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An Audio-Video Multipath Streaming Scheme with Media Synchronization Control: Application-Level QoS Assessment in a Wireless Ad Hoc Network

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SUMMARY This paper proposes the *MultiPath streaming scheme with Media Synchronization control* (*MPMS*) for audio-video transmission in wireless ad hoc networks. In many audio-video streaming applications, media compensate each other from a perceptual point of view. On the basis of this property, we treat the two streams as separate transport streams, and then the source transmits them into two different routes if multiple routes to the destination are available. The multipath transmission disturbs the temporal structure of the streams; in MPMS, the disturbance is remedied by media synchronization control. In order to implement MPMS in this paper, we enhance the existing Dynamic Source Routing (DSR) protocol. We compare the application-level QoS of MPMS and three other schemes for audio-video transmission by simulation with ns-2. In the simulation, we also assess the influence of the multipath transmission on other traffic. The simulation result shows that MPMS is effective in achieving high QoS at the application-level.

key words: *wireless ad hoc network, audio-video streaming, multipath routing, media synchronization, mutually compensatory property, QoS*

1. Introduction

The rapid advancement in portable computing platforms and wireless communication technology has led to significant interest in wireless ad hoc networks [1]. They are instantly deployable wireless networks without any base station or infrastructure support, where all nodes are capable of moving and can be connected dynamically in an arbitrary manner. Each mobile host acts as a router, which discovers and maintains routes to other hosts and forwards packets for them in the network.

Some applications of ad hoc networks require the ability to support real-time multimedia streaming such as live audio and video over the networks. Therefore, the realization of this type of service with high quality is highly demanded; nevertheless, it is very difficult to achieve high quality in ad hoc networks. Furthermore, achievable QoS of audio-video streaming in ad hoc networks with current technology has not been clarified sufficiently. The Internet Protocol (IP) suite plays an important role even in ad hoc networks. Then, we should devise an efficient streaming scheme and assess the quality in IP-based ad hoc networks.

Owing to the layered architecture of IP-based networks, its *Quality of Service* (*QoS*) also has a layered structure. We can identify six levels of QoS: *physical-level*, *link-level*, *network-level*, *end-to-end-level*, *application-level* and *user-level* [2]. The subjective quality (i.e., user-level QoS) is the most important to the users; it is closely related to application-level QoS. The preservation of the temporal structure of audio and video is essential to high application-level QoS of continuous media [2].

When we transmit the continuous media streams in ad hoc networks, the temporal structure of the streams can be disturbed largely by delay, its jitter and packet loss. In wireless networks such as IEEE 802.11, terminals share the same physical channel. The media access control (MAC) protocol for sharing usually has a carrier-sensing capability for collision avoidance and a retransmission-based error recovery mechanism for transmission errors in the wireless channel. Thus, network delay and its jitter easily increase. In order to preserve the temporal relation, we need the *media synchronization control* [3], which is application-level QoS control.

We identify three types of media synchronization: *intra-stream synchronization*, *inter-stream synchronization* and *inter-destination* (or *group*) *synchronization*. The intra-stream synchronization control is necessary for the preservation of the timing relation between *media units* (*MUs*) such as video frames in a single media stream; an MU is the information unit for media synchronization. The inter-stream synchronization is required for keeping the temporal relations among MUs in multiple media streams; synchronization between audio and video (i.e., *lip-sync*) is a typical example. The inter-destination synchronization adjusts the output timing of each MU multicast to two or more destinations so that the MU can be output simultaneously at all the destinations.

A variety of studies on continuous media transmission in wireless ad hoc networks have been reported. However, most of them do not assess the application-level or user-level QoS of audio and that of video together. In [4], for instance, Toh et al. treat audio transmission in ad hoc networks. They show experimental results of packet loss rate and delay jitter. Furthermore, they assess perceptual quality of the audio stream.

In wireless ad hoc networks, multiple paths can be useful in improving the effective bandwidth of communication pairs, responding to congestion and bursty traffic, and in increasing delivery reliability. However, we can find no study on multipath transmission of multimedia streams with the

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application-level or user-level QoS assessment.

In [5], Lee and Gerla propose *Split Multipath Routing* (*SMR*) protocol to avoid congestion and to use network resources efficiently. In SMR, the destination selects two routes that are maximally disjoint and informs the source of the routes. The source uses a simple per-packet allocation scheme for splitting traffic into two routes when there are two or more available routes to the destination. They compare the network-level QoS of SMR (such as packet delivery ratio and end-to-end packet delay) with that of DSR (Dynamic Source Routing) [6]. Also, they propose the scheme for discrete media transmission, and then it is not suitable for continuous media.

In [7], Mao et al. propose MultiPath Transport (MPT) of hierarchically coded multiple video streams in order to improve its quality. They employ an enhanced DSR protocol, which is called *Multipath DSR* (*MDSR*). MDSR selects multiple maximally disjoint routes from all the routes returned by a route query. They propose three Motion Compensated Prediction (MCP)-based video transport techniques for mobile ad hoc networks. These schemes take advantage of path diversity to achieve better performance. They evaluate the performance of these schemes by peak signal-to-noise ratio (PSNR) and the packet loss rate. However, they do not treat audio (i.e., they deal with video only) and also do not mention the temporal structure of the video stream.

In this paper, we propose the *MultiPath streaming scheme with Media Synchronization control* (*MPMS*) for audio-video transmission in wireless ad hoc networks. MPMS treats audio and video as two separate transport streams and sends the two streams to different routes if multipath routes are available.

It should be noted that the most important feature of multimedia is a synergy of component media; media compensate each other from a perceptual point of view. In a videotelephone system, for instance, even if the quality of the video stream is somewhat low, the voice stream with good quality can compensate for the degradation of the overall perceptual quality. We refer to this interdependency between component media as the *mutually compensatory property* [2]. This property of cross-modal influences between audio and video is also pointed out by ITU-T Recommendation J.148 [8], which details the requirements for the development of an objective multimedia perceptual quality model. In multimedia communications, QoS control by taking advantage of the property can provide good user-level QoS. From a mutually compensatory point of view, we assume that the audio stream has priority over the video one in route selection.

When the audio and video streams are transmitted into two different routes, the transfer delay of audio can differ from that of video; this difference disturbs inter-stream synchronization. Thus, in order to remedy the temporal structure disturbed by the multipath transmission, we employ media synchronization control.

We compare the application-level QoS of MPMS with that of three other schemes by simulation with ns-2. In the simulation, we also assess the influence of the multipath transmission on other traffic. MPMS tries to send audio and video streams separately into two different routes, and then the route for audio or video is likely to conflict with some route for other traffic. Furthermore, in order to maintain multiple routes, MPMS has to transmit more control packets than single-path schemes. Thus, the multipath transmission can affect other traffic in the network. The assessment of the influence is a subject to be studied in this paper.

The rest of the paper is organized as follows. Section 2 explains the multipath routing scheme adopted in this paper. Section 3 describes the principle of the media synchronization scheme. Section 4 illustrates a methodology for the QoS assessment, including the network configuration, simulation method and QoS parameters. The simulation results are presented and discussed in Sect. 5.

2. Multipath Routing Scheme

MPMS transmits audio and video streams separately into different routes if multipath routes are available. This strategy has two advantages. First, we can gain high user-level QoS because of the mutually compensatory property of the streams. Second, we can easily achieve intra-stream synchronization of the audio stream because a priority is given to the audio stream over the video one in route selection.

In MPMS, if more than one route is available, the source selects two routes out of them. One of the two routes has the shortest "distance" (e.g., hops) from the source to the destination among all the available routes, and the other is maximally disjoint from the first route. The former route is referred to as the *primary route*, and the latter is called the *secondary route*. The audio stream employs the primary route, and the video stream uses the secondary route. The source assigns these routes to packets according to their destination ports.

Furthermore, in order to achieve high application-level QoS, MPMS adaptively switches multipath transmission to single-path transmission and vice versa according to the network configuration. That is, in unsuitable situations for the multipath transmission, MPMS uses the single-path transmission. Thus, MPMS achieves at least the same QoS as that in the single-path transmission even in the worst case for the multipath transmission.

MPMS can utilize any routing algorithm for selecting candidates of multipath routes. Thus, we can attempt various implementation of MPMS. In this paper, as an example of MPMS, we enhance the existing DSR protocol.

2.1 Summary of DSR Protocol

DSR is a reactive routing protocol and operates entirely on demand. In the protocol, each node has no routing table. The source node decides an entire route to the destination and attaches the routing information to packets. Nodes relay the packets according to the information.

The DSR protocol consists of two mechanisms: *Route Discovery* and *Route Maintenance*. Let us explain these mechanisms below.

Route Discovery is initiated by a source whenever it has a data packet to send but does not have any routing information on the destination. To establish a route, the source floods the network with *Route Request* (*RREQ*) messages. When an RREQ reaches the destination, it sends a *Route Reply* (*RREP*) message containing path information back to the source. Until the source finds a route, the data packet is stored in the send buffer at the network layer.

For efficient flooding, each RREQ contains a sequence number that uniquely identifies the packet. When a node other than the destination receives a RREQ that is duplicate, it discards the RREQ.

Furthermore, the *route cache* maintained at each node records routes the node has learned. When intermediate nodes have some routes to the destination, they are allowed to send RREPs back to the source without forwarding RREQs to the destination.

Route Maintenance is a mechanism by which the sender of a packet detects network topology changes that render its route to the destination useless. When Route Maintenance indicates that a route is broken, the source is notified of it with a *Route Error* (*RERR*) packet. The source can then attempt to use another route to the destination already in its cache or can invoke Route Discovery again to find a new route.

2.2 Enhancement of DSR

The Route Discovery mechanism of DSR has the ability to return information on multipath routes. The route information is kept in the route cache of the source node. Thus, we enhance the route selecting mechanism from the cache.

When the source wants to send data to the destination, it searches its own route cache. If more than one route is available in the cache, it selects two routes out of them.

Figure 1 shows the algorithm for the route selection. In MPMS, the shortest hops route in the cache is selected as the primary route. If there are two or more routes which have the shortest hops, the source selects the earliest entry of the route cache among the shortest hops routes. This is because the route cache entries are created according to the arrival order of RREPs. That is, the earliest entry of the route cache can have the smallest round trip delay.

The secondary route is selected by comparing routes in the cache with the primary route. For each route in the cache, the source checks the number of the same nodes between the cached route and the primary route. Then, as the secondary route, the source selects the route which has the smallest number of common nodes with the primary route. If there are two or more routes which have the smallest number of common nodes, the source selects the shortest hops route among those routes. Furthermore, if there are two or more routes which have the shortest hops, the source selects the earliest entry of the route cache among the shortest hops routes.

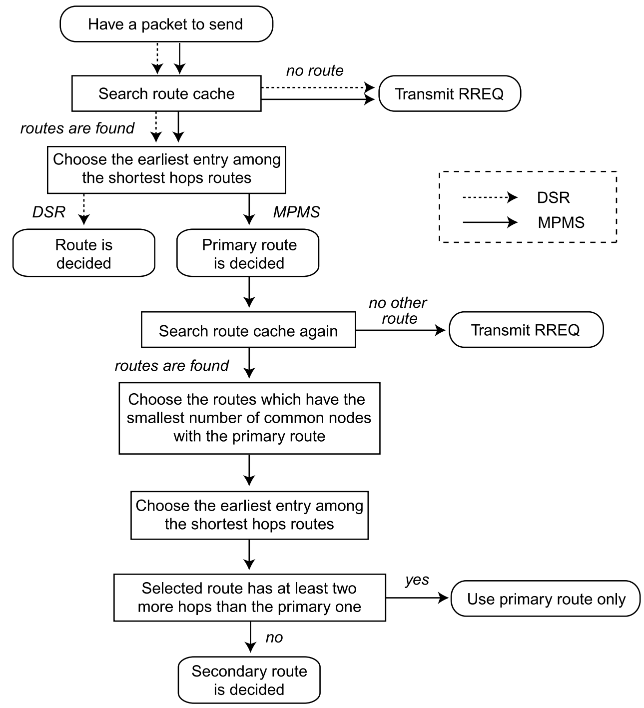

Fig. 1 Route selection algorithm.

Here, notice that multipath transmission may disturb the quality of inter-stream synchronization. As the difference in the number of hops between the primary route and the secondary one increases, the disturbance becomes more serious. Thus, we set a limitation on difference in the number of hops in selecting the secondary route. When the difference in the number of hops between the two routes is two or more, only the primary route is used for transmission of both media streams.

If only one route is found in the route cache, the source uses the route to deliver both of the streams; we regard the route as the primary route. Furthermore, the source sends an RREQ for searching the secondary route.

In the traditional DSR protocol, the source sends an RREQ when it wants to send data but has no route. In [6], Johnson et al. propose that the DSR protocol should have a backoff mechanism for sending RREQs. Once an RREQ is sent, the source enters a backoff period and cannot send RREQs until the end of the period. The source terminates the backoff period when it receives an RREP for the RREQ previously sent or when the period expires.

Our multipath scheme also employs a backoff mechanism. In our scheme, however, the source can send an RREQ when one of the two routes is broken, even if the other route is active; then, it enters a backoff period. In this case, the remaining route becomes the primary route, and the source transmits both streams over the primary route. Thus, for efficient multipath transmission, we enhance the backoff mechanism of the traditional DSR as follows.

First, when the source with our scheme has only the (new) primary route and then receives an RREP reporting the primary route, it does not terminate the backoff period. This is because the RREP is meaningless for the source. That is, when the source transmits a RREQ for searching a secondary route, an intermediate node on the primary route will reply a RREP reporting that route. Thus, when the source receives the RREP, it does not terminate the backoff period.

Second, when the source notices that the (new) primary route is broken, it terminates the backoff period. If the source employed the backoff mechanism of the traditional DSR protocol, it would remain in the backoff period. Then, it could not send any RREQ until the end of the period or the arrival of an RREP. RREQs and RREPs can drop owing to the collision with the media streams. Thus, if the source notices the primary route broken after sending an RREQ for finding a secondary route, and if the RREQ or an RREP for the RREQ drops, the source must wait for the end of the backoff period for sending a new RREQ. This decreases the throughput of media streams since the source does not send any packet when it has no route. Thus, our scheme specifies that the source terminates the backoff period at that time.

For simplicity of implementation, MPMS in this paper treats RREQs and RREPs in the same way as that in the traditional DSR. That is, the Route Discovery mechanism has not been optimized to choose the maximally disjoint path; for example, an intermediate node tells only one route when it replies an RREP without transmitting RREQs to the destination, and duplicate RREQs are discarded by intermediate nodes.

3. Media Synchronization Algorithm

In this paper, we employ the enhanced *Virtual-Time Rendering* (*VTR*) algorithm proposed in [9] for media synchronization control.

The original VTR algorithm [10] adaptively changes the buffering time according to the amount of delay jitter of MUs received at the destination. The enhanced VTR algorithm proposed in [9] also changes the buffering time according to MU loss for managing MU drop and retransmission. In this paper, however, we do not carry out retransmission-based error recovery at the application-level. The mechanism of increase in buffering time is utilized for error recovery at the MAC layer.

An MU, which is the information unit at the application layer, is divided into one or more MAC frames and transmitted into the network. Under the media synchronization control, if at least one of the MAC frames drops or its arrival delays beyond the time limit specified by the media synchronization algorithm, the MU cannot be reconstructed at the receiver, and then the receiver discards the remaining frames of the MU. The increased buffering time can increase the number of trials of frame retransmission for error correction at the MAC layer. Thus, the increase in the buffering time at the application layer is indirectly effective in the error recovery at the MAC layer.

The media synchronization control is effective in outputting MUs in order when the out-of-sequence of MUs occurs. The out-of-sequence of MUs means that an MU arrives earlier than another MU which is transmitted by the source before the former. Whether the destination exerts the media synchronization control or not, the application layer of the destination receives the same number of MUs as the one received by the network layer in the same order. This is because media synchronization control is application-level QoS control, and only the destination carries out the control. However, when the destination exerts the media synchronization control, it performs sequence-control of the MUs and can output them in the order of generation. On the other hand, if the network employs UDP as the transport protocol and if the destination does not carry out the media synchronization control, it just discards MUs received out of sequence.

In this paper, the audio is selected as the master stream and the video as the slave one since audio is more sensitive to intra-stream synchronization error than video. Only the master stream can change the buffering time for itself, and accordingly the slave stream changes it by the same amount at the same time. Furthermore, we also carry out inter-stream synchronization control.

4. Methodology for Quality Assessment

We assess the application-level QoS of MPMS by computer simulation with *ns-2* [11].

4.1 Network Configuration

In this paper, we first consider a grid topology network, which consists of 9 nodes as shown in Fig. 2. The interval between two vertical or horizontal adjacent nodes is constant, 20 m.

We formulate a detailed simulation model which is based on the *distributed coordination function* (*DCF*) [12] of the IEEE 802.11 wireless LAN. We employ the free space propagation model implemented in ns-2. Each node has an omni-directional antenna. The radio model uses system parameters similar to a commercial radio interface, Lucent Technologies' Orinoco 802.11b 11 Mbps PC card; that is, we assume IEEE 802.11b. The transmission speed is kept at 11 Mbps. In the simulation, the transmission range of each node is about 22.49 m. That is, the nodes are within

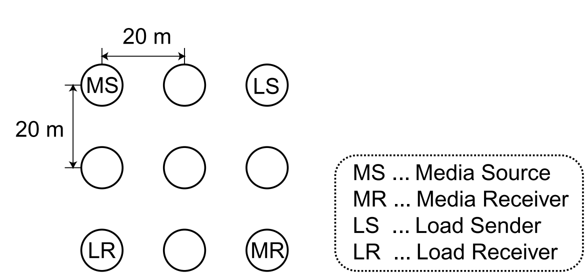

Fig. 2 Network configuration.

communication range of only their vertically or horizontally adjacent neighbors.

In this paper, we have chosen quite simple topology in order to examine the basic characteristics of MPMS. However, it imposes several limitations on the assessment result. For example, we cannot address the link failure problem caused by mobility. Therefore, we plan to extend our assessment so as to incorporate more realistic situations that are representative of the real world, such as more practical topology networks, mobile nodes and varying node distances.

4.2 Method of Simulation

We assume MS (Media Source) as the voice and video sources. MS unicasts the media streams to MR (Media Receiver) with RTP/UDP.

We use a voice stream of ITU-T G.711 μ-law and an MPEG1 video stream. Table 1 shows the specifications of the voice and video. Furthermore, we take the media capturing and encoding delay time into consideration in the simulation. The capture duration of a voice MU equals the inter-MU time, which is 40 ms in this paper, and the encoding time is negligible; therefore, we set the capturing and encoding delay time of each voice MU to 40 ms. On the other hand, the capture duration of a video MU is just a moment. However, it spends much time to encode a video frame. In this paper, we set the capturing and encoding delay time of each video MU to 8 ms, which is the same time as that in the experimental system in [9]†. Each MU leaves the source the capturing and encoding delay time after its timestamp. In addition, we assume that the capturing start time of the first voice MU and that of the first video MU are the same.

The parameter values in the enhanced VTR algorithm are set to the same values as those in [13]. That is, we set the *initial buffering time* $J_{\max}$ [9] and the *maximum allowable delay* $\Delta_{\rm al}$ [9] to 100 ms and 300 ms, respectively. We exert loosely-coupled inter-stream synchronization control [10].

In the simulation, we assess the application-level QoS of four schemes: MPMS, MPNC, SPMS and SPNC. The first two letters of each scheme mean the routing scheme; MP represents the multipath routing scheme, and SP (SinglePath) shows the traditional DSR. The last two letters display the existence (MS: Media Synchronization) or nonexistence (NC: No Control) of the media synchronization capability at the destination.

Table 1 Specifications of the voice and video.

item	voice	video
coding scheme	ITU-T G.711 μ-law	MPEG1 GOP I
image size [pixels]	—	192 × 144
original average MU size [bytes]	320	2000
original average MU rate [MU/s]	25.0	20.0
original average inter-MU time [ms]	40.0	50.0
original average bit rate [kbps]	64.0	320.0
measurement time [s]	120.0	

LS (Load Sender) and LR (Load Receiver) are used to handle a traffic flow of interference. We employ the traditional DSR for the load traffic. Note that both MP and SP share route caches with the traditional DSR for the load traffic. LS generates fixed-size IP datagrams of 1500 bytes each at exponentially distributed intervals and then transmits them to LR. The amount of the interference traffic is adjusted by changing the average of the interval. We refer to the average amount of the interference traffic as the *average load*.

4.3 QoS Parameters

We regard the quality of media synchronization as the major part of the application-level QoS in this paper. Thus, we need QoS parameters which reflect the media synchronization quality.

For the quality assessment of intra-stream synchronization for voice or video, we evaluate the *coefficient of variation of output interval*, which is defined as the ratio of the standard deviation of the MU output interval (i.e., presentation time interval of two MUs at the destination) of a stream to its average; this represents the smoothness of output of a media stream.

For the inter-stream synchronization quality, we calculate the *mean square error of inter-stream synchronization*, which is defined as the average square of the difference between the output time of each slave MU and its *derived output time*. The derived output time of each slave MU is defined as the output time of the corresponding master MU plus the difference between the timestamps of the two MUs [10].

For the assessment of transfer efficiency, we use the *MU loss rate*, which is the ratio of the number of MUs lost to the total number of MUs generated.

The *average MU delay*, which is the average time of *MU delay*, is a key measure for live media. The MU delay is defined as the time interval from the moment an MU is generated until the instant the MU is output.

Furthermore, we assess the *average number of route errors*, the *total use time of the send buffer in the source node*, and the *number of different nodes between two routes from the source to the destination* as the network-level QoS parameters.

The average number of route errors represents the average number of route destruction from MS to MR during a simulation run. When a route in use breaks, the intermediate node returns a route error packet to the source. The route destruction is detected by a transmission error at the MAC layer. MP can use a link on the MAC layer for two routes. Then, if the link which is shared by the two routes breaks, we add one (not two) to the number of route errors.

†In [9], JPEG is employed for video codec. On the other hand, this paper handles MPEG video. However, because of the GOP pattern in this paper, we have assumed that the capturing and encoding delay time of each MU is approximately the same as that of JPEG video in [9].

The total use time of the send buffer in the source node is the total amount of time during which there is at least one packet in the send buffer at the network layer. It reflects the total time when the source wants to send packets but has no route†. When it has no route, it keeps packets in the send buffer. Then, it sends buffered packets once a route is found.

The number of different nodes between two routes from the source to the destination shows the degree of independency of the two routes. This means the number of the secondary route nodes which are not included in the primary route. When the number of different nodes is one or more, the source sends the voice and video with two different routes.

5. Simulation Results

In this section, we first show the characteristics of multipath routing by the network-level QoS assessment. Then, we show the application-level QoS of the four schemes. Furthermore, we explain the influence of the multipath transmission on the load traffic.

Each symbol in the figures to be shown represents the average of 20 measured values which were obtained by changing the random seed for generating the interference traffic. We also show 95% confidence intervals of the QoS parameters in the figures. However, when the interval is smaller than the size of the corresponding symbol representing the simulation result, we do not show it in the figures.

5.1 Characteristics of Multipath Routing

The average number of route errors, the total use time of the send buffer, and the number of different nodes, which are assessed at the network-level, are not affected by the media synchronization control. Therefore, we do not distinguish between the existence of the media synchronization control and nonexistence of that in this subsection.

Figure 3 depicts the average number of route errors which occurred with transmitted packets from the source during a simulation run versus the average load. We see in this figure that the average number of route errors which occurred during a simulation run increases as the average load increases for the average loads lighter than about 400 kbps. This is because many collisions occur in the MAC layer on heavily loaded conditions, and then the intermediate nodes notice the link breakage more frequently.

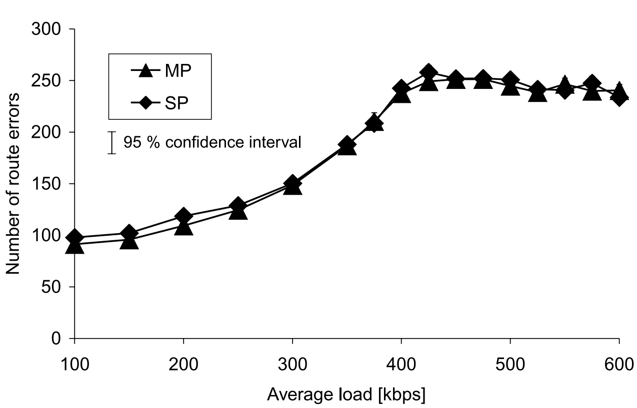

Fig. 3 Average number of route errors.

We also notice in Fig. 3 that the average number of route errors saturates for the average loads heavier than about 400 kbps. This is due to the limitation of the link layer queue at the source. On heavily loaded conditions, the MAC layer of the source node scarcely transmits a MAC frame because it detects other carriers on the wireless channel and waits the transmission opportunity of the frame until the network becomes clear. Thus, the link layer queue at the source node must keep many packets. When the queue becomes full, the application can hand no more packet to the MAC layer, and then packets are just dropped. This limits the packet transmission rate on the wireless channel. Therefore, the probability of route error occurrence becomes approximately constant. Owing to this, the average number of route errors saturates.

In Fig. 3, we find that for all the average loads here, the average number of route errors with MP is approximately the same as that with SP. Thus, MP suffers route destruction as much as SP.

Figure 4 shows the total use time of the send buffer in MS (the media source node) versus the average load. In addition, Fig. 5 depicts the number of different nodes between two routes in MP as a function of time when the average load is 350 kbps.

We notice in Fig. 4 that for all the average loads here, the total use time in SP is larger than that in MP. That is, in SP, the total periods when the source wants to send packets but has no route is longer than that in MP. The reason is as follows. As shown in Fig. 3, SP suffers route destruction as much as MP. In SP, the source always transmits both of the voice and video streams via the same route. Thus, if the route is invalidated, the source does not transmit the

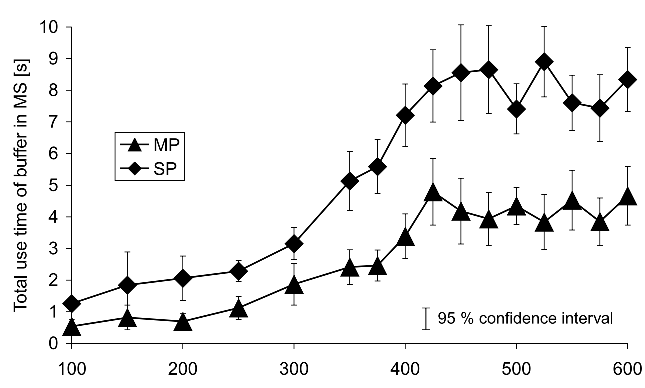

Fig. 4 Total use time of the send buffer in MS.

†Since DSR is a reactive routing protocol, the total amount of time during which the source has no route includes the period when the source has no packet to send. In addition, it is complicated to evaluate the total time when the source wants to send packets but has no route. Thus, we employ this parameter.

packets. On the other hand, in MP, the source uses two routes for transmitting the media streams. Even if one of the routes is invalidated, the source can transmit the media streams through the other route and then sends an RREQ to find other routes. Therefore, MP can decrease the period when the source has no route and then uses the send buffer of MS less frequently than SP.

In Fig. 4, we find that when the average load is heavier than around 400 kbps, 95% confidence intervals of the total use time are larger than those on lightly loaded conditions. That is, the total use time largely fluctuates on heavily loaded conditions, where we find that the total use time in MP ranges from about 2.5 to 5.5 seconds, while that in SP fluctuates between around 6.5 and 10 seconds. This is because the number of route errors saturates on that load condition as shown in Fig. 3.

We see in Fig. 5 that the number of different nodes between the primary route and the secondary route is one or more during almost all the simulation time. Thus, the source often transmits the voice and video streams on separate routes.

Furthermore, Fig. 6 displays the average available time of multiple routes in MP and that of a single route versus the average load. The average available time of multiple routes implies the average time when the multiple routes are used in MP during a simulation run of 120 seconds. On the other hand, the average available time of a single route means the average time when MP uses only the primary route. We find in this figure that for all the average loads here, the available time of multiple routes is larger than about 95 seconds, which is about 79% of the duration of a simulation run.

We also notice in Fig. 6 that the average available time of only a single route increases as the average load increases. This is because the number of route destruction increases as the average load increases as shown in Fig. 3.

In the topology, the minimum number of hops from the source to the destination is four. Thus, when there is no intermediate node common to the two routes, the number of different nodes between the primary route and the secondary one becomes three or more. In Fig. 5, we find that the number of different nodes is not always three or more; that is, the source does not always select the two routes which are maximally disjoint. This is because the Route Discovery mechanism in our implementation has not been optimized to choose the maximally disjoint path as discussed in Sect. 2.2.

5.2 Application-Level QoS Assessment

5.2.1 Transfer Efficiency

Figure 7 displays the MU loss rate of voice versus the average load, and Fig. 8 plots that of video likewise.

In Fig. 7, we see that when the average load is heavier than about 350 kbps, the MU loss rate of voice with MPMS is smaller than that with SPMS. We also find in this figure that for the average loads heavier than around 350 kbps, MPNC has smaller MU loss rates than SPNC. In addition, Fig. 8 shows similar relationships among all the schemes to those in Fig. 7. That is, the multipath routing scheme can reduce the MU loss rate than the traditional DSR on heavily loaded conditions. This is because the multipath routing scheme transmits the voice and video streams on two separate routes and then can reduce the amount of transferred data per route. Thus, the multipath routing scheme can reduce the MU loss rate.

On the other hand, in Figs. 7 and 8, we notice that the MU loss rate with MPMS is smaller than that with MPNC

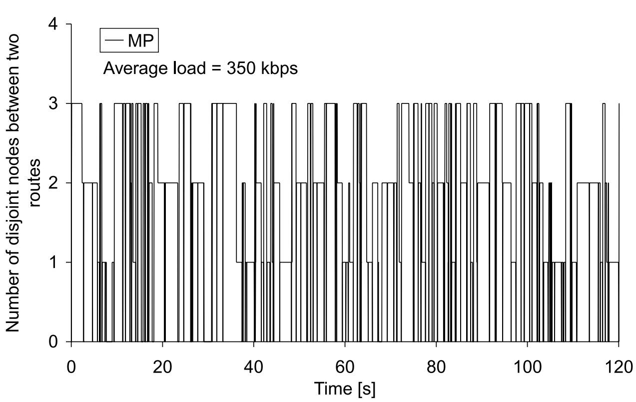

Fig. 5 Number of different nodes between two routes (average load = 350 kbps).

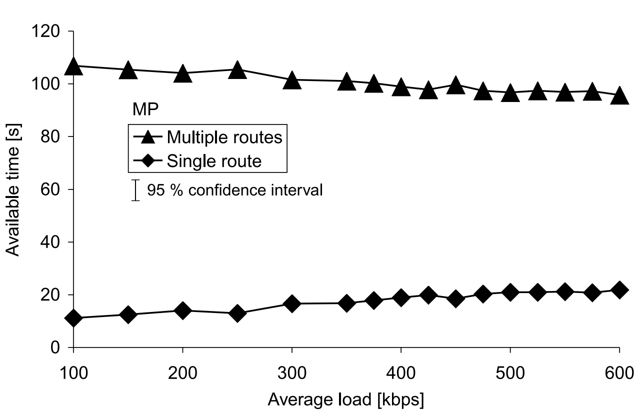

Fig. 6 Average available time of multiple routes.

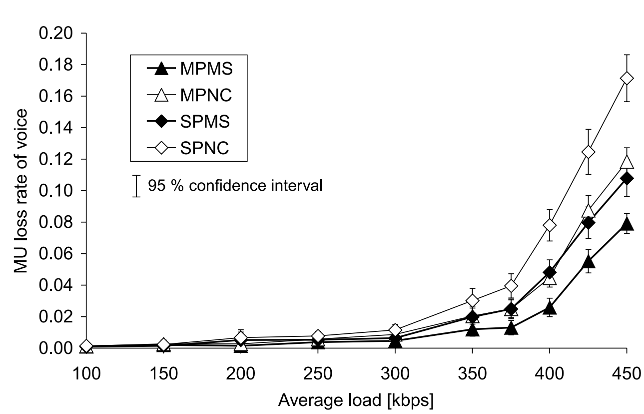

Fig. 7 MU loss rate of voice.

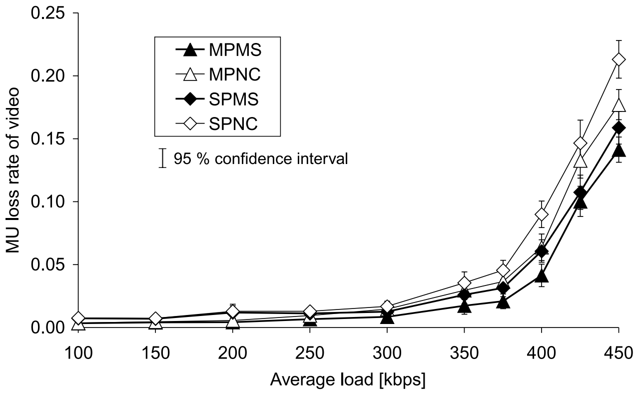

Fig. 8 MU loss rate of video.

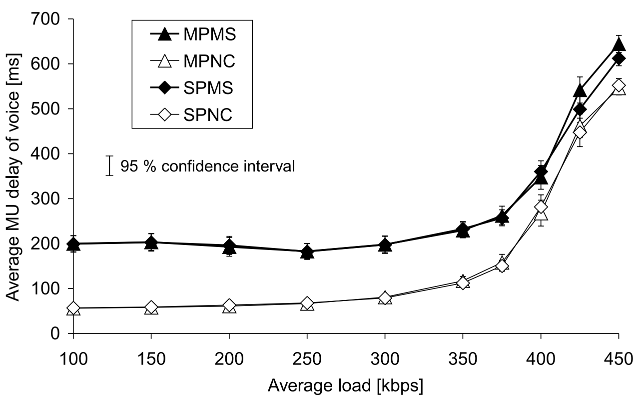

Fig. 9 Average MU delay of voice.

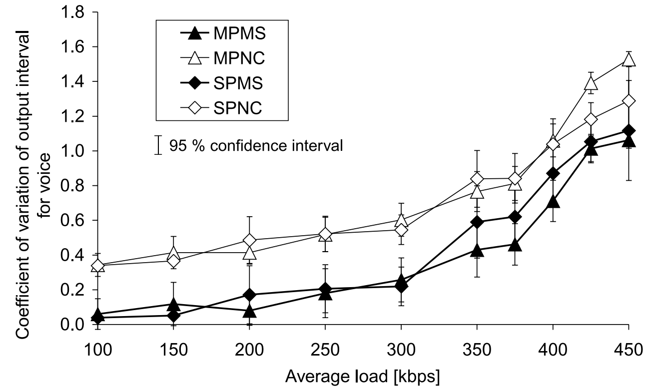

Fig. 10 Coefficient of variation of output interval for voice.

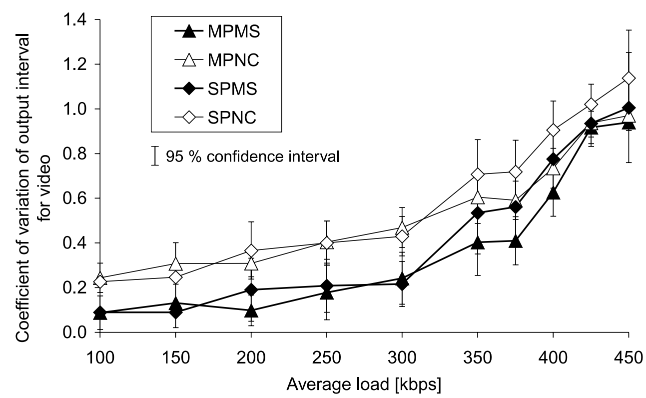

Fig. 11 Coefficient of variation of output interval for video.

when the average load is heavier than about 300 kbps, and SPMS has smaller MU loss rate than SPNC for the average loads heavier than around 300 kbps. That is, the media synchronization control can decrease the MU loss rate. This is because the sequence of MUs which arrive at the destination may be disturbed by route changes. When the destination exerts the media synchronization control, it performs the sequence-control of the MUs and can output them in the order of generation.

5.2.2 Real-Time Property

Figure 9 displays the average MU delay of voice. Since the relationship of the average MU delay of video between the schemes is similar to that in Fig. 9, we do not show it here.

In Fig. 9, we see that MPMS has larger MU delay than SPMS for the average loads heavier than around 425 kbps. The reason is as follows. At the destination, MPMS receives more MUs than SPMS because the MU loss rate with MPMS is smaller than that with SPMS as shown in Fig. 7. However, received MUs in MPMS tend to have larger transfer delays than that in SPMS on heavily loaded conditions because MPMS often uses more redundant routes than SPMS in order to perform multipath streaming. Thus, the average MU delay with MPMS is larger than that with SPMS.

5.2.3 Media Synchronization Quality

Figure 10 depicts the coefficient of variation of output interval for voice as a function of the average load. Figure 11 plots the coefficient for video versus the average load.

We can observe in Figs. 10 and 11 that for the average loads heavier than about 350 kbps, MPMS, which employs the multipath routing and the media synchronization control, has the smallest coefficient of variation among all the schemes. Thus, MPMS is effective in improving the intra-stream synchronization quality of the voice and video streams on heavily loaded conditions.

Recall that a possible drawback of the MP schemes is the degradation of the inter-stream synchronization quality because of different routes for voice and video. Now, let us study this problem. Figure 12 plots the mean square error of inter-stream synchronization versus the average load.

In Fig. 12, we find that for all the average loads here, MPMS has smaller inter-stream synchronization error than MPNC. Thus, the inter-stream synchronization control can reduce the inter-stream synchronization error.

We also see in Fig. 12 that when the average load is heavier than around 250 kbps, MPMS has larger mean square error of inter-stream synchronization than SPMS. Furthermore, the inter-stream synchronization error of

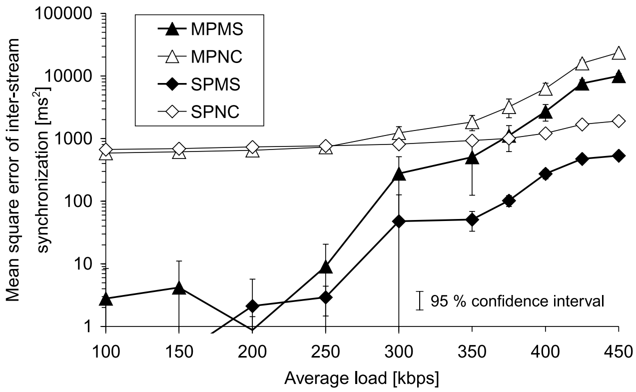

Fig. 12 Mean square error of inter-stream synchronization.

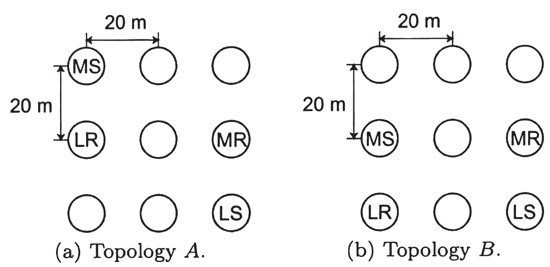

Fig. 13 Network configuration of other locations of nodes.

MPMS is larger than that of SPNC for the average loads heavier than about 400 kbps. However, it should be noted that the mean square error with MPMS is smaller than $25600\,\mathrm{ms}^2 = (160)^2\,\mathrm{ms}^2$ even on heavily loaded conditions. On the basis of the results in [14], the mean square errors smaller than $6400\,\mathrm{ms}^2 = (80)^2\,\mathrm{ms}^2$ means high quality of inter-stream synchronization, and those larger than $25600\,\mathrm{ms}^2$ correspond to asynchrony in lip-synch. Thus, MPMS can provide acceptable quality of inter-stream synchronization.

5.2.4 Effect of Locations of Nodes

We also assess other locations of nodes in the grid topology network with 9 nodes as shown in Fig. 13: Topology *A* and Topology *B*. Each network has a smaller number of the shortest hops from MS to MR than that in Fig. 2.

Figure 14 depicts the coefficient of variation of output interval for voice in Topology *A*. We see in this figure that for all the average loads here, the coefficient of variation with MPMS is the smallest among all the schemes. Thus, MPMS is also effective in Topology *A*.

On the other hand, Fig. 15 shows the coefficient of variation of output interval for voice in Topology *B*. We find in this figure that when the average load is heavier than about 800 kbps, the coefficients of variation with all the schemes fluctuate largely. However, MPMS has approximately the same values of the coefficient of variation as those with SPMS on the load condition. Thus, MPMS can achieve at least the same quality as that in SPMS.

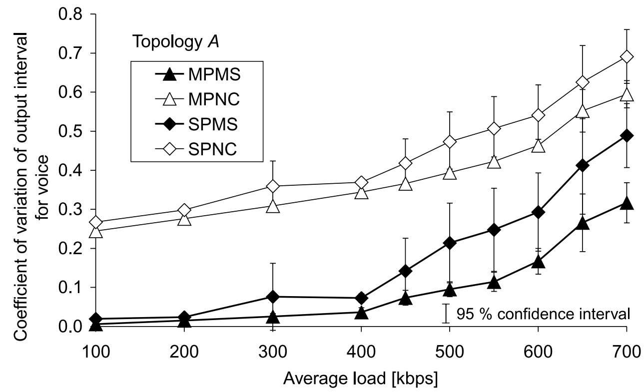

Fig. 14 Coefficient of variation of output interval for voice in Topology *A*.

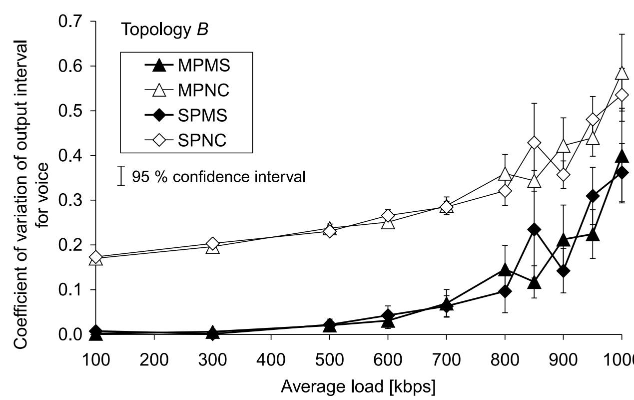

Fig. 15 Coefficient of variation of output interval for voice in Topology *B*.

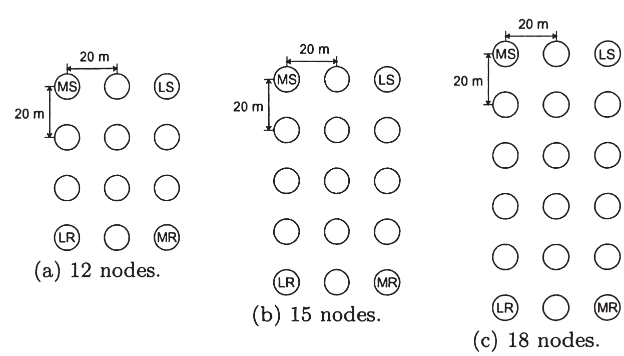

Fig. 16 Extended networks.

5.2.5 Influence of the Number of Nodes

We also assess the influence of the number of nodes on the application-level QoS of MPMS by extending the network in Fig. 2 in the vertical direction. The extended networks are shown in Fig. 16.

Figure 17 shows the MU loss rate of voice versus the number of nodes in the network when the average load is set to 350 kbps. In addition, Fig. 18 depicts the mean square error of inter-stream synchronization.

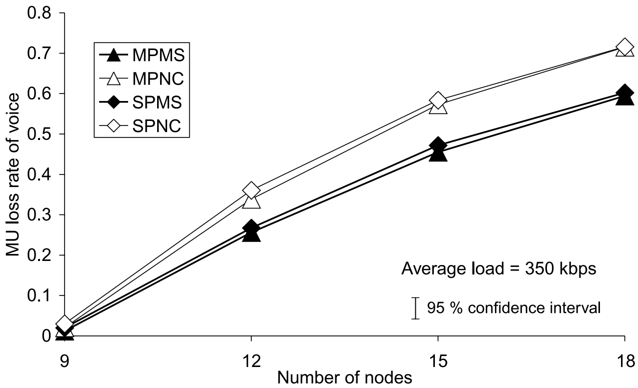

Fig. 17 MU loss rate of voice versus network size.

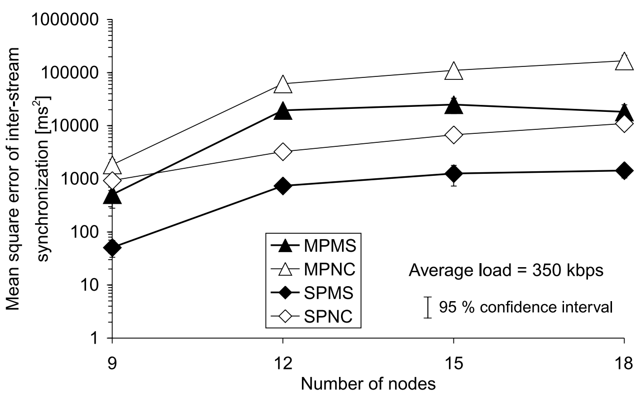

Fig. 18 Mean square error of inter-stream synchronization versus network size.

In Fig. 17, we find that for all the network sizes here, the MU loss rate of voice with MPMS is the same as or smaller than that with the other schemes. Thus, even if the network size becomes large, MPMS is effective in reducing MU loss rate. However, note that as the network size becomes large, the MU loss rate increases largely. When the number of nodes is 18, for instance, the MU loss rates of all the schemes are larger than about 0.6; it will not be acceptable for users.

We have also assessed the MU loss rate of voice for average loads lighter than 350 kbps. As a result, we found that even on lightly loaded conditions, the MU loss rate increases largely as the network size becomes large; for example, when the average load is 100 kbps, the MU loss rate of voice with MPMS in the 18 nodes network is about 0.32.

We notice in Fig. 18 that when the number of nodes is 12 or larger, MPMS has larger mean square error of inter-stream synchronization than SPNC and SPMS. However, the mean square error with MPMS is smaller than 25600 ms^2 for all the network sizes here. Thus, MPMS can provide acceptable quality of inter-stream synchronization in networks of those sizes.

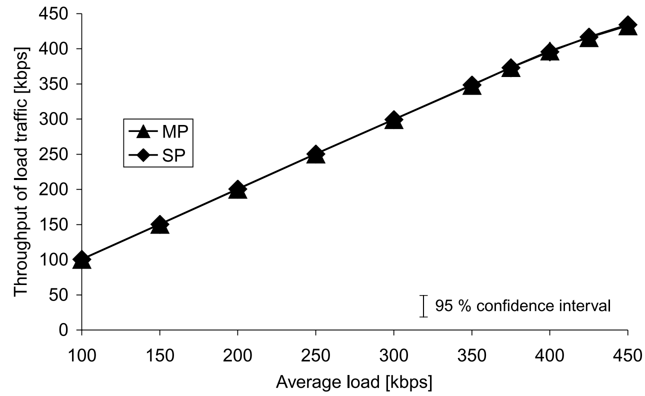

Fig. 19 Throughput of load traffic.

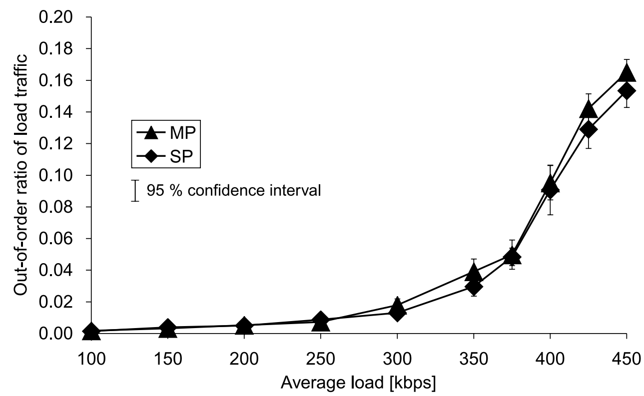

Fig. 20 Out-of-sequence ratio of load traffic.

5.3 Influence on Load Traffic

The multipath streams by MPMS can affect other traffic in the network. Therefore, we examine the statistics of the load traffic in the grid topology network with 9 nodes as shown in Fig. 2.

Figure 19 shows the throughput of the load traffic. This is defined as the average number of load information bits received in a second at LR. In addition, Fig. 20 means the out-of-sequence ratio of the load traffic. This indicates the ratio of load packets which arrived at LR out of sequence to the total number of packets transmitted from LS. Furthermore, Fig. 21 depicts the average number of route errors which occurred with transmitted packets from LS during a simulation run versus the average load. Note that in this subsection, we do not distinguish between the existence of the media synchronization control and nonexistence of that because the control does not affect these QoS parameters.

In Fig. 19, we find that for all the average loads here, the throughput of the load traffic in MP is almost the same as that in SP. That is, the routing protocols for the media streams scarcely affect the throughput of the load traffic.

On the other hand, in Fig. 20, we see that for the average loads heavier than around 300 kbps, the out-of-sequence ratio of the load traffic in MP is larger than that in SP. This is

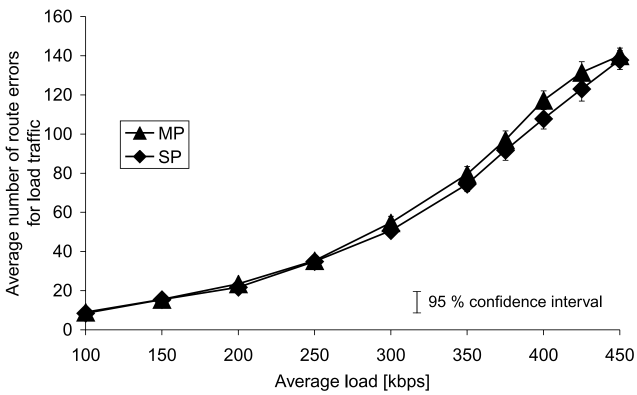

Fig. 21 Average number of route errors for load traffic.

because the number of route destruction for the load traffic in MP is larger than that in SP as shown in Fig. 21. Thus, the route for the load traffic in MP changes more frequently than that in SP; the out-of-sequence of packets occurs when the route changes. Therefore, the routing protocols for the media streams can affect the arrival order of packets at the load receiver. However, if the application has a sequence control buffer, the out-of-sequence of packets is not a critical problem.

6. Conclusions

In this paper, we proposed the MultiPath routing protocol for audio-video streaming with Media Synchronization control (MPMS) in wireless ad hoc networks. Then, we assessed the application-level QoS of MPMS by simulation. As a result, we found that MPMS reduces the MU loss rate on heavily loaded conditions and then has good intra-stream synchronization quality. On the other hand, the inter-stream synchronization quality of MPMS is lower than that of SPMS on heavily loaded conditions; however, the quality of MPMS is acceptable.

We next examined the effect of the locations of the source and destination nodes, and observed that MPMS can achieve at least the same quality as that in SPMS.

Also, we investigated the influence of the number of nodes on the application-level QoS. We then found that as the number of nodes increases, the application-level QoS of all the schemes degrades largely.

Furthermore, we assessed the influence of the multipath transmission on the load traffic. As a result, we noticed that the multipath streaming scheme causes more out-of-sequence packets than the single-path scheme. However, the throughput of the load traffic is hardly affected by the multipath streams.

As a next step of our research, we will study more efficient multipath routing schemes based on other routing protocols. In addition, we need to investigate QoS control schemes at the other layers such as QoS-aware MAC protocols and error recovery schemes at the transport layer.

We should assess the QoS in other network configurations which are representative of the real world, such as mobile nodes, many sources and destinations, and varying node distances. Furthermore, we need to assess the user-level QoS of MPMS and need to investigate the relationship between the user-level QoS and the application-level QoS. Through these studies, we will confirm the existence of the mutually compensatory property in MPMS. Also, we should assess QoS at the lower layers and investigate the relationship between application-level QoS and QoS at the lower layers.

Regarding the mobility of nodes, we have already assessed the influence in a grid topology network with a simple movement pattern of nodes [15]. From the results, we have found that MPMS is effective on reducing the audio MU loss rate in the network with mobile nodes; however, the optimization of MPMS for mobile networks is left for future work.

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