Performance Analysis of Synchronization Communication Protocols for Real-Time Multimedia Services

Tae Gue Kim* and Dong Ho Cho
*Dept. of Computer Engineering, KyungHee Univ. Kihwangup YonginKun Kyunggido, KOREA 449-701
PHONE: +82-331-280-2568
FAX: +82-331-282-4856
E-Mail: tgkim@nms.kyunghee.ac.kr

Abstract

In the real-time delivery of multimedia data streams over networks, the interruption of continuity in a single media stream and the mismatching of the data within the same time interval in multimedia data streams transferred in parallel on different channels are considered as the most serious synchronization problems. To handle those problems, several mechanisms such as synchronization marker, synchronization channels, and synchronization information method are proposed. In this paper, those mechanisms are analyzed and compared in various point of view by the computer simulation. According to the simulation results, it has been shown that not only the segmentation method but also synchronization channel method are superior to the synchronization marker method in view of the real-time transmission and quality of service. On the other hand, it could be seen that the segmentation method is superior to the synchronization channel method with respect to channel utilization.

I. Introduction

Synchronization in distributed multimedia systems is the technique for the receiving side to maintain the inter-media timing relationships of the transmitting side when the real-time multimedia data streams are transmitted through networks. This technique has an important role in maintaining quality of service(QoS) of the multimedia services over broadband integrated services digital network(B-ISDN) based on asynchronous transfer mode (ATM)[1]. Synchronization problem is closely related to the transmission scheme of multimedia data stream in a real-time multimedia system. For example, in the schemes such as office document architecture(ODA) and distributed communication architecture(DCA), there is no inter-media synchronization problem since different data types are integrated into single multimedia document and transmitted through single communication channel[2,3]. On the contrary, when individual interchange data elements(IDE) are transmitted through separate communication channel, the QoS of the multimedia services would be degraded fatally if synchronization control is not applied.

The scheme where individual IDEs are transmitted through the separate communication channel, however, has several advantages over the scheme where multimedia document is transmitted through single communication channel[4].

- The overall transmission delay could be reduced through the parallel transmission.
- Only the required media streams are retrieved, transmission capacity could be used more efficiently.
- As each media stream is transmitted through separate communication channel, there is no interference in the media interface devices.
- As each media stream could be transmitted through the best fit channel, bandwidth efficiency could be maximized.

Currently, most multimedia data stream standards allow the transmission of each media stream using separate communication channel if reconfiguration of multimedia stream is assured at the receiver side[4]. B-ISDN receiving most significant attention as a network which could provide multimedia services in the future also supports separate parallel channels[5,6]. Therefore, it is more desirable to use the scheme in which each media stream is transmitted through separate channel as a transmission mechanism of multimedia data stream in future B-ISDN environment. Especially, in real-time multimedia application, inter-media synchronization is needed since various types of data streams with different real-time constraints are incorporated. In the real-time multimedia applications over networks, the interruption of continuity in a single media stream and the mismatching of the data within the same time interval in multimedia data streams transferred in parallel on different channels are considered as most serious problems[7,8]. In the multimedia data transmission over the network, the random network delay especially makes these two problems, i.e. "continuity" and "mismatching" even more critical.

Recently, there are several studies on the transmission scheme of synchronization control information to solve the synchronization problems for multimedia communications in a real-time environment. The synchronization marker method[9] as well as the synchronization channel method[10] are proposed. Also, the scheme which uses the talkspurt and silence gap of voice stream as a synchronization information of video and data stream[10], is suggested.

However, comparison and objective performance evaluation of these synchronization mechanisms have not been performed yet. Therefore, in this paper, the performances of these synchronization mechanisms are evaluated through computer simulation. In section II, following this introduction, the constraints of synchronization in the ATM networks and synchronization control units are surveyed. Some transmission schemes of synchronization information are inspected in section III. The performances of these schemes are evaluated through computer simulation, and the simulations results are discussed in section IV. Finally, conclusions are made in section V.

II. Synchronization control in the ATM network and synchronization control unit

1. Synchronization Control in the ATM network

Distributed real-time applications require network support with variable bit rates for various types of data having different characteristics. For this reason, ATM network supports multi-connections through high capacity data links.

Thus, in ATM network, the multimedia data of an various applications are separated into multiple streams according to the variable transmission requirement of each medium. Each stream is transmitted on a separate connection[5,6].

A special connection called synchronization channel is used to transmit synchronization commands and parameters[10]. The information carried by this channel contains the control data(commands and parameters) as well as references to the beginning and the end of each control unit. In the case of real-time application, the synchronization channel makes it possible to perform sophisticated synchronization control with involved data stream relations[10]. However, the usage of a synchronization control channel has the drawback requiring more communication resources.

In fact, the required bandwidth will be increased by the introduction of this synchronization channel. Let S be the average time length of a synchronization unit and Ls the average number of bits required for the control of a single unit.

\[ R_s = \frac{L_s}{S} \]  

(1)
If the maximum number of bits required in order to control one unit is $L_{\text{max}}$, the peak bit rate will be greater than $L_{\text{max}}/S$. It could be noticed that the small number of control bits and the large time length of the synchronization control unit would save communication bandwidth. Then, the bandwidth limitation imposes a lower bound on the time length $S$ of the control unit[7,8].

2. Synchronization control based on the synchronization unit

Synchronization control of the real-time data delivery is essentially a rate matching problem of multimedia data streams. In order to display all the data streams in one unit at the same time, the stream which arrives first is delayed until the remaining ones are received. Thus, the synchronization delay will be introduced to the earlier arrivals. When traffic density is high and burstiness is large, the mismatching among all the data streams will gradually increase. The reason of this phenomenon is due to the fact that different network delays are introduced to each packets and frames. For each data stream, these random delays are the origin of gaps and jitters which may occur in the unit. The mismatching will not be recovered until the display of next unit is started. The scheme which introduces an intention delay at the beginning of each synchronization unit in order to filter the gaps, smoothes the jitters and decreases the mismatching at the end of a control unit[8]. That is, in a unit, the first packet of each data stream will be delayed intentionally before being played out. There are two intentional delay schemes as follows.

**Expanding Policy**: The receiver will wait for the late packets(or frames). There is no packet(or frame) discarding. The Expanding policy is more suitable for a video receiver.

**Ignoring Policy**: The receiver will preserve the timing at the expense of ignoring late packets(or frames). The receiver will drop late data. The Ignoring policy can only be used for a voice receiver because it is difficult to meet the high quality requirements of the video data.

Meanwhile, the effects of random network delay on the synchronization of synchronization unit are as follows. Assume $N$ data streams are included in a synchronization control unit. At the end of the control unit, $T_{i_j}$ is the mismatching tolerance between data streams $i$ and $j$ (i,j \leq S). The $t_i$ and $t_j$ are defined as the average random delay between the display of two adjacent packets(or frames) in the stream $i$ and $j$, respectively. Let $\mu(t)$ be the time slot of one i(j) data packet or frame. Then the synchronization unit with a time length $S$ is constrained by following relation[8].

$$\left(\frac{S}{t_i} - 1\right) \times 10^{-9} = \left(\frac{S}{t_j} - 1\right) \times 10^{-9} \times T_{i_j} = 10^{-9} \times (1S < N, 1j < N)$$

Equation (2) imposes an upper bound of the value $S$ for a synchronization unit. Therefore, $S$, the size of the synchronization unit in terms of time length, is bounded by both the expressions (1) and (2). Because of the bandwidth limitation on the control connection, $S$ can not be too small. On the other hand, the larger $S$ results in more QoS degradation due to synchronization loss at the end of a unit. Therefore, in order to maximize the efficiency of transmission channel and to maintain QoS reasonably, it is necessary not only to select the length of synchronization unit reasonably according to the characteristics of the application but also to manage it efficiently.

### III. Transmission Scheme of Synchronization Control Information

1. The synchronization marker method

In the scheme which uses synchronization marker, the sending side inserts synchronization markers into the each individual data stream from the real-time multimedia devices such as video cameras, microphones, and standardized databases. The receiver side uses these synchronization markers to synchronize the multimedia data streams. The scheme is depicted in Figure 1. As each data stream is transferred through separate communication channel, the network delays experienced during transfer are different from each other. Synchronization markers from each individual data stream of a synchronization unit arrive at the receiver at different time due to these differences in network delay. Therefore, the receiving end waits until synchronization markers from all streams of a multimedia application arrive. The receiving end performs synchronization through the user interfaces when all synchronization markers arrive.

![Fig. 1. Synchronization using synchronization markers](image)

This scheme could be applied with little modification to the conventional communication protocols. The overhead of this protocol is very small because there is no need for separate control channel and because synchronization is possible with a little additional information. The shortcomings of this protocol is that large buffer is needed due to differences in errors, jitters, and transmission rates of each connection. Another shortcomings of this protocols is that there is some interference at the interface device due to the modification of user data stream by the transmission system.

2. Synchronization channel method

The scheme which uses separate synchronization channel could support more complicated and sophisticated control than the scheme which uses synchronization marker. The concept of this scheme is depicted in Figure 2.

![Fig. 2. Synchronization using separate synchronization channels](image)
3. The synchronization using segmentation

The scheme which uses segmentation could provide synchronization without the use of separate synchronization information by defining synchronization unit (i.e., segment) based on the voice stream well. This scheme solves not only the interference problem of the scheme which uses synchronization marker but also the processing overhead of the synchronization channel method.

This scheme is based on the packetized voice transmission technique. As the voice traffic is very sensitive to the delay, the delay of the voice traffic channel should be bounded by the certain limits. If the delay and jitter constraints of the voice traffic stream is satisfied, the information on the length of talkspurt and silence gap extracted from the voice traffic stream could be used as a synchronization information of the text and video stream. For example, if the video stream is segmented according to the talkspurt and silence gap of the voice stream, it is possible to synchronize the video stream using the synchronization information extracted from the voice stream.

This scheme is depicted in Figure 3. Synchronization unit (segment) is composed of the talkspurts or the silence gap of the voice traffic, video and text data with the same length as the silence gap or the talkspurts of the voice traffic. Assume a multimedia application is composed of voice and video data with empty time periods. For example, if the voice stream is in the silence gap, all the video frames generated during this period constitute a segment.

![Fig. 3. Segmentation of the real-time voice and video](image)

In Figure 3, we denotes the identifier of voice stream as k, the identifier of video stream j, and the n-th segment in each stream as S_k^n and S_j^n.

Here, S_k^n and S_j^n, S_k^{n-1} and S_j^{n-1} denotes the case when the voice stream is silence gap and there exists only the video stream. In this case, the video data corresponding to the silence gap constitutes a segment.

There are two important features of this scheme[8,9]. First, as the segmentation is based on the voice data stream, the scheme is well suited to today's packet-switching voice system. Therefore, it is a very practical and reliable way to construct the segment on this model. On the other hand, any other scheme with a smaller segment size will break the talkspurts. This interruption is critical since user is always very sensitive to the voice medium. Second, since the non-empty component does not wait for the empty one for the display, the length of the empty component (e.g., silence gap) during the delivery could be adjusted dynamically according to the arrival of non-empty component, which is interrupted by the random network delay. As the empty voice component is always followed by a non-empty one, this scheme supports the continuity of data streams. Moreover, this scheme offers a more smooth representation of video content with less occurrence of gaps. Also, there is no need for extra control connection or insertion of synchronization marker because the segment is marked naturally by the voice data stream itself.

IV. Simulation results and discussion

1. Simulation environment

The cause of the synchronization problem in real-time multimedia system is the random delay experienced during the transfer of each media through the network. Thus, it is necessary to define the network delay in detail. In this paper, we have considered the network delay variables for the all packets (or frames) as independent, identically distributed random variable. Then, we have been in the network would have a very dynamic routing scheme. Also, to simplify the simulation process, it is assumed that the distribution of random network delay is exponential.

2. Traffic Modeling

(1) Modeling of data source

It is assumed that the distribution of data source is the poison process and that the average data rate is 1200 bps.

(2) Modeling of voice source

The behavior of the speech terminal could be modeled as two-state Markov process as shown in Figure 4.

![Fig. 4. Two-state Markov process model of voice traffic](image)

The voice source generates pattern of talkspurts and silence gaps based on the speech activity detector[12]. There are principal spurts and gap related to the talking, pauses, and listening patterns of a conversation. All spurts and gaps are independent and they have exponentially distributed durations. The transition probability from talkspurt to silent gap, γ, within a time slot is given by

\[ \gamma = 1 - \exp(-1/t_2) \]

where t_2 is the mean duration of a principal talkspurt in seconds.

This is essentially the probability that a talkspurt ends within γ seconds. Correspondingly, the transition probability from silence state to talking state during a time slot is

\[ \sigma = 1 - \exp(-1/t_3) \]

where t_3 is the mean duration of a principal gap. Then, we assume that mean duration of talkspurt, t_1, is 0.352 sec and that mean duration of silence gap, t_3, is 0.650 sec.

3. Simulation results and discussion

The performances of the scheme which uses synchronization marker as well as the scheme which uses separate synchronization channel are shown in Figure 7 and 8. Here, the synchronization degree is normalized on the average duration of talkspurt of the voice traffic. For example, synchronization degree 10, means that the duration of the synchronization unit is 1/10 of the average talkspurt duration. The scheme which uses segmentation is excluded in this comparison since the synchronization degree is fixed. Figure 7 shows the comparison of blocking probability of the voice and video traffic in each scheme as the function of
synchronization degree. It can be seen from the Figure 7 that if the synchronization degree is 1, the difference between two scheme is negligible. However, as the synchronization degree is increased, the performance differences between the two scheme becomes larger gradually. In the scheme which uses separate synchronization channel, the effect of synchronization degree on the blocking probability is not critical. On the other hand, in the scheme which uses synchronization marker, the blocking probability is increased significantly as the synchronization degree is increased. The cause of this phenomenon is as follows. When the synchronization unit is subdivided, the insertion of the synchronization marker cause the delay of user information. Then, since the synchronization markers experience different network delay, the delay jitter increases rapidly when transferred through network nodes. The average delay characteristics between the synchronization marker and the separate synchronization channel is depicted in Figure 8. In this Figure, the similar results mentioned above could be derived. In the scheme which uses synchronization marker, if the synchronization unit is subdivided, the average delay is increased rapidly. Since the number of synchronization marker inserted into the user data stream is increased as the synchronization degree is increased, the overall average delay is increased. On the other hand, in the scheme which uses separate synchronization channel, the effects of synchronization degree is negligible.

The channel utilization characteristics of three synchronization schemes are shown in Figure 9 as the function of average network delay. As the average network delay is increased, the channel utilization of the separate synchronization channel method is decreased rapidly. On the other hand, the channel utilization ratio of the synchronization marker method is not affected by the average network delay. Also, in the segmentation method, the channel utilization is sustained constantly since there is no need to transmit synchronization information.

Meanwhile, the intentional delay schemes of the control device also affect the performances of the synchronization scheme. In Figure 10, 11, 12, the simulation results of the each synchronization scheme are shown when ignoring policy is applied to the voice traffic and extending policy is applied to video traffic. It can be seen from Figure 10 that the blocking probability of the voice traffic could be reduced significantly by increasing the intentional delay of the video frame at the start of the each synchronization point. This phenomenon is due to the fact that the intentional delay of the video frame admits the larger delay variation between voice packets. In the comparison of each synchronization scheme, it would be shown that the blocking probability of the synchronization marker method is greatest due to the delay variation of the synchronization marker. On the other hand, the blocking probability of the segmentation method is almost negligible since it is not needed to transmit synchronization information.

The average delay characteristics of the synchronization schemes are shown in Figure 11. In this Figure, it could be seen that the average delay characteristics of the synchronization marker method is increased exponentially as the intentional delay of the video frame is increased. Such phenomenon could be explained by the fact that delay variance caused by the synchronization marker inserted into each data stream is increased as the intentional delay is increased. However, the average delay of separate synchronization channel and segmentation methodologies is not increased significantly since additional jitters or gaps due to the transmission of the synchronization information are not created. Also, it can be seen that the differences in the average delay is increased gradually as the intentional delay is increased. In the scheme which uses separate synchronization channel, the transmission capacity for user traffic is decreased by the use of separate synchronization channel. The effects of use of separate channel could be increased as the appropriate intentional delay is selected.

The channel utilization characteristics are shown in Figure 12. It could be seen from this Figure that the channel utilization of both the separate synchronization channel and the segmentation method is decreased slightly since additional delays and blocking are very small when the intentional delay of the video is increased. On the other hand, the channel utilization of the synchronization marker method is decreased rapidly since the delay jitter caused by the intentional delay of the video frame increases the blocking ratio.

As discussed above, in the scheme which uses the synchronization marker, the blocking probability is increased significantly as the synchronization degree is increased. Thus, this scheme could not meet the real-time synchronization constraints when sophisticated synchronization control is required by the application. Thus, this method is inappropriate as the synchronization mechanism of the real-time multimedia system.

Therefore, it could be seen that both the separate synchronization channel and segmentation methods are more desirable as synchronization methodologies for the future distributed real-time multimedia system.

V. Conclusions

In this paper, the performances of the synchronization scheme for the real-time distributed multimedia system are investigated. First, the basic problems of the synchronization control are surveyed in respect to the protocol structure, network resources, and the requirement of the control device. The transmission schemes of the synchronization information among network nodes distributed geographically could be classified into three schemes such as the synchronization marker, the separate synchronization channel, and the segmentation method. In this paper, the performances of these synchronization schemes are analyzed and compared through the computer simulation.

It could be seen from the simulation results that both the separate synchronization channel and the segmentation methodologies are superior to the scheme which uses synchronization marker in view of real-time transmission and QoS since the delay jitter and blocking is sustained very small. On the other hand, it could be seen those the performances of the segmentation scheme is superior to that of separate synchronization channel and synchronization marker in respect to the channel utilization.

References

Fig. 7. Blocking probability characteristics of each synchronization method vs. the synchronization degree

Fig. 8. Average delay characteristics of synchronization methods vs. the synchronization degree

Fig. 9. Channel utilization characteristics of synchronization methods vs. the network delay

Fig. 10. Blocking probability characteristics of each synchronization method vs. the intentional delay of video frames

Fig. 11. Average delay characteristics of each synchronization method vs. the intentional delay of video frames

Fig. 12. Channel utilization characteristics of each synchronization method vs. the intentional delay of video frames