

A Multimedia Synchronization Model and Its Implementation in Transport Protocols

Chun-Chuan Yang and Jau-Hsiung Huang, *Associate Member, IEEE*

Abstract—An implementation of a synchronization mechanism in transport protocol to support multimedia applications over a packet or cell switched network is proposed. In designing such a mechanism for practical use, ease of implementation and capability of handling random delay of packets are two key issues for success. Since the random delay of packet or cell switched networks makes synchronization among media more complicated after the transmission across the network, a model which considers the random transmission delay is hence required to specify the temporal relationship among media. Therefore, a real-time synchronization model is presented to satisfy this requirement in the paper. Based on the proposed synchronization model, a transport protocol, namely multimedia synchronization transport protocol, is designed and implemented. The performance measurement of such a protocol is also presented.

I. INTRODUCTION

WITH THE increasing power of modern computers and high-speed network technologies, the integration of multiple media into a single network application becomes possible; such as video conferencing, shared workspace and groupware systems [1]–[8]. For multimedia applications, synchronization is a very important issue which means the inter-media temporal relationship has to be satisfied across the network. To tackle such a problem, we have to first examine the characteristics of packet/cell switched network and the characteristics of media.

Two major problems of packet or cell switched network supporting real-time multimedia applications are random delay and lost packets, due in large part to buffer overflow. Hence, the temporal relationship among media may be destroyed at the receiver even if they were sent in synchronization by the sender. Worse yet, since each medium is usually transmitted through a separate connection and the quality of service of connections may be different; the delay characteristics of different media may be different. Thus, a synchronization mechanism should be implemented at the receiver to regain the synchronization among media.

Much work [9]–[13] has been done in describing the synchronization model for multimedia applications. Among those, Petri-net based models provide a good method to specify temporal relationships. A modified Petri-net model, object

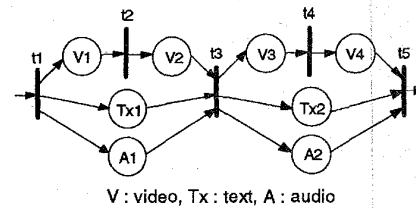


Fig. 1. An example of Petri net based synchronization model.

composition Petri net (OCPN) model [10], [11], has been proved to have the ability to specify any arbitrary temporal relationship among media and it may be applied to both stored-data applications and live applications. The eXtended OCPN (XOCPN) [12] can additionally specify the communication functions. In [13], user actions are also considered. However, when taking the real-time issue of multimedia and the random delay of packet/cell networks into consideration, OCPN/XOCPN and other Petri net based models are not sufficient to deal with the *late transmission* of packets. Here, we define *late transmission* of a packet to be that the packet fails to reach its destination in time (i.e., exceeding its real-time constraint) and should be dropped. We explain the insufficiency of OCPN by the example shown in Fig. 1 in the following.

We may treat the example in Fig. 1 as an MTV-like application in which there is music (audio) and corresponding video frames and text (subtitle) on the screen. In Fig. 1, we only show two cycles of a cyclic model of the application and the interpretation of transition t_1 to t_3 follows. Simultaneously, the application plays audio A_1 , displays two video frames V_1, V_2 orderly, and displays text Tx_1 . Fig. 1 shows that transition t_3 will be fired only after V_2, A_1 and Tx_1 finish playing. Similarly, transition t_5 will be fired only after V_4, A_2 , and Tx_2 finish playing.

In the following, we examine the characteristics of audio, video, and text data. For audio data, it is very jitter sensitive and cannot tolerate random delay between audio segments; hence, by holding A_2 too long after finishing A_1 will result in unrecognizable audio quality. For video frames, by dropping a small portion of video frames or by having a small random delay between video frames may only have a minor impact on human eyes and may be acceptable. For text data, it has the least jitter sensitivity compared to audio and video. Based on such an observation, we notice that audio data is much more jitter sensitive than video and text data.

Manuscript received August 7, 1994; revised April 5, 1995.

The authors are with the Communications and Multimedia Laboratory, Department of Computer Science and Information Engineering, National Taiwan University, Taipei, R.O.C., Taiwan (e-mail: jau@csie, nut.edu.tw).

Publisher Item Identifier S 0733-8716(96)00228-4.

In Fig. 1 based on OCPN model, if A_1 and Tx_1 have finished playing but V_2 is late due to network congestion, transition t_3 can not be fired until V_2 arrives and finishes playing. For this late transmission of V_2 , A_2 can not be played even if its data has arrived. However, this is improper since audio is very jitter sensitive as explained earlier. Therefore, a natural thought in handling this situation is to fire transition t_3 right after finishing A_1 to activate A_2 in order to maintain the audio quality. Clearly, the unfinished video frames in V_2 will be dropped.

However, any of the earlier proposed Petri net models does not allow firing transition t_3 before finishing V_2 . More precisely, they only describe the synchronization relationship under perfect network condition or require a known network delay bound for video-on-demand type of applications. In wide area packet/cell networks, it is impractical to make such an assumption since the upper bound of network delay is too large to support live applications such as video conference. Hence, a new model will be proposed in this paper and is named as *Real-Time Synchronization Model (RTSM)*. As will be explained in Section II, the definition and firing rules of RTSM are different from the Petri-net based models.

There is another question raised after defining the RTSM model. That is, where do we place the synchronization mechanism in the network architecture? We have two choices for the answer: 1) in the application, or 2) in the transport protocol. To place the synchronization mechanism in the application means that the application has to handle all the synchronization control for itself, which will introduce more overhead to the application. However, more complex synchronization relationships can be supported if the synchronization mechanism is implemented in the application. Multimedia titles and documents with complex synchronization relationships and no repeated cycles (as in Fig. 1) may be applications of this category.

On the other hand, we can place the synchronization mechanism in the transport protocol. A transport protocol supporting synchronization mechanism is trying to simulate an end-to-end connection as a circuit, where the receiving application receives data with the same temporal relationship as they were sent by the sender. Hence, the application at the receiver simply has to receive and play data since the synchronization is provided by the transport protocol. Thus, the design of applications is simplified. Applications such as video conference where a simpler synchronization relationship with repeated cycles may belong to this category. In this paper, we apply the proposed RTSM model in the design of a new transport protocol, namely *Multimedia Synchronization Transport Protocol (MSTP)* for synchronization purpose.

As to the part of the transport protocols, much work [14]–[24] has been done in discussing high-speed transport protocols. The goals of these protocols are focused on high-throughput or low-latency. The issue of real-time synchronization in multimedia applications is rarely addressed [25]–[29]. In [25], Huitema and Dabbous discussed the synchronization relationship between different types of data. However, that model did not address the real-time characteristics of media, and it could not achieve more complex synchronization rela-

tionship since it had only two types of data (RT, DT). In this paper, MSTP is presented to support more general services.

The remainder of this paper is organized as follows. In Section II, the RTSM and the concept of *key medium* and *time medium* are presented. The goal of key medium and time medium is to handle situations when data must be dropped due to late transmissions. Section III describes the architecture, the packet format, and the synchronization control mechanism of MSTP. The implementation and performance evaluation of MSTP are discussed in Section IV. Finally, Section V concludes this paper.

II. REAL-TIME SYNCHRONIZATION MODEL (RTSM)

Before introducing the definition of RTSM, we explain some terminologies used in the model. The elements in RTSM include *place*, *token*, and *transition*, as in the OCPN model [10]. The places are used to represent the medium unit (such as audio segments, video frames, or text strings) and their corresponding actions (such as playing audio, displaying video, or text on the screen). There could be one or zero *token* within a place to represent the state of the place. A place without token means the place is currently inactive. A place with a token is active and could be in one of the two states: token is *blocked* or *unblocked*. When a token is just added into a place, the corresponding action of the place can be executed, and before finishing the action, the token is blocked. The token in the place is unblocked after the action is finished. Transitions are used to represent the synchronization relationship between places. However, the places in RTSM are divided into two types further, and a new firing rule for RTSM is also proposed for the real-time issue. We explain the new idea in the following.

Considering the example in Fig. 1, in order to avoid the unexpected blocking due to late transmission, we need a mechanism to enforce some blocked transitions to be fired under the situation of late transmissions. In RTSM, there are two kinds of places as *regular places* and *enforced places*. To differentiate an enforced place from a regular place, a double circle is drawn for enforced place as shown in Fig. 2 where transition t_3 is fed by two regular places, V_2 and Tx_1 , and one enforced place, A_1 .

The firing rule about enforced places is that once an enforced place gets unblocked, the transition following it will be immediately fired regardless of the states of other places feeding this transition. Therefore, if A_1 becomes unblocked in Fig. 2, transition t_3 will be immediately fired regardless of the states of V_2 and Tx_1 . At the same time, tokens in the places before transition t_3 , such as V_1 , V_2 or Tx_1 , must be cleared since these places are obsolete due to the firing of t_3 . We call this action of clearing all obsolete tokens *backtracking*, since the action is backtracked from the fired transition in RTSM. In this way, the synchronization anomaly due to late transmission is solved. The following are the complete definition, firing rules and backtracking rules of RTSM. Notice that with the inclusion of enforced places and enforced firing, this synchronization model is different from Petri net in nature. However, there are still some similarities

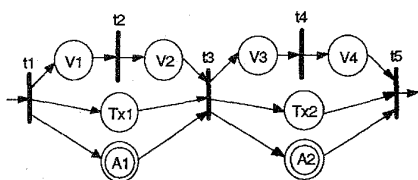


Fig. 2. An RTSM model for the example in Fig. 1.

between them, such as graphical presentation and definition of places and transitions.

A. Definition of RTSM

Definition 1: RTSM is a seven-tuple $\{T, P, E, A, D, R, M\}$, where

$T = \{t_1, t_2, \dots, t_n\}$	Transitions.
$P = \{p_1, p_2, \dots, p_m\}$	Regular places (single circles).
$E = \{e_1, e_2, \dots, e_s\}$	Enforced places.
$S = P \cup E$	All places.
$A: \{T \times S\} \cup \{S_T\}$	Directed arcs.
$D: S \rightarrow \text{Realnumber}$	Time duration of places.
$R: S \rightarrow \{r_1, r_2, \dots, r_k\}$	Type of media.
$M: S \rightarrow \{0, 1, 2\}$	State of places.

Each place may be in one of the following states:

0: no token	
1: token is blocked	Cross in the place.
2: token is unblocked	Dot in the place.

Definition 2: To initiate RTSM is to add a token to the initial place.

Definition 3: Firing rules of RTSM

Case (a): Transition t_i does not contain any enforced place in its input places.

- 1) Transition t_i fires immediately when each of its input places contains an unblocked token.
- 2) Upon firing, transition t_i removes a token from each of its input places and adds a token to each of its output places.
- 3) After receiving a token, a place p_j remains in the active state for the interval specified by the duration τ_j . During this interval, the token is blocked.

Case (b): Transition t_i contains at least one enforced place e_i in its input places.

- 1) When the token in any one of the enforced places e_i becomes unblocked, transition t_i is fired regardless of the states of other input places.
- 2) Upon firing, a set of *backtracking rules* is exercised to remove tokens from their input places. The backtracking rules will be explained in the following. Meanwhile, transition t_i adds a token to each of its output places.

To maintain the consistency of the progress of RTSM, a backtracking mechanism is required. For instance, considering a run-time case of Fig. 2: tokens in Tx_1 and A_1 are unblocked, and the token in V_1 is blocked, i.e., actions associated with places Tx_1 and A_1 are finished while the action with place V_1 is still in process. Since place A_1 is an enforced place, transition t_3 fires when the token in place A_1 becomes unblocked. To fire transition t_3 , we have to remove the tokens in the places before transition t_3 , and add a token into each

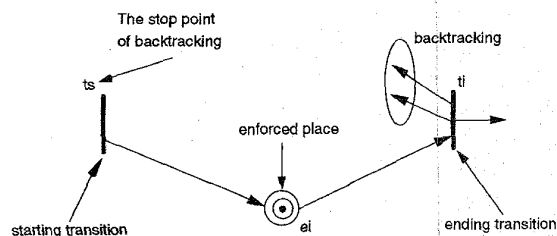


Fig. 3. The starting and ending transitions of the enforced place.

of the output places of transition t_3 : V_3 , Tx_2 and A_2 . That is, we have to remove not only the tokens in places Tx_1 and A_1 (place V_2 contains no token in this case), but also the token in place V_1 . By applying the firing rule case(a)-(2), which is used in the case of nonenforced firing, to this case is not enough, since it only removes tokens in the input places of the fired transition. Therefore, a backtracking action to remove tokens in the places before transition t_3 is required, and the backtracking action will remove the token in place V_1 in this case.

In summary, when an enforced firing occurs, there may exist some places which are in front of the fired transition and contain tokens within them, so we have to remove these obsolete tokens to match the correct state of the model. We may start the clearing action from the initial place of RTSM until reaching the just fired transition. However, backtracking is a more efficient way since it eliminates unnecessary traversing. As shown in Fig. 2, transition t_3 will be immediately fired as long as A_1 finishes playing regardless of the states of Tx_1 or V_2 . However, by firing t_3 , if there are tokens in Tx_1 , V_1 or V_2 , these tokens should be removed since their contents are skipped. In the following, we will describe the backtracking rules to remove tokens under such situations. The backtracking rules will first remove the token from the unblocked enforced place, then the following three rules should be exercised.

Backtracking rules of RTSM:

Rule 1: The action of backtracking stops at the starting transition t_s of the enforced place e_i or at the initial place as shown in Fig. 3.

Rule 2: During backtracking, if a place containing a token (either blocked or unblocked) is backtracked, then remove the token and stop the backtracking of this branch.

Rule 3: During backtracking, if a transition is reached with nonbacktracked output places, fire this transition to activate the nonbacktracked output places.

We use Fig. 4 as an example to explain the backtracking rules. As shown in Fig. 4(a), enforced place e_i becomes unblocked and will fire transition t_3 with place p_j being active. Applying rule (2) of the backtracking, the token in p_j will be removed as shown Fig. 4(b). Similarly, transition t_2 will be fired and a token will be added in p_k since it is a nonbacktracked output place as explained in rule (3).

In the following, we examine the effect of enforced places and the backtracking rules by looking at all possible cases. For an enforced place, we show that the relation between this enforced place and the rest of the RTSM must be in one of the

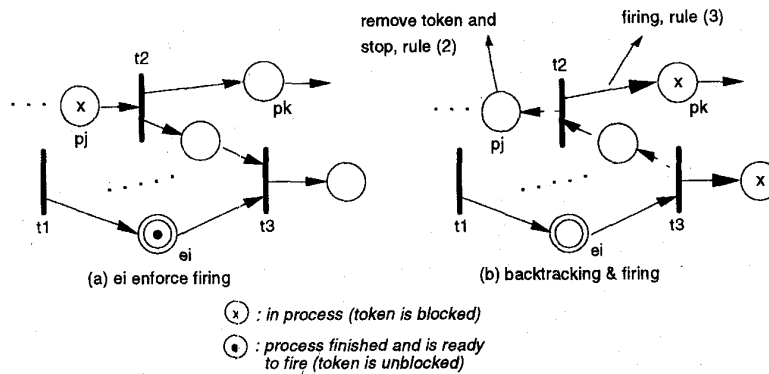


Fig. 4. The backtracking of the firing rules for RTSM.

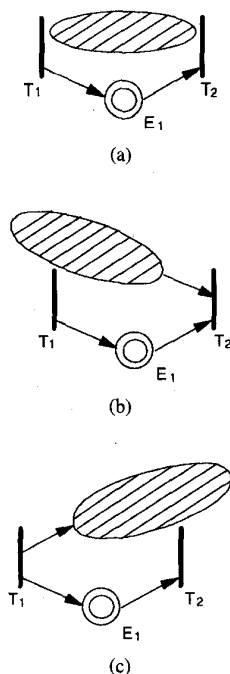


Fig. 5. Three relations between enforced places and the RTSM.

three cases as shown in Fig. 5. Fig. 5(a) shows that the places affected by the enforced place E_1 is bounded by transitions T_1 and T_2 , which are the starting and ending transitions of E_1 respectively. An example of this case is shown in Fig. 6(a). In this case, it means that all transitions and places affected by the enforced place are within the area bounded by T_1 and T_2 . Thus, the firing of T_2 due to the enforced place E_1 means that the unfinished work within the area bounded by T_1 and T_2 is late and has exceeded the real-time constraint. Hence, we have to exercise the backtracking rules to remove all tokens in this area after firing transition T_2 . Fig. 6(b) shows the result of Fig. 6(a) after backtracking and firing.

Case (b) of Fig. 5 shows that the places affected by E_1 reach beyond T_1 and an example of this case is shown in Fig. 7(a). This case is similar to case 5(a) where E_1 enforces a firing in transition T_2 means that all unfinished work in the branches

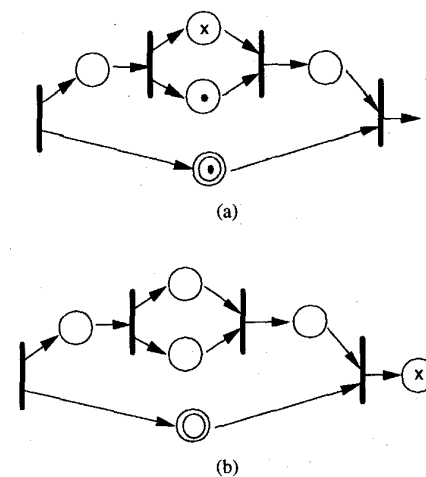


Fig. 6. Example of backtracking of Fig. 5(a).

coming into the area bounded by T_2 is also out of the real-time and should be discarded. Therefore, we remove all tokens in this area after firing transition T_2 . The result of Fig. 7(a) after backtracking and firing is shown in Fig. 7(b).

Lastly, case (c) of Fig. 5 shows that the affected places by E_1 reach beyond T_2 . In this case, not all places and transitions in the shaded area are fully controlled by the enforced place E_1 . An example of this case is shown in Fig. 8(a) where place p_j is not controlled by enforced place e_i for synchronization. Hence, the enforced firing of transition T_2 does not mean that we should remove all tokens in the shaded area. The solution to this situation is described in Rule (3) of the backtracking rules. That is, during backtracking, when meeting a transition T_i with output places which are not bounded in the range of backtracking, we enforce this kind of transitions to be fired. The result of Fig. 8(a) after backtracking and firing is shown in Fig. 8(b).

B. Concept of Key Medium and Time Medium

1) *Key Medium*: In this section, we will discuss how to assign enforced places to packets of different media. By definition, enforced places control the advance of time of multimedia applications. That is, if the places of a specific

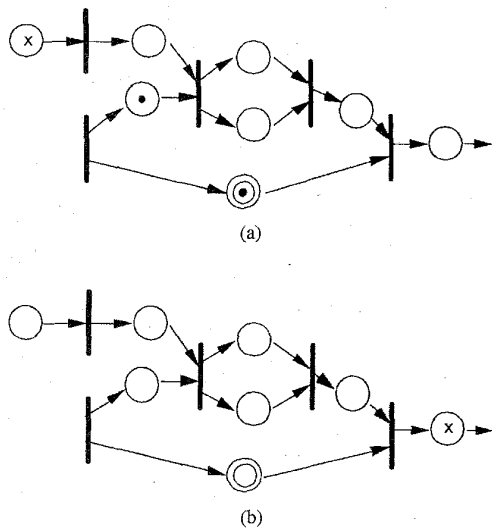


Fig. 7. Example of backtracking of Fig. 5(b).

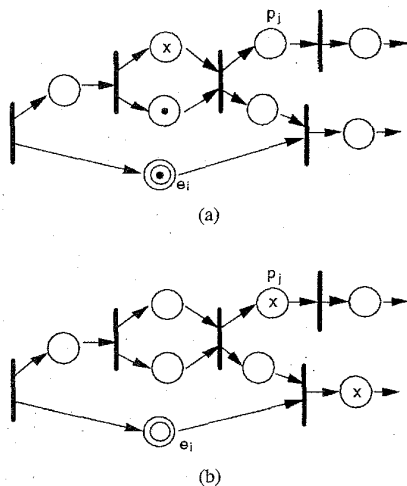


Fig. 8. Example of backtracking of Fig. 5(c).

medium, such as voice, are assigned as enforced places, then this medium will not be blocked due to late transmissions of other media for synchronization. Further, this medium will drive other media to advance in time if late transmissions happen to them. We denote this specific medium as the *key medium*. The concept of key medium is similar to that of master/slave policy proposed in [26], but is more compact for network applications.

Therefore, two factors should be considered in deciding the key medium. First is the importance of the medium and second is the jitter-sensitivity of the medium. For instance, if the quality of one medium is more important than that of others, then this medium should be chosen as the key medium. Or, if the importance of all media is the same, then the most jitter sensitive medium should be chosen as the key.

Note that all media except the key medium are subject to packet dropping due to late transmissions. Hence, if there is one medium requiring error free transmission, such as numerical data, then this medium should be either chosen as

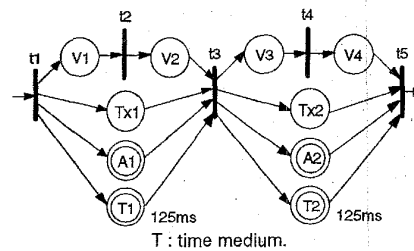


Fig. 9. Using time medium in RTSM model.

the key medium or should not be controlled by another key medium. Fig. 2 shows an example where text data and video data can be dropped due to late transmissions when audio data gets unblocked. Hence, in this case, error free transmission is not guaranteed to text and video transfer.

2) *Time Medium*: Since all media are transmitted across the network, it is also possible for the key medium to suffer a long delay such that its real-time constraint is exceeded. Therefore, what should be done if A_1 in Fig. 2 is still waiting for its packets due to late transmissions while its real-time constraint has expired? In such a case, it is reasonable to fire transition t_3 to activate A_2 , V_3 and Tx_2 instead of waiting for A_1 in order to maintain the quality of audio. To achieve this function, the introduction of key medium is not sufficient and we need a mechanism to express absolute time (the real-time constraint) in the model. For this goal, we introduce *time* as one of the media, named *time medium*, and can be placed in the synchronization model of multimedia applications.

Clearly, time medium is a *virtual medium* and normally contains a deterministic duration of time which specifies the real-time constraint between two transitions. Note that time medium can be assigned as regular places or enforced places. By assigning the places of time medium to be enforced places means that the transition following the enforced place of the time medium will be fired when the duration of time specified by the time medium expires, even if the places of other media are still blocked.

With the introduction of time medium, we can model the example of Fig. 2 to be that shown in Fig. 9. Note that in Fig. 9, both places of audio and time are assigned as enforced places. Fig. 9 shows that transition t_3 will be fired if either one of the following two cases is true. One is that A_1 has finished playing and becomes unblocked; or, 125 ms has passed by after the firing of transition t_1 . Hence, if A_1 is blocked for a long time due to late transmission, then A_1 will be skipped when time medium T_1 becomes unblocked. This resolves the problem of late transmissions of key medium.

III. MULTIMEDIA SYNCHRONIZATION TRANSPORT PROTOCOL (MSTP)

After the introduction of RTSM, we explain how to implement RTSM in the proposed MSTP protocol in this section.

A. The Architecture of MSTP

MSTP adopts a multi-protocol architecture [31] as shown in Fig. 10. For an application with multiple media, the MSTP

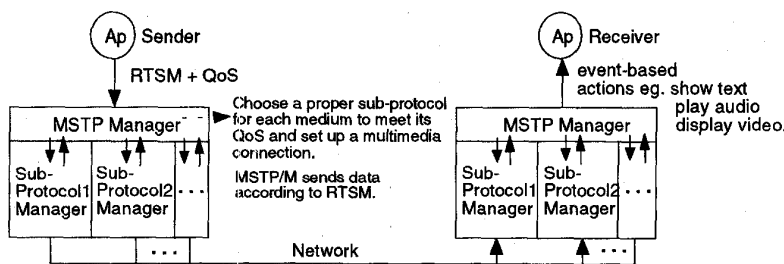


Fig. 10. The architecture of MSTP.

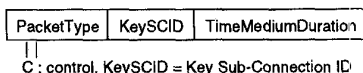


Fig. 11. The format of the control packets.

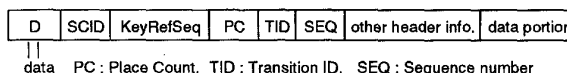


Fig. 13. The format of the data packets.

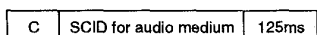


Fig. 12. The format of the control packet for the example in Fig. 9.

manager sets up a single multimedia connection; and for each medium, it employs a sub-protocol to set up a sub-connection with its own quality of service (QoS). Each sub-protocol manager manages a sub-protocol connection, which is actually a transport layer connection, and the function of the MSTP manager is to keep track of the operation of all sub-protocol connections such as setup and release of each sub-protocol connection. Note that all sub-protocol connections managed by a single MSTP manager belong to a single multimedia application. The purpose of MSTP is to provide a synchronized data stream from sender to receiver. That is, the sender sends data according to a synchronization relationship and MSTP will deliver data to the application at the receiver conforming to the original synchronization relationship.

Briefly speaking, the MSTP manager at the sender (sending MSTP manager) accepts RTSM and QoS for each medium from the application program, and sets up a multimedia connection with the MSTP manager at the receiver (receiving MSTP manager). After the connection is set up, the sending MSTP manager will send packets according to RTSM, and the receiving MSTP manager will deal with out-of-synchronization situations and deliver proper packets to the receiver application. In this way, applications at both sides do not need to deal with the control for synchronization because MSTP manager will handle it. This simplifies the design of applications programs.

B. Packet Format of MSTP

There are two kinds of packets sent by the MSTP manager: control packets and data packets. Control packets are used to send control information to set up, release and change configuration of connections. The synchronization related configurations are: 1) which sub-connection is the key medium sub-connection (denoted by KeySCID), and 2) the duration of the time medium (denoted by TimeMediumDuration) which is accompanied with the key medium. The sending MSTP

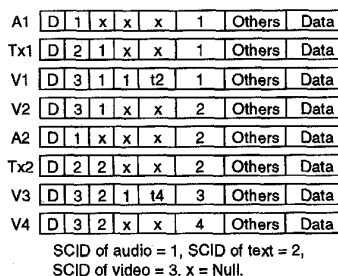


Fig. 14. The format of data packets.

manager sends control packets to inform the receiving MSTP manager about current configuration of the connection. Fig. 11 shows the format of control packets. Note that the format only addresses the issues of synchronization and does not show other header information such as port numbers, QoS, etc. The control packet for the example in Fig. 9 is shown in Fig. 12 which has to be sent to the receiving MSTP manager before the connection is set up.

The packet format of data packets is shown in Fig. 13. The field SCID (subconnection identifier) is used to indicate which sub-connection this packet belongs to. The KeyRefSeq field indicates the sequence number of the key medium packet with which this data packet has to be synchronized. Notice that the KeyRefSeq fields of key medium packets is null. Fields PC and TID are used for packets of places which do not feed into the same transition with the key medium, such as V₁ and V₃ in Fig. 9. Field PC indicates the total number of places that feed into the transition specified in the field TID.

For example, in Fig. 9, the packet containing V₁ will have PC = 1 and TID = 2, which means transition t₂ is fed by only one medium. Hence, transition X will be fired when PC number of packets with TID = X have arrived. Field SEQ indicates the sequence number of this packet in this sub-connection. Other header information includes port numbers, QoS, etc. The data portion field contains the real data of the packet.

We again use the example in Fig. 9 for illustration. The data packets for the example are shown in Fig. 14. Note that fields

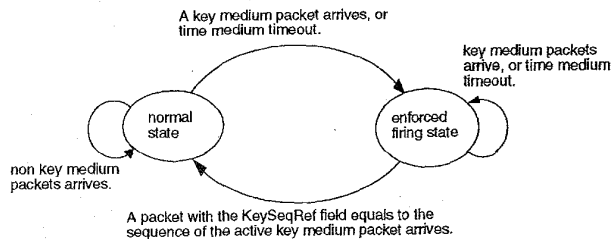


Fig. 15. State transition diagram for the sub-protocols with error control.

PC and TID for Tx_1, Tx_2, V_2 , and V_4 are null since they feed into the same transitions with the key medium.

C. Interactions Between MSTP Manager and Sub-Protocol Managers

In the following, we will explain how the receiving MSTP manager decide when to fire a transition which is fed by at least one key medium packet (an enforced place). Normally, this transition will be fired after the key medium packet has finished playing. However, this requires the receiving MSTP manager to estimate the time to finish playing of the packet and thus introduces more computing overhead and also needs a timer for triggering the event. In order to avoid the overhead, we adopt a looser policy. That is, the receiving MSTP manager will fire the transition right after the arrival of next key medium packet. For example, transition t_3 in Fig. 9 is enforced to be fired after A_2 arrives, which means if any of V_1, V_2 or Tx_1 has not yet arrived when A_2 arrives, it is considered to be out-of-synchronization and will be dropped. We will explain in detail the operations of the MSTP manager in Section III-D.

We classify sub-protocols into two categories, with error control and without error control. Sub-protocols which use acknowledgments and retransmission mechanisms for lost packets, if real-time constraints are not violated, belong to protocols with error control. Sub-protocols without error control will not retransmit lost packets. Actions of sub-protocol managers, such as send, receive, and discard packets, are under control of the MSTP manager.

For a sub-protocol manager with error control, it accepts send requests from the MSTP manager and sends packets with proper header information to the corresponding sub-protocol manager at the receiver. Notice that the receiving MSTP manager does not know the RTSM at the sender, so there is no way to inform sub-protocols at the receiver about the sequence number of the next packet to receive once a transition is enforced to fire. For example, if the application RTSM is as shown in Fig. 9, once transition t_3 is enforced to fire, the video sub-protocol at the receiver could not decide the sequence number of next packet to receive since it does not know how many video places (packets) are accompanied with A_1 by not knowing the source RTSM. What we only know in such a situation is the sequence number of the currently active key medium packet, i.e., A_2 in the example.

Therefore, we define two states for non key medium sub-protocols with error control: 1) *normal state*, and 2) *enforced firing state*. Initially, the sub-protocol is in the normal state, and it must deliver the arriving packet in sequence. In the

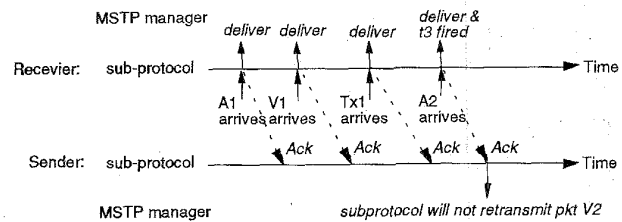


Fig. 16. An example of which MSTP manager controls actions of sub-protocol.

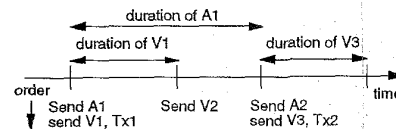


Fig. 17. Sending process of the example in Fig. 9.

normal state, the sub-protocol is aware of the sequence number of the next packet to receive, and thus it could perform the operations of error control. However, in the enforced firing state, which means a transition was just enforced to fire, the sub-protocol only knows the sequence number of the currently active key medium packet as explained in the last paragraph. Thus it works as a protocol without error control, and accepts the arriving packet as long as the KeyRefSeq field of the packet is equal to the sequence number of currently active key medium packet. The transitions between the normal state and enforced firing state are illustrated in Fig. 15.

The MSTP manager could also prevent sub-protocols from retransmitting the out-of-synchronization packets. We illustrate this by an example as shown in Fig. 16. In this example, although the sender does not receive the ACK of V_2 , it will not retransmit V_2 since the ACK of A_2 has arrived and which will invalidate V_2 because the sender knows the synchronization relationship between them. This retransmission mechanism will obtain a better performance than that of the Slack-ARQ in [30]. The selection of sub-protocols depends on the QoS of each medium. In MSTP, the selected sub-protocol for the key medium receives the best service than those of other media.

D. Synchronization Control in MSTP

In this sub-section, we explain how MSTP works to support the synchronization services. Recall that the receiving MSTP manager does not know the original RTSM, but deals with synchronization control according to the information contained in packet headers. Actions taken by the MSTP manager at the sender and receiver will be presented.

1) *Actions of the MSTP Manager at the Sender*: After the multimedia connection is set up, the sending process of the sending MSTP manager is simple. As mentioned in Section III-A, it sends packets according to RTSM specified by the application program. It traverses the RTSM and sends packets of the place being visited [11] by making a send request to the corresponding sub-protocol of the packet. The sending process of the example in Fig. 9 can be easily realized as shown in Fig. 17.

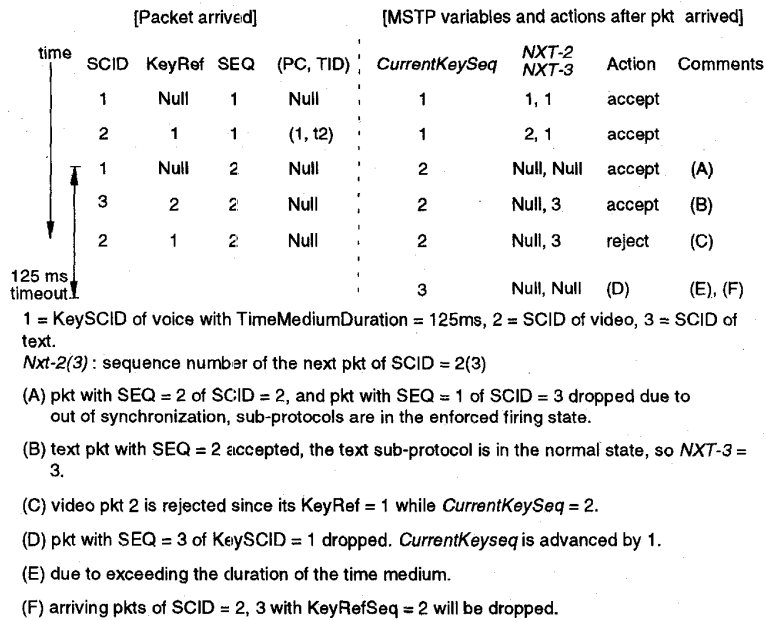


Fig. 18. A run-time case for example in Fig. 9.

2) *Actions of the MSTP Manager at the Receiver:* After the multimedia connection is set up, The receiving MSTP manager knows the current configuration of the multimedia connection, i.e., KeySCID and TimeMediumDuration, according to the control packet sent by the sending MSTP. The receiving MSTP manager maintains a set of status variables for each connection for synchronization purpose. These variables include:

- 1) two variables to save current configuration: *KeySCID* and *TimeMediumDuration*,
- 2) *CurrentKeySeq* to indicate the sequence number of the currently active key medium packet,
- 3) a set of variables to keep the sequence number of next-to-receive packet for each sub-protocol, *Nxt-SCID*.
- 4) pairs of variables (*APC*, *TID*) to store the number of packets which feed into transition *TID* in RTSM. *TID* is the ID of the transition, and *APC* is the number of arrived packets whose *TID* field = *TID*.

Besides, a timer is needed for the use of time medium. When the receiving MSTP manager receives one packet from the sub-protocol, it will execute proper actions according to the synchronization algorithm which is briefly explained as following. Any nonkey medium packet with $KeyRefSeq = CurrentKeySeq$ meets the synchronization relationship with the key medium and can be accepted. The (*APC*, *TID*) pairs are used to synchronize the places which feed into the same transition *TID*. The value of *APC* is increased by one every time a packet with *TID* field = *TID* arrives, and the transition is fired when the *APC* value is equal to the *PC* field of the arriving packet. That is, all the packets that feed into the transition are ready and should be delivered to the application program.

If the time medium is used, a timeout event means that the next key medium packet is too late to be accepted, so that we

advance *CurrentKeySeq* by one and switch the state of sub-protocols to the enforced firing state as explained in Section III-C. For applications without use of key medium, RTSM model can be easily implemented in MSTP protocol by using fields *PC* and *TID* in the packet format.

3) *An Example:* We use a run-time case of Fig. 9 to explain the algorithm mentioned above. An arriving pattern and corresponding actions are shown in Fig. 18. In this example, late transmissions of both key-medium and nonkey media are considered. The comments of actions taken are provided in the figure.

IV. IMPLEMENTATION AND PERFORMANCE MEASUREMENT

We have implemented a prototype system using MSTP protocol and built a simulation system for the wide area network (WAN) environment in order to evaluate the performance of the MSTP protocol. In this section, we describe the system architecture for the simulation and present the results of the performance measurement of RTSM compared with other models. We also discuss the impact of the intra- and inter-frame video coding schemes on the implementation of RTSM and MSTP at the end of this section.

A. The Application RTSM and the Implemented Sub-Protocols

For simplicity, the application we built is a video phone system which contains audio and video data. The monitor display is shown in Fig. 19. The audio data is encoded by the PCM encoding scheme, and the resolution of the video frame is 180×140 (QCIF format). The RTSM for the application is shown in Fig. 20. Note that the number of video places corresponding to one audio place is random due to the following reason. At the sender, the application gets

TABLE I
QoS PROVIDED BY THE GATEWAY PROCESSES

sub-connection ID	medium type	Quality of Services		
		mean delay	standard deviation of delay	loss prob.
1	Audio	100 ms	20 ms	0
2	Video	120 ms	100 ms	0.01

Ps. The generated delay times are forced to be within the range $[0.5*\text{mean}, 4*\text{mean}]$.

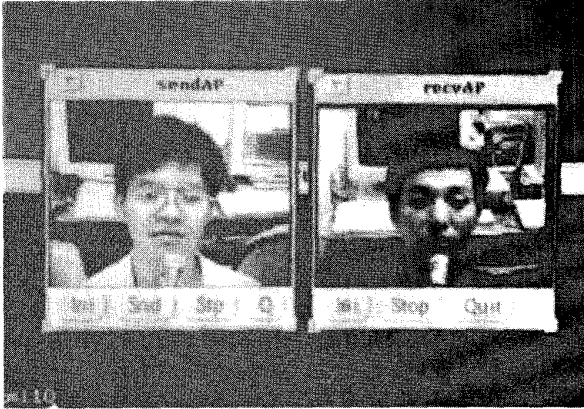


Fig. 19. The display of the application on the monitor.

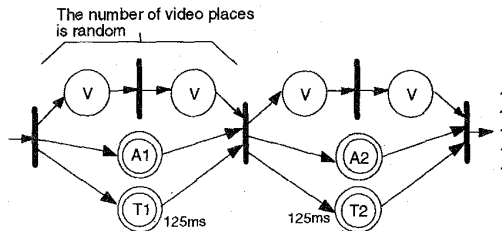


Fig. 20. The RTSM for the implemented application.

one audio packet from the audio device every 125 ms (i.e., 1 Kbytes audio data), and during this 125 ms, the application gets from the video frame grabber one or two video frames, which is determined by the run-time processing overhead of the operating system. The RTSM of the application is not static but dynamic since the number of video frames corresponding to one audio packet is nondeterministic due to the nonrealtime nature of UNIX. Therefore, the application must inform the sending MSTP manager at the running time about the synchronization relationship between media packets, which is achieved by a procedure call to the MSTP manager. The MSTP manager then fills in proper values in the fields of outgoing MSTP packets, and send them to the receiver via the sub-protocols.

We implemented two kinds of sub-protocols in the system: with and without error control. In the sub-protocol with error control, the arriving packets are accepted if they arrive in sequence. That is, no gaps in arriving packets are allowed. Hence, the acknowledgments and retransmissions of lost packets are necessary. Nonetheless, if this sub-protocol is used in MSTP, gaps between packets are still possible when enforced

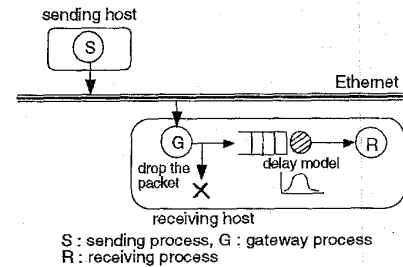


Fig. 21. The WAN simulated environment.

firing occurs. In the sub-protocol without error control, a packet is accepted as long as its sequence number is greater than that of any previously received packet. Gaps are thus allowed in the sub-protocols.

We also define the QoS which are provided by the underlying network environment in the simulation system. The parameters of the QoS include the distribution of the delay behavior of packets and the packet loss probability. In the implementation, we built a simulation environment for the WAN that provides the requested QoS.

B. The WAN Simulated Environment

In order to simulate the behavior of a WAN on an Ethernet, we add a *gateway process* at each destination host to perform proper operations on every packet it receives, which will result in a similar behavior as if the packet is transmitted across a WAN. We illustrate this configuration in Fig. 21. A gateway process implements a normally distributed delay model and a packet loss probability for each packet. That is, a packet received by the gateway process will be dropped with a probability, or be delayed for some time before forwarding the packet to the receiving process. The delayed time is generated by the delay model.

Since the application we built contains two media, two sub-connections are hence required. Therefore, we implemented two gateway processes, one for each sub-connection. The QoS provided by each gateway process are listed in Table I. Notice that the delay value generated by the normal distribution is forced to be within the range $[0.5*\text{mean}, 4*\text{mean}]$ in the program to make the delay values more realistic. As shown in Table I, we provide the audio sub-connection a better QoS since audio is the key medium of the application. The video packets may be dropped by the gateway process with probability 0.01, so that if the selected video sub-protocol requires error control, retransmission of dropped packets may be necessary, depending on whether MSTP is used.

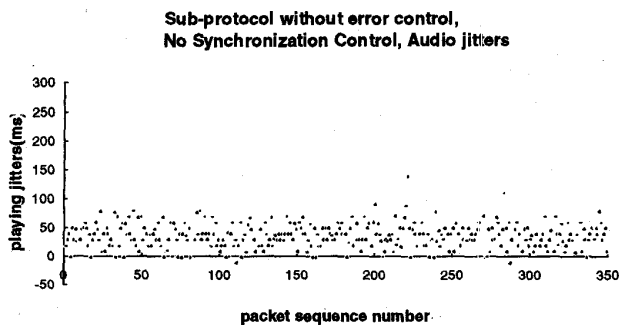


Fig. 22. The playing jitters for audio without synchronization control, and transmitted by a sub-protocol without error control.

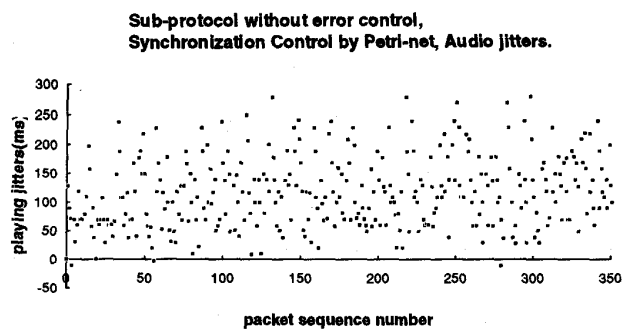


Fig. 24. The playing jitters for audio with synchronization control by Petri-net, and transmitted by a sub-protocol without error control.

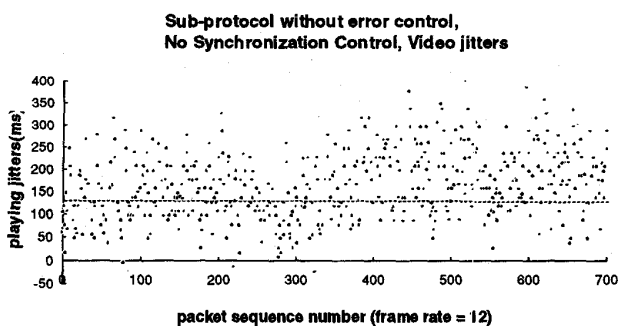


Fig. 23. The playing jitters for video without synchronization control, and transmitted by a sub-protocol without error control.

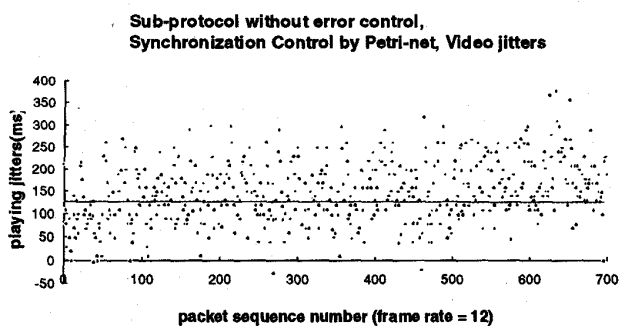


Fig. 25. The playing jitters for video with synchronization control by Petri-net, and transmitted by a sub-protocol without error control.

C. Performance Measurement

The criterion to evaluate the performance of a synchronization control mechanism is the jitter between the actual playing time and the proper playing time for each packet. In order to measure the jitter, we have to record the actual playing time, and to estimate the proper playing time for each packet. The proper playing time for a packet is estimated by its sending time.

Because the video packet may be dropped by the gateway process with probability 0.01, the choice of the sub-protocol for video data will affect the performance of the system. In the following subsections A and B, we show the performance measurement of the system without considering the time medium. Sub-protocols without and with error control are discussed in these subsections, respectively. The performance effect of the time medium is presented in subsection C.

1) *Sub-Protocols Without Error Control:* The results of performance measuring are divided into three parts for comparisons: 1) without any synchronization control, 2) synchronization control by a Petri-net model such as OCPN, and 3) synchronization control by the RTSM with key medium. Video packets dropped by the gateway process are considered as lost and never retransmitted by the sender since the sub-protocol adopted is without error control. The packets of audio data are never dropped by the gateway process since its packet lost probability is 0 as listed in Table I.

No synchronization control: No synchronization control means that there is no synchronization operation at the re-

ceiver, but the sender still has to send packets according to the synchronization model. The receiver just plays the packets delivered by the sub-protocols. The jitters of the playing times in this case are shown in Figs. 22 and 23, one for each medium. The frame rate measured equals to 12 frames per second (fps). In the application, we implemented a smoothing buffer of 125 ms in the application at the receiver to smooth out the unavoidable delay jitter. This 125 ms delay introduced by the smoothing buffer is acceptable for video phone application. Hence, the jitters of most audio packets are within the acceptable range (smaller than 125 ms) as shown in Fig. 22. But as we can see in Fig. 23, the jitters of more than 50% of video packets are above 125 ms and many of those are above 300 ms; hence a serious out-of-synchronization phenomenon will be observed.

Synchronization control using the Petri-net model: In this case, the synchronization control of OCPN model is applied at the receiver in order to synchronize the packets. The jitters of audio and video packets are shown in Figs. 24 and 25. Since the audio packets usually arrive earlier than their corresponding video packets, the playing of audio packets is normally delayed for synchronization purpose. Fig. 25 shows that the audio packets may be seriously delayed such that the jitters of many audio packets are much greater than 125 ms. The quality of audio is hence destroyed and becomes unrecognizable.

Key medium control by the RTSM: We apply the synchronization control of RTSM at the receiver in this case and

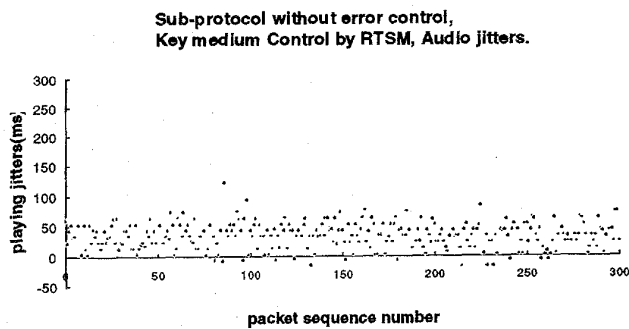


Fig. 26. The playing jitters for *audio with key medium control by RTSM*, and transmitted by a sub-protocol *without error control*.

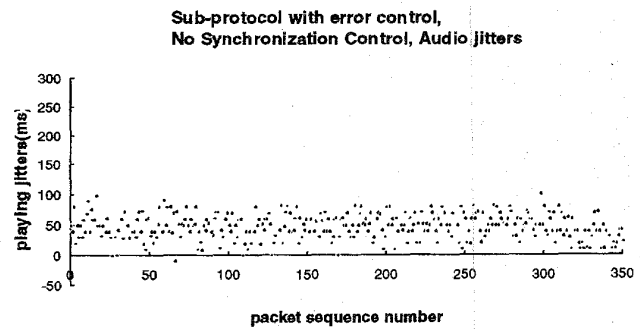


Fig. 28. The playing jitters for *audio without synchronization control*, and transmitted by a sub-protocol *with error control*.

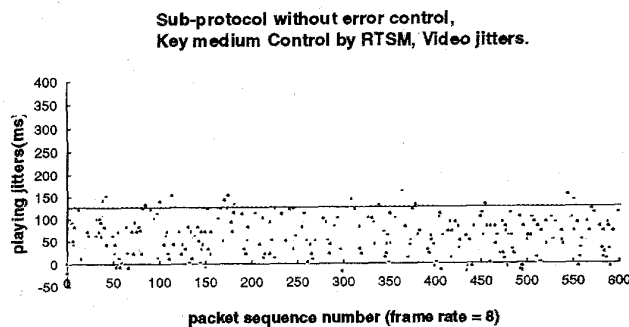


Fig. 27. The playing jitters for *video with key medium control by RTSM*, and transmitted by a sub-protocol *without error control*.

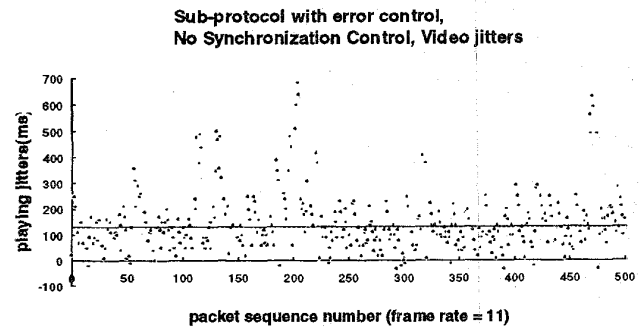


Fig. 29. The playing jitters for *video without synchronization control*, and transmitted by a sub-protocol *with error control*.

the results are shown in Figs. 26 and 27. The playing of audio packets is never delayed as in the case of OCPN because the enforced firing of the key medium in RTSM. The video packets of late transmission will be dropped by the receiving MSTP manager to maintain the quality of the key medium, i.e. the quality of audio. Therefore, the synchronization between audio and video is achieved and the voice quality is maintained. We can see the synchronization effect of the RTSM by comparing Fig. 27 with Fig. 25 for video, Fig. 26 with Fig. 24 for audio.

The cost of the RTSM synchronization control is the decreased video frame rate from 12 fps to 8 fps in this case. However, this is reasonable since the out-of-synchronization video packets should be discarded instead of being displayed. Therefore, the result shows that the synchronization control by the RTSM can achieve the synchronization relationship between media and still maintain the necessary quality of the key medium.

2) *Sub-Protocols With Error Control*: Packets must be delivered to the application in sequence for sub-protocols with error control, and therefore lost packets must be retransmitted. In the implementation, a negative ACK (NACK) will be sent if a packet lost situation is detected to reduce the retransmission delay. We assume the NACK/ACK packets are never lost. In the following, we show the results of performance measuring for different synchronization control methods when sub-protocols with error control are adopted.

No synchronization control: The jitters for audio and video packets are shown in Figs. 28 and 29 respectively. The measured video frame rate equals 11 fps. The jitters of audio,

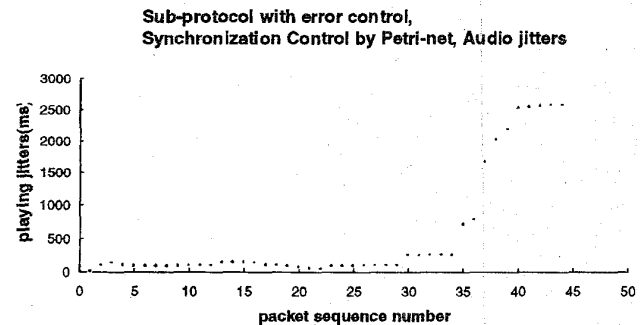


Fig. 30. The playing jitters for *audio with synchronization control by Petri-net*, and transmitted by a sub-protocol *with error control*.

shown in Fig. 28, are similar to those shown in Fig. 22 since we do not drop any audio packet in the gateway process. However, by comparing Figs. 29 and 23, the range of jitters of video packets is enlarged, up to 700 ms, because of the retransmissions of the dropped packets. This means that when adopting sub-protocols with error control, the situation of out-of-synchronization will be worse if we do not exercise any synchronization control.

Synchronization control using the Petri-net model: In this case, the video packets of sequence numbers 34 to 39 experience a continuous dropping and retransmission as shown in Fig. 31, so the jitters increase continuously and sharply. On the other hand, the audio packets are delayed in order to synchronize with video; hence the pattern of jitters for audio is similar to that of video as shown in Fig. 30.

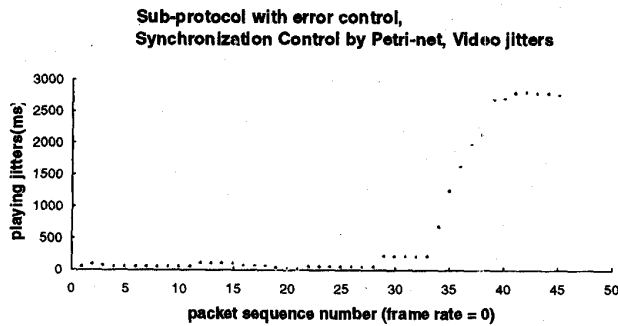


Fig. 31. The playing jitters for video with synchronization control by Petri-net, and transmitted by a sub-protocol with error control.

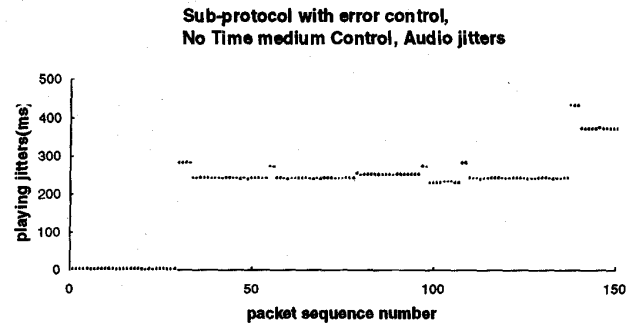


Fig. 34. The playing jitters for *audio without time medium control*, and transmitted by a sub-protocol with error control.

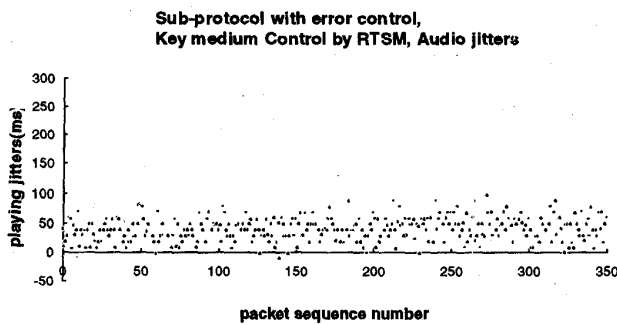


Fig. 32. The playing jitters for audio with key medium control by RTSM, and transmitted by a sub-protocol with error control.

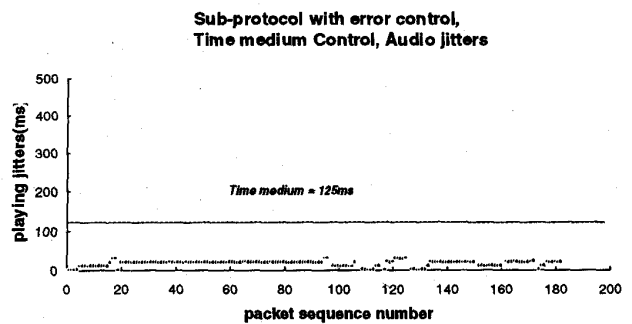


Fig. 35. The playing jitters for *audio with time medium control*, and transmitted by a sub-protocol with error control.

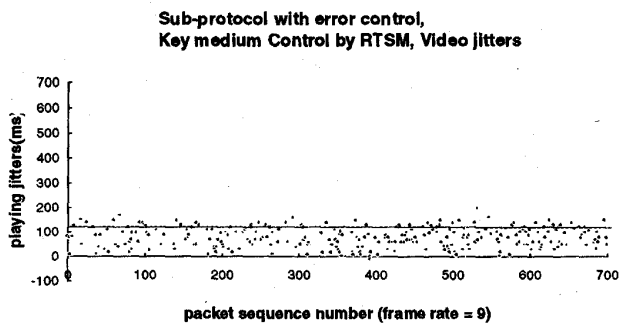


Fig. 33. The playing jitters for video with key medium control by RTSM, and transmitted by a sub-protocol with error control.

This result shows that we can not just hold the audio packets to wait for the corresponding video packets for synchronization purpose. Consequently, we need a policy and mechanism, such as RTSM and MSTP, to control the acknowledgment and retransmission of packets for real-time applications.

Key medium control by the RTSM: As shown in Figs. 32 and 33, the jitters of both audio and video packets are kept small while the synchronization between them is still achieved. Similarly, some video packets will be dropped due to late transmissions such that the video frame rate is decreased from 11 fps to 9 fps.

3) *The Effect of Time Medium:* As mentioned in Section II, the time medium is used to deal with cases where the key

medium suffers a long delay. In this experiment, we do not include video data in the model and introduce a 0.05 packet loss probability for audio packets to show the impact of time medium. In Fig. 34, we show the jitters for the case without the control of time medium. It shows that the jitter will have a jump every time a packet is dropped and retransmitted. That is, once an audio packet is delayed due to retransmission, all the following packets are also delayed.

The result of applying the time medium control on audio packets is shown in Fig. 35. When applying the time medium, packets that suffer jitters more than 125 ms are discarded and not played. As we can see from the figure that although delay jitters are inevitable, but they are kept small enough such that a smoothing buffer with 1 Kbytes, which introduces an initial delay of 125 ms, is enough to smooth out the jitters. Also notice that only few audio packets are dropped such that the audio quality remains good. This result shows that by applying time medium control can resolve the problem of increasing jitters of the key medium.

D. Discussion

In the implementation for performance measurement, the intra-frame coding, motion JPEG, is adopted so that the discarding of late video packets by RTSM will not result in error-propagation over their following video packets. However, it will not be the case if an inter-frame coding such as MPEG is used. Because once we discard some video packets which belong to an I-frame in MPEG coding scheme,

the whole group of frames normally there are several video frames in a group of frames headed by that I-frame will be damaged.

The solution to the problem depends on the requirement of the application for video quality. If the application considers video as the most important medium and can not tolerate any loss of video packets, we have to provide enough bandwidth (peak rate assignment) for video traffic through bandwidth reservation process and requires the network layer protocol to provide a better QoS (e.g., end-to-end delay) for video during connection setup. We also have to choose video as the key medium and apply no time medium control in the RTSM model of the application so that no video packets are skipped by enforced firing of time medium. However, the real-time constraint of video packets may not always be satisfied because reliable transmission requires the lost packets to be retransmitted. Thus, there must be a compromise between the real-time and error-free requirements.

On the other hand, if the application can tolerate some video packets to be dropped, we still can improve the video quality by reserving enough bandwidth and requiring a better QoS for video if possible. But the quality improvement is only limited in reducing the probability of late transmissions of video packets, and resulting in less dropping of video packets. That is, the probability of error-propagation as in MPEG can be reduced, but cannot be totally avoided.

V. CONCLUSION

In this paper, we consider the impact of random delay in packet/cell networks on synchronization among media of multimedia applications. As we can see from the above performance measurements, MSTP clearly meets the design goal of RTSM. For RTSM, more than one key medium can be extended easily in concept, but the implementation will be much more complicated such that the implementation overhead may be too high. Moreover, the time medium can be replaced by a timestamping mechanism to be more precise in timing at the cost of higher overhead.

The RTSM model for the applications may be *static* such as multimedia documents and titles, or *dynamic* such as the video phone or conference system. The dynamic RTSM will depend on the run-time environment as the system we implemented. For static RTSM, formal languages or scripts may be used to describe the synchronization relationship. However, for dynamic RTSM, it is the obligation of the application to inform the sending MSTP manager about the synchronization relationship between media via the primitives provided by the protocol at the running time. The detail discussion of the representation of RTSM for different types of applications is not included in this paper. In summary, the contributions of this paper are listed in the following.

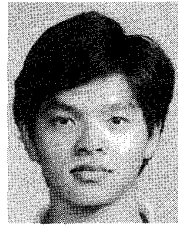
- 1) An RTSM with enforced places to deal with late transmissions is proposed.
- 2) The concept of key medium in RTSM for synchronization control is suggested.

- 3) The concept of time medium associated with key medium to deal with late transmissions of key medium is suggested.
- 4) An implementation method of RTSM in transport protocol MSTP to achieve synchronization services is provided.
- 5) The implementation and performance measurement show the feasibility of MSTP in synchronization control for distributed multimedia applications.

REFERENCES

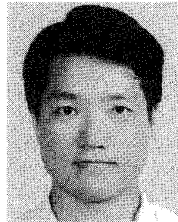
- [1] J.-H. Huang, C.-C. Yang, W.-H. Tseng, C.-W. Lee, B.-J. Tsaur, L.-K. Chuang, and W.-J. Liu, "Design and implementation of multimedia conference system on broadcast networks," in *Proc. IEEE 18th Conf. Local Comput. Networks (LCN)*, 1993, pp. 337-341.
- [2] T.-M. Chen, C.-C. Yang, W.-H. Tseng, I.-C. Chang, J.-H. Huang, C.-C. Lin, M.-S. Lee, N.-J. Fon, and S. Kaw, "Design and implementation of a desktop computer supported cooperative work system," in *Proc. IEEE Int. Conf. Consumer Electron. (ICCE)*, 1994, pp. 174-175.
- [3] M. R. Macedonia and D. P. Brutzman, "Mbone provides audio and video across the internet," *IEEE Comput. Mag.*, pp. 30-36, Apr. 1994.
- [4] W. H. F. Leung, T. J. Baumgartner, Y. H. Hwang, M. J. Morgan, and S. C. Tu, "A software architecture for workstations supporting multimedia conferencing in packet switching networks," *IEEE J. Select. Areas Commun.*, vol. 8, no. 3, pp. 380-390, Apr. 1990.
- [5] P. V. Rangan and D. C. Swinehart, "Software architecture for integration of video services in the etherphone system," *IEEE J. Select. Areas Commun.*, vol. 9, no. 9, pp. 1395-1404, Dec. 1991.
- [6] S. Masaki, H. Yamaguchi, Y. Hayashi, T. Nishimura, and K. Shimamura, "Multimedia handling scheme in a groupware system for B-ISDN," in *Proc. IEEE GLOBECOM*, 1992, pp. 747-751.
- [7] M. Arango, et al., "The touring machine system," *Commun. ACM*, pp. 68-77, Jan. 1993.
- [8] Y. V. R. Reddy, K. Srinivas, V. Jagannathan, and R. Karinithi, "Computer support for concurrent engineering," *IEEE Comput. Mag.*, pp. 12-15, Jan. 1993.
- [9] R. Steinmetz, "Synchronization properties in multimedia systems," *IEEE J. Select. Areas Commun.*, vol. 8, no. 3, pp. 401-412, Apr. 1990.
- [10] T. D. C. Little and A. Ghafoor, "Synchronization and storage models for multimedia objects," *IEEE J. Select. Areas Commun.*, vol. 8, no. 3, pp. 413-427, Apr. 1989.
- [11] ———, "Multimedia synchronization protocols for broadband integrated services," *IEEE J. Select. Areas Commun.*, vol. 9, no. 9, pp. 1368-1382, Dec. 1991.
- [12] N. U. Qazi, M. Woo, and A. Ghafoor, "A synchronization and communication model for distributed multimedia objects," in *Proc. ACM Multimedia*, 1993, pp. 147-155.
- [13] B. Prabhakaran and S. V. Raghavan, "Synchronization models for multimedia presentation with user participation," in *Proc. ACM Multimedia*, 1993, pp. 157-166.
- [14] T. F. La Porta and M. Schwartz, "Architectures, features, and implementation of high-speed protocols," *IEEE Network Mag.*, pp. 14-22, May 1991.
- [15] W. Doeringer, A. Dykeman, M. Kaiserswerth, B. Meiser, H. Rudin, and R. Williamson, "A survey of light-weight transport protocols for high-speed networks," *IEEE Trans. Commun.*, vol. 38, no. 11, pp. 2025-2039, Nov. 1990.
- [16] R. M. Sanders and A. C. Weaver, "The Xpress transfer protocol (XTP)—A tutorial," *Quart. Pub. ACM SIGCOMM*, vol. 20, no. 5, pp. 67-80, Oct. 1990.
- [17] D. R. Cheriton and C. L. Williamson, "VMTP as the transport layer for high-performance distributed systems," *IEEE Commun. Mag.*, vol. 27, no. 6, pp. 37-44, June 1989.
- [18] D. R. Cheriton, "VMTP: A transport protocol for the next generation of communication systems," *ACM Comput. Commun. Rev.*, pp. 406-415, 1986.
- [19] D. D. Clark, M. L. Kamert, and L. Zhang, "NETBLT: A high throughput transport protocol," *ACM Comput. Commun. Rev.*, pp. 353-359, 1988.
- [20] Z. Haas, "A communication architecture for high-speed networking," in *Proc. IEEE INFOCOM*, 1990, pp. 433-441, San Francisco.
- [21] A. N. Netravali, W. D. Roome, and K. Sabnani, "Design and implementation of a high-speed transport protocol," *IEEE Trans. Commun.*, vol. 38, pp. 2010-2024, Nov. 1990.

- [22] T. F. La Porta and M. Schwartz, "The multistream protocol: A highly flexible high-speed transport protocol," *IEEE J. Select. Areas Commun.*, vol. 11, pp. 519-530, May 1993.
- [23] ———, "Performance analysis of MSP: feature-rich high-speed transport protocol," *IEEE/ACM Trans. Networking*, pp. 740-753, Dec. 1993.
- [24] J. Kurose, "Open issues and challenges in providing quality of service guarantees in high-speed networks," in *ACM SIGCOMM*, pp. 6-15, 1992.
- [25] C. Huitema and W. Dabbous, "Minimal complexity for the simplest protocol," in *Proc. IEEE INFOCOM*, 1991, pp. 1481-1489.
- [26] D. P. Anderson and G. Homsy, "A continuous media I/O server and its synchronization mechanism," *IEEE Comput. Mag.*, pp. 51-57, Oct. 1991.
- [27] A. Cambell, G. Coulson, F. Garcia, and D. Hutchison, "A continuous media transport and orchestration service," in *ACM SIGCOMM*, pp. 99-110, 1992.
- [28] D. Ferrari, A. Gupta, M. Moran, and B. Wolfinger, "A continuous media communication service and its implementation," in *Proc. IEEE GLOBECOM*, 1992, pp. 220-224.
- [29] H. Schulzrinne, "A transport protocol for audio and video conferences and other multiparticipant real-time applications," MIT Tech. Rep., July, 1992.
- [30] B. J. Dempsey, J. Liebeherr, and A. C. Weaver, "A new error control scheme for packetized voice over high-speed local area networks," in *Proc. IEEE 18th LCN*, 1993, pp. 91-100.
- [31] J.-H. Huang and S.-H. Lee, "MHTP—A multimedia high-speed transport protocol," in *Proc. IEEE GLOBECOM*, 1992, pp. 1363-1368.



Chun-Chuan Yang received the B.S. degree in computer and information science from National Chiao-Tung University, Hsin-Chu, Taiwan, R.O.C., in 1990. He is a Ph.D. student in the Department of Computer Science and Information Engineering at National Taiwan University, Taipei, Taiwan.

His current research topics include multimedia network protocols, high-speed and real-time protocols, admission control, flow control, and multimedia synchronization control.



Jau-Hsiung Huang (A'89) received the B.S. degree in electrical engineering from National Taiwan University, Taiwan, in 1981, and the M.S. and Ph.D. degrees in computer science from the University of California, Los Angeles, CA, USA, in 1985 and 1988, respectively.

Since 1988, he has been a member of the faculty in the Department of Computer Science and Information Engineering, National Taiwan University, where he is currently a professor. He co-founded the Communications and Multimedia Laboratory in the department in 1990 to launch research in the areas of distributed multimedia systems and virtual reality. He has published over 40 technical papers in the areas of multimedia networking, high-speed networking, parallel and distributed systems, and performance evaluation of computing systems.