

PAPER

Application-Level and User-Level QoS Assessment of Audio-Video IP Transmission over Cross-Layer Designed Wireless Ad Hoc Networks

Toshiro NUNOME^{†a)}, Member, Shuji TASAKA^{†b)}, Fellow, and Ken NAKAOKA^{††c)}, Member

SUMMARY This paper performs application-level QoS and user-level QoS assessment of audio-video streaming in cross-layer designed wireless ad hoc networks. In order to achieve high QoS at the user-level, we employ *link quality-based routing* in the network layer and *media synchronization control* in the application layer. We adopt three link quality-based routing protocols: *OLSR-SS (Signal Strength)*, *AODV-SS*, and *LQHR (Link Quality-Based Hybrid Routing)*. OLSR-SS is a proactive routing protocol, while AODV-SS is a reactive one. LQHR is a hybrid protocol, which is a combination of proactive and reactive routing protocols. For application-level QoS assessment, we performed computer simulation with ns-2 where an IEEE 802.11b mesh topology network with 24 nodes was assumed. We also assessed user-level QoS by a subjective experiment with 30 assessors. From the assessment results, we find AODV-SS the best for networks with long inter-node distances, while LQHR outperforms AODV-SS for short inter-node distances. In addition, we also examine characteristics of the three schemes with respect to the application-level QoS in random topology networks.

key words: wireless ad hoc network, audio-video streaming, QoS, cross-layer design, link quality-based routing, media synchronization control

1. Introduction

The recent advance of wireless networks is leading a paradigm shift, from wireless services to ubiquitous services. *Mobile ad hoc network (MANET)* [1] has been receiving much attention for realization of the ubiquitous wireless services. The MANET is an autonomous wireless network without any infrastructure support. Each mobile node in MANET acts as a router, which discovers and maintains routes to other nodes 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.

The Internet Protocol (IP) suite plays an important role even in ad hoc networks. Owing to the layered architec-

ture of IP-based networks, its *QoS (Quality of Service)* 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 user-level QoS is also referred to as *subjective QoS* or *perceptual QoS*, which is the most important since the users are the ultimate recipients of the services. ITU-T defines the user-level QoS as *QoE (Quality of Experience)* [3].

A variety of studies on audio-video 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; however, they do not consider video streams. In [5], Xiang et al. assess subjective quality of an audio stream in an ad hoc network with Internet connection. They employ the E-model [6] and do not perform subjective experiments. Liang et al. perform application-level QoS assessment of audio and video streams in a single-hop ad hoc network in [7]. They assess *PSNR (Peak Signal-to-Noise Ratio)* and end-to-end delay for video. The PSNR, which represents spatial quality of video streams, is popularly used as an application-level QoS parameter in the literature [7]–[10]. For audio, they employ *PESQ (Perceptual Evaluation of Speech Quality)* [11] and the E-model. However, they assess audio and video separately and then do not consider cross-modal effect between them. They also do not assess user-level QoS.

The preservation of the temporal structure of audio and video is essential to high QoS at the application-level in audio-video transmission [2]. When we transmit the audio-video streams in ad hoc networks, the temporal structure of the streams can be disturbed largely by delay, its jitter and packet loss.

In order to preserve the temporal relation, we need the *media synchronization control* [12], which is application-level QoS control. However, only by the application-level QoS control, it is difficult to achieve high QoS of audio and video streams in ad hoc networks.

The cross-layer design architecture [13] is expected as an approach to high quality communication in ad hoc networks. The architecture exploits interaction among more than two layers. Although the layered architecture in IP-

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[†]The authors are with the Graduate School of Engineering, Nagoya Institute of Technology, Nagoya-shi, 466-8555 Japan.

^{††}The author is with Digital Technology Research Center, SANYO Electric Co., Ltd., Gifu-ken, 503-0195 Japan.

a) E-mail: nunome@nitech.ac.jp

b) E-mail: tasaka@nitech.ac.jp

c) E-mail: ken.nakaoka@sanyo.com

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based networks has some advantages such as reduction of network design complexity, it is not well suited to wireless networks. This is because the nature of the wireless medium makes it difficult to decouple the layers.

There are many studies on the cross-layer design architecture for multimedia streaming [9], [10], [14]–[17]. In [9] and [14], the number of hops maintained by the routing protocol is used for selecting the video coding rate to the network capacity. If there are many hops from the sender to the receiver, the approach reduces the coding rate at the sender. It is a cross-layer design between the network and application layers. Abd El Al et al. propose an error recovery mechanism for real-time video streaming that combines FEC and multipath retransmission in [10]. This scheme determines strength of the error correction code and a quantization parameter for video encoding according to the number of hops. In [15], Frias et al. exploit the multipath routing protocol for scheduling prioritized video streams and best effort traffic. They schedule the traffic on the basis of the number of multiple routes. In [16], Nunome and Tasaka propose the *Multi-Path streaming scheme with Media Synchronization control (MPMS)*. It 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, media synchronization control is employed.

While the above approaches refer to cross-layering between the network and application layers, in [17], Setton et al. explore a new framework for cross-layer design that incorporates adaptation across all layers of the protocol stack: application, transport protocols, resource allocation, and link layer techniques.

It should be noted that all of the previous studies mentioned above do not evaluate the user-level QoS of transmitted multimedia streams. Furthermore, these studies except for [16] consider video only and do not assess its temporal quality.

In the cross-layer design architecture, the routing protocol is an essential component. The *link quality-based routing* is one of the most promising approaches to establishment of routes with high quality and high throughput. It has been studied as *QoS routing* [18] and *multi-rate aware routing* [19], [20]. It can avoid using links with low data rates by taking account of link quality such as signal strength and link utilization level for route selection; this implies a cross-layer design among the network and lower layers.

The aim of this paper is to achieve high user-level QoS of audio and video streams transmitted over ad hoc networks. The cross-layer design with media synchronization control and the link quality-based routing can be one of the most effective solutions for this purpose.

In this paper, we assess application-level QoS and user-level QoS of audio-video streaming with media synchronization control and link quality-based routing protocols in a wireless ad hoc network. We adopt three link quality-based routing protocols: *OLSR-SS (Signal Strength)* [21], *AODV-*

SS [22], and *LQHR (Link Quality-Based Hybrid Routing)* [23]. *OLSR-SS* is a modified version of *OLSR* [24], which is a proactive routing protocol. *AODV-SS* is a reactive protocol based on *AODV* [25]. *LQHR* is a hybrid protocol, which is a combination of proactive and reactive routing protocols. We clarify advantages and disadvantages of the three types in audio-video streaming with media synchronization control.

The rest of this paper is organized as follows. Section 2 explains link quality-based routing protocols for ad hoc networks. Section 3 illustrates a methodology for the QoS assessment, including the network configuration, simulation method, QoS parameters, and user-level QoS assessment. The assessment results are presented and discussed in Sect. 4.

2. Link Quality-Based Routing

A variety of studies on link quality-based routing protocols have been reported. As in traditional hop-based routing protocols, they can be classified into three categories: proactive, reactive, and hybrid. We then give an overview of the three types of protocols.

2.1 Proactive Routing Protocol

The proactive routing protocol periodically exchanges the routing information between nodes. The protocol performs well for fixed or low mobility networks.

In [21], Itaya et al. propose two techniques of multi-rate aware routing for improving the stability of communication. The first technique is employment of a threshold for *signal strength (SS)* of received routing packets. It is used to avoid routing packets via unreliable neighbors with poor radio links. The second technique is *synchronous update (SU)* of routing tables. It is used to avoid loops due to mismatch in timing of route updates. The techniques can be implemented as modifications to conventional routing protocols. They have implemented these techniques into *OLSR*. Although the first technique can be applied to reactive routing protocols, they have implemented nothing in [21].

As the proactive routing protocol for the comparison in this paper, we employ the scheme proposed in [21] with a little modification. The threshold for signal strength is kept constant for simplicity; in this paper, we denote the threshold by T_h . Furthermore, we assume that the time synchronization among the nodes is performed completely, because the simulation environment can get the global time synchronization automatically. We refer to the scheme as *OLSR-SS*, although it is called *OLSR-SS-SU* in [21].

2.2 Reactive Routing Protocol

The reactive routing protocol discovers routing paths when the source wants to send data; that is, it works on demand. It is appropriate for the use in highly mobile networks.

For example, in [26], Fan proposes high throughput reactive routing in multi-rate ad hoc networks. He modifies the AODV protocol in order to select suitable links with high data rates. In the scheme, the routing cost is calculated on the basis of MAC delay, which is equal to total delay of RTS/CTS/DATA/ACK communication. However, the scheme needs the information on the transmission speed of each link; that is, it is not a pure reactive scheme.

On the other hand, in [22], Budke et al. evaluate the QoS extensions for supporting real-time multiplayer game applications in IEEE 802.11 mobile ad hoc networks. They select AODV and add signal strength monitoring for *Route Request (RREQ)* packets. That is, the scheme can be regarded as a reactive version of the scheme proposed in [21]; thus, we refer to the scheme as *AODV-SS*.

In this paper, as the reactive routing protocol for the comparison, we specify AODV-SS as follows. When an intermediate node receives RREQ, it decides whether the packet should be forwarded or not by received signal strength. If the received signal strength at the intermediate node is lower than the threshold T_h , which is the same as that in OLSR-SS, the node drops the packet.

2.3 Hybrid Routing Protocol

The hybrid routing protocol is a combination of proactive and reactive routing protocols.

Nakaoka et al. propose LQHR in [23]. In LQHR, each node maintains routing information produced by an existing proactive routing protocol and measures link quality between the neighboring nodes. When a source node makes a communication request which needs high quality links, it selects a route to the destination node by referring to the link quality on an on-demand basis.

LQHR takes account of link quality representing both reliability and the link utilization level of each node. In [27], we revise the LQHR algorithm in order to overcome difficulties related to networks with many route selections. In this paper, we employ the revised algorithm.

LQHR consists of two modules:

- **Quality Measurement (QM) Module**
The QM module produces and maintains routing information by means of a proactive routing protocol; for example, OLSR is employed in [23]. It also periodically measures the link quality between adjacent nodes. The link quality is represented as a vector whose components are some quality parameters.
- **Route Selection (RS) Module**
The RS module selects a route to the destination node by referring to the link quality, which is measured by the QM module, on an on-demand basis when a communication request is made at a node.

On having a communication request, the source node sends a *Route Quality Request (RQReq)* message to each of the *possible next-hop nodes*. The possible next-hop node is a candidate of the next-hop node on the route to the des-

tinuation. The nodes receiving the RQReq message refer to the destination address and then forward it to each of their own possible next-hop nodes. The RQReq message is forwarded up to *last-hop nodes*. The last-hop node means the single-hop neighbor node to the destination. Once the RQReq message reaches the last-hop node, it forwards back a *Route Quality Response (RQRsp)* message, via the series of the possible next-hop nodes the RQReq message has gone through, finally to the source node; thus a route from the source to the destination is selected. The RQRsp messages are chosen and discarded on the way to the source node on the basis of the link quality of each forwarding node. See [23] and [27] for details.

3. Methodology of QoS Assessment

We assess the application-level QoS and the user-level QoS of audio-video streaming in ad hoc networks with the three schemes of link quality-based routing: LQHR, OLSR-SS, and AODV-SS. For this purpose, we performed computer simulation with *ns-2 (network simulator version 2)* [28].

Figure 1 shows the schematic diagram of the QoS assessment. We refer to the transmission unit at the application-level as a *Media Unit (MU)*; we define a video frame as a video MU and a constant number of audio samples as an audio MU. From the practical audio and video streams, we get traffic trace files for the simulation. The files include each MU size and inter-MU time. In addition, the file for video also includes the picture type of each video MU. In the simulation, we take into consideration the capturing and encoding delay time before the transmission procedure in order to emulate the audio-video streaming inputted real-time. With the traffic trace files and a simulation scenario, ns-2 outputs time charts in which the output timing of each MU is described. We can achieve application-level QoS parameter values by the time charts. Furthermore, for the user-level QoS assessment, the audio-video player plays the practical audio-video stream with the output timing obtained from the time charts.

3.1 Network Configuration

In this paper, we consider a simple mesh topology network as a first step to the study on the characteristics of the three schemes of link quality-based routing with media synchronization control in ad hoc networks. The network consists of 24 nodes as shown in Fig. 2. Each node has an omnidirectional antenna. We employ the shadowing model [29] as the propagation model in the simulation. In the model, received signal strength at the receiver is determined by the following equation:

$$\left[\frac{P_r(d)}{P_r(d_0)} \right]_{\text{dB}} = -10\beta \log \left(\frac{d}{d_0} \right) + X_{\text{dB}} \quad (1)$$

If $P_r(d)$ exceeds the threshold of received signal strength, the packet can be received. Here, β means path

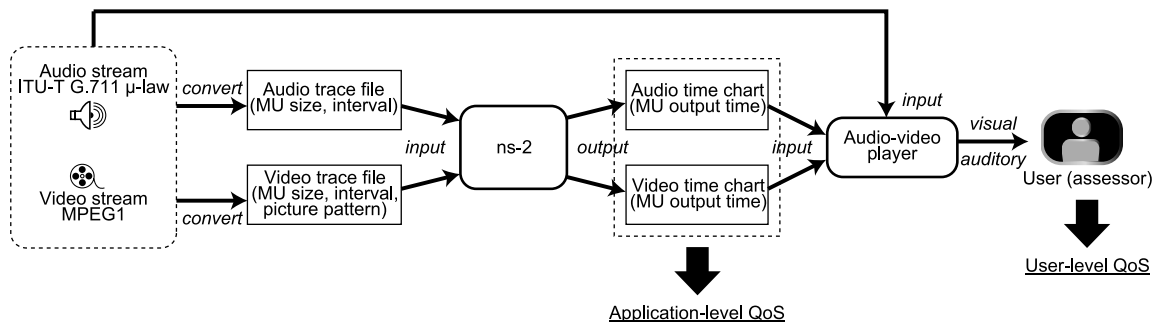


Fig. 1 Schematic diagram of QoS assessment.

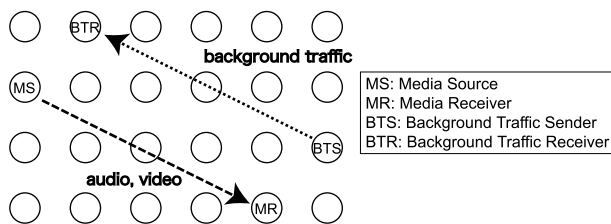


Fig. 2 Network configuration.

loss exponent and is set to 2 in the simulation. d_0 is close-in distance and is set to 1.0. X_{dB} shows a Gaussian random variable; the average and the standard deviation are set to 0 and 4.0, respectively. These are default values in ns-2. The model does not consider propagation errors or fading.

In the simulation, we assume seven patterns of the mesh topology by changing the distance between two vertically or horizontally adjacent nodes; we refer to the distance as the *inter-node distance*.

In mesh topology networks, there are many available routes; therefore, the networks are suitable for the assessment of the behavior of routing schemes. However, it should be noted that as a next step of this study, we need assessment in more practical topology networks like those with many mobile nodes.

We formulate a detailed simulation model which is based on the *distributed coordination function (DCF)* [30] of the IEEE 802.11b [31]. The transmission speed is automatically changed from 2 Mb/s to 11 Mb/s by means of the rate adaptation mechanism. In this paper, we employ *ARF (Automatic Rate Fallback)* [32]. The transmission speed is controlled for each link, and broadcast frames are transmitted at 2 Mb/s. The maximum number of trials of frame retransmission is set to four. The RTS/CTS mechanism is not used in the simulation, because it has been reported that the conventional RTS/CTS mechanism does not work well in ad hoc networks [33], [34].

Because the received signal strength changes dynamically in the shadowing model, the communication range of each node fluctuates in time and is determined by the transmission speed. In the simulation, a node can receive a packet with probability 0.95 when the distance between the node and the sender is 34.54 m at 11 Mb/s, 48.36 m at 5.5 Mb/s, and 62.17 m at 2 Mb/s. These values are calculated by the

Table 1 Specifications of the audio and video.

item	audio	video
coding scheme	ITU-T G.711 μ -law	MPEG1 GOP IPPPP
image size [pixels]	–	320 × 240
original average MU size [bytes]	320	2708
original average MU rate [MU/s]	25.0	15.0
original average inter-MU time [ms]	40.0	66.7
original average bit rate [kb/s]	64.0	325.0
measurement time [s]	120.0	

threshold program, which is included in ns-2.

3.2 Method of Simulation

In Fig. 2, we assume *MS (Media Source)* as the audio and video sources. *MS* transmits the media streams to *MR (Media Receiver)* with RTP/UDP. We use an audio stream of ITU-T G.711 μ -law and an MPEG1 video stream, which has been prepared by encoding a part of Japanese news program. Table 1 shows the specifications of the audio and video.

In the simulation, we assume real-time captured audio-video stream. Thus, we take the media capturing and encoding delay time into consideration. The capture duration of an audio 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 audio MU to 40 ms. On the other hand, the capture duration of a video MU is just a moment. However, it takes much time to encode a video frame. Furthermore, in MPEG, the captured frame is buffered in the frame buffer for its predictive coding. Thus, in this paper, we set the capturing and encoding delay time of each video MU to 74 ms; each MU leaves the source the capturing and encoding delay time after its timestamp. This value includes capturing, buffering and encoding delay for a picture. We assume that the encoding delay is 7.3 ms, which is approximately the same as that of JPEG video in [35]. We also consider that the buffering delay is the same as the frame interval, 66.7 ms.

We exert media synchronization control with the enhanced VTR algorithm [35] as a first step of the cross-layer

study. The parameter values in the enhanced VTR algorithm are set to the same as those in [36]. That is, we set the *initial buffering time* J_{\max} [35] and the *maximum allowable delay* Δ_{al} [35] to 100 ms and 300 ms, respectively.

In the simulation, if MR cannot receive a picture, the succeeding P-pictures are discarded until the next I-picture appears for preserving spatial quality of the video stream; that is, the spatial quality does not degrade over the network.

Each simulation runs for 145 seconds. The source starts to generate audio and video streams at time 21 from the beginning of the simulation. In LQHR, the route is requested one second before starting audio and video streams; that is, the source generates an RQReq packet to the destination at time 20. In addition, LQHR periodically renews the route every five seconds after sending the first RQReq. For a fair comparison, AODV-SS also searches the route one second before starting to generate the streams by transmitting a dummy packet.

In this paper, LQHR employs the received signal strength as a link quality instead of Signal-to-Noise Ratio (SNR). This is because the simulation by the original ns-2 cannot consider the strength of background noise and therefore cannot calculate SNR. The threshold value for signal strength T_h is set to -62.7 dBm, which is the threshold for acceptable signal strength at 11 Mb/s in the simulation, for all the three schemes.

The decision mechanism of the optimal T_h value is out of scope in this paper, because we focus on basic characteristics of the three schemes. However, for example, a method for optimizing the threshold value discussed in [21] can be used in the three schemes.

BTS (Background Traffic Sender) and *BTR (Background Traffic Receiver)* are used to handle an independent interference traffic flow for the media streams. We also employ the same routing scheme as that for the media transmission. BTS generates fixed-size IP datagrams of 1500 bytes each at exponentially distributed intervals and then sends to BTR. BTS starts to generate the traffic at time 20. 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*. We set the average load to 100 kb/s in the simulation.

The route for audio-video transmission and that for background traffic are established autonomously and individually. Thus, the two routes are not always in parallel and can intersect each other. Furthermore, owing to the characteristics of the wireless radio, even if the two routes do not cross, they can affect each other.

3.3 Application-Level and Lower-Level QoS Parameters

In order to assess the application-level QoS of the media streams, we need to examine the intra-stream and inter-stream synchronization quality.

For the quality assessment of intra-stream synchronization for audio 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., the presentation time interval of two MUs at the destination) of a stream to its average; this represents the smoothness of the output stream.

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

As a measure of transfer efficiency, we assess the *average MU rate*, which is the output rate of MUs. Here, the discarded MUs are not included into the output MUs.

The *average MU delay*, which is the average 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.

In addition, we also assess the behavior of the three routing schemes. For this purpose, we employ the *percentage of the number of hops*, the *percentage of selected transmission speed*, and the *number of control packets for routing*. The percentage of the number of hops shows the relative frequency of the number of hops from the source to the destination. The percentage of selected transmission speed represents the relative frequency of the transmission speed for all the links. These parameters show characteristics of the selected routes.

The number of control packets for routing means the total number of the routing packets, such as route request packets, route reply packets, and topology information packets. It shows the routing overhead.

3.4 User-Level QoS Assessment

In this paper, we assess user-level QoS of the audio-video stream transferred with the three schemes by a subjective experiment. It was conducted as follows.

For subjective assessment, we made *stimuli*, which are objects to be evaluated, by actually outputting the audio and video MUs with the output timing obtained from the simulation. Each stimulus lasts 120 seconds.

We put the stimuli in a random order and presented them to 30 assessors, using a laptop PC with headphones. The laptop PC is equipped with a 12-inch XGA (1024 × 768 pixels) LCD display. The assessors are male and female. They were in their twenties and non-experts in the sense that they were not directly concerned with audio and video quality as a part of their normal work.

In this paper, we utilize the *mean opinion score (MOS)* as the user-level QoS parameter. The MOS value is obtained by the *rating-scale method*, where an assessor classifies stimuli into a certain number of categories each assigned an integer. We adopted the following five categories of impairment: “imperceptible” assigned integer 5, “perceptible but not annoying” 4, “slightly annoying” 3, “annoying” 2, “very annoying” 1. The integer value is regarded as a sub-

jective score. A MOS value for a stimulus is the averaged score over all assessors.

In audio-video streaming in ad hoc networks, its quality can fluctuate quite widely. In the rating-scale method, each assessor is supposed to give a subjective score for a stimulus. However, it is difficult for the assessors to give the average of the perceived quality at the end of each stimulus because of the temporal fluctuation. Thus, we asked the assessors to give a score for each fragment of a stimulus as stated below; we then averaged the scores to obtain a MOS value after the experiment.

The MOS values are calculated as follows. While a stimulus is presented to each assessor, he/she classifies every instantaneous quality into one of the five categories of impairment according to his/her subjective assessment. The assessor inputs a score by the laptop PC's keyboard whenever his/her classification changes from a score that had been input immediately before. The input score is kept until the assessor changes it to another; it is sampled every 0.1 second. We then regard the average of the sampled scores of the stimulus as the subjective score of the assessor. The MOS value of the stimulus is obtained by taking an average of the scores over all the assessors.

The assessors assessed stimuli for the three routing schemes. For each routing scheme, there were four stimuli, which correspond to the inter-node distances of 20 m, 25 m, 30 m, and 35 m. It took about 40 minutes for an assessor to finish all assessment which includes the presentation of the original audio-video stream, a stimulus for practice, and $3 \times 4 = 12$ stimuli.

4. Assessment Results

In this section, we first show the application-level QoS and the user-level QoS of the three schemes. We then present the statistics of the behavior of the routing schemes. Finally, we investigate the application-level QoS in random topology networks.

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 interference traffic. We also show 95% confidence intervals of the measured values 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.

4.1 Application-Level QoS of Audio and Video Streams

In this section, we also evaluate the application-level QoS with original AODV and that with original OLSR.

Figure 3 depicts the coefficient of variation of output interval for audio as a function of the inter-node distance. Figure 4 plots the coefficient for video versus the inter-node distance.

We see in Fig. 3 that when the inter-node distance is shorter than 30 m, the coefficient of variation of output interval for LQHR is the smallest among the three link quality-

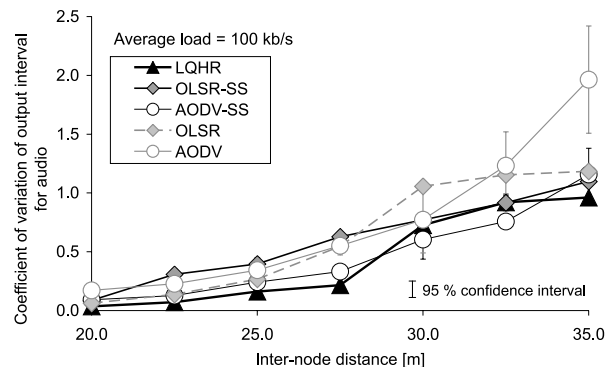


Fig. 3 Coefficient of variation of output interval for audio.

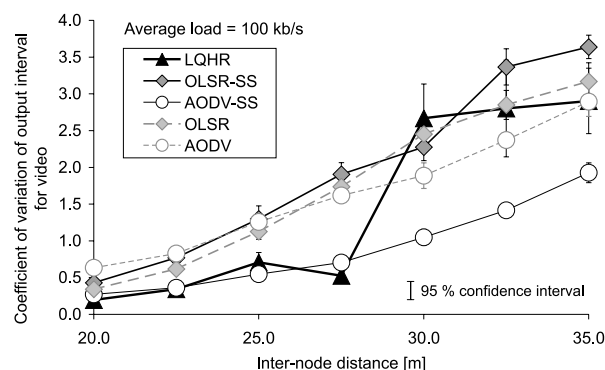


Fig. 4 Coefficient of variation of output interval for video.

based schemes. In Fig. 4, we also find that for most of the inter-node distances smaller than 30 m, the coefficient for LQHR is the smallest. This is because LQHR can select appropriate routes owing to the combination of the two routing strategies: periodical acquisition of link quality and on-demand route discovery.

On the other hand, we notice in Figs. 3 and 4 that when the inter-node distance is equal to or longer than 30 m, the coefficient of variation with LQHR suddenly becomes large. The reason is as follows. The implementation of LQHR in this paper is an enhanced version for networks with many nodes [27]. In the enhancement, we optimize the algorithm for comparatively dense networks by means of a heuristic approach. The enhanced algorithm restricts the selection of the highest quality links for the route; those links often have very short distances to the receivers. If those links are used, there are huge number of hops, or the RReq packets cannot reach the destination. The mechanism can avoid the situations. However, when the network becomes sparse, the limitation cannot work well. This is because links with excessive quality do not exist in the sparse networks, and then the limitation may remove adequate links from the candidates. Therefore, the performance of LQHR suddenly decreases in those networks.

In Figs. 3 and 4, we also find that for almost all the inter-node distances, OLSR-SS has approximately the same or larger coefficients than the other link quality-

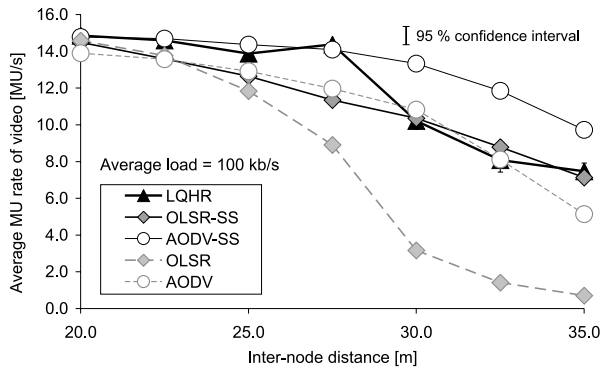


Fig. 5 Average MU rate of video.

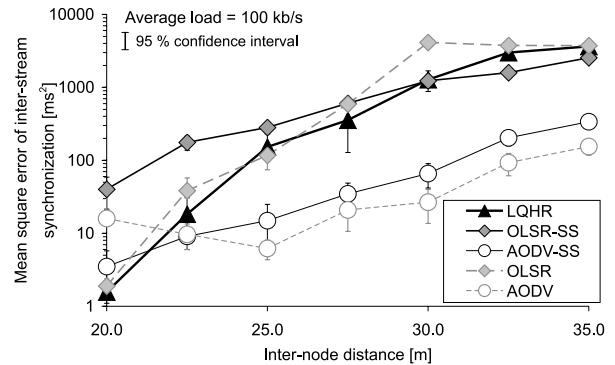


Fig. 7 Mean square error of inter-stream synchronization.

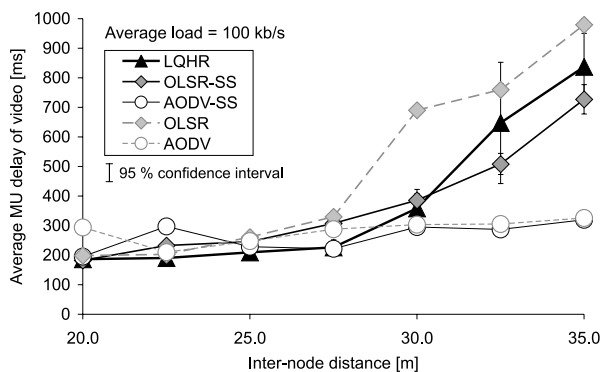


Fig. 6 Average MU delay of video.

schemes. OLSR-SS renews its routing information periodically, and the periodical update is done on a distributed basis. Thus, the output timing of the media streams is disturbed owing to mismatch of the routing information.

In Fig. 4, we notice that when the inter-node distance is equal to 30 m or longer, the coefficient for video with AODV-SS is the smallest among the three link quality-based schemes. This is due to the higher average MU rate described below.

Figure 5 displays the average MU rate of video versus the inter-node distance. In this figure, we see that AODV-SS has approximately the same or higher MU rate of video than the other schemes. This is because AODV-SS can avoid congestion by dynamical update of the route. However, in AODV-SS, the source starts to find the route when it initiates the generation of audio and video streams; although in the simulation, for a fair comparison, the source starts to find the route one second before. Furthermore, AODV-SS employs a mechanism of incremental route search [25]. Therefore, at the start of audio-video streaming, AODV-SS loses some packets. On the other hand, the hybrid approach (namely, LQHR) can transmit packets by using a proactively selected route even if the route is not found immediately.

Figure 6 displays the average MU delay of video. Since the relationship of the average MU delay of audio between the schemes is similar to that in Fig. 6, we do not show it here.

In Fig. 6, we find that for the inter-node distances equal to 30 m or longer, the MU delay with AODV-SS is the smallest among the three link quality-based schemes. This is because AODV-SS immediately stops using routes with unstable links because of its reactive property. AODV-SS renews the route whenever it notices route disconnection, which is detected as the excess of the MAC retry limit. In the unstable route, congestion is caused by the retransmission delay at the MAC layer; the node cannot send further packets and then the queue becomes full. The scheme can avoid congestion because it can stop to use the unstable route immediately.

On the other hand, the proactive approach and the hybrid one, namely, OLSR-SS and LQHR, continue to use the selected route during the routing update interval, which is set to five seconds in the simulation, and then congestion occurs.

In Figs. 3 through 6, we can observe that the application-level QoS with the threshold for received signal strength (namely, AODV-SS and OLSR-SS) is better than that without the threshold (namely, original AODV and original OLSR, respectively). Therefore, the link quality-based routing protocols are effective in the improvement of the application-level QoS of the audio-video streaming.

Figure 7 plots the mean square error of inter-stream synchronization versus the inter-node distance. In this figure, we can confirm that in the whole range of the inter-node distance considered here, the mean square errors of inter-stream synchronization for all the schemes are smaller than $6400 \text{ ms}^2 (= 80^2 \text{ ms}^2)$, which is a threshold of high inter-stream synchronization quality reported by Steinmetz [37].

4.2 Effect of Parameters for Routing Schemes on Application-Level QoS

In this section, we assess the effect of the parameters for the three link quality-based schemes on the application-level QoS.

Figure 8 depicts the coefficient of variation of output interval for video when the threshold of the received signal strength T_h is changed from -62.7 dBm to -60.0 dBm ; that is, the routing protocols are required to employ higher qual-

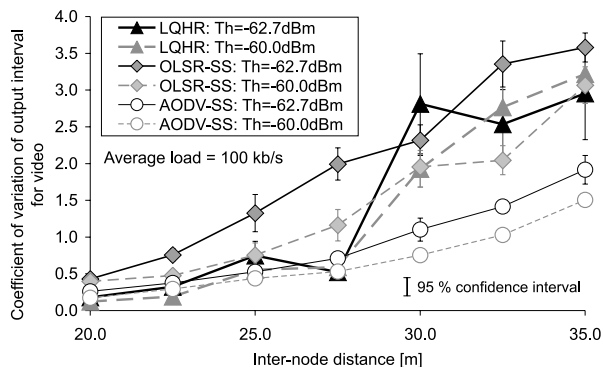


Fig. 8 Coefficient of variation of output interval for video with another value of T_h .

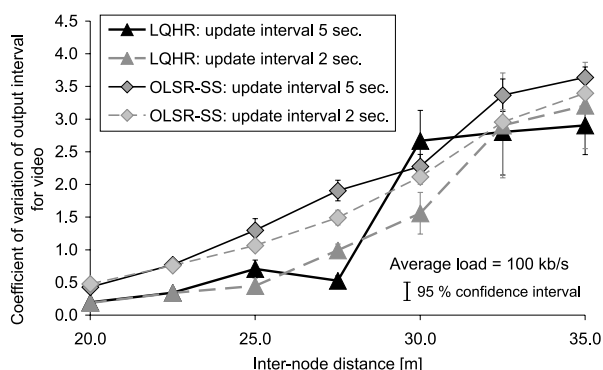


Fig. 9 Coefficient of variation of output interval for video with another value of route update interval.

ity links. We find in this figure that AODV-SS and OLSR-SS can decrease the coefficient for video with changing T_h from -62.7 dBm to -60.0 dBm. This is because the protocols use higher quality links when we employ severe values of T_h .

On the other hand, we notice in Fig. 8 that the relationships among the three schemes with $T_h = -60.0$ dBm are approximately the same as those with $T_h = -62.7$ dBm.

Figure 9 depicts the coefficient of variation of output interval for video when the route update interval of the proactive and hybrid routing protocols is changed from five seconds to two seconds. We find in this figure that as the route update interval decreases, the coefficient for video with OLSR-SS decreases. This is because more appropriate routes can be used when the small route update interval is employed. We also see in this figure that LQHR with short route update interval can achieve smaller coefficient than that with large route update interval in some inter-node distances. In addition, the relationships between the two protocols are almost the same in the two route update intervals.

4.3 User-Level QoS of the Audio-Video Stream

From the results of the application-level QoS assessment, we can observe that AODV-SS and OLSR-SS is superior to the original schemes. Therefore, we perform the user-level

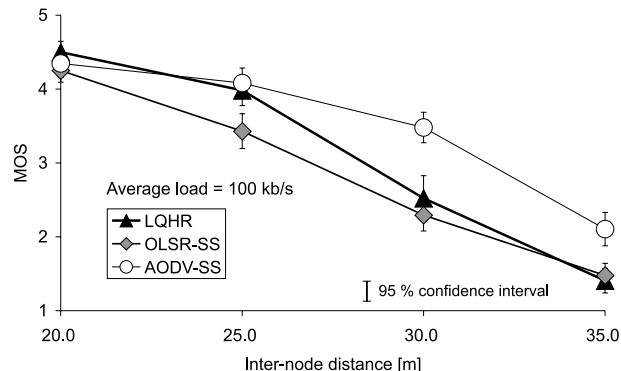


Fig. 10 MOS value for the audio-video stream.

Table 2 Average number of disconnections of audio-video route in AODV-SS.

inter-node distance [m]	number of disconnections
20	10.20
22.5	15.67
25	30.20
27.5	42.63
30	72.27
32.5	133.13
35	211.40

QoS assessment of audio-video streaming for the three link quality-based routing schemes.

Figure 10 shows the MOS value for the audio-video stream. In this figure, we find that for inter-node distances not shorter than 25 m, the MOS value of AODV-SS is the largest among the three schemes. In addition, we see that LQHR has the best value when the inter-node distance equals 20 m. In short inter-node distance networks, LQHR can achieve high quality of audio output as shown in Fig. 3. On the other hand, AODV-SS can get higher MU rates of video than the other schemes especially in long inter-node distance networks as shown in Fig. 5. Thus, we have learned that the result of the user-level QoS assessment is consistent with that of the application-level QoS assessment. As a next step of this study, we need to clarify the relationships between the two QoS by means of *QoS mapping*.

4.4 Statistics of the Behavior of Routing Schemes

Table 2 shows the average number of disconnections of the audio-video route in AODV-SS. The disconnected route must be renewed, and then the number of route disconnections means the frequency of route updates.

When the route is updated every five seconds in OLSR-SS and LQHR, the number of route updates during the audio-video transmission in a simulation run is $120/5 = 24$. We find in Table 2 that the frequency of route updates in AODV-SS is more than OLSR-SS or LQHR when the inter-node distance is equal to or longer than 25 m.

Figure 11 depicts the percentage of the number of hops in the audio-video route. The percentage of selected transmission speed for the audio-video stream is shown in

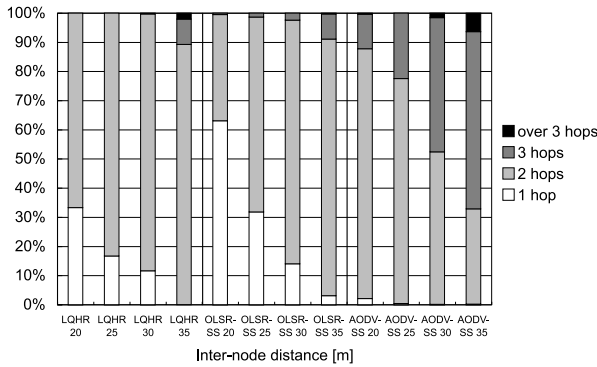


Fig. 11 The percentage of the number of hops in audio-video route.

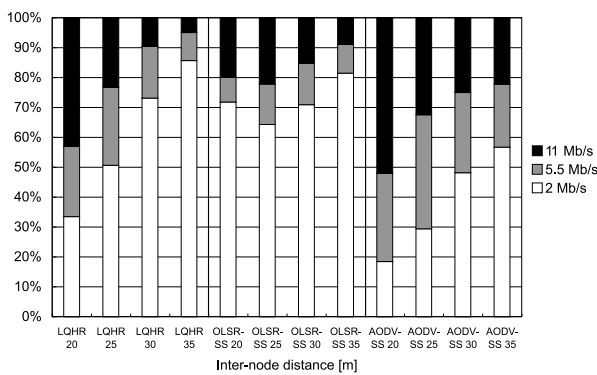


Fig. 12 The percentage of selected transmission speed for audio-video stream.

Fig. 12.

We notice in Fig. 11 that AODV-SS selects more hops than LQHR and OLSR-SS. This is because AODV-SS dynamically discovers routes in a purely on-demand way.

In Figs. 11 and 12, we can observe that the selected transmission speed is closely related to the number of hops; AODV-SS selects higher transmission speeds than the other schemes. In addition, LQHR may not select routes with higher speed links compared to AODV-SS. This is because LQHR is not optimized well; as discussed earlier, the protocol may not select appropriate links especially in the sparse networks. We need to modify the mechanism more efficiently.

Figure 13 shows the number of routing packets during a simulation run. We can observe in this figure that for the inter-node distances equal to 30 m or shorter, the number of routing packets with LQHR is the largest among the three schemes. This is because LQHR adds a mechanism of on-demand route searching to the link-state routing mechanism in the original OLSR.

In Fig. 13, we also find that when the inter-node distance is equal to or longer than 32.5 m, the number of routing packets in AODV-SS is the largest. This is because it is hard to discover stable routes in AODV-SS when the distance between the nodes becomes longer. On the other hand, the routing overhead of OLSR-SS is hardly affected by the inter-node distance owing to the periodical transmission of

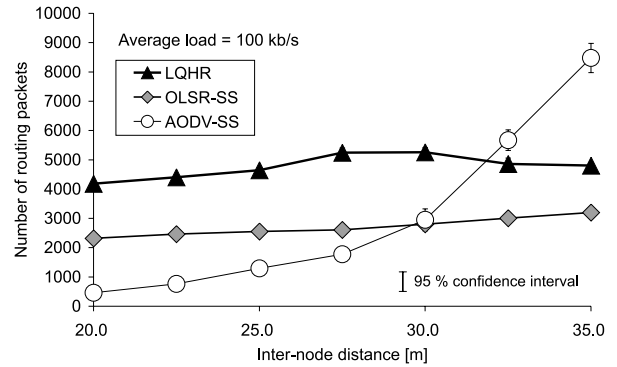


Fig. 13 Number of control packets for routing.

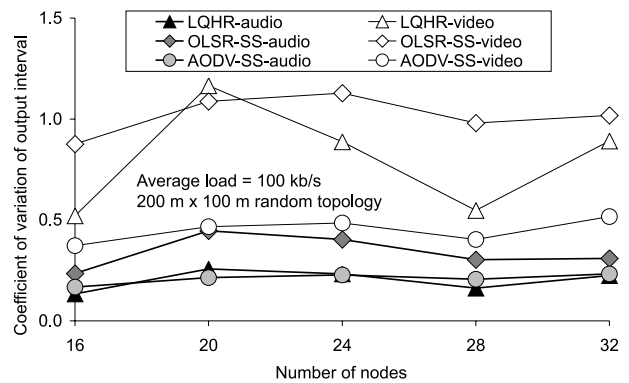


Fig. 14 Coefficient of variation of output interval in random topology networks.

the control packets.

From the above observation, we find that AODV-SS basically achieves high performance particularly when the inter-node distance is long. On the other hand, LQHR can achieve high QoS in networks with short inter-node distances, although it has a room for improvement. OLSR-SS has smaller routing overhead than the other schemes in networks with long inter-node distances.

4.5 Application-Level QoS in Random Topology Networks

In the simulation, we also evaluated the application-level QoS of the audio-video stream in random topology networks.

We randomly distributed 16, 20, 24, 28, or 32 nodes in a 200 m × 100 m area. The nodes are fixed; that is, they do not move. We randomly picked up four nodes from the nodes as MS, MR, BTS and BTR nodes, respectively. Other conditions for the simulation are the same as those described in Sect. 3.

Figure 14 depicts the coefficient of variation of output interval in the random topology networks. 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 interference traffic and for generating topology.

In Fig. 14, we do not depict the 95% confidence intervals. This is because the coefficient of variation largely differs from seed to seed. It is difficult to assess the application-level QoS in random topology networks because of their randomness. An appropriate method for QoS assessment in those networks is one of our future studies.

We find in Fig. 14 that the coefficient of variation for audio with AODV-SS is approximately the same as that with LQHR and is smaller than that with OLSR-SS. In addition, we see that the coefficient of variation for video with AODV-SS is the smallest, and that with LQHR is the second smallest for most of the number of nodes considered here.

We also notice in Fig. 14 that when the number of nodes is 20, the coefficient of variation of output interval for video with LQHR is large. The reason is that long pauses of video occurred in a few simulation runs owing to the loss of I-pictures. We employ MPEG1 as the video stream, and the picture pattern is IPPPP. If MR cannot receive an I-picture, the succeeding P-pictures are discarded until the next I-picture appears. Therefore, the temporal quality degrades quite largely, and then the coefficient becomes large.

The random topology networks with small number of nodes have similar characteristics to the mesh topology networks with long inter-node distances. That is, the networks are sparse, and then high quality links are few.

From the above discussion, even in the random topology networks, AODV-SS and LQHR can improve application-level QoS of the audio-video stream.

5. Conclusions

In this paper, we assessed the application-level QoS and the user-level QoS of audio-video streaming in a cross-layer designed wireless ad hoc network with media synchronization control at the application-level and link quality-based routing protocols at the network-level. As a result, we found that AODV-SS, which is a reactive scheme, can achieve better application-level QoS and user-level QoS than the other schemes in networks with long inter-node distances. However, it takes long time to search route when the source has no route.

When the inter-node distance is short, LQHR can achieve high QoS because of the combination of the proactive link quality acquisition and the reactive route discovery. However, LQHR is not optimized well and has a room for improvement. Thus, as a next step of our research, the modification of the LQHR protocol is necessary.

While this paper does not assume QoS control mechanism in the MAC layer, IEEE 802.11e [38] has been expected for QoS provision. Romdhani and Bonnet present a cross-layer routing protocol which is based on the cooperation between the AODV routing protocol and the IEEE 802.11e EDCA MAC protocol in [39]. We have a plan to investigate the efficiency of the IEEE 802.11e in the cross-layer design architecture for audio-video streaming.

In addition, we must assess the QoS of the three schemes in the practical propagation model of the wireless

channel.

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works. Dr. Nunome is a member of the IEEE.

Toshiro Nunome received the B.S., M.S. and Ph.D. degrees from Nagoya Institute of Technology, Nagoya, Japan, in 1998, 2000 and 2006, respectively. From 2000 to 2001, he was with Pioneer Corp. In 2002, he joined Nagoya Institute of Technology as a Research Associate. Now, he is an Assistant Professor in the Department of Computer Science and Engineering, Graduate School of Engineering. His research interests include media synchronization, multi-cast communications and wireless ad hoc networks.



ing Scholar in the Department of Electrical Engineering at the University of California, Los Angeles. His current research interests include wireless networks, multimedia QoS and multimedia communication protocols. He is the author of a book entitled *Performance Analysis of Multiple Access Protocols* (MIT Press, Cambridge, MA, 1986). Dr. Tasaka is a member of the IEEE, ACM and Information Processing Society of Japan.

Shuji Tasaka received the B.S. degree in electrical engineering from Nagoya Institute of Technology, Nagoya, Japan, in 1971, and the M.S. and Ph.D. degrees in electronic engineering from the University of Tokyo, Tokyo, Japan, in 1973 and 1976, respectively. Since April 1976, he has been with Nagoya Institute of Technology, where he is now a Professor in the Department of Computer Science and Engineering, Graduate School of Engineering. In the 1984–1985 academic year, he was a Visiting



wireless communication.

Ken Nakaoka received the B.E. and M.E. degrees in electronic engineering from Nagoya University in 1993 and 1995, respectively. He joined Sanyo Electric Co., Ltd. in 1995, and he has been engaged in the development of mobile multimedia communication systems. From 1997 to 2000, he has been with ATR Adaptive Communications Research Laboratories. He is presently a Senior Staff in Digital Technology Research Center. His current research interests include Wireless Ad Hoc Networks and ITS