actually speaking, and during these times the voice source generates cells. A voice source is "inactive" when the speaker is silent, during these times the voice source does not generate cells. Speech is packetized to form cells with 44 octets of voice information. The cells generated by voice sources are fed into a common queue and transmitted over the link on a first-come first-served basis. This is called statistical multiplexing, as shown in Fig. 1.

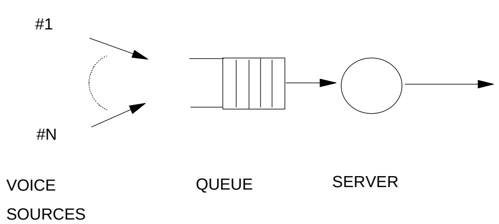

Figure 1. Statistical multiplexing

When the queue reaches the threshold for congestion control, cells are discarded according to a specific cell discarding (CD) algorithm. CD refers to organizing voice cells into high priority (more significant) and low priority (less significant) cells; this approach allows for dropping of the low-priority cells during congestion[4][5]. The receiver considers the cells which have not been received, within the delay time of the buffer memory, to be missing and employs a missing cell recovery technique to enhance speech quality.

In normal conversation, the duration of talkspurts fits the exponential distribution very well, while the duration of silence is not as well approximated by the exponential distribution[6]. However, for analytical purpose, the distributions of the duration of both talkspurts (mean: 352 ms) and silences (mean: 650 ms) of a voice source are assumed to be exponential[7]. The burst model is limited only to voice sources. This model incorporates statistical multiplexing, DSI, and a CD with no priority scheme.

The burst model is a mathematical model that characterizes cell losses for voice traffic in ATM and produces the distribution of equilibrium cell loss rates for a given set of ATM network parameters.

Let the number of voice sources be N and the link capacity C bps, respectively. Let M active voice sources saturate the network link. Let $T(r)$ be the equilibrium cell loss rate when r voice sources are active. Then, $T(r)$ is given as

$$T(r) = 0 \qquad \text{if } r \le M \qquad (1.1)$$

$$T(r) = \frac{r - M}{r} \qquad \text{if } r > M \qquad (1.2)$$

If the number of active voice sources, r, is greater than the link load of 100%, M, (r - M) cells will be lost, and if not, then no cells are lost. The equilibrium cell loss rate, $T(r)$, is a function of the number of active voice sources as well as the link capacity.

The probability of having r active voice sources with N voice sources, can be obtained using the binomial distribution. As N increases, the event of having r active voice sources with N voice sources is similar to the event of having r tails by tossing N coins, which is a Bernoulli trial.

Of interest is the distribution of cell loss rate for a given number of voice sources, N, and the link capacity, C bps. Once the distribution of cell loss rate is obtained, it is straightforward to generate cell losses for an ATM network without simulating an ATM network. The distribution of cell loss rate is used to find the mean cell loss rate, the standard deviation of cell loss rate, and is used for the evaluation of speech quality of an ATM network.

From the above discussion, the cell loss rate will have discrete values and the probability of different cell loss rates can be calculated as follows. Let T be the cell loss rate and Pr(T) the probability of the cell loss rate T, respectively, then,

$$\Pr(T) = \sum_{r=0}^{M} C_r^N \left[\frac{D_a}{D_a + D_i}\right]^r \left[\frac{D_i}{D_a + D_i}\right]^{N-r}$$

$$\text{if } T = 0 \text{ and } r \le M \qquad (2.1)$$

$$\Pr(T) = C_{\frac{M}{1-T}}^N \left[\frac{D_a}{D_a + D_i}\right]^{\frac{M}{1-T}} \left[\frac{D_i}{D_a + D_i}\right]^{N-\frac{M}{1-T}}$$

$$\text{if } T = \frac{r-M}{r} \text{ and } M < r \le N \qquad (2.2)$$

where, D_a is the mean active duration

D_i is the mean inactive duration

III. RESULTS

Figures 2 and 3 show the mean cell loss rate and its standard deviation as a function of percent link loads, respectively. The burst model approximates well the distribution of cell loss rates as the link capacity increases.

In order to obtain the relationship between SNRseg and cell loss rate, 64Kbps PCM was used for two female (FT1, FT2) and two male speech (MT1, MT2). The duration of test speech was as short as 1.5 seconds. Assuming the cell loss pattern of short segments could be regarded as random, the relationship between SNRseg and cell loss rate was determined with the random model. SNRseg correlates well with cell loss rate. The proportional factors between normalized SNRseg and cell loss rate are calculated. SNRseg decreases by 2.73% (0.8dB) as cell loss rate increases by 1%.

The normalized SNRseg's can be calculated for the STS-3C link of 155Mbps using the cell loss rates from the burst model. According to the central limit theorem, the distribution of cell loss rates is well approximated with a normal distribution for large N. To obtain the worst SNRseg for an error of 0.3%, the maximum cell loss rates are calculated as $\mu + 3\sigma$, where μ is the mean cell loss rate and σ is its standard deviation. Normalized SNRseg for the worst case as well as for the average case for STS-3C are shown in Fig. 4. As shown in Fig. 4, the SNRseg decreases by 5% (1.5dB) at the link load of 98% for the STS-3C link with an error of 0.3%.

IV. CONCLUSION

A mathematical model, called the burst model, has been developed to evaluate speech quality with cell losses in ATM. The burst model has been validated using computer simulation and it predicts cell loss behavior in ATM very accurately as link capacity increases.

From the random model, it has been determined that normalized SNRseg is inversely proportional to cell loss rate with a factor of 2.73.

With the mean cell loss rate and its standard deviation obtained using the burst model for STS-3C, normalized SNRseg has been determined as a function of link loads. SNRseg decreases by 1.5dB at the link load of 98% for the STS-3C link with an error of 0.3%.

REFERENCES

[1] H. Nagabuchi and N. Kitawaki, "Evaluation of Coded Speech Quality Degraded by Cell Loss in ATM Networks," Elect. and Comm. in Japan, Part 3, vol. 75, No. 9, pp. 14-24, 1993.

[2] S. Q. Li, "Study of Information Loss in Packet Voice Systems," IEEE Trans. on Comm., vol. 37, No. 11, pp. 1192-1202, Nov. 1989.

[3] K. Sriram, R. McKinney and M. Sherif, "Voice Packetization and Compression in Broadband ATM Networks," IEEE J. on Selec. Areas in Comm., vol. 9, No. 3, pp. 294-304, June, 1991.

[4] N. Yin, S. Q. Li and T.E. Stern, "Congestion Control for Packet Voice by Selective Packet Discarding," in Proc. GLOBECOM '87, Tokyo, Japan, vol. 3, pp. 1782-1786, Nov. 1987.

[5] D. W. Petr, L.A. DaSilva, Jr. and V. S. Frost, "Priority Discarding of Speech in Integrated Packet Networks," IEEE J. on Selec. Areas in Comm., vol. 7, No. 5, pp. 644-656, June, 1989.

[6] P. T. Brady, "A Model for Generating On-off Speech Patterns in Two-way Conversations," Bell Sys. Tech. J., vol. 48, pp. 2445-2472, Sept. 1969.

[7] H. Heffes and D. Luncantoni, "A Markov Modulated Characterization of Packetized Voice and Data Traffic and Related Statistical Multiplexer Performance," IEEE J. on Selec. Areas in Comm., vol. 4, No. 6, pp. 856-867, June, 1986.

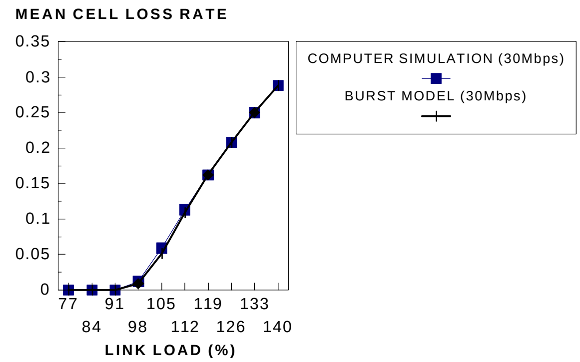

Figure 2. Mean cell loss rate as a function of percent link load

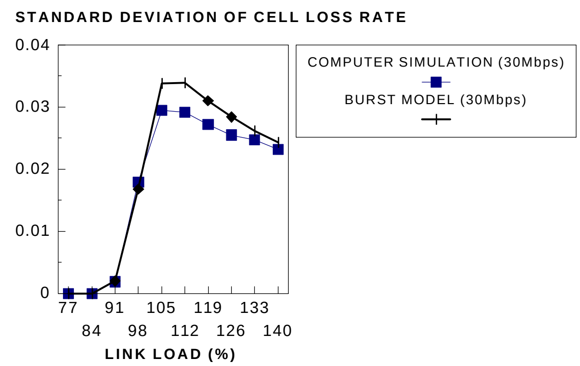

Figure 3. Standard deviation of cell loss rate as a function of percent link load

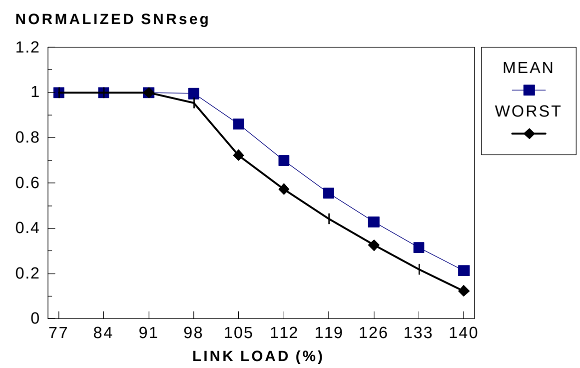

Figure 4. Normalized SNRseg for STS-3C link of 155Mbps