LETTER

An Audio-Video Multipath Streaming Scheme for Ad Hoc Networks: The Effect of Node Mobility

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SUMMARY This letter studies the effect of node mobility on application-level QoS of audio-video multipath streams in wireless ad hoc networks. The audio-video streams are transmitted with the *MultiPath streaming scheme with Media Synchronization control (MPMS)*, which was previously proposed by the authors. We perform computer simulation with a grid topology network of IEEE 802.11b including two mobile nodes. The simulation results show that MPMS is effective in achieving high application-level QoS in mobile networks as well.

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

1. Introduction

Some applications of wireless ad hoc networks [1] require the ability to support live audio-video streaming over the network. Thus, the realization of this type of service with high quality is highly demanded.

The authors have proposed the *MultiPath streaming scheme with Media Synchronization control (MPMS)* for audio-video transmission in wireless ad hoc networks [2]. MPMS treats audio and video as two separate transport streams and sends the two streams to different routes if multipath routes are available. Furthermore, in order to remedy the temporal structure of the media streams disturbed by the multipath transmission, we employ *media synchronization control* [3].

In [2], we assess the *application-level QoS (Quality of Service)* [4] of MPMS in fixed grid topology networks. However, mobile networks are not considered there. The mobility is a key characteristic of ad hoc networks. Thus, we need to assess the application-level QoS of MPMS in mobile ad hoc networks.

Many studies on ad hoc networks employ the random waypoint model [5] for QoS assessment of transmitted data. The model is useful for QoS assessment without consideration of the temporal structure of transmitted data. However, the temporal structure must be considered in the application-level QoS assessment of audio-video transmission. In the random waypoint model, the network configuration dynamically changes in each simulation run, and then assessment results of the temporal structure of audio-video streams are largely different from run to run. In [6], we perform a preliminary simulation of audio-video streaming in the random waypoint network. As a result, we have found that in many simulation runs, the application-level QoS is not acceptable owing to drastic changes of network conditions. That is, audio-video streaming in situations represented by the random waypoint model is impractical. Thus, as a first step toward this kind of study, networks with some specific topology are practical for the application-level QoS assessment of audio-video streams.

In this letter, we investigate the influence of the node movement on the application-level QoS of MPMS by simulation. To assess the basic characteristics of MPMS in mobile networks, we employ a grid topology network with mobile nodes. We also refine the route selection algorithm of MPMS.

2. MPMS

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* [4] 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.

When the audio and video streams are transmitted into two different routes, the transfer delay of audio usually differs from that of video; the difference disturbs *inter-stream synchronization.* Thus, in order to remedy the temporal structure, MPMS employs the enhanced *Virtual-Time Rendering (VTR)* algorithm [7] for media synchronization control.

In what follows, we show an outline of the routing strategy of MPMS. See [2] for details.

2.1 Routing Strategy

MPMS can utilize any routing algorithm for selecting candidates of multipath routes. As an example of MPMS, we enhance the existing DSR (Dynamic Source Routing) protocol [8].

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,

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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.

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 situations unsuitable for the multipath transmission, MPMS uses the single-path transmission. Thus, even in the worst case for the multipath transmission, MPMS achieves at least the same QoS as that in the single-path transmission.

2.2 Improvement on Route Selection Algorithm

The route selection algorithm of MPMS in [2] can choose a secondary route which contains all the primary route nodes. That is, the secondary route can have only redundant nodes against the primary route†.

In this letter, we refine the route selection algorithm. If a cached route includes all the primary route nodes, the route selection algorithm rejects the route as the secondary route.

3. Methodology for QoS Assessment

We assess the application-level QoS of MPMS by computer simulation with *ns-2* [9].

3.1 Network Configuration

In this letter, we consider a simple grid topology network as a first step to the study on the characteristics of MPMS in mobile ad hoc networks. The network consists of 18 nodes as shown in Fig. 1. The media streams are transmitted from MS (Media Source) to MR (Media Receiver); an independent interference traffic flow for the media streams is transmitted from LS (Load Sender) to LR (Load Receiver). Except for MR and LR, the interval between two vertical or horizontal adjacent nodes is constant, 20 m. In grid topology networks, multiple routes are available in almost all source/destination pairs. That is, the network topology corresponds to an effective case for multipath streaming schemes. It should be noted that as the next step of this study, we need assessment in more practical topology networks, such as many mobile nodes and varying node distances.

MR and LR move along the dotted rectangle depicted in this figure. In order to assess the influence of the node mobility, the movement speed of each node is set

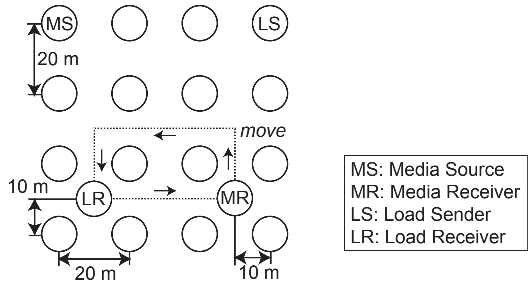

Fig. 1 Network configuration.

to 0 m/s (motionless), 1.0 m/s (= 3.6 km/h, walking speed), 2.0 m/s or 4.0 m/s. In every case of the movement speed here, MR and LR come back to their original positions when the simulation time becomes 120 seconds (i.e., the end of a simulation run). That is, when the movement speeds are set to 1.0, 2.0 and 4.0 m/s, each node rounds the rectangle once, twice and four times during a simulation run, respectively.

We formulate a detailed simulation model which is based on the *distributed coordination function (DCF)* [10] of the IEEE 802.11. The transmission speed is kept at 11 Mbps (i.e., IEEE 802.11b). The communication range of each node is set to about 22.49 m. We set the carrier sensing range [11], within which a transmitter triggers carrier sense detection, to about 44.98 m. The RTS/CTS mechanism is not used in the simulation. The maximum number of trials of frame retransmission is set to seven.

3.2 Method of Simulation

We assume MS as the voice and video sources. MS unicasts the media streams to MR with RTP/UDP.

We use a voice stream of ITU-T G.711 μ-law and an MPEG1 (GOP I) video stream. An *Media Unit (MU)* is the information unit for media synchronization. The size of each video MU (i.e., a video frame) changes from MU to MU. Each voice MU has a constant size. The original MU rates of the voice is 25 MU/s, and that of the video is 20 MU/s. The original bit rate of the voice stream is 64 kbps, and the original average bit rate of the video stream is 320 kbps.

The parameter values in the enhanced VTR algorithm are set to the same values as those in [2].

In the simulation, we assess the application-level QoS of two schemes: MPMS and SPMS. SPMS shows the traditional DSR (i.e., single-path routing) with the media synchronization control [2].

LS and LR are used to handle load traffic for the media streams. We employ 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. We refer to the average amount of the load traffic as the *average load*.

†Owing to the limitation of the difference in the number of hops between the two routes, if a candidate of the secondary route has two or more redundant nodes, it is not used.

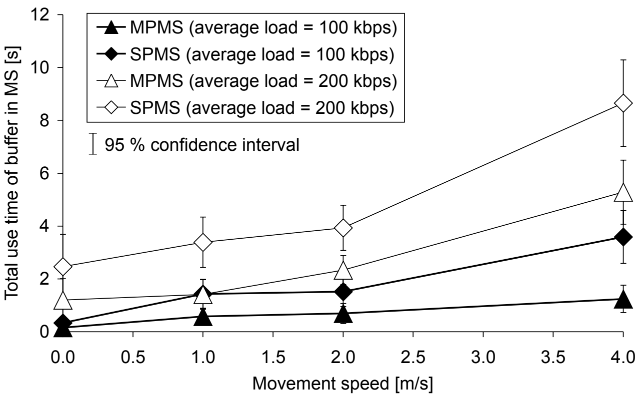

Fig. 2 Total use time of the send buffer in MS.

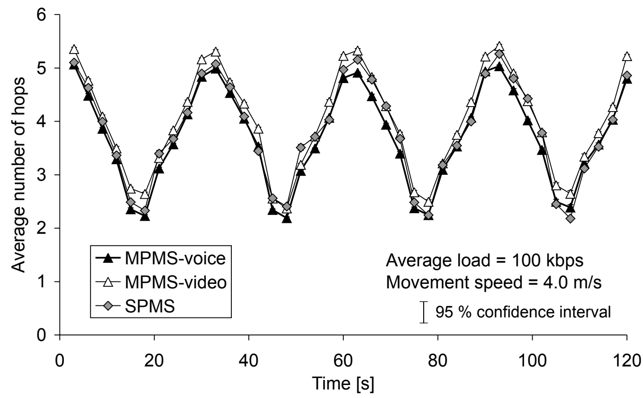

Fig. 3 Average number of hops as a function of time.

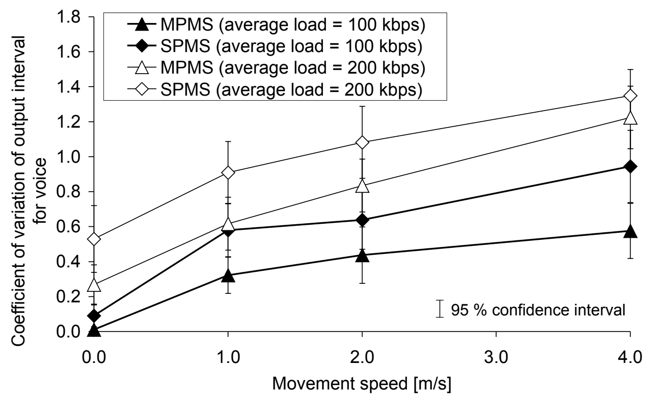

Fig. 4 Coefficient of variation of output interval for voice.

3.3 QoS Parameters

In this paper, we employ the three QoS parameters: the *coefficient of variation of output interval*, the *mean square error of inter-stream synchronization*, and the *total use time of the send buffer in the source node*. See [2] for details.

4. Results of QoS Assessment

In this section, we first show the results of the network-level QoS assessment. Then, we present the results of the application-level QoS parameters.

Each symbol in the figures to be shown represents the average of 30 measured values which were obtained by changing the random seed for generating the load traffic. We also show 95 % confidence intervals of the measurement results in the figures.

4.1 Network-Level QoS Assessment

Figure 2 shows the total use time of the send buffer in MS versus the movement speed. It reflects the total time when the source wants to send packets but has no route. This figure shows the results on two different load conditions: 100 kbps and 200 kbps.

We find in Fig. 2 that MPMS has smaller values of the total use time of the send buffer than SPMS. That is, MPMS can decrease the period when the source has no route in the mobile topology network.

In Fig. 2, we notice that the total use time of the send buffer in each scheme increases as the average load and the movement speed increase. This is because the probability of route destruction increases as the average load and the movement speed increase.

Figure 3 depicts the average number of hops on the route versus time when the average load and the movement speed are set to 100 kbps and 4.0 m/s, respectively. This figure plots the average number of hops every three seconds; the measurement was made for 120 seconds after the capturing of the first MU.

We find in Fig. 3 that the average number of hops on the route for the voice stream in MPMS is smaller than that for the video stream. This is because a priority is given to the voice stream over the video one in route selection of MPMS.

In Fig. 3, we see that the average number of hops on the route in SPMS is approximately equal to or slightly larger than that on the route for the voice stream in MPMS for almost all the simulation time. This is because MPMS has more chances to find the shortest hops route than SPMS.

4.2 Application-Level QoS Assessment

Figure 4 depicts the coefficient of variation of output interval for voice as a function of the movement speed. It represents the smoothness of output of a media stream. Since the relations of the coefficients for video among the schemes are similar to those for voice in Fig. 4, we do not show the video ones here. In this figure, we find that MPMS has smaller values of the coefficient of variation than SPMS. This is because MPMS can decrease the period when the source has no route.

We also notice in Fig. 4 that the coefficients increase as the average load increases, and as the movement

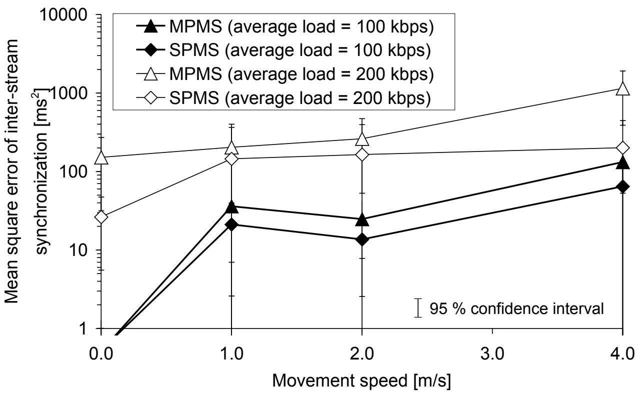

Fig. 5 Mean square error of inter-stream synchronization.

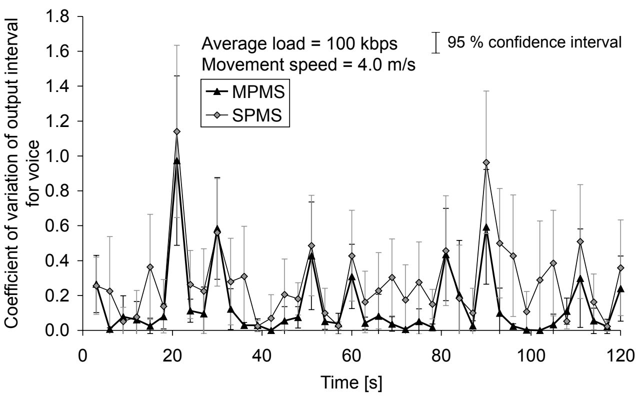

Fig. 6 Coefficient of variation of output interval for voice as a function of time.

speed becomes higher. This is because the route destruction occurs frequently on those conditions.

Figure 5 plots the mean square error of inter-stream synchronization versus the movement speed. We see in this figure that for almost all the movement speeds and average loads here, MPMS has larger mean square errors of inter-stream synchronization than SPMS. However, it should be noted that the mean square error with MPMS is smaller than $6400\ \mathrm{ms}^2 = (80)^2\ \mathrm{ms}^2$. On the basis of the results in [12], the mean square errors smaller than $6400\ \mathrm{ms}^2$ means high quality of inter-stream synchronization in lip-synch. Thus, MPMS can provide high quality of inter-stream synchronization.

Figure 6 depicts the coefficient of variation of output interval for voice versus time when the average load and the movement speed are set to 100 kbps and 4.0 m/s, respectively. In this figure, we show the results in the same way as those in Fig. 3.

In Fig. 6, we find that the coefficient of variation for voice with MPMS is smaller than that with SPMS especially in the periods of around time 63 to time 79 and time 96 to time 105. As shown in Fig. 3, the number of hops from MS to MR decreases in those periods, and the average number of hops for the voice stream in MPMS is smaller than that in SPMS. Thus, when the average number of hops from MS to MR decreases, MPMS gets an advantage over SPMS in the intra-stream synchronization quality of the voice stream.

We have also assessed the coefficients as a function of time in other movement speeds. As a result, we found the same relationships between the two schemes as those on the 4.0 m/s speed condition.

5. Conclusions

In this letter, we investigated the influence of node movement on the application-level QoS of MPMS. As a result, we found that MPMS is effective in improving the application-level QoS in mobile networks as well, though the application-level QoS degrades as the movement speed increases. In particular, MPMS gains an advantage over SPMS in the intra-stream synchronization quality when the number of hops from the source to the destination decreases.

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