

QoE Enhancement of Audio-Video IP Transmission in Cross-Layer Designed Ad Hoc Networks

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1. Introduction

The *QoE* (*Quality of Experience*), which is perceptual quality for the users, is the most important *QoS* (*Quality of Service*) among those at all levels since the users are the ultimate recipients of the services. Even in *mobile ad hoc networks* (*MANET*), provision of high *QoE* is one of the most important issues.

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 cross-layer design architecture (Srivastava & Motani, 2005) 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-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. The number of hops maintained by the routing protocol is used for selecting the video coding rate to the network capacity (Gharavi & Ban, 2004), (Zhao et al., 2006). 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. (2006) have proposed an error recovery mechanism for real-time video streaming that combines FEC and multipath retransmission. This scheme determines strength of the error correction code and a quantization parameter for video encoding according to the number of hops. Frias et al. (2005) 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. Nunome & Tasaka (2005) have proposed the *MultiPath 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; it is application-level *QoS* control.

While the above approaches refer to cross-layering between the network and application layers, Setton et al. (2005) have explored 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 QoE of transmitted multimedia streams. Furthermore, these studies except for (Nunome & Tasaka, 2005) consider video only and do not assess its temporal quality.

The routing protocol is an essential component in ad hoc networks. 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* (Zhang & Mouftah, 2005) and *multirate aware routing* (Lin et al., 2003), (Seok et al., 2003). 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 chapter is to achieve high QoE 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 chapter, we assess application-level QoS and QoE 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)* (Itaya et al., 2005), *AODV-SS* (Budke et al., 2006), and *LQHR (Link Quality-Based Hybrid Routing)* (Nakaoka et al., 2006). OLSR-SS is a modified version of OLSR (Clausen & Jacquet, 2003), which is a proactive routing protocol. AODV-SS is a reactive protocol based on AODV (Perkins et al., 2003). 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 quality of the audio-video stream can fluctuate largely in ad hoc networks, and then it is difficult to assess the QoE. That is, the assessment method is one of the important research issues. We employ a continuous time assessment method of QoE in audio-video transmission proposed in (Ito et al., 2005); it utilizes the *method of successive categories* (Tasaka & Ito, 2003), which is a psychometric method, continuously in time.

The rest of this chapter is organized as follows. Section 2 explains link quality-based routing protocols for ad hoc networks. We introduce the continuous time assessment method of QoE in Section 3. Section 4 illustrates a methodology for the QoS/QoE assessment, including the network configuration, simulation method, QoS parameters, and QoE assessment method. The QoS assessment results are presented and discussed in Section 5. Section 6 discusses the result of QoE assessment.

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.

Itaya et al. (2005) have proposed 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 (Itaya et al., 2005).

As the proactive routing protocol for the comparison in this chapter, we employ the scheme proposed in (Itaya et al., 2005) with a little modification. The threshold for signal strength is kept constant for simplicity; in this chapter, 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 (Itaya et al., 2005).

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, Fan (2004) 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, Budke et al. (2006) 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 (Itaya et al., 2005); thus, we refer to the scheme as *AODV-SS*.

In this chapter, 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. (2006) propose LQHR. 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. We revise the LQHR algorithm in order to overcome difficulties related to networks with many route selections.

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 (Nakaoka et al., 2006). 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.

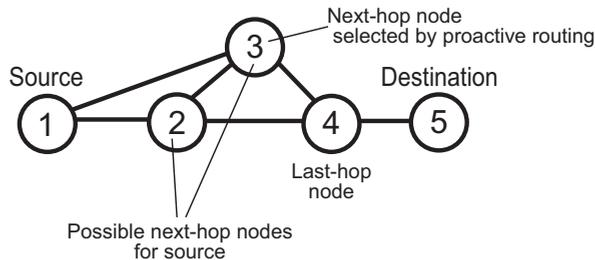


Fig. 1. Example of route discovery in LQHR.

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 destination. For example, in Fig. 1, we assume that node 1 is the source node and that node 5 is the destination node. Then, nodes 2 and 3 are the possible next-hop nodes for node 1.

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. In Fig. 1, node 4 is the last-hop node to node 5.

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.

In this chapter, we impose two restrictions on the algorithm of LQHR in order to overcome problems related to networks with many route candidates; many RQReq and RQRsp packets are generated, and then the effectiveness of the route discovery mechanism may degrade. One restriction is for the possible next-hop nodes, and the other is for the last-hop nodes.

At first, the revised algorithm restricts the possible next-hop nodes. The original LQHR algorithm sends RQReq packets to all the possible next-hop nodes. However, if there are many possible next-hop nodes, this is not a good strategy because the node will generate many RQReq packets, which cause congestion. Thus, the revised algorithm sends RQReq packets to only r_1 nodes which has higher link quality than other nodes. In this chapter, we set the value of r_1 to 5.

In addition, we also employ the following condition for the possible next-hop nodes. When link quality between two nodes is very high at each node, the two nodes may be

geometrically close to each other. If the routing algorithm selects such links, the route will have a large number of hops. Thus, a node does not send RQReq packets to a possible next-hop node in which the link between the two nodes is one of the best r_2 links. In this chapter, we set the value of r_2 to 3.

Next, the revised algorithm also restricts the last-hop nodes. In some topology, there are a large number of last-hop nodes. However, it may not be true that all the candidates of last-hop nodes have good quality links to the destination node. Thus, as the last-hop node, the algorithm permits only nodes with the link to the destination whose quality is larger than the threshold T_h .

3. Continuous time assessment of QoE

In this chapter, we employ the method of continuous time QoE assessment in (Ito et al., 2005). This section describes the method, which utilizes the method of successive categories continuously in time.

3.1 Method of successive categories

For a start, we introduce four types of measurement scales. With the psychometric methods, the human subjectivity can be represented by a measurement scale. We can define four types of the measurement scales according to the mathematical operations that can be performed legitimately on the numbers obtained by the measurement; from lower to higher levels, we have *nominal*, *ordinal*, *interval*, and *ratio* scales (Guilford, 1954). Since almost all the statistical procedures can be applied to the interval scale and the ratio scale, it is desirable to represent the QoE by an interval scale or a ratio scale. With the psychometric methods used in (Tasaka & Ito, 2003), we can represent QoE by an interval scale.

In the method of successive categories, a subjective score is measured by the *rating scale method* (Guilford, 1954), in which subjects classify each stimulus into one of a certain number of categories. Here, a stimulus means an object for evaluation, such as audio and video. Each category has a predefined number. For example, "excellent" is assigned 5, "good" 4, "fair" 3, "poor" 2, and "bad" 1. However, in the strict sense, we cannot use the assigned number for assessing the QoE since the assigned number is an ordinal scale.

In order to obtain an interval scale as the QoE metric, we first measure the frequency of each category with which the stimulus was placed in the category by the rating-scale method. With the *law of categorical judgment* (Tasaka & Ito, 2003), we can translate the frequency obtained by the rating-scale method into an interval scale. We can apply almost all the operations to the scale.

3.2 The Law of Categorical Judgment

The law of categorical judgment makes the following assumptions. Let the number of the categories be $m + 1$. When stimulus j ($j = 1, \dots, n$) is presented to an assessor, a psychological value designated by s_j occurs on an interval scale in him/her. For the $m + 1$ categories, their *boundaries* have values on the interval scale. We denote the upper boundary of category g ($g = 1, \dots, m + 1$) by c_g and define $c_0 \triangleq -\infty$ and $c_{m+1} \triangleq +\infty$. The assessor sorts n stimuli into the $m + 1$ categories ($n > m + 1$) by comparing s_j with c_g . If $c_{g-1} < s_j \leq c_g$, then stimulus j is classified into category g . The categories can be arranged in a rank order, in the sense that each stimulus in category g is judged to have a psychological value which is "less than" the one

for any stimulus in category $g + 1$. This statement holds for all values of g from 1 to m . The variable c_g is normally distributed with mean t_g and standard deviation d_g . Also, the variable s_j is normally distributed with mean R_j and standard deviation σ_j . Then, we can consider R_j as an interval scale.

Since the law of categorical judgment is a suite of assumptions, we must test goodness of fit between the obtained interval scale and the measurement result. Mosteller proposed a method of testing the goodness of fit for a scale calculated with Thurstone's law (Mosteller, 1951). The method can be applied to a scale obtained by the law of categorical judgment. In this chapter, we use Mosteller's method to test the goodness of fit.

3.3 Continuous time QoE assessment with the method of successive categories

We utilize the method of successive categories continuously in time. The audio-video stream for evaluation is partitioned into many *fragments* each with time length Δ . For example, a stream with total length T is divided into T/Δ or $T/(\Delta + 1)$ fragments. We regard each fragment as a stimulus and utilize the method of successive categories for all stimuli (fragments). That is, assessors classify the current fragment into one of the categories every Δ . Then, we apply the law of categorical judgment to the result for all fragments.

Since the assessor only has to judge which one of the categories is the most appropriate for the stimulus every Δ , the method imposes little burden on the assessor. Moreover, by setting the number of categories to 3 or 5, the assessors can continuously enter their judgment in an input device by their fingers without directing their attention to the device. In addition to this, by utilizing the law of categorical judgment, we can obtain values of QoE metric in the form of the interval scale.

4. Methodology of QoS/QoE assessment

We assess the application-level QoS and the QoE 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*).

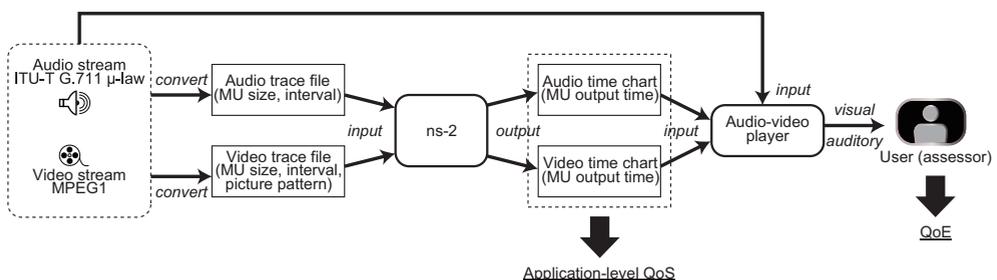


Fig. 2. Schematic diagram of QoS/QoE assessment.

Figure 2 shows the schematic diagram of the QoS/QoE 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 QoE assessment, the audio-video player plays the practical audio-video stream with the output timing obtained from the time charts.

4.1 Network configuration

In this chapter, we consider a simple mesh topology network to assess 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. 3. Each node has an omnidirectional antenna. We employ the shadowing model (Rappaport, 1996) as the propagation model in the simulation. In the model, received signal strength at the receiver is determined by the following equation:

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

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

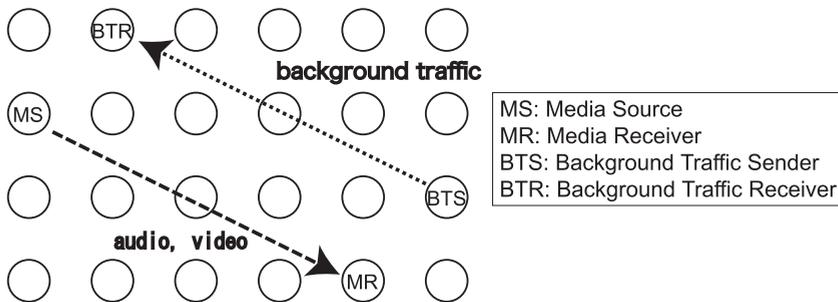


Fig. 3. Network configuration.

We formulate a detailed simulation model which is based on the *distributed coordination function (DCF)* of the IEEE 802.11b. The transmission speed is automatically changed from 2 Mb/s to 11 Mb/s by means of the rate adaptation mechanism. In this chapter, we employ

ARF (*Automatic Rate Fallback*) (Kameran & Monteban, 1997). 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 (Xu et al., 2002), (Ray et al., 2003).

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 *threshold* program, which is included in ns-2.

4.2 Method of simulation

In Fig. 3, 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 have been prepared by encoding a part of Japanese news program. Table 1 shows the specifications of the audio and video.

In the simulation, 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 chapter, 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 chapter, 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 (Tasaka et al., 2000). We also consider that the buffering delay is the same as the frame interval, 66.7 ms.

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	

Table 1. Specifications of the audio and video.

We exert media synchronization control with the enhanced VTR algorithm (Tasaka et al., 2000). The parameter values in the enhanced VTR algorithm are set to the same as those in (Nunome & Tasaka, 2004). That is, we set the *initial buffering time* J_{\max} (Tasaka et al., 2000) and the *maximum allowable delay* Δ_{al} (Tasaka et al., 2000) 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 chapter, 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 chapter, because we focus on basic characteristics of the three schemes. However, for example, a method for optimizing the threshold value discussed in (Itaya et al., 2005) 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.

4.3 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.

4.4 QoE assessment

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

We made *stimuli* for subjective assessment 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.

A subjective score is measured by the rating-scale method. 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 subjective score.

In audio-video streaming in ad hoc networks, its quality can fluctuate quite largely. 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.

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 one second. The sampled value is assumed as a subjective score for the fragment for the one second.

In this chapter, we utilize the method of successive categories in order to obtain an interval scale as the QoE metric. We first measure the frequency of each category with which the fragment of the stimulus was placed in the category by the rating-scale method. With the law of categorical judgment, we can translate the frequency obtained by the rating-scale method into an interval scale. We then perform Mosteller's test, which tests the goodness of fit between the obtained interval scale and the measurement result. The interval scale of which we have confirmed the goodness of fit is referred to as the *psychological scale*.

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.

5. Assessment results of application-level and lower-level QoS

In this section, we first show the application-level QoS of the three schemes. We then present the statistics of the behavior of the routing schemes.

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.

5.1 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 4 depicts the coefficient of variation of output interval for audio as a function of the inter-node distance. Figure 5 plots the coefficient for video versus the inter-node distance.

We see in Fig. 4 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-based schemes. In Fig. 5, 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. 4 and 5 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 chapter is an enhanced version for networks with many nodes. 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

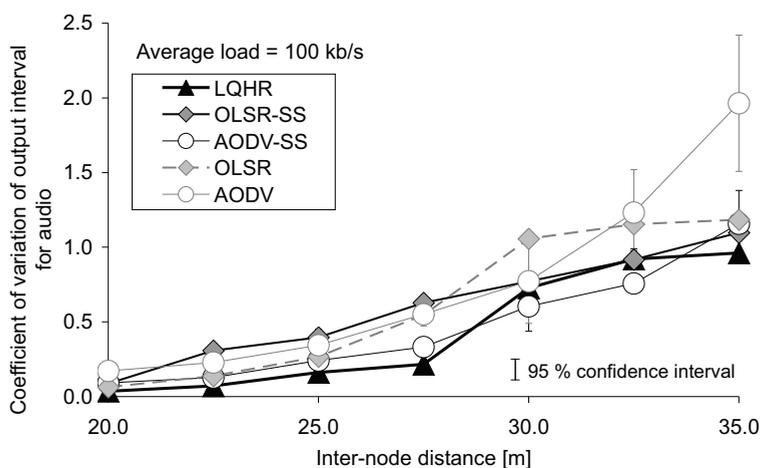


Fig. 4. Coefficient of variation of output interval for audio.

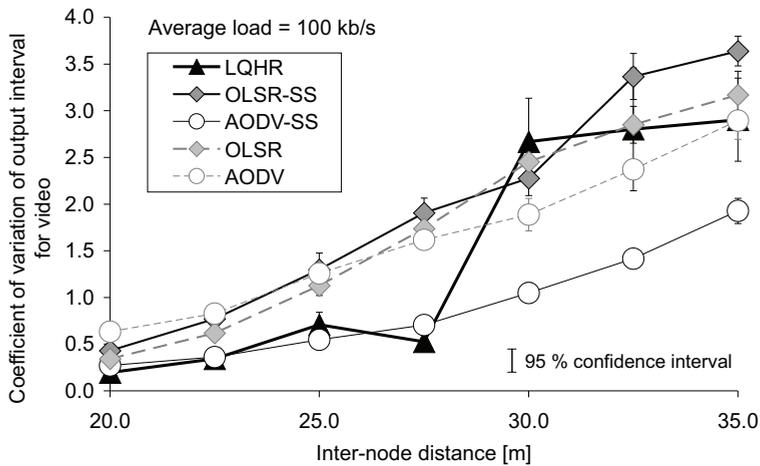


Fig. 5. Coefficient of variation of output interval for video.

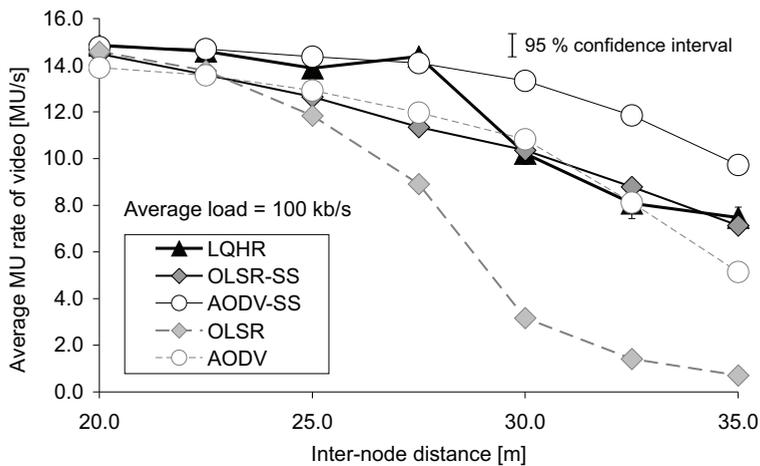


Fig. 6. Average MU rate of video.

RQReq 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. 4 and 5, 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-based 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.

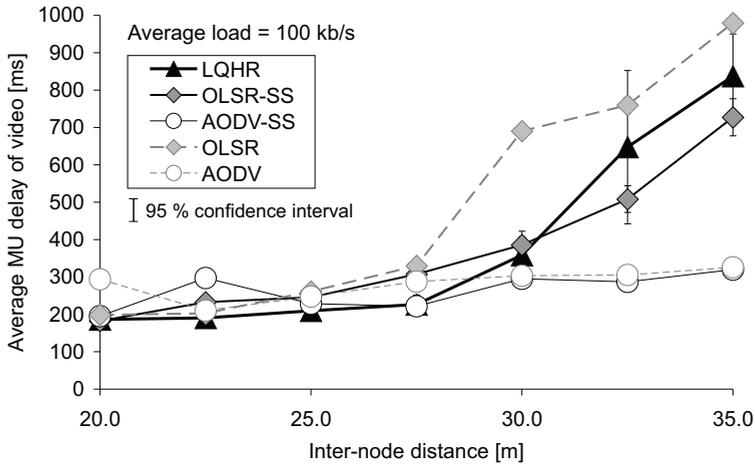


Fig. 7. Average MU delay of video.

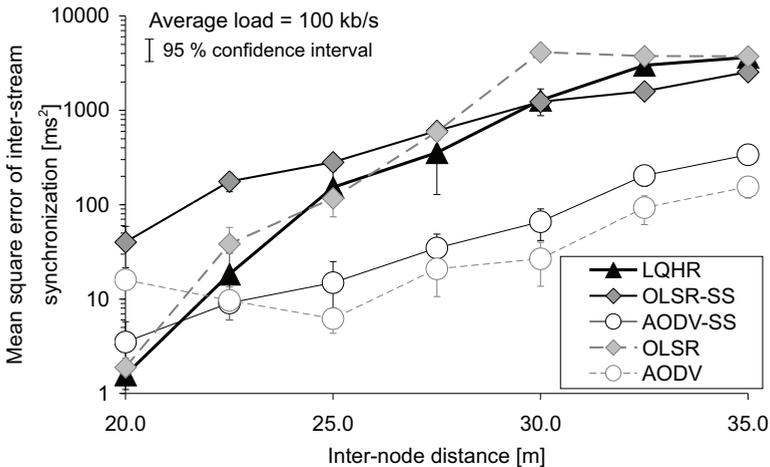


Fig. 8. Mean square error of inter-stream synchronization.

In Fig. 5, 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 6 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 (Perkins et al., 2003). 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 7 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. 7, we do not show it here.

In Fig. 7, 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. 4 through 7, 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 8 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 ms^2 ($= 80^2 \text{ ms}^2$), which is a threshold of high inter-stream synchronization quality reported by Steinmetz (Steinmetz, 1996).

5.2 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 Tab. 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.

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

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

Figure 9 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. 10.

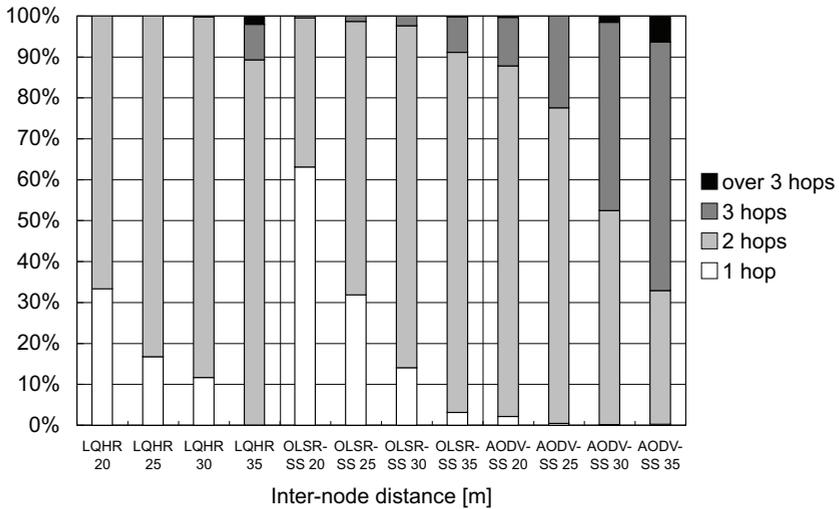


Fig. 9. The percentage of the number of hops in audio-video route.

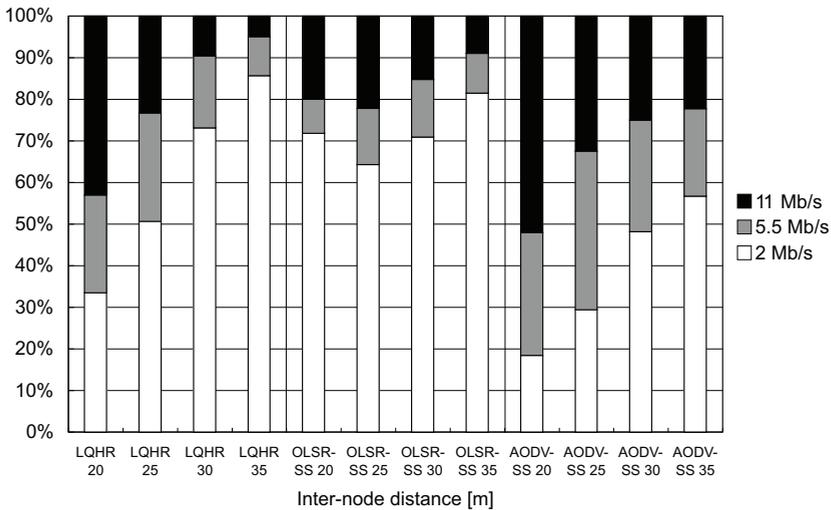


Fig. 10. The percentage of selected transmission speed for audio-video stream.

We notice in Fig. 9 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. 9 and 10, 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.

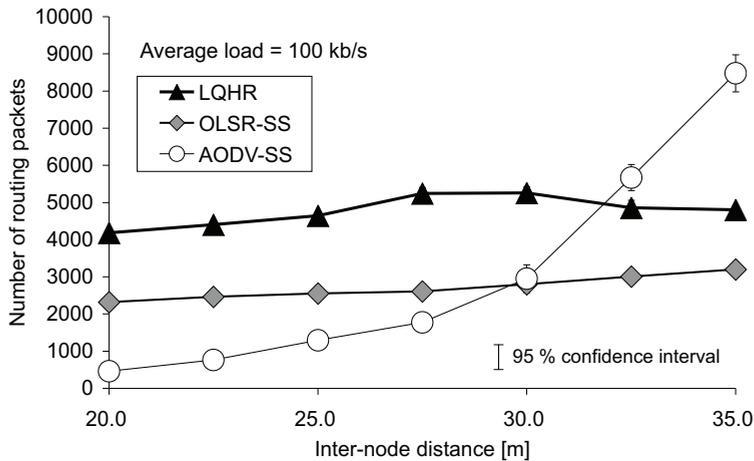


Fig. 11. Number of control packets for routing.

Figure 11 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. 11, 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 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.

6. QoE assessment result

In this section, we show the result of QoE assessment of the three schemes: AODV-SS, OLSR-SS, and LQHR.

6.1 Calculation for all the inter-node distances

We first calculate the psychological scale for all the inter-node distances employed in the assessment. We processed the result in the period of time 30 through 120. As a result of the Mosteller's test, we found that the null hypothesis that obtained interval scale fits the observed data can be rejected at significance level 0.01. This is because the obtained scale does not fit well for all the schemes for the inter-node distance 20 m, OLSR-SS for the inter-node distance 25 m, and AODV-SS for the inter-node distance 30 m.

We checked the fragments which give large errors of Mosteller's test. As a result, by removing about 27 % of the fragments, we saw that the hypothesis cannot be rejected. Figure 12 depicts the psychological scale versus the elapsed time for the inter-node distance 20 m.

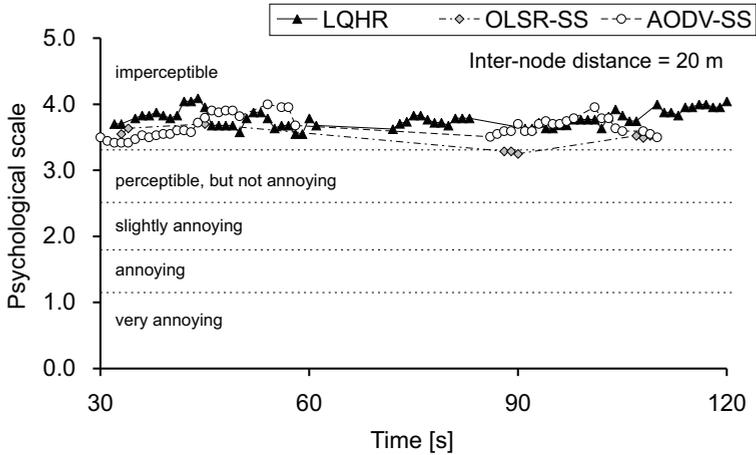


Fig. 12. Psychological scale for inter-node distance 20 m.

Note that we can select any origin of an interval scale. In this chapter, for convenience, we regard the minimum value of the psychological scale for the inter-node distance 35 m as the origin for all the inter-node distances.

Horizontal dotted lines in Fig. 12 show boundaries between the categories. Note that the lower bound of category 1 is $-\infty$, and the upper bound of category 5 is ∞ .

In Fig. 12, the removed fragments are not shown; there are a lot of removed fragments especially for OLSR-SS.

6.2 Calculation for each inter-node distance

Because the observed data can be categorized by the inter-node distances, we individually calculate the psychological scale for each inter-node distance.

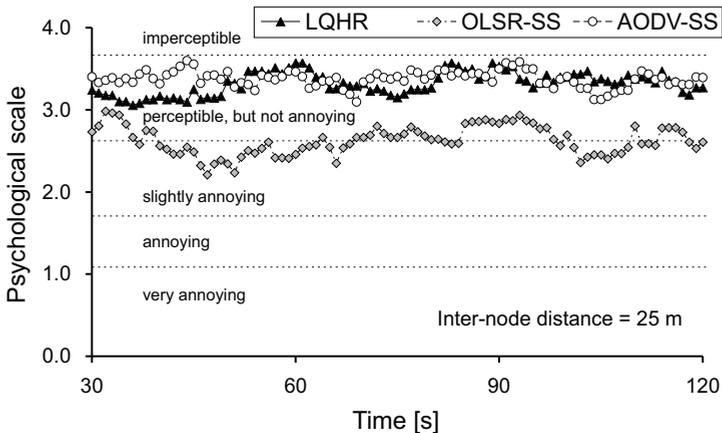


Fig. 13. Psychological scale for inter-node distance 25 m.

For the inter-node distance 20 m, we could not obtain the psychological scale. This is because the output quality of audio-video does not largely degrade for all the schemes, and

then no assessor classified the stimuli into category 1, “very annoying”. It can be observed in Fig. 12 that all the schemes have high output quality; almost all the fragments are categorized as category 5, “imperceptible”.

On the other hand, for the inter-node distance 25 m, as a result of the Mosteller’s test, we found that the null hypothesis cannot be rejected at significance level 0.01. Therefore, we consider that the obtained interval scale for this inter-node distance is appropriate for the QoE metric. Figure 13 plots the psychological scale versus the elapsed time for the inter-node distance 25 m.

For the inter-node distance 30 m, by removing about 8 % of the fragments, we found that the hypothesis cannot be rejected. Figure 14 plots the psychological scale. In the case of this inter-node distance, the quality severely changes from seed to seed, i.e, from assessor to assessor. Thus, it is more difficult for the case than the others to fit the interval scale to the obtained score.

For the inter-node distance 35 m, we saw that the hypothesis cannot be rejected by removing about 5 % of the fragments. Figure 15 indicates the psychological scale.

Comparing Fig. 15 to Figs. 13 and 14, we find that the ratio of the width of category 4, “perceptible, but not annoying” to that of category 3, “slightly annoying” for the inter-node distance 35 m is smaller than that for the inter-node distance 25 m or 30 m. This is because there are few fragments which have high quality for the inter-node distance 35 m, and then assessors did not classify the stimuli into high categories.

We notice in Figs. 13 through 15 that AODV-SS achieves higher QoE than OLSR-SS for all the inter-node distances. We also see in these figures that LQHR has approximately the same QoE as AODV-SS for inter-node distance equal to 25 m; however, when the inter-node distance is 35 m, the QoE of LQHR is almost the same as that of OLSR-SS. This is because LQHR can achieve appropriate routes in short inter-node distances, while LQHR is not optimized well for long inter-node distances.

7. Conclusions

In this chapter, we assessed the application-level QoS and QoE of audio-video streaming in a cross-layer designed wireless ad hoc network with media synchronization control at the

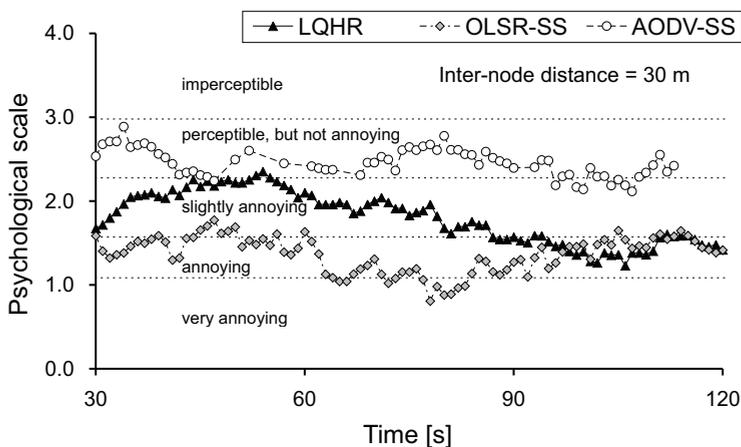


Fig. 14. Psychological scale for inter-node distance 30 m.

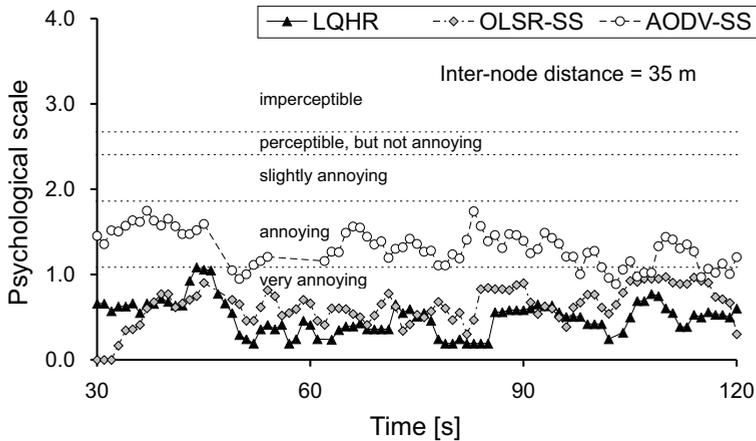


Fig. 15. Psychological scale for inter-node distance 35 m.

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 QoE 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 QoE/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.

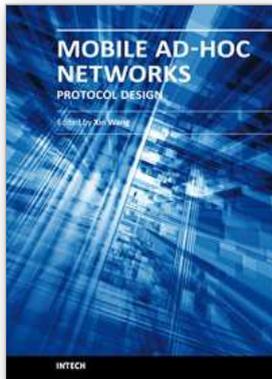
While this chapter does not assume QoS control mechanism in the MAC layer, IEEE 802.11e has been expected for QoS provision. Romdhani & Bonnet (2005) present a cross-layer routing protocol which is based on the cooperation between the AODV routing protocol and the IEEE 802.11e EDCA MAC protocol. We have 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 QoE of the three schemes in the practical propagation model of the wireless channel.

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Being infrastructure-less and without central administration control, wireless ad-hoc networking is playing a more and more important role in extending the coverage of traditional wireless infrastructure (cellular networks, wireless LAN, etc). This book includes state-of-the-art techniques and solutions for wireless ad-hoc networks. It focuses on the following topics in ad-hoc networks: quality-of-service and video communication, routing protocol and cross-layer design. A few interesting problems about security and delay-tolerant networks are also discussed. This book is targeted to provide network engineers and researchers with design guidelines for large scale wireless ad hoc networks.

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