

PACKING DENSITY OF VOICE TRUNKING USING AAL2*

Chunlei Liu Sohail Munir Raj Jain

**New Address: Raj Jain, Washington University in Saint Louis,
jain@cse.wustl.edu, <http://www.cse.wustl.edu/~jain>**

Sudhir Dixit

Nokia Research Center

3 Burlington Woods Dr., Suite 250, Burlington, MA 01803

email: sudhir.dixit@research.nokia.com

Abstract

AAL2 has been adopted in ITU-T and ATM Forum to reduce the packing delay for voice trunking. A parameter called "Timer.CU" is used in AAL2 to avoid prolonged delay for any packet. However, the AAL2 documents do not discuss how to set the Timer.CU value. In this paper, we establish a Markov chain model to analyze the AAL2 packing process. We find that the packing density and link efficiency depend significantly on the Timer.CU value. The analysis of this paper gives an algorithm to calculate a reference Timer.CU value to achieve desired link efficiency. Simulation results that verify the analysis are also presented.

1 Introduction

Since the emergence of computer networks, efforts have been made to transfer voice over networks[2, 3, 7]. Starting as a technical novelty, Internet telephony is now becoming a big business. However, the quality of Internet phone still remains a problem. Because telephony is a real-time application, delay, among other quality measurements, is the most important factor that affects the quality of voice. If a packet arrives late, its contents become obsolete and have to be discarded. According to ITU-T Recommendation G.114[5], an end-to-end delay of 0 to 150 ms is acceptable for most user applications. A delay of 150 to 400 ms is acceptable provided that administrators are aware of the transmission time impact on the transmission quality of user applications, but any delay above 400 ms is unacceptable for general network

*This research was sponsored in part by US Army CECOM Contract DAAB08-98-C-B305. This paper is available on-line at <http://www.cis.ohio-state.edu/~jain/papers/density.htm>

planning purposes.

The problem of delay becomes more severe when efficient compression/decompression methods are used. For example, in order to fill an ATM cell which has 48-byte payload, the ITU-T G.711 (64kbps) codec needs 6 ms, but the more efficient ITU-T G.723.1 (5.3kbps) codec needs 72 ms. Notice this 72 ms does not include the propagation delay, queueing delay, etc, that the cell must undergo when it travels through the networks.

ATM Adaptation Layer 2 (AAL2) has been designed to reduce the packing delay. It is described in ITU-T Recommendation I.366.2[6] and ATM Forum specification "ATM Trunking using AAL2 for Narrowband Services"[8]. The idea is to multiplex voice packets from several sources into one ATM cell so that the time to fill a cell can be reduced significantly. Figure 1 illustrates a scenario of voice packets from three sources being packed into cells.

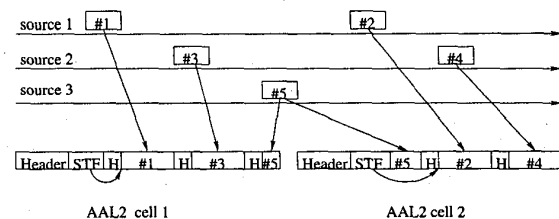


Figure 1: AAL2 cell packing

Also shown in Figure 1 is the format of AAL2 cells. Every cell has a standard 5-byte ATM header. After the header is the Start Field (STF) that indicates where in the payload the next complete packet starts. For every packet, there is

a 3-byte mini-header (H) which includes the Channel ID (CID), Length Indicator (LI), User-to-User Indication (UUI) and Header Error Control (HEC) of the packet.

However, the above packing mechanism has one problem. Suppose the first packet is put into a cell and is waiting for the arrival of other packets to complete the cell. But after an extended period, no other packets arrive. This packet will suffer a prolonged delay.

To avoid such prolonged delay, a parameter called "Timer_CU" is proposed in [6]. When the packing begins, a timer is set to this parameter value. If the cell is not completely packed within the time period determined by this Timer_CU value, the timer expires and the partially packed cell will be sent.

Zhang[9] analyzed the impact of Timer_CU value on packet delay variation (PDV), and found that the Timer_CU value is not relevant in the total PDV calculation. While the statement about PDV is correct, we found that the Timer_CU value has to be set appropriately since it significantly affects the link efficiency. If the Timer_CU value is too small, more partial cells are likely to be sent and the link efficiency will be low. If it is too large, some packets will suffer prolonged delay and the voice quality will degrade.

In this paper, we establish a Markov chain model to analyze the AAL2 packing process using ITU-T G.723.1 voice encoding. The Markov analysis reveals the correlation between successive cells and gives a formula for calculating the packing density based on the Timer_CU value and the number of voice sources in the system. To validate our analytic result, a simulation was launched to actually implement the AAL2 packing process. The results of the simulation perfectly match our analysis. The comparison is presented in section 5.

2 Simulation Model

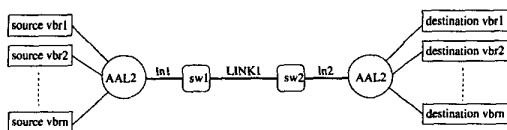


Figure 2: Simulation model

Figure 2 shows our simulation model and the packing process. On the left, a number of variable bit rate (VBR) voice sources are connected to the source AAL2. The corresponding VBR destinations are shown on the right. Voice sources

send voice signal in the form of packets to the source AAL2. The source AAL2 keeps a working cell. Arriving packets are packed into the working cell. Once the working cell is packed, it is sent to the switch.

When the first packet is put in the cell, a timer is set to Timer_CU parameter value. If the cell is not fully packed within Timer_CU time, the cell "expires" and is sent even though it is partially packed.

If an arriving packet can not fit in the remaining space in the working cell, the bytes that can be fit in are put into the available space, and the cell is dispatched. The remainder of the packet is put in the next cell and the timer is initialized to Timer_CU.

The cells go through links and switches in the networks and arrive at the destination AAL2. The destination AAL2 unpacks the cells and dispatches each packet to its destination according to the channel identifier(CID) contained in the mini-header of the packet.

3 Voice Model and Packet Arrival Pattern

Human voice consists of alternating talkspurts and silence intervals. It has been found that talkspurt lengths and silence intervals are exponentially distributed[1]. In a commonly accepted model, the talkspurts have a mean length of 352 ms and silence intervals have a mean length of 650 ms [4].

There are a number of standards for coding voice. ITU-T G.723.1 is currently the most widely used encoding standard. During talkspurts, G.723.1 sends out a 20-byte packet every 30 ms. During silence periods, no packets are sent. In this paper, we assume all voice sources use G.723.1 as the encoding method.

Compared with the talkspurt length and silence period, the 30 ms packet length is short. This means, during a talkspurt, the voice source emits a series of packets at 30 ms intervals. Therefore, the arrivals are highly predictable.

Suppose that the number of voice sources in the system is N and that a packet has just been received from a voice source. We want to calculate the probability of no packet arrival from all sources in the next τ ms, where τ is the Timer_CU value.

Consider one voice source first. During an average 352 ms talkspurt and 650 ms silence cycle, 12 packets need to be sent as shown in Figure 3.

Let s denote the number of packets generated during a talk-

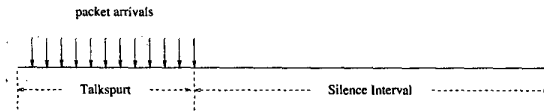


Figure 3: packet arrival pattern from one source

spurt interval. In order to have no packet arrival in τ ms, the starting point of this τ ms cannot fall in the τ ms interval before any of these s arrivals. The probability of no packet arrival from this source is

$$\frac{1002 - s\tau}{1002}$$

Here we assume $\tau < 30$. Since the average number of packets in a talkspurt is 12, the probability of no packet arrival from one source is

$$p = \frac{1002 - E[s]\tau}{1002} = \frac{1002 - 12\tau}{1002}$$

Voice sources are assumed to be independent. After the receipt of the first packet in the cell, the probability of no arrivals from all other sources within τ ms is

$$R_0 = p^{N-1}$$

and the probability of one packet arrival from all other sources is

$$R_1 = (N-1)p^{N-2}(1-p)$$

4 Calculation of Packing Density

Using the probabilities R_0, R_1 , we can calculate the average packing density.

The average number of bytes in an AAL2 cell depends on whether there is a remainder from the last cell and how many packets are received since the first packet was put in the cell. Let r_n be the remainder length left from the $(n-1)$ -th cell. Then this r_n is exactly the STF field in the n -th AAL2 cell. Since each packet has a 3-byte header and a 20-byte payload, the remainder is always shorter than 23 bytes, i.e., $0 \leq r_n \leq 22$.

Event $r_n = 0$ happens only when cell $n-1$ expires or when $r_{n-1} = 1$ and cell $n-1$ does not expire.

Event $r_n = 1$ happens only when $r_{n-1} = 2$ and cell $n-1$ does not expire.

Event $r_n = 2$ happens only when $r_{n-1} = 3$ and cell $n-1$ does not expire.

.....

Event $r_n = 22$ happens only when $r_{n-1} = 0$ and cell $n-1$ does not expire.

Therefore, $\{r_n\}$ forms a Markov chain. Consider the stationary state where all $\{r_n\}$ have the same probability distribution. Let r denote the random variable for the remainder length, and denote

$$\pi_i = P\{r = i\}, \quad i = 0, \dots, 22$$

and

$$Q_i = P\{\text{timer_CU expires} \mid r = i\}, \quad i = 0, \dots, 22.$$

Since an AAL2 cell has 47 byte payload, and for G.723.1 a CPS packet requires 23 bytes, it happens that for all $i = 0, \dots, 22$,

$$\begin{aligned} Q_i &= P\{\text{less than 2 packets are received in } \tau \text{ ms}\} \\ &= R_0 + R_1. \end{aligned}$$

Denote

$$Q = R_0 + R_1,$$

then the transition matrix P is

$$\begin{pmatrix} Q & 0 & 0 & 0 & \dots & 0 & 1-Q \\ 1 & 0 & 0 & 0 & \dots & 0 & 0 \\ Q & 1-Q & 0 & 0 & \dots & 0 & 0 \\ Q & 0 & 1-Q & 0 & \dots & 0 & 0 \\ \vdots & \vdots & \vdots & \vdots & \ddots & \vdots & \vdots \\ Q & 0 & 0 & 0 & \dots & 0 & 0 \\ Q & 0 & 0 & 0 & \dots & 1-Q & 0 \end{pmatrix}$$

In the stationary state, we have $\pi = \pi P$, so

$$\begin{aligned} \pi_0 &= Q + \pi_1(1-Q), \\ \pi_i &= \pi_{i+1}(1-Q), \quad i = 1, 2, \dots, 21 \\ \pi_{22} &= \pi_0(1-Q). \end{aligned}$$

Therefore,

$$\pi_1 = \pi_0(1-Q)^{22}, \quad \pi_0 = \pi_0(1-Q)^{23} + Q,$$

and

$$\begin{aligned} \pi_0 &= \frac{Q}{1 - (1-Q)^{23}}, \\ \pi_i &= \pi_0(1-Q)^{23-i}, \quad i = 1, 2, \dots, 22. \end{aligned}$$

Using these probabilities, the average number of bytes in an AAL2 cell is:

$$C = \pi_0(23R_0 + 46R_1 + 47(1 - Q)) + \sum_{i=1}^{22} \pi_i(iR_0 + (i + 23)R_1 + 47(1 - Q)).$$

Because each 20-byte voice packet has a 3 byte mini-header and each AAL2 cell has a 6-byte overhead, the packing density is:

$$D = \frac{C}{53} \times \frac{20}{23} \times 100\%.$$

Notice that

$$\frac{47}{53} \times \frac{20}{23} \times 100\% = 77.11\%$$

is the maximum possible density.

5 Simulation and Comparison

To validate our analysis, we used a simulation program to actually implement the AAL2 packing process. The simulation model is described in Section 2. The results of 11 simulations are summarized in Table 1. The first column is the Timer.CU value used in the simulation. The second column is the total number of cells being sent. Column 3, 4 and 5 respectively are the number of cells that have 0, 1, 2 or more packet arrivals since the packing of the first packet. The actual density is listed in the last column.

τ	total cells	rcvd 0 pkt	rcvd 1 pkt	rcvd 2+ pkt	density (%)
0.5	5621	4079	1246	296	46.93
1	5028	2339	1787	902	56.00
2	4286	949	1462	1875	64.78
3	4077	479	960	2638	69.00
4	3705	154	626	2925	72.43
5	3645	69	315	3261	74.35
6	3587	26	206	3355	75.42
8	3470	8	66	3396	76.54
10	3716	0	10	3706	77.05
12	3779	0	0	3779	77.11
14	3632	0	0	3632	77.11

Table 1: Simulation results: number of packets received within τ ms

We calculate the corresponding $R_0, R_1, R_{2+} = 1 - R_0 - R_1$, and compare them with our analysis results. In Figure 4, the analytic probabilities are shown with solid lines and simulation data are shown with "+"s.

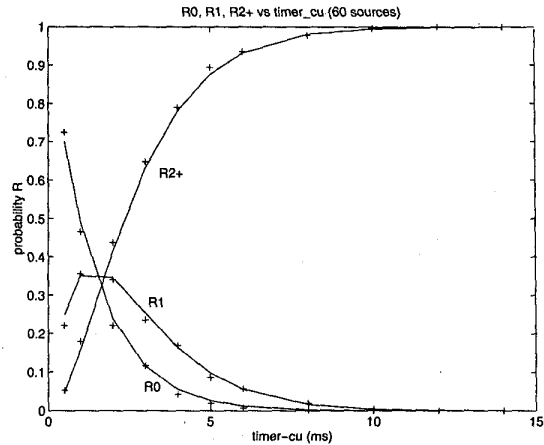


Figure 4: R_0, R_1 and R_{2+} for different Timer.CU values. 60 sources. solid line: calculation; "+" : simulation.

The resulting packing density for different Timer.CU values is shown in Figure 5. Again the analytic calculation is shown with solid line and the simulation data are shown with "+"s. The simulation data match the analytic calculation perfectly.

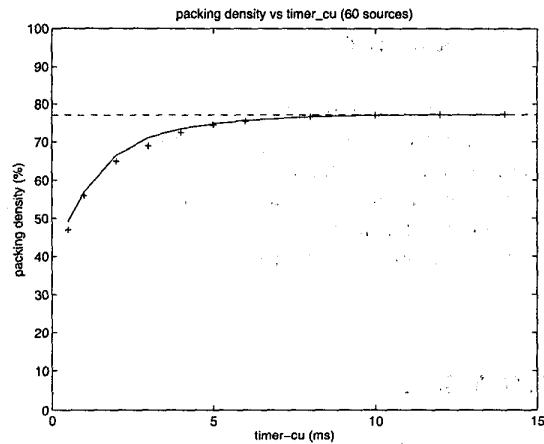


Figure 5: Packing density for different Timer.CU values. 60 sources. solid line: calculation; "+" : simulation.

One application of the analytic formula for density is to find the appropriate Timer.CU value to reach the desired packing density. Figure 6 shows the needed Timer.CU values in order to reach 90% and 95% of the maximum packing density.

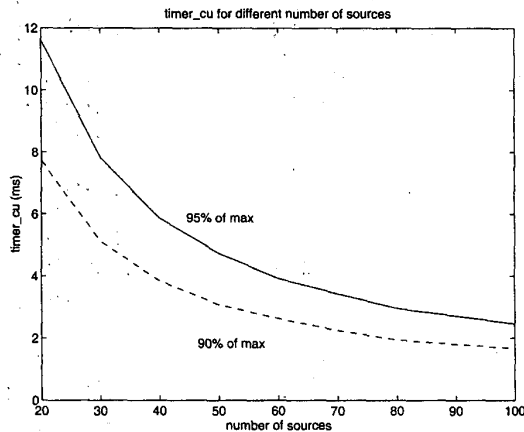


Figure 6: Timer_CU values to reach 90% and 95% of the maximum packing density.

Timer_CU value also has an impact on the end-to-end delay. Increasing Timer_CU value may cause longer delay. However, this impact is relatively small compared to other components of the total delay. In practice, we should take into account both the desired link efficiency and maximum acceptable delay. The value calculated by the above method can be used as a reference.

6 Conclusion

In conclusion, we find that Timer_CU value has significant impact on link efficiency. An appropriate choice of Timer_CU value depends on the number of voice sources and delay requirements. The analysis of this paper gives an algorithm to calculate a reference Timer_CU value to achieve the desired link efficiency.

References

- [1] Paul T. Brady, A model for generating ON-OFF speech patterns in two-way conversations. *Bell System Technical Journal*, Vol 48, pp 2445-2472, Sept 1969.
- [2] Danny Cohen, Specifications for the Network Voice Protocol, *ISI/RR-75-39*, USC/Information Sciences Institute (4676 Admiralty Way, Marina del Rey, CA 90292), March 1976, Available from DTIC (AD #A023506).
- [3] Danny Cohen, Issues in Transnet Packetized Voice Communication, *Proceedings of the Fifth Data Commu-*

nications Symposium, Snowbird, Utah, September 1977 (IEEE Catalog No. 77CH1260-9C), pp. 6-10/13.

- [4] Shuang Deng, Traffic characteristics of packet voice, *IEEE International Conference on Communications*, Vol 3, pp 1369-1374, 1995.
- [5] ITU-T Recommendation G.114, One-Way Transmission Time, Feb 6, 1996.
- [6] ITU-T Recommendation I.366.2, B-ISDN ATM Adaptation Layer Type 2 Specification, November 1996.
- [7] D. T. Magill, Adaptive speech compression for packet communication systems, Conference record of the IEEE National Telecommunications Conference, pp. 29D-1 - 29D-5, 1973.
- [8] Rajakulasingam Raviraj, ATM Trunking using AAL2 for Narrowband Services, *ATM Forum Straw Ballot STR-VTOA-LLTAAT2-1.0*, October 1998.
- [9] Kevin Zhang, Packet Delay Variation in Voice Trunking using AAL2, *ATM Forum contribution 98-0630*, October 4-9, 1998, Australia.