A Lightweight SCTP for Partially Reliable Overlay Video Multicast Service for Mobile Terminals

Jinsuk Baek, Paul S. Fisher, Minho Jo, and Hsiao-Hwa Chen, *Fellow, IEEE*

***Abstract*—In this article, a video multicast protocol for multi-homed mobile terminals is proposed as an alternative stream control transmission protocol (SCTP) for partially reliable multicast services. It works with overlay peer-to-peer video multicast facility in the application layer. For a multi-homed mobile terminal, an error burst may occur when a handover is in process in the primary path switching procedure. The key issue concerned in this protocol is the ability to predict packet loss and to retransmit the lost packets as soon as a mobile terminal completes its primary path switching procedure. This property controls the delay sensitivity of transmissions. Conversely, the protocol can tolerate partial loss in video transmission as long as the loss is limited to a relatively short error burst. In addition, it reduces the message overhead significantly and provides a scalable communication mechanism for multicast applications. The performance improvement of the proposed protocol comes from 1) the estimation of temporal velocity of mobile terminals with lost packet prediction in a long error burst, and 2) the requirement for each mobile terminal to indicate which packets can be safely discarded from its agent.**

***Index Terms*—Overlay multicast, partially reliable multicast, scalable multicast, stream control transmission protocol, video multicast.**

I. Introduction

MULTICASTING is the only efficient scalable solution capable to deliver a single message to multiple recipients using the same IP address [1]. Multicasting must be realized in different layers of the Internet protocol stacks, including MAC layer, IP network layer, and application layer. In the MAC layer (e.g., IEEE 802.11a/b/g/e wireless LAN) [2], [3], a multicast sender (e.g., the access point of an infrastructure-based wireless LAN) transmits multicast frames locally using a simple broadcast mechanism, which transmits the frames at a low fixed rate without any back-off process. Moreover, it does not require multicast receivers (e.g., a mobile terminal) to send feedback, such as ACKs and NAKs, to their senders. This results in unreliable multicast frame transmissions.

Manuscript received August 11, 2009; revised November 15, 2009 and March 22, 2010; accepted June 08, 2010. Date of publication June 21, 2010; date of current version October 15, 2010. This work was supported in part by the Brain Korea 21 program; Ministry of Education, Science, and Technology; the Korean Government; and the National Science Council of Taiwan (NSC98-2219-E-006-011). The associate editor coordinating the review of this manuscript and approving it for publication was Dr. Zhihai (Henry) He.

J. Baek and P. S. Fisher are with the Department of Computer Science, Winston-Salem State University, Winston-Salem, NC 27110 USA.

M. Jo is with the Graduate School of Information Management and Security, Korea University, Seoul, Korea (e-mail: minhojo@korea.ac.kr).

H.-H. Chen is with the Department of Engineering Science, National Cheng Kung University, Tainan City 70101, Taiwan (e-mail: hshwchen@ieee.org).

Color versions of one or more of the figures in this paper are available online at http://ieeexplore.ieee.org.

Digital Object Identifier 10.1109/TMM.2010.2053523

The IP network layer supports multicasting via a multicast routing algorithm (e.g., DVMRP and PIM) and a group membership protocol (e.g., IGMP). However, IP multicasting [4]–[6] does not guarantee reliable datagram delivery due to its best-effort transmission. It also requires all deployed routers to be upgraded with a multicast capability. Therefore, a more practical approach is an overlay peer-to-peer multicast facility [7]–[14] supported by the application layer. Owing to its simple deployment and centralized management for all multicast states at the end system, the overlay multicasting has been considered as a promising approach to provide a reliable multicast capability. The success of overlay multicasting depends on the transport layer mechanism, since it uses a reliable unicast mechanism to provide a reliable multicast service, as well as an unreliable one when the application does not require a reliable service.

Unfortunately, the conventional overlay multicast architecture does not scale well, since the application of the multicast sender (MS) has to handle the separate sessions for each connection. Multiple multicast agents (MAs) have been deployed for scalable services to alleviate this problem [7]. The MAs can be used to relay the multicast data from an MS to multicast receivers (MRs). Each MA buffer temporarily maintains all the packets it has received recently from the MS. The MA buffer performs local error recovery for all the MRs in its group. Such a design, defined as the logical tree topology of the application layer, achieves the scalability by distributing the MS retransmission workload among multiple MAs.

A variety of networks (e.g., LAN, WLAN, WiMAX, and 3GPP) have currently been deployed. The next-generation network is converging to a unified all-IP-based network. Many mobile terminals are configured in multi-homed environments by simply installing two or more network interfaces to support different protocols [15]. The data transfer services offered by traditional transport protocols, such as TCP and UDP, are inadequate for these environments. Instead, stream control transmission protocol (SCTP) [16] has been proposed as an appropriate transport layer protocol supporting overlay multicast, owing to its multi-streaming and multi-homing features. The multi-homing feature enables an MR to use more than one IP address to support more than one communication path; that is, a primary path together with several alternative paths in a single multicast session. As a result, the multi-homing feature can maximize the utilization of the multi-homed environment to increase network availability.

It should be noted that the legacy SCTP is a connection-oriented transport protocol providing reliable end-to-end message delivery via selective ACK (SACK), flow control, and congestion control. This creates a tradeoff in video multicast service for MRs. First, the legacy SCTP incurs an unavoidable overhead in transmitting loss-tolerant, delay-sensitive, and real-time video data packets. At the same time, video data are loss-tolerant only when the data packet losses are independent because of link errors at the physical layer or frame collisions at the MAC layer. We should also note that the legacy SCTP was designed originally for wired networks. Therefore, the primary path switching is not conducted until the current primary path fails. Thus, the MR will not change its primary path until its new path is activated and its new IP address is updated for the MS. During this time period, the MR is in its unreachable status, causing continuous packet losses. This link failure can easily occur, especially when an MR performs a vertical handover between heterogeneous access networks. Finally, for SCTP, each MR is required to send a SACK to the MA for each packet it has correctly received. As a result, the MA's ability to handle these SACKs limits the number of MRs participating in a reliable overlay multicast session. Thus, this compromises the scalability.

In this paper, we propose an improved SCTP considering partial loss-tolerant and delay-sensitive characteristics in video data transmission. The proposed protocol focuses on the interactions between an MA and its MRs when the MR is in dual-homing mode. Under this condition, the MR is in an overlapping area between different network domains, and multiple communication paths are available. Due to its optimistic primary path switching approach, an MR can easily experience continuous packet losses when the MR performs an SCTP handover. The proposed protocol requires each MR to adaptively decide if retransmission is needed when it detects the availability of multiple paths, to provide a partially reliable transmission. This decision is made based on the expected velocity and expected moving distance of the MR in the overlap area.

If continuous packet losses are expected, the MR sends feedback to the MA. The contents of the feedback packet contain 1) the largest packet sequence number it has successfully received, and 2) how many consecutive packets will be lost. This single message therefore requests retransmission for these packets. After receiving this message, the MA immediately retransmits the packets to the MR, without waiting for other messages for the same packet from the other MRs. If continuous packet losses are not expected, the MRs also send cumulative ACKs at fixed and infrequent intervals. This packet indicates which packets can be safely discarded from the MA buffer. This cumulative ACK also differs from the conventional cumulative ACK, since it tolerates intermittent packet losses, as we consider partially loss-tolerant characteristics of video data.

The remainder of this paper is outlined as follows. Section II reviews the existing transport layer protocols to provide overlay and reliable multicast services. Section III introduces our new protocol. In Section IV, we evaluate the performance of the proposed protocol, followed by Section V that draws the conclusions of this paper.

II. Related Works

A. Overlay Multicast

The Class D group address model is currently employed to support IP multicasting. However, this causes several implementation issues, including 1) globally unique address allocation for each multicast group, 2) secure admission control at the multicast sender, and 3) scalable maintenance of the packet forwarding table at the multicast routers. These issues make the ISPs unable to support multicast services. Numerous proposals [17]–[19] providing multicast services based on the unicast mechanism have been introduced to deal with these limitations. Multicast RTP (M-RTP) [17] is one of these solutions. It sets up a multicast session using a set of unicast RTP sessions. It employs real-time control protocol (RTCP) to monitor the QoS of the RTP data packets. It also restricts the replication of data packets for security purposes. However, it incurs relatively expensive routing cost and wide bandwidth usage compared to IP multicasting. In [18], multiple routes were configured from sender to receivers to improve the throughput of the end-to-end sessions. However, non-negligible redundant flow exists in the same link. In [19], the authors proposed a protocol, called "multicast extension to open the shortest path first" (MOSPF), in which recursive unicast was used for scalability at the existing multicast routing protocol to avoid the packet forwarding state at the router.

B. Reliable Multicast

Many transport layer protocols [20]–[25] were proposed to provide a reliable multicast service. All protocols provide reliable service based on different retransmission control schemes, different feedback scheduling methods, and different strategies to determine which network entity should buffer packets for retransmission and how long these packets should be retained. However, all of the protocols aimed to provide bulk data transfer services, rather than multimedia multicasting services. Moreover, they aimed to provide multicast service for single-homed receivers.

C. SCTP

The performance of unicast-based, overlay, and reliable multicast is dependent largely on the transport layer technology as the basis for data transfer. Many mobile terminals are configured to allow the multi-homed environment. Thus, SCTP has been considered as a promising approach to support overlay and reliable multicast in many proposals [26], [27].

SCTP provides connection-oriented reliable transmission over the IP core network via SACK, flow control, congestion control and avoidance, as well as failure detection and recovery. It also provides faster transmission than TCP, since the multi-stream mechanism is designed to overcome the head-of-the-line blocking problem of TCP. It divides the overall SCTP message flow into sub-flows. A partial ordering of the message is performed within each of the sub-flows. This message-oriented feature prevents the message of one sub-flow from interfering with another sub-flow that would otherwise result in transmission delay. A further improved version of SCTP in terms of congestion control and flow control has been proposed in [28] and [29], respectively. More importantly, the multi-homing feature of SCTP increases network availability, since it enables an MR to switch its primary path to an alternative path when it detects a primary path failure. The primary path switching scheme should be judiciously designed to provide a seamless handover. Otherwise, it can have a significant negative impact on the performance of the SCTP session. This issue has led to numerous proposals [30]–[34] to determine the appropriate network conditions for the primary path switching. Comprehensively, all existing switching schemes depend essentially on the threshold values to recognize the

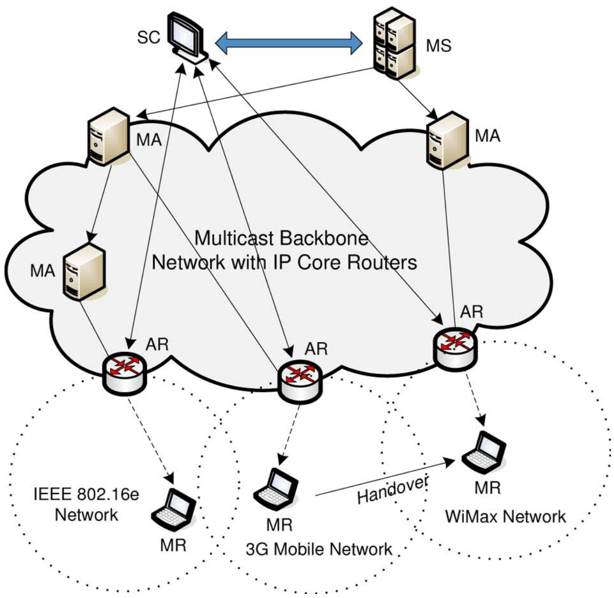


Fig. 1. Network configuration in overlay video multicast service.

current network conditions. However, it is practically difficult and computationally expensive to find such values in a dynamic network environment.

III. Lightweight SCTP

A. Overlay Multicast Architecture

First, let us consider an overlay multicast tree configuration in a wireless network which consists of a multicast sender (MS), a session controller (SC), multicast agents (MAs), access routers (ARs), and multicast receivers (MRs). The configuration is shown in Fig. 1. The SC is a functional entity having a dedicated communication channel with an MS for session control. The MA can be used to relay multicast data from MS to MRs. AR is used to manage the session in the access network to support mobility. For data delivery, the MS transmits multicast data to MRs in cooperation with MAs in the network. The MS assigns an appropriate MA for an MR and maintains this information in its data forwarding table. Accordingly, the MR receives the data packets from the MA. All retransmissions are performed via MA-to-MR communications. For control purposes, the MR informs SC of a session to join and leave. To monitor the session, each MR reports its status (e.g., link layer handover with an access point or IP layer handover with AR) to AR or SC. This will eventually be aggregated at the SC. The logical tree construction and peer-to-peer lookup mechanism [35] among application network entities can be supported in a wireless network.

The retransmitted packets, based on SACK at the MR, are worthless to the MR after their lifetimes have expired. They can only increase network congestion. Nonetheless, retransmission is required when an MR incurs continuous packet losses over a significantly long period of time. Therefore, we need a different transport layer protocol to provide partially reliable multicast service for video data. The transport layer protocol must be designed to tolerate intermittent packet losses only to fit its loss-tolerant but delay-sensitive characteristics. Instead, the packets, which are continuously lost, must be retransmitted to the MR as quickly as possible to maintain the QoS requirements. We begin the discussions by considering the following three facts.

1) Packet losses can happen from time to time, because each MR adopts an optimistic primary path switching approach. Therefore, packet losses tend to occur in long error bursts during the SCTP handover process. Based upon the time sensitivity of video packets, retransmission must be done within a very narrow time window. As a result, retransmission without considering timing simply results in bandwidth hogging, as the non-timely retransmitted packets are discarded by the MR. An MR should predict how many packets will be lost before it performs primary path switching to minimize this negative impact.
2) The most current wireless links are generally reliable, but link errors remain to be a major problem. Therefore, packet losses can be independent and are not correlated with previous transmission failures. This might not be true if the packet losses are due to buffer overflow in a router. In such a case, the packet losses occur during short error bursts, separated by relatively long periods of successful transmission. However, if the length of an error burst is not significantly long, these packets do not need to be retransmitted, especially when the packets are for video data transmission and have loss-tolerant characteristics. Even with packet retransmission, such retransmitted packets may be worthless, since the video stream will not backup to accommodate the retransmitted packets, unless the video stream is being viewed with a small delay time by the MR.
3) The MA buffer maintains all the packets it has recently transmitted to perform error recovery for all dependant MRs. The buffer size of the MA for a specific multicast session is limited. Therefore, the MA must periodically discard packets from its buffer. Discarding packets too late results in an inefficient use of limited buffer space on the MA. Conversely, discarding packets, which might still be needed, is unacceptable for a reliable multicast service. Therefore, the MRs should inform the MA which packets can be safely discarded from its retransmission buffer.

B. Packet Loss Scenarios

Since the proposed protocol can be applied in a straightforward manner to the multi-alternative routing paths environment, we assume that each MR has only one alternative path to simplify our description. The optimistic primary path switching procedure performs a sequence of actions as follows.

1) The MR detects a link-up of a new path and configures a new IP address at the network layer.
2) The MR sends an Add-IP message to the SC. This will be forwarded to the MS.
3) The MR detects link-down condition in its current primary path.
4) The MR sends a Primary-Switching message to the SC. The MS assigns a new MA for an MR and updates its data forwarding table. The new MA also updates its data forwarding table.
5) The MR sends a Delete-IP message to the SC. The MS removes the information of MR from its data forwarding table.

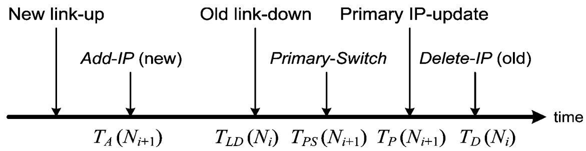


Fig. 2. Primary path switching scenario with expected elapsed times.

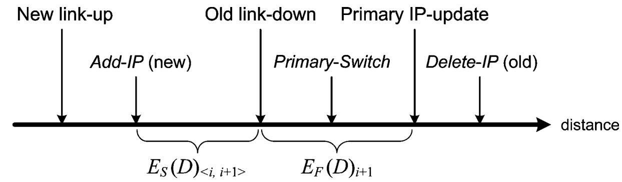


Fig. 3. Primary path switching scenario with expected moving distance.

The alternative path is used for an MA-to-MR data packet after action (4), while the primary path is still used for the MR-to-MA data packets prior to processing action (5). That is, for receiving data packets, the MR immediately uses its alternative path as a primary path as soon as it explicitly requests a primary path switch. However, the MR cannot receive a data packet until 1) it sends the Primary-Switching message, and 2) the MA updates the data forwarding table contained in its kernel. The MA-to-MR data packets, destined for the MR's old IP address, are lost during this period. In contrast, to send data packets, the MR does not use the alternative path until it explicitly deletes its current primary path. Even though the MR sequentially sends the Primary-Switching and Delete-IP messages using the shortest allowed intervals, the MR-to-MA control packets are also lost during this period.

Most primary path switching schemes require an MR to perform actions (4) and (5) before action (3) if certain network conditions have been satisfied. However, fast moving MRs still can experience a link-down event in its primary path, even before sending a Primary-Switching message to the SC. To resolve this problem, our protocol requires the MR to predict how many packets will be received successfully in the overlap area and how many packets will be lost during the aforementioned critical period. This prediction is performed by taking into account the expected moving speed of the MR when it is in the overlap area.

C. Expected Moving Speed of MR

We can estimate the moving speed of the MRs based on the elapsed moving time and the moving distance at the current network. Let us define $T_A(N_i)$ as the time when an MR sends the ADD-IP message to the session controller (SC) in its current network N_i. As an MR records the times whenever it moves to a different network, it keeps a series of $T_A(N_i)$ such that $2 \leq i \leq n$, assuming that it visited n networks. Therefore, the elapsed moving time $\triangle T_A(N_i)$ of the MR at the current network N_i can be obtained by

$$\triangle T_A(N_i) = T_A(N_{i+1}) - T_A(N_i). \tag{1}$$

The estimated speed of the MR at a given network N_i can now be obtained by dividing the moving distance $D(N_i)$ the MR traversed in the current network N_i by the elapsed moving time $\triangle T_A(N_i)$. Due to the difficulty in obtaining the actual moving distance $D(N_i)$, we use an expected distance $E(D)_i$ in each network N_i. This will be described later. Even though expected moving distance $(E(D)_i)$ can differ from the real moving distance $(D(N_i))$, it still works well for the proposed protocol. Overall performance of the protocol does not depend on exact measurements, because our interest is in the relative moving speed among the MRs. The critical estimation is for $E(D)_i$ to be consistently shorter than $D(N_i)$, thereby not anticipating the required switch prior to the switch occurring. The elapsed moving time $(\triangle T_A(N_i))$ can be changed frequently from network to network. At the same time, fast moving MRs tend to keep their current movement speeds. Therefore, the proposed protocol considers a limited number of recent historical speeds, rather than a full history. As a result, the expected moving speed $E(S)_{i+1}$ of the MR in its next network N_{i+1} is obtained by considering the recent $i-1$ records. This will obey

$$E(S)_{i+1} = \sum_{j=k}^{i} \omega_s(N_j) \frac{E(D)_j}{\triangle T_A(N_j)}, \quad k \geq 2 \tag{2}$$

where $\omega_s(N_j)$ is the weight for network N_j. Depending on the most recent speed of the MR, the proposed protocol adjusts the size of k and puts more weight (say 0.875) on the most recent speed of the MR. We also define S_T as a pre-configured threshold time $(S_T > 0)$ to differentiate MRs in terms of velocity. This is used to determine the relative speed of the MRs and γ as a fast speed coefficient $(\gamma \geq 1)$. Each MR can now be classified into three different categories, including 1) fast moving MRs with large $E(S)$ values such that $E(S) \geq \gamma S_T$, 2) slow moving MRs with small $E(S)$ values such that $S_T \leq E(S) < \gamma S_T$, and 3) slow moving MRs having a ping-pong movement pattern with extra-small $E(S)$ values such that $E(S) < S_T$. Based upon these observations, each MR determines when the packet retransmission should be requested.

D. Prediction of the Number of Packet Losses

Even though the video data are loss-tolerant, the long error burst is unacceptable, because it compromises the QoS of a multicast session. As we mentioned earlier, this can occur at an MR when it performs the handover involved with the primary path switching. The proposed protocol requires each MR to predict the number of packets that would be lost during the execution of the primary path switching procedure to cope with the long error burst. Once this is known, it requests for retransmission of those packets before it experiences a long error burst. We first consider a fast moving MR, which has a large expected moving speed $(E(S))$ value such that $E(S) \geq \gamma S_T$. Figs. 2 and 3 show the primary path switching scenarios between networks N_i and N_{i+1}, considering the time and distance domains, respectively.

Here, we define the expected elapsed time $T_S(N_{i+1})$ as the gap between "the time that an MR sends Add-IP message to the session controller (SC) at a new network N_{i+1} (the time $T_A(N_{i+1})$)", and "the time that the MR detects link-down for its current primary path (the time $T_{LD}(N_i)$)" for network N_i. Thus, we have

$$T_S(N_{i+1}) = T_{LD}(N_i) - T_A(N_{i+1}). \tag{3}$$

We also define the expected elapsed time $T_F(N_{i+1})$ as the gap between "the time that an MR detects the link-down of its current primary path at a new network N_{i+1} (the time $T_{LD}(N_i)$)", and "the time that the primary path switching procedure is completed (the time $T_P(N_{i+1})$)" for network N_{i+1}, or

$$T_F(N_{i+1}) = T_P(N_{i+1}) - T_{LD}(N_i). \tag{4}$$

As the times $T_{LD}(N_i)$ and $T_P(N_i)$ are unknown at time $T_A(N_{i+1})$, we use the expected link-down time $(E(T_{LD})_i)$ and expected primary path switching completion time $(E(T_P)_i)$. This can be expressed as

$$E(T_{LD})_i = \frac{E_S(D)_{\langle i,i+1\rangle}}{E(S)_{i+1}} \tag{5}$$

and

$$E(T_P)_i = \frac{E_S(D)_{\langle i,i+1\rangle} + E_F(D)_{i+1}}{E(S)_{i+1}}. \tag{6}$$

In the new network N_{i+1}, the number of expected lost data packets $E(LP)_{i+1}$ at the MR in the primary path switching procedure now can be expressed as

$$E(LP)_{i+1} = T_F(N_{i+1})\frac{R}{L} \tag{7}$$

where R is the current data reception rate (in bits per second) of the MR, and L is the data packet size of the multicast session.

Conversely, the MR successfully receives $E(P)_{i+1}$ data packets during time $T_S(N_{i+1})$ such that

$$E(P)_{i+1} = T_S(N_{i+1})\frac{R}{L}(1 - P_f) \tag{8}$$

where P_f is the probability that an MR does not successfully receive the multicast data packets due to frame collision, wireless link error, or a router buffer overflow.

E. Cumulative ACK With Prediction

In our protocol, each multicast agent (MA) is required to maintain in its buffer all those packets it has recently received from a multicast sender (MS). They perform local error recovery for all their MRs. As the MAs have limited buffer sizes, we need a mechanism that allows the MAs to periodically discard packets from their buffers in a safe manner.

The SCTP provides such a mechanism employing SACKs for every successfully received packet. The SACK packet format includes the cumulative ACK field, namely cumulative transmission sequence number (TSN) ACK field. The value of the TSN ACK field is the highest consecutive TSN that an MR has seen. This lets the MA discard all packets from its buffer that have been cumulatively acknowledged by all of its MRs.

Unfortunately, this conventional cumulative ACK does not provide a fast mechanism to safely discard packets from the MA's buffer, because 1) the MR has to send SACK packets with start and end values for the range of TSNs that have been received relative to the cumulative TSN ACK value, whenever it receives an out-of-order data packet. However, all of them have the same cumulative TSN ACK value; 2) This SACK means that the MR has correctly received packets up to the one included in the cumulative TSN ACK field. That is, this does not allow the MR to send the SACK, if there is a lost packet in the range $[0, \text{ cumulative TSN ACK value}]$. For video multicast services, this should be allowed since we consider loss-tolerant characteristics of video data, as long as the packet losses are intermittent events.

We define a new type of feedback, namely cumulative ACK with prediction of the number of lost packets $(CACKwP)$, to mitigate this problem. It is noted that $CACKwP$ is different from the conventional cumulative ACK, because 1) it contains a maximum transmission sequence number (max_TSN) that each MR has received, and 2) it also contains the range of TSNs that need to be retransmitted from the MA.

Both calculations are done at Add-IP message sending time $(T_A(N_{i+1}))$ in the given network N_{i+1} to satisfy the timing constraints of the retransmission. Consequently, the MR i transmits $CACKwP$ to its MA with the information at time $T_A(N_{i+1})$ shown in the equation at the bottom of the page. The values of second and third parameters can be given by

$$\begin{aligned} \min\_TSNwP_i(N_{i+1}) &= \max\_TSN_i(N_i) \\ &\quad + E(P)_{i+1} + 1 \qquad (9) \\ \max\_TSNwP_i(N_{i+1}) &= \min\_TSNwP_i(N_{i+1}) \\ &\quad + E(LP)_{i+1} - 1. \qquad (10) \end{aligned}$$

Conversely, all MRs periodically send cumulative ACK without predicting the number of lost packets $(CACKwP)$ to their MAs. This ACK includes only the $\max\_TSN$ filed. The period is dependent upon $\max\_MA(i)$. This is the maximum number of packets the MA of the MR i is willing to maintain in its buffer. Once we define t_d as the multicast session duration and S as packet receiving rate (in packets per second), MR i sends a similar $CACKwoP$ packet after each packet $\max\_MA(i)$ such that

$$\begin{aligned} \max\_TSN_i &= k \ \max\_MA(i) \\ \text{for} \quad k &= 1, 2, 3, \ldots, \frac{St_d}{\max\_MA(i)}. \end{aligned} \tag{11}$$

Assume that the MA handles n MRs in a multicast session and there is no pending packet for retransmission. The MA now can discard up to packet D, or

$$D \le \min\Big\{\max\_TSN_i(N_i) \Big| \ 1 \le i \le n \text{ and } 1 \le j \le N\Big\}. \tag{12}$$

This lets MAs discard from their buffers all the packets that have been acknowledged by all MRs. The $CACKwP$ and $CACKwoP$ guarantee that the MA will always have in its buffer all the packets that can be requested by any of its MRs for the same multicast session, since it denotes the MRs receiving at least up to packet D without continuous lost packets. Of course, MRs that leave the multicast session without notice can

$$CACKwP_i\Big\{\max\_TSN_i(N_i), \min\_TSNwP_i(N_{i+1}), \max\_TSNwP_i(N_{i+1})\Big\}$$

disrupt the multicast session for all MRs. With our protocol, the MA will use a timeout mechanism to detect such situation.

For the slow moving MRs, which have small expected moving speed $(E(S))$ values such that $S_T \leq E(S) < \gamma S_T$, it is relatively difficult to predict the number of successfully received packet $(E(P)_{i+1})$ accurately, since the MRs tend to stay for a longer time than the fast moving MRs in the overlap area between two networks. Therefore, we require those MRs to perform the prediction at time $t_{LD}(N_i)$, such that $t_{LD}(N_i) \approx T_{LD}(N_i)$, but $t_{LD}(N_i) < T_{LD}(N_i)$.

Accordingly, the content of the $CACKwP_i$ packet for slow moving MRs will obey the rule shown in the equation at the bottom of the page. The $\min\_TSNwP_i(N_{i+1})$ can be given by

$$\min\_TSNwP_i(N_{i+1}) = \max\_TSN_i(N_i) + 1. \tag{13}$$

Let $\min\_TSN_{MA}(i)$ be the transmission sequence number that an MA first transmits to MR i after completing the primary path switching procedure at time $T_P(N_{i+1})$. The MA has to control its retransmission before transmitting the packet $\min\_TSN_{MA}(i)$ to minimize duplicate data packet receptions at the MR i. This consideration has been included in the MA algorithm.

Finally, we do not perform the prediction for the MRs having extremely low expected moving speeds $(E(S))$, such that $E(S) < S_T$. These MRs are likely to display irregular movement patterns. For these MRs, some mobility prediction support will be needed with the assistance from the network layer protocols. However, this prediction algorithm is beyond the scope of this paper. We have however developed such an algorithm. The details of MR and MA algorithms for an arbitrary network N_{i+1} are described in Algorithms 1 and 2, respectively.

Algorithm 1: MR Algorithm: Retransmission Request for MR i with prediction at network N_{i+1}

Join multicast session
begin
  Calculate $E(S) = E(S)_{i+1}$
  Send $CACKwoP_i$ at every $\max\_MA(i)$ packets interval
  switch $E(S)$ **do**
    case $(E(S) \geq \gamma S_T)$
      Find $\max\_TSN_i(N_i)$
      $\min\_TSNwP_i(N_{i+1}) = \max\_TSN(N_i) + E(P)_{i+1} + 1$
      $\max\_TSNwP_i(N_{i+1}) = \min\_TSNwP_i(N_{i+1}) + E(LP)_{i+1} - 1$
      Send $CACKwP_i$ to its MA at time $T_A(N_{i+1})$
      Break
    end
    case $(S_T \leq E(S) < \gamma S_T)$
      Find $\max\_TSN_i(N_i)$
      $\min\_TSNwP_i(N_{i+1}) = \max\_TSN_i(N_i) + 1$
      $\max\_TSNwP_i(N_{i+1}) = \min\_TSN_i(N_{i+1}) + E(LP)_{i+1} - 1$
      Send $CACKwP_i$ to its MA at time $t_{LD}(N_i)$
      Break
    end
    case $(E(S) < S_T)$
      No predicted retransmission request
      Break
    end
  end
end
Leave multicast session

Algorithm 2: MA Algorithm: Retransmission Request From n MRs With Prediction

Join multicast session
Set $\max\_MA$
begin
  switch *event* **do**
    case *Data packet from MS arrives*
      Store packet in buffer
      Break
    end
    case $CACKwP_i$ from an MR i *arrives*
      Set $D = \min\{\max\_TSN_i | \ 1 \leq i \leq n\}$
      Remove up to packet D from its buffer
      if $\min\_TSNwP_i \geq \min\_TSN_{MA}(i)$ **then**
        No retransmission
      end

$$CACKwP_i\Big\{\max\_TSN_i(N_i), \min\_TSNwP_i(N_{i+1}), \max\_TSNwP_i(N_{i+1})\Big\}$$

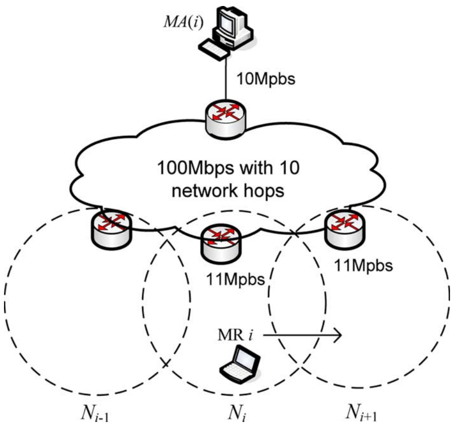


Fig. 4. Network model.

else

Immediately retransmit packets in range

$[\min\_TSNwP_i, \max\{\max\_TSNwP_i,$
$\min\_TSN_{MA}(i)-1\}]$ to MR i

end

Break

end

case $CACKwoP_i$ *from an MR arrives*

Set $D = \min\{\max\_TSN_i | 1 \leq i \leq n\}$

Remove up to packet D from its buffer

Break

end

end

end

Leave multicast session

IV. Performance Evaluation

Let us consider a network topology as shown in Fig. 4 to illustrate the performance improvement of the proposed protocol. Assume that MR i has two different NICs, which use IEEE 802.11b as their wireless links, while the current $MA(i)$ for MR i has one NIC, which uses Ethernet as a wired link. Let us also assume that these two hosts are interconnected through a 100 Mbps IP core network, each having an average of ten network hops.

We have developed an RTT generator to observe the variations of round trip times between the two hosts under this topology. With this generator, the effective bandwidth for the two paths has a range between 0.5 Mbps to 1.5 Mbps. The propagation delays for the multicast receiver (MR) at a given time also have been proportionally changed based upon the MR's physical distances to its access routers (ARs). Based on the statistical random multiplexing of the packet switched routers, the queuing delay has been generated as a Poisson random process. It has an average 50 ms over time. Finally, we do not consider the processing delay, because this is relatively small and it is constant compared to other delays. Based on the generated round trip times, we can also generate the packet loss probability of each MR by employing the well-known equation of TCP throughput [36], where MSS is the maximum segment packet size, RTT_{MR} is the round trip time from MR to its MA, and P is the packet loss probability of the MR. Thus we have

$$Throughput(\Phi) = \frac{MSS}{RTT_{MR}\sqrt{P}}. \quad (14)$$

Considering fast moving MRs, we assume that the MRs maintain their current speed in their dual-homing status. Their movement directions are also assumed steady, except for the MRs that may exhibit a ping-pong movement pattern. Slow moving MRs are defined to have a speed of less than 10 km/h. As we set the speed coefficient γ to four, the MRs that are faster than 40 km/h are considered to be fast moving MRs. We summarize the experimental parameters as follows. We set the minimum speed coefficient γ to four, historical speed threshold S_T to 10 km/h, critical period δ to 3 s, data packet size L to 1 Kb, packet sending rate S to 140 kbps, retransmission timer value of MA RT_{MA} to 150 ms, expected movement distance $E(D)_i$ to 1000 meters, expected movement distance in an area $E(D)_{\langle i,i+1\rangle}$ to 400 meters, and wireless link speed LS to $10^4\ \mathrm{m/s}$, respectively.

Due to the difficulty in obtaining the actual moving distance, we use an expected moving distance of MRs that considers the geographical network formation as shown in Fig. 5. For a given network N_i, these calculations can be performed by the corresponding AR. This has parameter values such as R_{i-1}, R_i, R_{i+1}, $r_{\langle i-1,i\rangle}$, $r_{\langle i,i+1\rangle}$, α_{i-1}, and α_i. This information will be given to the MR through a router advertisement message. We consider two different expected distances, including 1) expected movement distance $E(D)_i$ in the current network N_i, and 2) $E(D)_{\langle i,i+1\rangle}$ in the overlap area between networks N_i and N_{i+1}. As we consider only a non-winding path for non-slow moving MRs, the theoretical movement distance D_i for MR i at the network N_i can be bounded by

$$\sqrt{\min\_D_i} \leq D_i \leq 2R_i \quad (15)$$

where $\min\_D_i$ is equal to the equation at the bottom of the page. That is, the minimum movement distance can be obtained when an MR enters the network N_i at the coordinate A or A', and leaves at the left most point of the area of network N_{i+1}. On the other hand, the maximum movement distance can be obtained

$$\left(2R_i - r_{\langle i,i+1\rangle} - \frac{r_{\langle i-1,i\rangle}}{2}\right)^2 + R_{i-1}^2 - \left(R_{i-1} - \frac{r_{\langle i-1,i\rangle}}{2}\right)^2$$

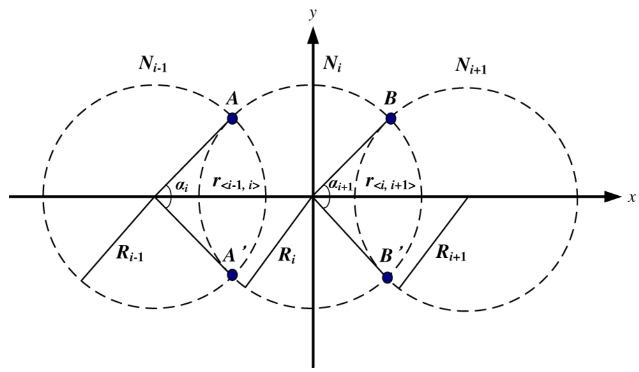


Fig. 5. Model for calculating expected movement distance of an arbitrary MR.

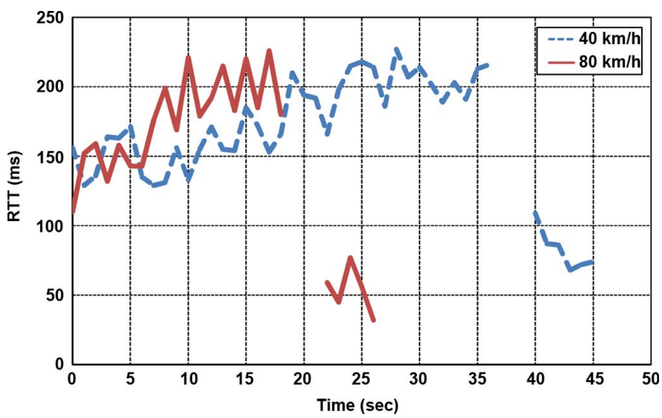


Fig. 6. RTTs variations of MR at 40 km/h and 80 km/h.

when the entering point is the coordinate A or A', and the leaving point is the coordinate B or B'.

Under the same assumption, the theoretical movement distance $D_{\langle i-1,i\rangle}$ and $D_{\langle i,i+1\rangle}$ for MR i can be also bounded by (16) and (17), respectively:

$$0 \leq D_{\langle i-1,i\rangle} \leq 2\sqrt{r_{\langle i-1,i\rangle}R_{i-1} - \frac{r^2_{\langle i-1,i\rangle}}{4}} \tag{16}$$

$$0 \leq D_{\langle i,i+1\rangle} \leq 2\sqrt{r_{\langle i,i+1\rangle}R_{i} - \frac{r^2_{\langle i,i+1\rangle}}{4}}. \tag{17}$$

That is, the minimum movement distance in the overlap area between networks N_{i-1} and N_i can be obtained when the MR enters the overlap area at the coordinate A or A' and leaves the area at the same coordinate. On the other hand, the maximum movement distance for the same area is obtained when the MR enters the area at the coordinate A and leaves at A' or vice versa. The movement distance in the overlap area between networks N_i and N_{i+1} can be calculated in a similar way as shown in (17).

In contrast, the expected movement distance $E(D)_i$ in an arbitrary network N_i can be calculated based upon the four intersection points A, A', B, and B' among the networks N_{i-1}, N_i, and N_{i+1}. When all ARs are evenly deployed, that is, $R_{i-1} = R_i = R_{i+1} = R$, $r_{\langle i-1,i\rangle} = r_{\langle i,i+1\rangle} = r$, and $\alpha_{i-1} = \alpha_i = \alpha$, the four coordinates can be simplified into

$$\begin{aligned} A &= \left[-\left(R-\frac{r}{2}\right), \sqrt{R^2-\left(R-\frac{r}{2}\right)^2}\right] \\ A' &= \left[-\left(R-\frac{r}{2}\right), -\sqrt{R^2-\left(R-\frac{r}{2}\right)^2}\right] \\ B &= \left[R-\frac{r}{2}, \sqrt{R^2-\left(R-\frac{r}{2}\right)^2}\right] \\ B' &= \left[R-\frac{r}{2}, -\sqrt{R^2-\left(R-\frac{r}{2}\right)^2}\right]. \end{aligned} \tag{18}$$

Therefore, the expected movement distance $E(D)_i$ in an arbitrary network N_i can be represented by the average of the longest movement distance and the shortest movement distance, or

$$E(D)_i = \frac{\sqrt{\left(2R-r-\frac{r}{2}\right)^2 + R^2 - \left(R-\frac{r}{2}\right)^2} + 2R}{2}. \tag{19}$$

In the same environment, the expected movement distance $E(D)_{\langle i,i+1\rangle}$ in the overlap area between networks N_i and N_{i+1} will obey (20) representing the average length of the chord between B and B', or

$$E(D)_{\langle i,i+1\rangle} = \sqrt{R^2-\left(R-\frac{r}{2}\right)^2 + \left(\frac{r}{2}\right)^2}. \tag{20}$$

Based upon these calculations, the expected distances are available to the MRs. They will allow each MR to calculate its expected speed $E(S)$ in a given network. Given the network topology in Fig. 5, we consider that an MR is currently located in network N_i and performs handover to network N_{i+1}. Also, the MR performs the primary path switching procedure when it is in the overlap area between the networks N_i and N_{i+1}. Fig. 6 shows how RTTs of MRs change when the MRs move toward the network N_{i+1}. We consider two MRs moving at speeds of 40 km/h and 80 km/h, respectively. At first, we can observe that the 40 km/h speed MR and 80 km/h speed MR spend about 36 s and 18 s in the overlap area, respectively. Note that the RTTs of both MRs are temporary unavailable. As we set the new NIC activation delay to two seconds, and the new primary path update delay plus one-way transit time to send a Primary-Switch message to one second, the MR has an unreachable status for 3 s in both cases. We define this as critical period δ. After completing the primary path switching procedure, the MR starts with a better RTT, because it uses a new primary path that will have a shorter RTT as it approaches to the AR of the new network.

Analyzing the performance of our protocol, we are cognizant of the fact that a packet transmitted from the MA is lost due to the optimistic primary path switching when an MR performs a handover process. Notification of the packet status between the corresponding entities is not instantaneous. The retransmission delay is important in evaluating the performance of our protocol. We evaluated the retransmission delay of our protocol and compared it with the SCTP protocol. For a fair comparison, we only considered the retransmission delay for the lost packets that are included in the long error burst during the critical period.

Based on the numerical results shown in Fig. 6, the temporal average round trip time, $RTT_{\langle MA(i),MRi\rangle}$, between the MR i and its $\mathrm{MA}(i)$, is less than 100 ms as the MRs are relieved from the critical period. Therefore, the temporal average one-way transit time, $OTT_{\langle MA(i),MRi\rangle}$, between the same entities is less than 50 ms. In addition, the average inter-packet arrival time $\Delta_T[PSN, PSN+1]$ at the MR is about 7 ms.

The SCTP protocol operates as follows. $\mathrm{MA}(i)$ transmits a data packet to its MR i and waits for the packet's SACK message until its retransmission timer RT_{MA} expires. Once the timer RT_{MA} expires, it retransmits the packet. The retransmission delay for the first lost packet in an error burst, D_{RET}, at $\mathrm{MA}(i)$) thus conforms to the following equation:

$$D_{RET} = RT_{MA} + OTT_{\langle MA(i),MRi\rangle}. \tag{21}$$

Consider the case of the SCTP protocol. The total retransmission delay for all lost packets in all error bursts is D_{SCTP}. At $\mathrm{MA}(i)$, it can be represented by the duration of the critical period δ_i, data reception rate $R(i)$, data packet size L, multicast session duration t_d, and the elapsed time to travel across one network domain. This results in the following equation:

$$D_{SCTP} = \left(\left\lfloor \frac{t_s}{\frac{E(D)}{E(S)}} - 1 \right\rfloor\right) \times \left(D_{RET} + \triangle_T\left[PSN, PSN+1\right]\left\lfloor \frac{R(i)}{L} \right\rfloor \delta_i\right). \tag{22}$$

Conversely, our protocol enables an MA to immediately retransmit the requested packet with a $CACKwP$ message without waiting for the retransmission timer to expire. Therefore, the total retransmission delay for all lost packets of MR i, D_{L_SCTP}, conforms to (23) at the bottom of the page.

In both (22) and (23), the first term represents the frequency of the primary path switching of the MR, while the second term represents the expected delay caused by one primary path switching procedure. Fig. 7 shows the total retransmission delay for an MR for both protocols. As shown, the delay at the MRs is reduced by about 35 s for one gigabyte of video data. In addition, our predicted retransmission request is more important for the fast moving MRs, as the difference linearly increases as a function of speed.

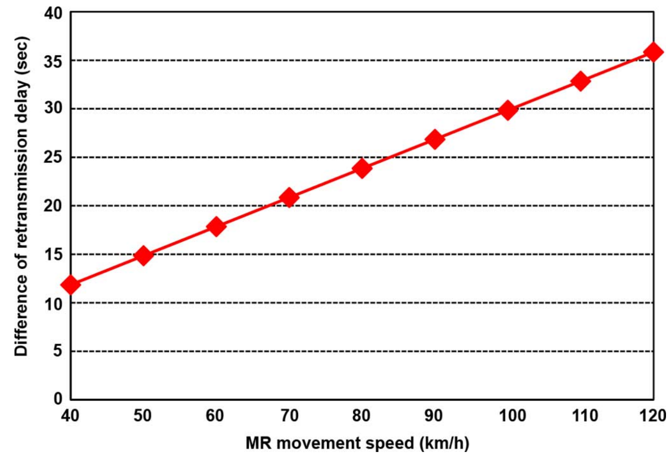


Fig. 7. Minimum difference of total retransmission delay for an MR versus different moving speeds.

This is the minimum difference between the two protocols in terms of retransmission delay, because the difference is evaluated only when the MR performs the primary path switching procedure. When we consider the entire multicast session with m packets transmission, for the MR having 1) packet loss probability P_{indep} caused by link error, 2) packet loss probability P_{cont} due to the buffer overflow of a router, and 3) the average length of error burst $A(b)$, the SCTP protocol introduces additional retransmission delay. This can be calculated by (24) at the bottom of the page, where the first term represents the retransmission delay for continuous packet losses, while the second term represents the same for the independent packet losses. When we set the packet loss probability P to 0.01 ($P_{cont} = 0.009$, $P_{indep} = 0.001$) and $A(b)$ to ten, the MR with SCTP has about 7 min and 16 s retransmission delay. This is unnecessary if we consider the timing constraint of the video data. In addition, the MR will experience additional delay because the MA is required to re-start its congestion control in the slow start phase whenever its retransmission timer expires.

Another advantage of the proposed protocol is that the MRs are not required to transmit unnecessary feedback messages to their MA. This will allow an MA to handle far fewer feedback messages sent by its MRs. We compare our protocol to the legacy SCTP, which requires all MRs to transmit SACKs for each successfully received packet.

With our protocol, the loss probability of MR i must be considered separately. The first is the loss probability due to link

$$D_{L\_SCTP} = \left(\left\lfloor \frac{t_s}{\frac{E(D)}{E(S)}} - 1 \right\rfloor\right)\left[\left(D_{RET} - RT_{MA}\right) + \triangle_T\left[PSN, PSN+1\right]\left\lfloor \frac{R(i)}{L} \right\rfloor \delta_i\right] \tag{23}$$

$$m\frac{P_{cont}}{A(b)}\left\{D_{RET} + \triangle_T\left[PSN, PSN+1\right]\left[A(b)-1\right]\right\} + mP_{indep}D_{RET} \tag{24}$$

errors or the buffer overflow of the router. The other is the loss probability due to the optimistic primary path switching method of the MR.

In the case of link errors or router buffer overflow, which results in independent packet losses or relatively shorter error bursts, the number of $CACKwP_i$ messages is not counted in our protocol. In the case of primary path switching, which results in continuous packet losses, a single $CACKwP_i$ message will request multiple packet retransmissions. Therefore, the number of $CACKwP_i$ messages sent by MR i can be represented by multicast session duration t_d and elapsed time to travel across one network domain. This results in the following equation:

$$N(MR\_CACKwP_i) = \left(\left\lfloor \frac{t_d}{\frac{E(D)}{E(S)}} \right\rfloor - 1 \right). \quad (25)$$

In addition, our protocol requires each MR to send the $CACKwoP_i$ message with a $\max\_MA(i)$ interval and an additional $CACKwoP_i$ message for the last received packet. When the session involves transmission of m packets, the number of $CACKwoP_i$ messages sent by each MR is given by

$$N(MR\_CACKwoP_i) = \left(\left\lfloor \frac{m}{\max\_MA(i)} \right\rfloor + 1 \right). \quad (26)$$

As a result, the total number of feedback messages, $N(MR_{L\_SCTP})$, that MR i transmits to its MAs is given by

$$N(MR_{L\_SCTP}) \leq \left\lfloor \frac{m}{\max\_MA(i)} \right\rfloor + \left\lfloor \frac{t_d}{\frac{E(D)}{E(S)}} \right\rfloor. \quad (27)$$

In contrast, the total number of SACK messages sent by each MR with an SCTP, where all MRs acknowledge all packets received, is given by

$$N(MR_{SCTP}) = m. \quad (28)$$

When all MAs of MR i have the same initial buffer size during the session, that is, $\max\_MA(i)$ is replaced with a constant $\max\_MA$, the minimum difference $N(MR)\triangle_{\min}$ between the numbers of feedback messages for the two protocols conforms to the inequality given in the following:

$$N(MR)\triangle_{\min} \geq m - \left\lfloor \frac{m}{\max\_MA(i)} \right\rfloor - \left\lfloor \frac{t_d}{\frac{E(D)}{E(S)}} \right\rfloor. \quad (29)$$

Fig. 8 shows the results when we consider an MR moving at 80 km/h. We set the number of transmitted packets m to 1 000 000. This roughly represents a transfer of one gigabyte with a packet size of one kilobyte. As we can see from Fig. 8, the minimum difference in the number of feedback messages from an MA between our protocol and SCTP is more than 990 000, as long as $\max\_MA$ is set to be larger than 100.

We show that the reduced number of feedback messages sent by each MR directly affects the scalability of the MAs. We posit that the MA can handle up to M MRs in a multicast session. For MA i, let λ_i be the average arrival rate of MRs and μ_i be the average departure rate of MRs. It is noted that $\lambda_i = \lambda$ and $\mu_i = \mu$, if $0 \leq i \leq M$. Under this condition, the probability P_k that there is exactly k MRs under MA i in a specific multicast session is given by

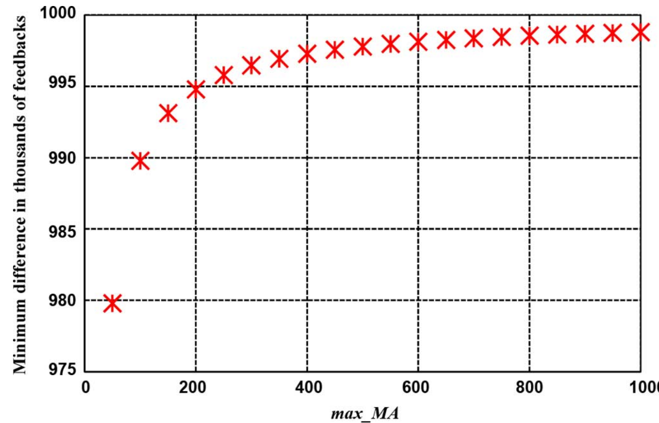


Fig. 8. Minimum $N(MR)\triangle_{\min}$ for an MR moving at 80 km/h.

$$P_k = \left(\frac{\lambda}{\mu}\right)^k P_0, \quad \text{for} \quad 1 \leq k \leq M. \quad (30)$$

Since $\sum_{k=0}^{M} P_k = 1$, we have $P_0 = 1 - (P_1 + P_2 + \cdots + P_M)$. Inserting the values given in (30), we obtain

$$P_0 = 1 - P_0 \left[\left(\frac{\lambda}{\mu}\right) + \left(\frac{\lambda}{\mu}\right)^2 + \cdots + \left(\frac{\lambda}{\mu}\right)^M \right].$$

Now, if $\lambda = \mu$, we have $P_0 = 1 - P_0(M)$ or $P_0 = 1/M + 1$. If $\lambda \neq \mu$, we obtain $P_0 = 1 - P_0[\sum_{k=1}^{M} (\lambda/\mu)^k]$ or

$$P_0 = \frac{1}{\sum_{k=0}^{M} \left(\frac{\lambda}{\mu}\right)^k}.$$

Therefore, we have

$$P_0 = \begin{cases} \frac{1}{M+1}, & \lambda = \mu \\ \frac{1}{\sum_{k=0}^{M} \left(\frac{\lambda}{\mu}\right)^k}, & \lambda \neq \mu. \end{cases} \quad (31)$$

While the average number of MRs, $E(MR)$, in a multicast session obeys (32), the $E(MR)$ should be calculated based on different mobility conditions, or

$$E(MR) = \sum_{k=0}^{M} k P_k. \quad (32)$$

If $\lambda = \mu$, we get

$$\begin{aligned} E(MR) &= \sum_{k=0}^{M} k P_k = \sum_{k=0}^{M} k \left(\frac{\lambda}{\mu}\right)^k P_0 \\ &= \sum_{k=0}^{M} k \frac{1}{M+1} \\ &= 0 + \frac{1}{M+1} + \frac{2}{M+1} + \cdots + \frac{M}{M+1} \\ &= \frac{\frac{M(M+1)}{2}}{M+1} = \frac{M}{2}. \end{aligned} \quad (33)$$

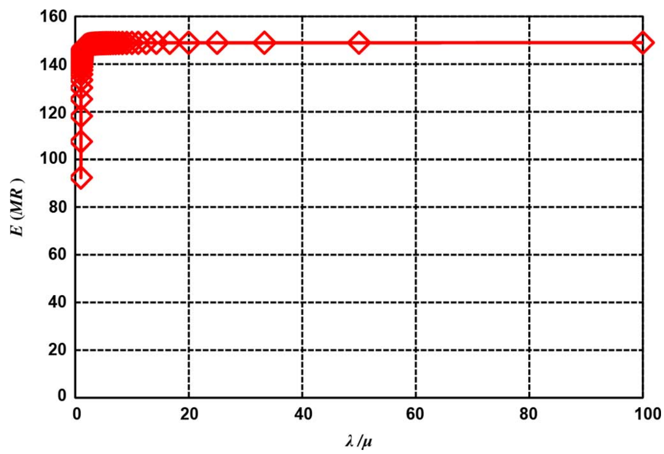


Fig. 9. Distribution of E(MR) with λ/μ variations.

If $\lambda \neq \mu$, we get

$$E(MR) = \sum_{k=0}^{M} kP_k = \sum_{k=0}^{M} k\left(\frac{\lambda}{\mu}\right)^k \frac{1}{\sum\limits_{k=0}^{M}\left(\frac{\lambda}{\mu}\right)^k} = \sum_{k=0}^{M} k\left(\frac{\lambda}{\mu}\right)^k \frac{1-\left(\frac{\lambda}{\mu}\right)}{1-\left(\frac{\lambda}{\mu}\right)^M}. \tag{34}$$

Now, let us consider $S = \sum_{k=0}^{M} k\,(\lambda/\mu)^k$, which can be rewritten into

$$S = 1\left(\frac{\lambda}{\mu}\right)^1 + 2\left(\frac{\lambda}{\mu}\right)^2 + 3\left(\frac{\lambda}{\mu}\right)^3 + \cdots + M\left(\frac{\lambda}{\mu}\right)^M \tag{35}$$

and

$$\left(\frac{\lambda}{\mu}\right) S = 1\left(\frac{\lambda}{\mu}\right)^2 + 2\left(\frac{\lambda}{\mu}\right)^3 + 3\left(\frac{\lambda}{\mu}\right)^4 + \cdots + M\left(\frac{\lambda}{\mu}\right)^{M+1}. \tag{36}$$

Then, we have

$$\begin{aligned} S &= \left\{\frac{\left(\frac{\lambda}{\mu}\right)\left[1-\left(\frac{\lambda}{\mu}\right)^M\right]}{\left[1-\left(\frac{\lambda}{\mu}\right)\right]^2}\right\} - \frac{1}{1-\left(\frac{\lambda}{\mu}\right)} - \left\{\frac{M\left(\frac{\lambda}{\mu}\right)^{M+1}}{\left[1-\left(\frac{\lambda}{\mu}\right)\right]}\right\} \\ &= \left\{\frac{\left(\frac{\lambda}{\mu}\right)\left[1-\left(\frac{\lambda}{\mu}\right)^M\right]}{\left[1-\left(\frac{\lambda}{\mu}\right)\right]^2}\right\} - \left\{\frac{1+M\left(\frac{\lambda}{\mu}\right)^{M+1}}{\left[1-\left(\frac{\lambda}{\mu}\right)\right]}\right\}. \end{aligned} \tag{37}$$

Now, $E(MR)$ becomes

$$\begin{aligned} E(MR) &= \frac{1-\left(\frac{\lambda}{\mu}\right)}{1-\left(\frac{\lambda}{\mu}\right)^M} \times S \\ &= \frac{\left(\frac{\lambda}{\mu}\right)}{1-\left(\frac{\lambda}{\mu}\right)} - \frac{1+M\left(\frac{\lambda}{\mu}\right)^{M+1}}{1-\left(\frac{\lambda}{\mu}\right)^{M+1}}. \end{aligned} \tag{38}$$

Therefore, if $M > \lambda$, we can obtain

$$E(MR) = \begin{cases} \frac{M}{2}, & \lambda = \mu \\ \frac{\left(\frac{\lambda}{\mu}\right)}{1-\left(\frac{\lambda}{\mu}\right)} - \frac{1+M\left(\frac{\lambda}{\mu}\right)^{M+1}}{1-\left(\frac{\lambda}{\mu}\right)^{M+1}}, & \lambda \neq \mu. \end{cases} \tag{39}$$

In contrast, if the average arrival rate is greater than the capacity of the MA, then $E(MR)$ can be represented by

$$E(MR) = \sum_{k=0}^{\mu} (M-k) MC_k \left(\frac{\lambda}{\mu}\right)^k \times \left[1-\left(\frac{\lambda}{\mu}\right)\right]^{M-l}, \quad M < \lambda. \tag{40}$$

The effective arrival rate λe is defined as $\lambda_i(1-P_M)$. It is equal to λ, provided $M > \lambda$ and $P_M \rightarrow 0$. The numerical results for $E(MR)$ with different rates of λ and μ are shown in Fig. 9.

When the MA is involved in $N(AR)$ network domains, the number of $CACKwoP$ messages which the MA receives from its $E(MR)$ MRs can be calculated by multiplying $E(MR)$ and the number of $CACKwoP$ messages sent by each MR. Therefore, the number of $CACKwoP$ messages, $N(MA\_CACKwoP)$, is given by

$$N(MA\_CACKwoP) \leq E(MR)\left(\left\lfloor \frac{m}{\max\_MA} \right\rfloor + 1\right). \tag{41}$$

On the other hand, the number of $CACKwP$ messages, $N(MA\_CACKwP)$, which the MA receives from its $E(MR)$ MRs is given by

$$N(MA\_CACKwP) = \sum_{k=0}^{N(AR)-1} E(MR)P_h \tag{42}$$

where P_h is the probability of handover for $E(MR)$ MRs over the $N(AR)$ network in which the MA is involved. Assume that we have a uniform probability distribution. Then, $N(MA\_CACKwP)$ is equal to $2E(MR)$. Finally, the total number of feedback messages, $N(MA_{L\_SCTP})$, that the MA receives from its $E(MR)$ MRs can be expressed as

$$\begin{aligned} &N(MA_{L\_SCTP}) \\ &\quad = N\big(MA\_CACKwP\big) + N\big(MA\_CACKwoP\big). \end{aligned} \tag{43}$$

Conversely, the total number of SACKs messages sent by $E(MR)$ MRs for the same MA with an SCTP is given by

$$N(MA_{SCTP}) = E(MR)m \tag{44}$$

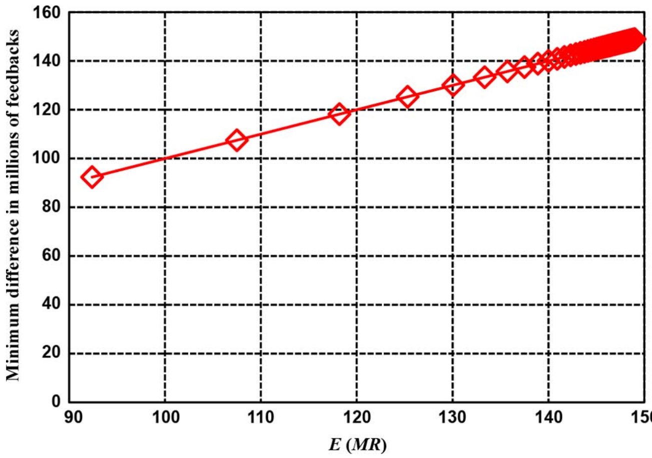


Fig. 10. Minimum $N(MA)\triangle_{\min}$ for different numbers of $E(MR)$.

resulting in the minimum difference $N(MA)\triangle_{\min}$ for both protocols as follows:

$$N(MA)\triangle_{\min} \geq E(MR)\left[m-2-\left(\left\lfloor\frac{m}{\max\_MA}\right\rfloor+1\right)\right]. \quad (45)$$

Fig. 10 shows the numerical results for the minimum difference in the number of feedback messages sent by the MRs for our protocol and SCTP when M is set to 150. As the number of MRs under the MA approaches to the maximum capacity M, the difference is more than 140 million messages. This difference becomes more prominent as the number of MAs increases. This result indicates that our protocol provides an efficient buffer management for the MA, by reducing the number of feedback messages sent by the MRs. This feature provides scalability, since each MA can handle more MRs.

V. Conclusion

The characteristics of video transmissions include the tolerance of short duration packet losses but the intolerance to long duration packet losses. These facts imply that all current protocols are unsuitable. Thus, we proposed a new protocol to support mobile devices. We performed performance analysis and provided numerical results to substantiate the claims of this proposed protocol. We showed that the proposed new peer-to-peer protocol for multi-homed mobile devices is resistant to the transmission failures resulted from primary path switching procedures. We characterized mobile devices as fast, medium and slow movers through wireless network topologies. Lost packets are reduced by anticipating the network path switch, namely to calculate the quantity of packets that will be lost at anticipated network locations. This enables the MRs to receive these packets immediately after the network path switch without delay. At the same time, we improved the packet discarding process at the MA based on a new ACK packet via CACK, which contains the maximum sequential number the MR has received. When the MA receives each of these new ACKs, it discards these identified packets from its buffer. This reduces the packet overhead dramatically and allows a particular MA to control a larger number of MRs. Within this framework, we have improved the QoS by better managing packet loss and delivery.

References

[1] J. Baek and J.-F. Paris, "A heuristic buffer management and retransmission control scheme for tree-based reliable multicast," *ETRI J.*, vol. 27, no. 1, pp. 1–12, Feb. 2005.

[2] *IEEE 802.11 v/D0.07, art 11: Wireless Medium Access Control (MAC) and physical layer (PHY) specifications: Amendment v: Wireless Network Management*, Draft Supplement to IEEE 802.11 Std., Draft 0.07, 2007.

[3] IEEE Computer Society, *02.11: Wireless LAN MAC and PHY Specifications Amendment 8,* 2005.

[4] B. Whetten, L. Vicisano, R. Kermode, M. Handley, S. Floyd, and M. Luby, "Reliable multicast transport building blocks for one-to-Many bulk-data transfer," *IETF RFC 3048*, Jan. 2001.

[5] D. Thaler, "Interoperability rules for multicast routing protocols," *IETF RFC 2715*, Oct. 1999.

[6] M. Kim, H. Radha, J. Y. Lee, and H. Choo, "An efficient multicast-based binding update scheme for network mobility," *KSII Trans. Internet Inf. Syst.*, vol. 2, no. 1, pp. 23–36, Feb. 2008.

[7] H. O. Al-Misbahi and A. Y. Al-Aama, "The overlay multicast protocol (OMP): A proposed solution to improving scalability of multicasting in MPLS networks," in *Proc. Int. Conf. Wireless Broadband and Ultra Wideband Communications*, Sydney, Australia, Aug. 2007, pp. 79–84.

[8] J. Baek, P. S. Fisher, M. Jo, and H.-H. Chen, "A new reliable overlay video transport protocol for multicast agents in wireless mesh networks," *Wiley Int. J. Commun. Syst.*, Apr. 20, 2009, DOI: 10.1002/dac.1009.

[9] J. Baek and J.-F. Paris, "A scalable recovery tree construction considering packet losses correlation for reliable multicast," *KSII TTrans. Internet Inf. Syst.*, vol. 2, no. 2, pp. 82–102, Apr. 2008.

[10] S. Banerjee, C. Kommareddy, K. Kar, B. Bhattacharjee, and S. Khuller, "Construction of an efficient overlay multicast infrastructure for real-time applications," in *Proc. IEEE INFOCOM*, San Francisco, CA, Apr. 2003, pp. 1521–1531.

[11] G.-I. Kwon and J. W. Byers, "ROMA: Reliable overlay multicast with loosely coupled TCP connections," in *Proc. IEEE INFOCOM*, Hong Kong, China, Mar. 2004, pp. 385–395.

[12] Y. Li, S. Peng, and W. Chu, "K-tree trunk and a distributed algorithm for effective overlay multicast on mobile ad hoc networks," in *Proc. Int. Symp. Parallel Architectures, Algorithms, and Networks*, Sydney, Australia, May 2008, pp. 53–58.

[13] T. Morin, "Requirements for multicast in layer 3 provider—provisioned virtual private networks (PPVPNs)," *IETF RFC 4384*, Apr. 2007.

[14] S. Y. Shi and J. S. Turner, "Routing in overlay multicast networks," in *Proc. IEEE INFOCOM*, New York, Jun. 2002, pp. 1200–1208.

[15] J. Jo and J. Cho, "Cross-layer optimized vertical handover schemes between mobile WiMAX and 3G networks," *KSII Trans. Internet Inf. Syst.*, vol. 2, no. 4, pp. 171–183, Aug. 2008.

[16] R. Stewart *et al.*, "Stream control transmission protocol (SCTP) dynamic address reconfiguration," *IETF Internet Draft, draft-ieft-tsvwg-addip-sctp-08.txt*, Jun. 2004.

[17] R. Cohen and G. Kaempfer, "A unicast-based approach for streaming multicast," in *Proc. IEEE INFOCOM*, Anchorage, AK, Apr. 2001, pp. 440–448.

[18] Y. Zhu, B. Li, and J. Guo, "Multicast with network coding in application-layer overlay networks," *IEEE J. Select. Areas Commun.*, vol. 22, no. 1, pp. 107–120, Jan. 2004.

[19] B. Zhang and H. T. Mouftah, "Providing multicast through recursive multicast," *IEEE Commun. Mag.*, vol. 43, no. 1, pp. 115–121, Jan. 2005.

[20] S. Floyd *et al.*, "A reliable multicast framework for lightweight sessions and application-level framing," *IEEE/ACM Trans. Netw.*, vol. 5, pp. 784–803, Dec. 1997.

[21] B. Whetten and G. Taskale, "The overview of reliable multicast transport protocol II," *IEEE Netw.*, vol. 14, no. 1, pp. 37–47, Feb. 2000.

[22] K. Guo and I. Rhee, "Message stability detection for reliable multicast," in *Proc. IEEE ICC*, New Orleans, LA, Jun. 2000, pp. 814–823.

[23] Z. Xiao, K. P. Birman, and R. Renesse, "Optimizing buffer management for reliable multicast," in *Proc. Int. Conf. Dependable Systems and Networks*, Washington, DC, Jun. 2002, pp. 187–202.

[24] K. P. Birman *et al.*, "Bimodal multicast," *ACM Trans. Comput. Syst.*, vol. 17, no. 2, pp. 41–88, May 1999.

[25] M. Costello and S. McCanne, "Search party: Using randomcast for reliable multicast with local recovery," in *Proc. IEEE ICC*, Vancouver, BC, Canada, Jun. 1999, pp. 1256–1264.
[26] O. Jung, J. Park, and S. Kang, "Design requirement of overlay multicast session manager," in *Proc. Int. Conf. Advanced Communication Technology*, Phoenix Park, Korea, Feb. 2005, pp. 269–272.
[27] F. K. Yong, W. T. Chee, and S. Ramadass, "M-SCTP: Transport layer multicasting protocol," in *Proc. National Computer Science Postgraduate Colloq.*, Penang, Malaysia, Jun. 2005, pp. 1–4.
[28] G. Ye, T. N. Saadawi, and M. J. Lee, "Improving stream control transmission protocol performance over lossy links," *IEEE J. Select. Areas Commun.*, vol. 22, no. 4, pp. 727–736, May 2004.
[29] K. J. Lee, S. S. Nam, and B. I. Mun, "SCTP efficient flow control during handover," in *Proc. IEEE WCNC*, New Orleans, LA, Apr. 2006, pp. 69–73.
[30] D. P. Kim, D. H. Lee, S. J. Koh, and Y. J. Kim, "Adaptive primary path switching for SCTP handover," in *Proc. 10th Int. Conf. Advanced Communication Technology*, Phoenix Park, Korea, Feb. 2008, pp. 900–902.
[31] M. Dahal and D. Saikia, "RTT based congestion control and path switching scheme for SCTP," in *Proc. Int. Conf. Communication Technology*, Guilin, China, Nov. 2006, pp. 1–4.
[32] S. Kashihara, T. Nishiyama, K. Iida, H. Koga, Y. Kadobayashi, and S. Yamaguchi, "Path selection using active measurement in multi-homed wireless network," in *Proc. 2004 Int. Symp. Applications and the Internet*, Tokyo, Japan, Jan. 2004, pp. 273–276.
[33] Y. Qiao *et al.*, "SCTP performance issue on path delay differential," *Lecture Notes Comput. Sci.*, vol. 4517, pp. 43–54, Jun. 2007.
[34] J. Funasaka, K. Ishida, H. Obata, and Y. Jutori, "A study on primary path switching strategy of SCTP," in *Proc. Workshop Cooperative Computing, Internetworking and Assurance*, Chengdu, China, Apr. 2005, pp. 536–541.
[35] M. Shin and W. A. Arbaugh, "Efficient peer-to-Peer lookup in multi-hop wireless networks," *KSII Trans. Internet Inf. Syst.*, vol. 3, no. 1, pp. 5–25, Feb. 2009.
[36] J. Mahdavi and S. Floyd, TCP-friendly Unicast Rate-Based Flow Control, Nov. 2009. [Online]. Available: http://www.psc.edu/networking/papers/tcp_friendly.html.

Jinsuk Baek received the B.S. and M.S. degrees in computer science and engineering from Hankuk University of Foreign Studies (HUFS), Seoul, Korea, in 1996 and 1998, respectively, and the Ph.D. degree in computer science from the University of Houston (UH), Houston, TX, in 2004.

He is an Associate Professor of computer science at the Winston-Salem State University (WSSU), Winston-Salem, NC. He is the director of Network Protocols Group at the WSSU. His research interests include wireless sensor networks, multicast protocols, mobile computing, network security, and multimedia communications.

Dr. Baek has served and is serving as an active member for various technical activities in the computer network and communication field. Currently, he is serving as an editor of the *KSII TIIS Journal*, the technical committee of IEEE VTC 2010, ICONI 2010, and a reviewer for four leading journals, including *Wiley Wireless Communications and Mobile Computing*, IEEE TRANSACTIONS ON MOBILE COMPUTING, IEEE TRANSACTIONS ON BROADCASTING, and *Hindawi Journal of Computer Systems, Network, and Communications*.

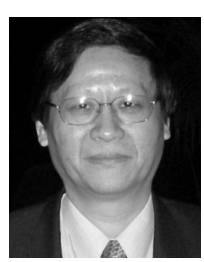
Paul S. Fisher received the B.A. and M.A. degrees in mathematics from University of Utah, Salt Lake City, and the Ph.D. degree in computer science from Arizona State University, Tempe.

He is R. J. Reynolds Distinguished Professor of Computer Science at the Winston-Salem State University (WSSU), Winston-Salem, NC. He is the director of High Performance Computing Group at the WSSU. He has written and managed more than 100 proposal efforts for corporations and DoD involving teams of 1 to 15 people. He worked as a consultant to the U.S. Army, U.S. Navy, U.S. Air Force, and several companies over the years. In the 1990s, he commercialized an SBIR-funded effort and built Lightning Strike, a wavelet compression codec, and then sold the company to return to academe. His current research interests include wired/wireless communication protocols, image processing, and pattern recognition.

Minho Jo received the B.S. degree in industrial engineering from Chosun University, Gwangju, Korea, and the Ph.D. degree in computer networks from the Department of Industrial and Systems Engineering, Lehigh University, Bethlehem, PA, in 1994.

He worked as a Staff Researcher with Samsung Electronics, Korea, and was a Professor at the School of Ubiquitous Computing and Systems, Sejong Cyber University, Seoul, Korea. He is now a Research Professor at the Graduate School of Information Management and Security, Korea University, Seoul. His current interests lie in the area of wireless sensor networks, RFID, wireless mesh networks, security in communication networks, machine intelligence in communications, Wireless Body Area Networks (WBAN), and ubiquitous and mobile computing.

Prof. Jo is Executive Director of the Korean Society for Internet Information (KSII) and Board of Trustees of the Institute of Electronics Engineers of Korea (IEEK), respectively. He is Founding Editor-in-Chief and Chair of the Steering Committee of *KSII Transactions on Internet and Information Systems*. He serves as an Editor of IEEE NETWORK. He is Editor of the *Journal of Wireless Communications and Mobile Computing*, and Associate Editor of the *Journal of Security and Communication Networks* published by Wiley, respectively. He serves on an Associate Editor of the *Journal of Computer Systems, Networks, and Communications* published by Hindawi. He served as Chairman of IEEE/ACM WiMax/WiBro Services and QoS Management Symposium, IWCMC 2008. He is the TPC Chair of IEEE Vehicular Technology Conference 2010 (VTC 2010-Fall). He is General Chair of International Ubiquitous Conference, and Co-Chair of the International Conference on Ubiquitous Convergence Technology. He is Technical Program Committee of IEEE ICC 2008 and 2009 and IEEE GLOBECOM 2008 and 2009 and TPC Chair of CHINACOM 2009 Network and Information Security Symposium.



Hsiao-Hwa Chen (S'89–M'91–SM'00–F'10) received the B.Sc. and M.Sc. degrees from Zhejiang University, Hangzhou, China, and the Ph.D. degree from the University of Oulu, Oulu, Finland, in 1982, 1985, and 1991, respectively.

He is currently a Distinguished Professor in the Department of Engineering Science, National Cheng Kung University, Tainan City, Taiwan. He has authored or coauthored over 400 technical papers in major international journals and conferences, six books, and more than ten book chapters in the areas of communications.

Prof. Chen served as the general chair, TPC chair, and symposium chair for many international conferences. He served or is serving as an Editor or/and Guest Editor for numerous technical journals. He is the founding Editor-in-Chief of *Wiley's Security and Communication Networks Journal* (www.interscience.wiley.com/journal/security). He is the recipient of the best paper award in IEEE WCNC 2008 and a recipient of the IEEE Radio Communications Committee Outstanding Service Award in 2008. He is a Fellow of IET and a Fellow of BCS.