區域網路中之分封電話效能分析
Performance Analysis of Packet Telephony over LANs†

Rong-Tsung Sheu and Jean-Lien C. Wu

Department of Electronic Engineering
National Taiwan University of Science and Technology
43, Keelung Road, Section 4, Taipei, Taiwan, R.O.C. 10772
TEL: (886)-(2)-2737-6373 FAX: (886)-(2)-2737-6424
Email: jcw@nlhyper.et.ntust.edu.tw

Abstract

The framework of multimedia communications over LANs with non-guaranteed quality-of-service (QoS) has been proposed in H.323, which has been applied to a variety of multimedia communications, including LANs as well as MANs, WANs, enterprise networks and the Internet through gateway interfaces. The main body of H.323 includes the specification of the real-time transport protocol (RTP) and the RTP control protocol (RTCP) for real-time applications over Internet. It is essential to offer the quality-of-service (QoS) of packet telephony on smoothing the bursty traffic over real-time packet transmission in LANs since the media often were rendered unintelligibly due to delay jitters. This paper studies the performance of packet telephony in terms of packet delay and packet loss probability over the CSMA/CD LANs. Results of the analysis show that in a CSMA/CD LAN environment, packet telephony services still maintain a higher trunking efficiency than that in T1 under the conditions that a certain level of packet delay and loss probability is tolerated and a suitable voice compression technology, such as G.729 and GSM, is adopted.

Keywords: *Quality-of-Service (QoS), Packet Telephony, Carrier Sense Multiple Access (CSMA), Trunking Efficiency, Order Statistics.*

1. Introduction

In general, voice telephony technologies can be categorized as circuit-switched and packet-switched. Traditional circuit-switched networks are designed for point-to-point real-time voice communication. Nowadays, packet-switched networks have been adapted increasingly for the growing needs of digital communications for integrated services over Internet. An important contribution to this growth is Internet telephony. Critical to more prevalent use of Internet telephony is how to provide smooth cooperation with the existing telephone network. This interoperability has come through the use of Internet telephony gateway (ITG), which performs protocol translation between the Internet and the public switched telephone network (PSTN). In other words, the packet-switched telephony is becoming more and more popular not only among researchers, engineers, or students but also any one who might want to make conversations via the Internet. Currently, a variety of ITGs have already been installed to provide integrated services between the Internet and the PSTN [1][2].

The ITU Recommendation H.323 [3][4][5] is now the dominant standard for the framework of multimedia communications over LANs and can be used in any packet-switched network, regardless of the ultimate physical layer. The next generation LAN switches and switching hub devices are expected to serve with very higher transmission rate, cell-oriented transport and multimedia integrated transferring capability, hence the design of efficient media access protocol for integrated multimedia applications over LANs draws the attention of researches [6]. LANs were originally designed mainly to handle non-real time data traffic, now they are being used increasingly to carry real-time traffic. The Internet telephony technology and their attractive market will evolve changes in the office operations as well as the application of packet telephony services over LANs. As high-speed LAN technology is expected, packet telephony over LANs is capable of providing both low-cost interface from PC to voice devices (such as telephony gateway) and low-cost voice transmission between LANs and WANs.

The conventional PBX industry will face the challenge that the packet telephony is provided over a high-speed LAN. However, in order to provide satisfactory packet telephony in LANs, a number of technical bottlenecks including packet delay, packet loss and packet transmission synchronization await to be overcome. Although both Poisson and 2-state models are often used for their simulations based on the assumption of traffic model, the method for analyzing both distribution and control for their system resource varies due to different system characteristics. In a circuit-switched system, resources are distributed statically to users. Therefore, the performance analysis usually does not emphasize on resources allocation, but on the blocking rate. Common analytical methods include using the Erlang-B formula for determining the blocking rate in the lost call cleared system, and using the Erlang-C formula for determining the blocking rate in the lost call delayed system. On the other hand, in a packet-switched system, resources are mainly allocated dynamically to users. Therefore, the performance measure emphasizes the buffer overflow and the packet delay. The MMPP and fluid-flow are two most common models used for the performance analysis. In regard to the analysis of buffer overflow, both the buffer and the system capacity are taken into consideration and the probability distribution of the buffer overflow is used to capture the control factors of the system. Regarding the analysis of packet delay, the point-to-point delay, the media access delay and the system I/O process delay, etc., are considered. The extent of analysis

† This research was sponsored by National Science Council and Chung-Hwa Telecom Lab. under grant NSC 88-2213-E-011-038.

usually varies according to the scope of measurement.

A key issue in the design of packet telephony system is the delay tolerance with packet smoothing. Analytical work performed in [7] employed a simplified model by transmitting a tagged packet with a target delay to guarantee the on-time arrival of at least 99% of the packets. The assumptions include that networks are synchronized and the emission time of the first packet is known. Obviously, both the target delay and the network synchronization have significant impact on system performance. Senders and receivers are connected via a network, which adds delays to packets. These delay components often consist of propagation delay as well as internal network queueing and transmission delays. From the existing related works, methods of delay analysis can be categorized into three classes. First, regarding the delay jitter analysis in capturing voice source traffic behavior, Fulton and Li [8] presented a method to fit the first-order and second-order statistical functions to approximate the delay jitter experienced by a stationary traffic stream multiplexed at a major communication node. Similar works, but in the continuous bit rate multiplexor case, can be found in [9], where the mathematical model of jitter distribution and some simple asymptotic results on per-stream behavior are provided. Second, regarding the packet delay analysis, simple analysis is traditionally introduced in general queueing theory [10][11], more advanced analyses can be found in [12][13], where Interrupted Bernouilli processes [14] and Markov-modulated Bernouilli processes were considered. Third, the analysis of end-to-end delay was proposed in [15] by going through a number of configurations and adding one or more delay components to each end-to-end transmission path.

With the trend of digital transmission of voice telephony, it has spurred much interest in the integration of multimedia communications. A number of papers[1][3][16] have addressed the issue of voice and data traffic over CSMA networks. These studies considered network configurations that restrict the number of simultaneously active telephone calls to 100-200 nodes in the absence of data traffic. In practice, these systems must be capable of supporting a much larger number of active telephones. Gonsalves and Tobagi [16] provided the comparison of performance of voice and data with different LAN protocols including Ethernet, Token Bus, and Expressnet.

The rest of the paper is organized as follows. Section 2 describes the traffic model used for modeling voice sources. Section 3 presents the performance analysis of the carrier sense multiple access/collision detection (CSMA/CD) protocol used in the packet telephony over LANs. The CSMA/CD process model is applied to simplify and reduce the solution complexity in deriving the packet delay. Finally, we proceed to present a simple order statistics method to analyze the voice packet dropping probability. With the order statistics approximation, the probability of packet loss in 1% is obtained. In Section 4, the simulation model is described. Results of performance analysis are examined in Section 5. Finally, the paper is concluded in Section 6.

2. The Modeling of Voice Sources

The statistical analysis has increasingly shown that the traffic streams in modern broadband networks exhibit long-range, or self-similar, characteristics [17][18], the consequence of this discovery has lead to a revival of interest in non-standard queueing systems. However, the asymptotic form often provides the known results for such systems with a long tail distribution, simulation is thus required to learn about the rest of the distribution. For the case of voice traffic over LANs, we assumed that the traffic is of Poisson distribution. As to the voice user activity model [11], it is usually assumed that the voice traffic generated by users (i.e. endpoints) follows an alternating pattern of active, or talkspurt (ON), intervals, typically averaging 0.4 to 1.2 seconds in length, followed by inactive, or silence (OFF), intervals averaging 0.6 to 1.8 seconds in length. The voice source is modeled as an Interrupted Bernoulli Process (IBP) [12] with two states, talkspurt and silence. The parameter α denotes the rate of transition out of the talkspurt state; β is defined to be the rate of transition out of the silence state. Thus, the mean length of talkspurt interval is $1/\alpha$ sec in length. The mean length of silence interval is $1/\beta$ sec in length. The probabilities P_{ON} and P_{OFF} that a voice user is in talkspurt state and in silence state are given by $P_{ON}=\frac{\beta}{\alpha+\beta}$ and $P_{OFF}=\frac{\alpha}{\alpha+\beta}$, respectively.

While in talkspurt state, voice packets are assumed to be transmitted randomly, obeying a Poisson process with average rate. In a CSMA/CD LAN, all nodes will sense the beginning of an idle period at most b time units after the end of a transmission. Here, b is in units of packet duration, if τ is the propagation delay in seconds, C is the capacity in bit per second, and L is the average packet length, then $b = \tau C/L$. The expected time between two transmissions is at most an additional time of $1/G$. G is the mean successful attempts per time unit to be offered by the CSMA/CD protocol [19]. A packet will collide with some later packet with probability at most $1\text{-}e^{-bG}$ and it will be successfully transmitted with probability at least e^{-bG} and will occupy 1 time unit.

3. Performance Analysis

In this section, we first use the CSMA/CD state diagram to analyze the voice packet delay, and apply the order statistic theorem to estimate the voice packet loss probability in the packet telephony system.

A. Delay analysis using the CSMS/CD state diagram

We now focus on real-time transmission services. Collided packets arise due to the sharing of the multiple access link capacity with packets belonging to multiple nodes. The retransmission delay of a collided packet is the time that it spends in the transmission period until the packet is transmitted successfully. The retransmission delay depends on the number of users, the link capacity and the traffic generation rate of different sources. However, the retransmission delay must be bounded or minimized in order to satisfy the requirement of real-time applications.

The mean packet delay can be defined as the elapsed time from the instant the packet is ready to the time the packet is transmitted successfully. The state diagram of the CSMA/CD process is shown in Figure 1. The three states, "Carrier

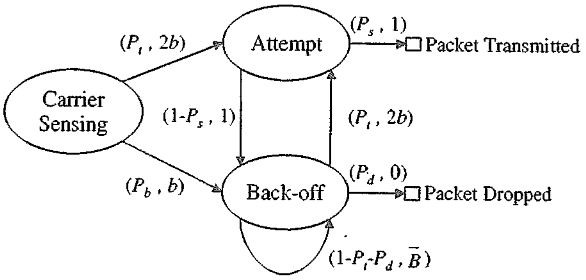


Figure 1. The state diagram of the CSMA/CD process

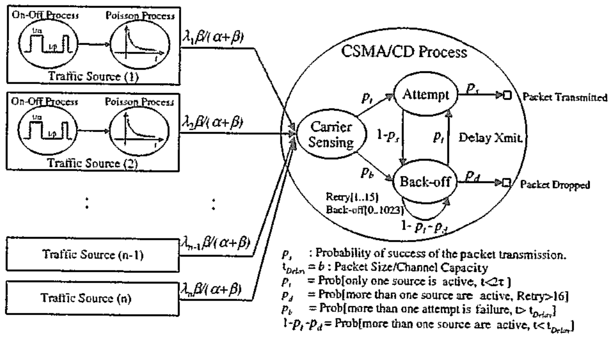


Figure 2. The CSMA/CD process simulation model

Sensing" and "Attempt", "Back-off", along with the associated delay values and the transition probabilities define the process used to obtain an expression for the mean packet delay. A station with packet ready for transmission is said to be in the "Carrier Sensing" state. If a station senses the link idle for $2b$ time units after the arrival of a packet, it enters the "Attempt" state to transmit the packet otherwise it enters the "Back-off" state. If the packet is transmitted successfully in the "Attempt" state, the station finishes its transmission in one time unit, where if the packet incurs a collision with others it immediately enters the state "Back-off". If the first attempt is not successful for busy period $\overline{B}$ with probability P_b, the station enters the "Back-off" state, where it remains until the packet attempt is complete or the retry time expires with probability P_d.

Now let $P_t = (1+bG)e^{-bG}$ denote the probability that only one packet succeeds during an idle interval and upon the first attempt. Let us examine the packet transmission under the condition that there are n users always ready to transmit. If each user transmits a packet in talkspurt during a contention slot with probability P_{ON} with the average packets generation rate λ , then $G = N\lambda\ P_{ON}$ denotes the average total packet arrival rate for N voice sources.

By analyzing a large number of such traffic sources over best-effort LANs, the resulting traffic process often exhibits CSMA properties. It might be useful to recall that the successful transmission probability of 1-Persistent CSMA derived by Kleinrock and Tobagi [20]. $\overline{B}$ is the expected duration of the busy period, q_0 is the probability of zero packets accumulated at the end of a transmission period and P_s is the probability of success of the packet, $\overline{B}$, q_0 and P_s can be cited from [20] and are given as

$$q_0 = (1+bG)e^{-G(1+b)},$$

$$\overline{B} = \frac{1+2b-(1-e^{-bG})/G}{q_0}, \text{ and}$$

$$P_s = P\{\text{success}\}$$

$$= \frac{G(1+G+bG(1+G+bG/2))e^{-G(1+2b)}}{G(1+2b)-(1-e^{-bG})+(1+bG)e^{-G(1+b)}}.$$

Using the above expressions, the probability P_b that a contention period has at least one attempt is equal to the probability that some packets arrive during the $\overline{B}+b$ time units of the contention periods, therefore, $P_b = 1 - q_0$. The probability P_d that a back-off period results in that the retry time expires is the same as the probability that no packets is transmitted from the 1st to the Rth attempt during the back-off period. This value can be expressed as

$$P_d = 1 - P(r \le R)$$
$$= 1 - \sum_{r=0}^{R} P_s(1-P_s)^{R-r}. \tag{3-1}$$

From Figure 2, the mean packet delay T_D can be obtained as follows

$$T_D = P_t(2b + E[A]) + P_b(b + E[B]), \tag{3-2}$$

where $E[A]$ is the additional mean delay accumulated each time the state "Attempt" is entered, and $E[B]$ is the delay caused by each backlog in the state "Back-off". $E[A]$ and $E[B]$ are expressed respectively by

$$E[A] = (1-P_s)(1+E[B])+P_s \tag{3-3}$$

and

$$E[B] = P_t(2b+E[A])+(1-P_s-P_d)(\overline{B+E[B]}). \tag{3-4}$$

Solving the two equations, we have

$$E[A] = 1-(1-P_s)\frac{P_t(2b+1)+(1-P_t-P_d)\overline{B}}{P_tP_s+P_d} \tag{3-5}$$

and

$$E[B] = \frac{P_t(2b+1)+(1-P_t-P_d)\overline{B}}{P_tP_s+P_d}. \tag{3-6}$$

B. The estimation of packet loss probability

In the delay sensitive traffic, each voice packet is time stamped to check with the delay constraint D_{max}. A packet will be discarded at the receiver when the measured packet life-time is more than D_{max}. In this section, the probability distribution of the interarrival time and the probability of packet dropping with time constraint will be determined. The process of packet arrival can be thought of as a Poisson process so that the probability of interarrival time is of gamma distribution. Let t_i be the arrival time of packet i, t_{D_i} be the interarrival time between t_i and t_{i-1} and t_i is an r.v. with gamma distribution. Therefore, the density function $f_D(t)$ and the distribution function $F_D(x)$ of t_{D_i} are expressed as follows

$$f_D(t) = e^{-\lambda t}\frac{\lambda(\lambda t)^{s-1}}{\Gamma(s)},\ t>0 \text{ and } s>0 \tag{3-7}$$

$$F_D(x) = P(t_{D_i} \le x)$$

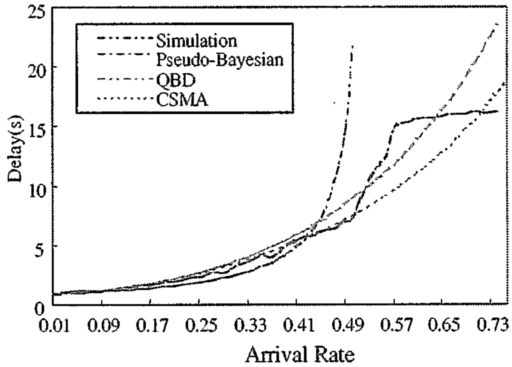


Figure 3. Mean packet delay vs. arrival rate

$$= \int_0^x e^{-\lambda t}\frac{\lambda(\lambda t)^{s-1}}{\Gamma(s)}dt = 1-e^{-\lambda x}\Big|_{s=1},\ s>0 \qquad (3\text{-}8)$$

with $\Gamma(s)$ being the gamma function. Here $s = 1$ because the change of packet number in the interval $[t_i, t_{i-1}]$ is 1. From equation (3-8), it is obvious that the r.v. of packet delay t_{D_i} has the memoryless property of the exponential distribution. Hence, the probability that the packet delay t occurs only in conjunction with a preceding mean packet delay interval T_D is expressed as follows

$$F_D(x) = P(x \le t) = 1 - e^{-\lambda x/T_D} \qquad (3\text{-}9)$$

If $t_{D_1}, t_{D_2}, \dots, t_{D_n}$ are the items of a random samples of size n from a continuous type distribution, we let the random variables $t_{S_1} < t_{S_2} < \cdots < t_{S_n}$ denote the ordered waiting time intervals of that samples t_{D_i}. The probability $G_D(y)$ that the ith order statistic $t_{S_i} \le t$ occurs can be obtained by using the order statistics theorem [21], and is expressed as

$$G_D(y) = P(t_{S_i} \le y) = \sum_{k=i}^{n}\binom{n}{k}F_D(y)^k(1-F_D(y))^{n-k} \qquad (3\text{-}10)$$

Finally, let T be the maximum packet life-time and the packet dropping rate be less than 1%, the packet loss probability $H_D(T)$ can be expressed as follows

$$H_D(T) = P(t_{S_i} > T) = \sum_{k=0}^{ceil(0.99n)}\binom{n}{k}F_D(T)^k(1-F_D(T))^{n-k} \qquad (3\text{-}11)$$

For large population, i.e. $n >> 1$, the probability of binomial distribution $b(n, p)$ is difficult to evaluate but in turn can be approximated by a normal distribution probability. Thus, $b(n, p)$ can be approximated using normal distribution $N(np, np(1-p))$ and we have

$$H_D(T) = P(t_{S_i} > T) \approx \Phi\left(\frac{0.99n - np)}{\sqrt{np(1-p))}}\right)$$

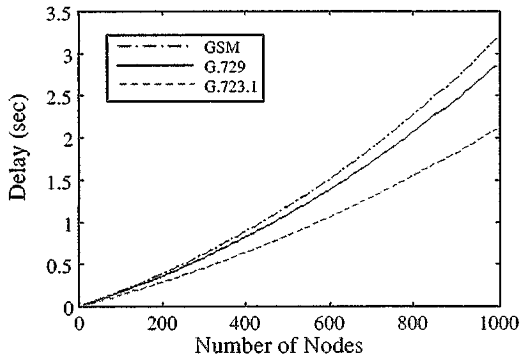
$$= \Phi\left(\frac{0.99n - nF_D(T))}{\sqrt{nF_D(T)(1-F_D(T))}}\right) \qquad (3\text{-}12)$$

4. Simulation Model

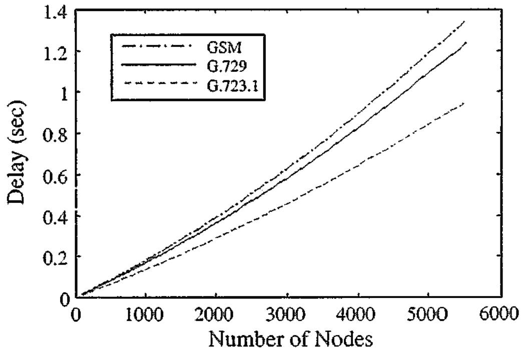
In this section, we perform the performance simulation of packet telephony over LANs with the simulation model as shown in Figure 2. Two kinds of traffic models are used: the On-Off and Poisson. The On-Off traffic sources have been used extensively in recent studies since they can be used to model standard voice sources. An On-Off traffic source is modeled here as a two-state Markov modulated process.

The population must be large enough in order to be modeled as a Poisson process for the long-range traffic sources, the value of population n must be designed easily to adjust for adapting the requirement of traffic simulation. For example, a large number of voice source may be assigned to 3,000 or 5,000 for the simulation of a 100 Mbps LAN.

N traffic sources are first generated by the On-Off process and then fed into the Poisson process. In each On-Off process, the duration of talkspurt and silence is generated with the ratio of $\beta\ /\ (\alpha\ +\ \beta\ )$ (e.g. 1.0/1.35). During the talkspurt period, these packets are modeled by a Poisson process. All traffic sources are generated to simulate the voice arrivals from the packet telephony terminals and fed into the CSMA/CD process. In the CSMA/CD process, the 1-Persistent CSMA functions of carrier sensing and collision detection [20] are included.



(a). Packet delay on 10-Base Ethernet

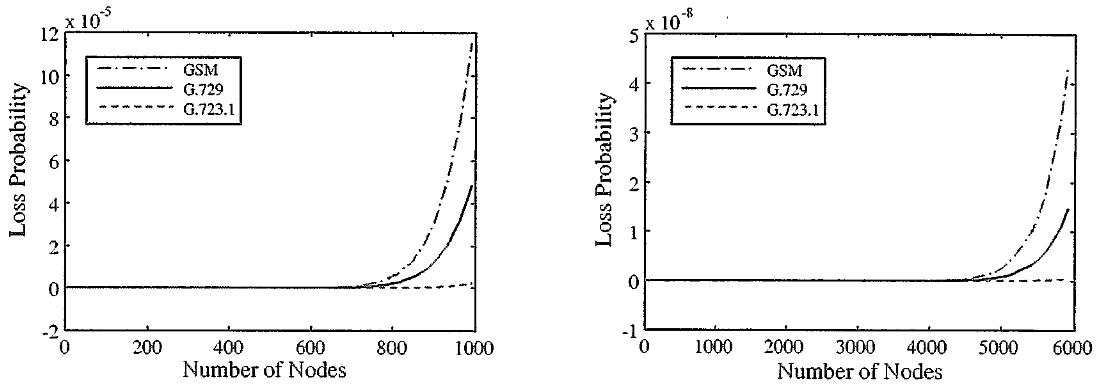


(b) Packet delay on 100-Base Ethernet

Figure 4. Comparison of packet delay for G.723.1, G.729 and GSM

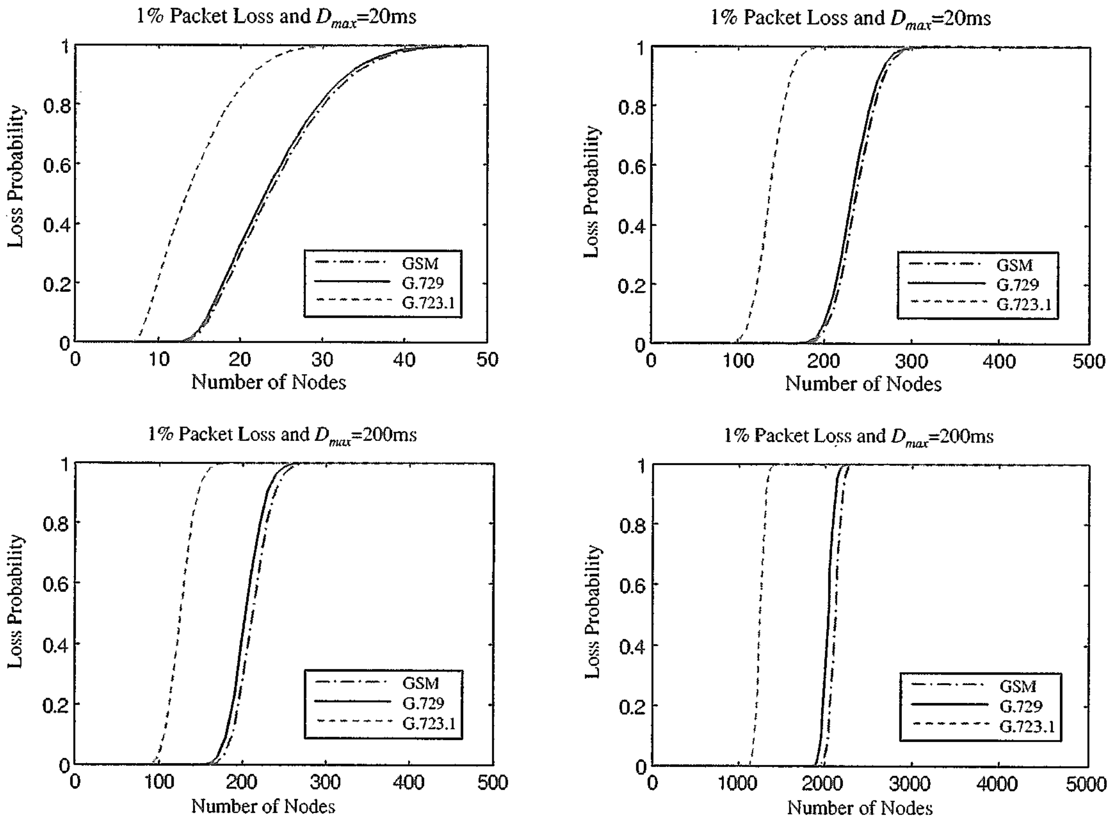
5. Numerical Results

The delay analysis of the exponential back-off algorithm derived by Huang et al. [22] is quite complicated and its upper- and lower-bound, labeled "Pseudo-Bayesian" in Figure 3, are close to our simulation results labeled with



(a) 10-Base Ethernet (b) 100-Base Ethernet

Figure 5. Comparison of packet loss-probability without delay constraint



(a) 10-Base Ethernet (b) 100-Base Ethernet.

Figure 6. Comparison of packet loss rate with delay constraint

"CSMA". The results shows that the CSMA/CD process model is much more accurate in analyzing the packet delay for telephony services over LANs.

The comparison of packet delay for G.723.1, G.729 and GSM is shown in Figure 4.. It is obvious that the delay varies significantly with the network transmission rate and the traffic loads. Namely, packet delay is found to be sensitive to the number of traffic sources with respect to different transmission rates. The results show that about 100 ms delay experienced in 160 nodes with an arrival rate of 8 kbps over a 10-Base Ethernet, while more than 1620 nodes experienced 100 ms delay for a 100-Base Ethernet. This result is intuitively true, the more the number of nodes, or the higher the traffic loads, the longer the packet delay.

From Figure 4, the trunking efficiency of packet telephony can be achieved are about 16.0 (160/10 users/Mbps) and 16.2 (1620/100 users/Mbps) over 10-Base and 100-Base Ethernets, respectively, the trunking efficiency of T1 is calculated as 15.54 (24/1.544 users/Mbps). In general, the higher the bandwidth of a LAN, the more the admitted number of users. Nevertheless, there is always a trade-off between the QoS provided to the users and the cost. To achieve a solution, perceptible by users, the codec schemes, such as G.723, G.729 or GSM, to be used and the reduced QoS due to the burstiness caused by packet delay have to be considered.

Based on the analytical results, we can clearly observe that if the voice compression scheme such as G729 is adopted, the 10 Mbps and 100 Mbps Ethernet can provide a trunking efficiency of 16.0 and 16.2, respectively, which is better than 15.4 in T1. However, this does not imply that providing over LANs is a good solution. The issue shall be evaluated from an objective point of view. The quality of voice provided by T1 is a standard in the telephone industry and has been used for a number of years and widely accepted. Providing packet telephony services over LANs should take into consideration the degree of acceptance by the users. For example, using voice compression will increase the bandwidth utilization as well as the trunking efficiency, relatively reduces the quality of voice to a certain degree. Furthermore, the packet delay time thereof is about 200~300 ms which is longer than that of T1 (30-50 ms) [6]. Of course these quality factors certainly can be overcome using high-speed LANs. For example, high-speed LANs provide much higher bandwidth, thereby reduces the deterioration of the quality of voice by compression. Moreover, providing packet telephony services over LANs with the voice quality equivalent to that of T1 is feasible if a priority class control can be added on the control of the MAC communication protocol to adjust the retransmission mechanism and control the packet loss rate to be less than 1%.

In Figure 5., the packet loss rate will not be too high according to the packet life-time without delay constraint, on the other hand, the packet delay will increase rapidly at heavy loads. Therefore, adopting an appropriate packet-dropping scheme to adjust the real-time voice traffic can greatly improve the packet delay so that the quality of voice will only be affected slightly. From Figure 6, the data indicate that adopting packet-dropping scheme with loss rate of 1% can achieve the improvement on delay and the number of voice users. Considering a delay constrained by 200ms and packet dropping rate of 1%, with a 10 Mbps Ethernet, the voice capacities of the codec schemes of G.723, G.729 and GSM are 110, 175 and 180, respectively. At 100 Mbps, they have the capacities of 1200, 1850 and 2020, respectively.

6. Conclusions

In this paper we investigate the traffic delay problems encountered when users request for telephony services in a LAN environment. We analyzed and simulated the CSMA/CD process model to understand the delay of packet telephony service over LANs. The delay analysis is based on the concept of the CSMA/CD process model, it is a simple method for packet delay analysis of multimedia communication over LANs. One can use the results of the CSMA/CD process model to derive the packet delay from large traffic sources N with k retransmissions and still obtain very accurate results.

The CSMA/CD process model enables us to rapidly calculate the relationship among the number of users, the packet delay and the loss rate. This property is helpful in the design of the admission control mechanism. Results of the delay analysis proposed in this paper match quite well with simulation results. In the mean time, we can use this method to fine tune the parameters in the MAC protocol with respect to the retransmission mechanism, hence to improve the performance. In the future, we will extend our study of the delay and performance control mechanisms related to AAL2, and locate a novel bit-dropping mechanism to enhance best-effort support for IP-over-ATM. We believe that the LAN-based packet telephony services will share the market of conventional telephone services with the unceasing innovation of high speed LAN technology in the near future.

國科會計畫編號：NSC 88-2213-E-011-038

References

[1] J. Rosenberg and H. Schulzrinne, "Internet Telephony Gateway Location", Proceedings of IEEE INFOCOM98, Vol. 3. Sanfrancisco, Apr. 1998, pp. 488 ~ 496.

[2] D. Minoli and E. Minoli, Delivering Voice over IP Network, John Wiley & Sons, Inc., 1998.

[3] ITU-T Recommendation H.323v2: Packet-based Multimedia Communications Systems, Feb. 1998.

[4] ITU-T Recommendation H.225: Media Stream Packetization and Synchronization on Non-Guaranteed Quality of Service LANs, May 1996.

[5] ITU-T Recommendation H.245: Control Protocol for Multimedia Communication, March 1997.

[6] T. J. Kostas, M. S. Borella, L. Sidhu, G. M. Schuster, J. Grabiec and J. Mahler, "Real-Time Voice Over Packet-Switched Networks", IEEE Network Magazine, Jan./Feb. 1998, pp. 18 ~ 27.

[7] F. Alvarez-Cuevas, M. Bertran, F. Oller and J. M. Sliga, "Voice Synchronization in Packet Switching Networks", IEEE Network Magazine, Sept.. 1993, pp. 20 ~ 25.

[8] C. A. Fulton and S. Li, " Delay Jitter First-Order and Second-Order Statistical Functions of General Traffic on High-Speed Multimedia Networks" IEEE/ACM Transactions on Networking,, Vol. 6, No. 2, Apr. 1998, pp. 150 ~ 163.

[9] A. Privalo and K. Sohraby, "Pre-Stream Jitter Analysis in CBR ATM Multiplexors", IEEE/ACM Transactions on Networking, Vol. 6, No. 2, Apr. 1998, pp. 141 ~ 149.

[10] J. N. Daigle, Queueing for Telecommunications, Reading, MA: Addison-Wesley, 1992.

[11] M. Schwartz, Broadband integrated Networks, New Jersey, Prentice Hall, 1996.

[12] I. Widjaja, M. Neuts and J. M. Li, "Conditional Overflow Probability and Profile Curve for Congestion Detection in ATM Networks", Proceedings of IEEE INFOCOM96, Sanfrancisco, Apr. 1996, pp. 970 ~ 977.

[13] H. Heffes and D. M. Lucantoni, "Markov Modulated Characterization of Packetized Voice and Data Traffic and related Statistical Multiplexer Performance", IEEE J. Select. Area in Commun., Vol. 4, 1986, pp. 856-868.

[14] K. M. Elsayed and H. G. Perros, "A computationally efficient algorithm for characterizing the superposition of multiple heterogeneous interrupted Bernoulli processes", Proceedings of the 2nd International Workshop on the Numerical Solution of Markov Chains, 1995, pp. 443 ~ 462.

[15] S. Dravida and K. Sriram, "End-to-End Performance Models for Variable Bit Rate over Tandem Links in Packet Network", IEEE J. Select. Area in Commun., Vol. 7, No. 5, June. 1989, pp. 657 ~ 669

[16] T. A. Gonsalves and F. A. Tobagi, "Comparative Performance of Voice/Data Local Area Network", IEEE J. Select. Area in Commun., Vol. 7, No. 5, June. 1989, pp. 657 ~ 669

[17] A. T. Andersen and B. F. Nielsen, "A Markovian Approach for Modeling Packet Traffic with Long-Range Dependence", IEEE/ACM Transactions on Networking, Vol. 16, June. 1998, pp. 719 ~ 732.

[18] D. Heyman and T. Lakshman, "What are the implications of long-range dependence for VBR video traffic engineering?", IEEE/ACM Transactions on Networking, Vol. 4, June. 1996, pp. 301 ~ 317.

[19] D. Bertsekas and R. Gallager, Data Networks, 2nd ed., Englewood Cliffs, NJ: Prentice Hall, 1992.

[20] L. Kleinrock and F. A. Tobagi, "Packet Switching in Radio Channels: Part I-Carrier Sense Multiple-Access Modes and Their Throughput-Delay Characteristics", IEEE Transaction on Commun., Vol. COM-23, NO. 12, Dec. 1975, pp. 1400 ~ 1416.

[21] L. Blank, Statistical Procedures for Engineering, Management, and Science, McGraw-Hill, 1980.

[22] J. C. Huang and T. Berger, "Delay Analysis of the Modified 0.487 Contention Resolution Algorithm", IEEE Trans. Inform. Theory, IT-31, 1985, pp. 264 ~ 273.