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The Outbound Voice Traffic Reduction Scheme in the Push-To-Talk Environment

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ABSTRACT: *The push-to-talk technology provides a walkie-talkie like service for group communication. In this paper, we propose an efficient outbound voice traffic reduction scheme in the push-to-talk environment when network congestion happens. The Open Mobile Alliancepush-to-talk over cellular (OMA PoC) architecture, the standard for this kind of group communication, is used as a base network in this paper.The PoC server plays an important role in copying and forwarding of received RTP media packets and the outbound voice traffic of this PoC Server tends to be affected by other traffic when network congestion happens. We extend the functionality of the Controlling PoC Server to accommodate the location information of PoC Clients by co-operating the location management server, which also communicates with the location reporting and packet forwarding agent node. By minimizing the outbound voice traffic of the PoC server, we improve the overall performance in terms of the end-to-end quality which is derived from the E-model. Simulation results show that the proposed scheme outperforms the existing PoC scheme when network congestion happens.*

Keywords: *Push-to-talk, group communication, OMA PoC architecture, voice traffic, network congestion*

I. INTRODUCTION

Recently many Internet applications have been used widely in our lives for the proliferation of smart phones. Especially the concept of the VoIP has been used in many applications such as Skype, Viber, Voxer, TiKL *etc*., which have potentials to replace the traditional voice communication through the public switched telephone network. There are many terms such as mVoIP, VoWLAN, and VoLTE that show the proliferation of this kind of packetized voice communication. Among applications which support VoIP, there are several VoIP applications such as Voxer, and TiKL which support the group communication as well as one-to-one communication, which is also known as the push-to-talk (PTT or P2T) [1, 2, 3]. This push-to-talk technology supports the half duplex communication among end-to-end users like the trunked radio system (TRS). Nextel communications was the first company to provide the push-to-talk service commercially using iDEN.

The Open Mobile Alliance (OMA) has standardized the technology for the push-to-talk over cellular (PoC) to support group communication among mobile subscribers through wireless networks since 2005. The OMA PoC architecture defines functional entities for the network configuration with the SIP/IP core including several SIP proxies and SIP registrars to support the push-to-talk service [1]. The control plane protocol is used as a signaling protocol which extends some of SIP procedures for the group communication [4] and the user plane protocol is used to support media packet transmission and control the talk right [5].

The well-known problem in VoIP applications including PTT or OMA PoC is that the VoIP capacity is limited in wireless networks due to the small periodic voice data. Several researches on the VoIP system capacity in the IEEE 802.11 wireless LAN (WLAN) have shown that it has a very small threshold compared to the link bandwidth due to its medium access control (MAC) protocol [6, 7]. In addition, we observed that the outbound traffic of a PTT/PoC server, which receives a talker's voice data as an inbound traffic and transmits its copy to other group members, is seriously influenced by the other network traffic when network congestion happens. We will show the performance degradation of media packet flows through the simulation study.

In this paper, we propose an efficient scheme which reduces the outbound traffic of a PoC server, by which it decreases the packet delay and the packet loss caused by the network congestion. An efficient mechanism is provided in this paper to detect PoC Clients' registration and exchange control messages to manage the location information of PoC Clients. Also the PoC Server functionality is extended to use this location information of PoC Clients to reduce the outbound voice traffic. Simulation results show the performance improvement by the reduction of the outbound traffic. The simulation was performed using *ns-3* [8].The mean opinion score is used as the main performance metric to measure the end-to-end quality, which can be derived from the E-model.

In Section 2, we describe the OMA PoC standard related to this paper. In Section 3 and Section 4, we present the proposed scheme to reduce the outbound voice traffic in the OMA PoC environment and show the performance improvement through the simulation study, respectively. Finally, we give a conclusion in Section 5.

II. OMA PoC

In this section, we describe the OMA PoC standard related to the proposed scheme. The OMA PoC standard is mainly divided into the control plane protocol [4] which is a signaling protocol similar the SIP [9] and the user plane protocol [5] which carries user's voice traffic based on RTP [10].

2.1 Session Establishment

Now we describe some of the control plane protocol related to the proposed scheme. Fig.1 shows the brief OMA PoC architecture which includes functional entities only related to the proposed scheme. The service logic for SIP sessions are implemented in the application server using SIP/UDP/IP. The application server functionality is implemented by the PoC server when the SIP/IP Core for the PoC service is according to 3GPP/3GPP2 IP Multimedia Subsystem (IMS) [4]. Thus the SIP/IP Core and PoC Server functionalities may be in one physical entity. Media packets carrying users' voice data and the talk/media burst control for managing the talk right are transferred between PoC Clients and a PoC Server using RTP/UDP/IP [5].

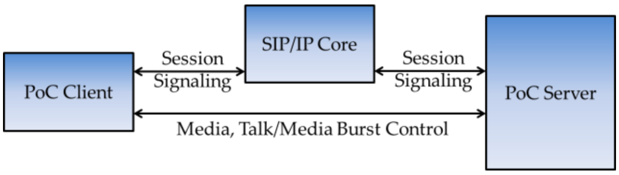

Figure 1. A brief OMA PoC architecture

The PoC Server performs either the Controlling PoC Function or the Participating PoC Function. In this paper, we call the PoC Server with the Controlling PoC Function and the Participating PoC Function as the Controlling PoC Server and the Participating PoC Server in short. The Controlling PoC Server mainly performs the management of PoC sessions such as the session establishment and the media burst control [4].The Participating PoC Server performs relays the Talk Burst and Media Burst Control messages between the PoC Client and the Controlling PoC Server and may relay RTP media packets from the Controlling PoC Server.

Each PoC Client should register to their Participating PoC Server prior to participating in the PoC session according to rules and procedures of RFC 3261 [9] with extended headers including PoC feature tags [4]. Fig.2 shows the registration procedure of the PoC Client. In the SIP REGISTER request of the PoC Client, information such as the SIP URI and IP address of the PoC Client can be found. This information is used in the proposed scheme to keep the location information of PoC Clients.

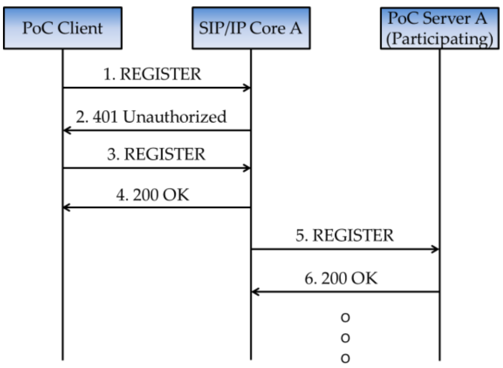

Figure 2. Registration procedure

PoC Session establishment is also made according to rules and procedures of RFC 3261 with extended headers including PoC feature tags as shown in Fig. 3 [4]. For simplicity, messages from/to the SIP Core are excluded from Fig.3 and only messages from/to OMA PoC entities are shown in the figure. Dotted arrows represent MBCP (media burst control protocol) messages,

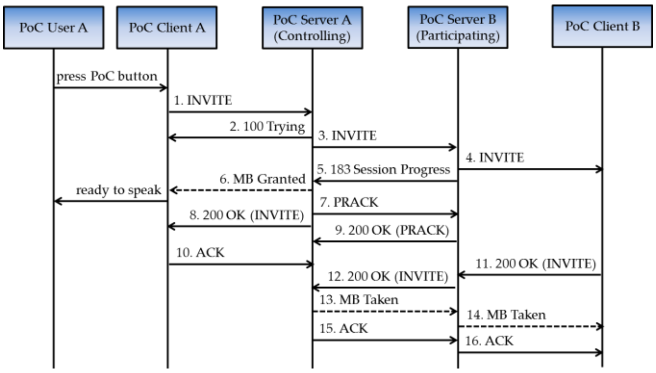

Figure 3. Ad hoc PoC Session establishment on-demand Session

which manages the talk right of PoC Clients [5]. There are four kinds of PoC Sessions: 1-1, ad-hoc, pre-arranged, and chat [4, 11]. In this paper, we are interested in the ad-hoc PoC Session and the pre-arranged PoC Session. In the ad-hoc PoC Session, the group information can be found from the recipient list in the SIP INVITE request. In the pre-arranged PoC Session, the group information is maintained by the Controlling PoC Server. Thus, we can find the group information when the SIP INVITE request is arrived at the PoC Server. A PoC Session can also be classified into the on-demand session and the pre-established session according to the time of the session establishment. The on-demand session is started when a user initiates the PoC Session with his/her recipient list [4]. The pre-established PoC Session is another method for the session establishment, which first makes a parameter negotiation to establish a PoC Session and RTP packet transmission is performed if required [4].

The determination of the PoC Server role of either Controlling PoC Function or Participating PoC Function takes place during the PoC Session setup and lasts for the duration of the whole PoC Session. In ad hoc PoC group sessions, the controlling PoC server is the PoC server of the inviting user. In pre-arranged PoC group sessions, the controlling PoC server is the PoC server hosting the pre-arranged PoC group. That is, the Controlling PoC server is the PoC server of the domain that owns the URI that identifies the pre-arranged PoC group.

2.2 Media Transfer

After either the ad hoc or the pre-arranged PoC Session is established, media packets including voice data are transferred between the PoC Server and PoC Clients. These media packets are transported by RTP/UDP/IP [5]. The PoC Server forwards received RTP Media packets towards all other PoC Clients that are not on hold [5], that is, the received packet is copied and forwarded to other PoC Clients in a group. The basic RTP header of media packets is described in RFC 3550 [10] and additional RTP payload formats for various audio/video codecs such as G.729A, GSM, EVRC and AMR are described in several RFC documents [12, 13, 14]. From the SSRC field of the RTP header, the PoC Server and PoC Clients can find who sends the RTP packet.

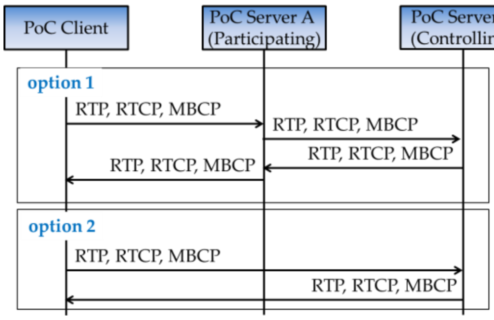

Figure 4. Media transfer options

Fig.4 shows two options that transfer voice data between PoC clients. In Option 1, the Participating PoC Server is in charge of transferring voice packets and also provides the filtering function and the transcoding function among different codecs. However the performance may be degraded due to the increased packet delay compared to Option 2 in which voice packets and other related packets are transmitted directly through the Controlling PoC Server. In this paper, we only consider Option 2 to support group communication. The incoming traffic and the outgoing traffic of a PoC Server are unbalanced because of its one-to-many communication pattern. Thus problems such as the traffic handling at a PoC Server and packet transmission to recipient PoC Clients occur especially when network congestion happens.

III. Voice Traffic Reduction Scheme

In this section, we describe the basic concept and the algorithms used for the proposed scheme.

3.1 Basic Concept

Fig.5 shows the basic concept of the proposed scheme. We consider the situation of network congestion like Fig.5(a) which can make the transmission problem. For group communication, the Controlling PoC Server copies received RTP media packets from a PoC Client which has the media burst right and forwards those copies to all other PoC Clients in a group. Those RTP media packets are forwarded through the Internet with other packets generated by network applications such as FTP, email, web, *etc*. The nature of the Internet is basically the best-effort network due to the characteristics of the layer 3 protocol, IP. Thus this small amount of periodic RTP media packets can be easily damaged by other packets travelling the Internet, which will be shown in Section 4 through the simulation study. Dotted arrows in Fig.5 (a) indicate the possible packet loss while forwarding RTP media packets from the Controlling PoC Server to other PoC Clients.

The concept of the proposed scheme is shown in Fig.5 (b). Two functional entities have an important role in implementing the proposed scheme. The first one is the Controlling PoC Server, which exists in the OMA PoC standard. Its functionality will be extended by including a function to manage the location information of PoC Clients with the collaboration of a location management server which is newly introduced for the proposed voice traffic reduction scheme. The other one is the location reporting and packet forwarding (LRPF) agent functionality, which supports to find OMA PoC Clients in the same network and manages received RTP media packets. The LRPF agent functionality can be easily implemented in routers, gateways or wireless access points in order to reduce the outbound voice traffic from the Controlling PoC Server. For example, a gateway and an access point are used in Fig.5 (b). Both the gateway and the access point are assumed that they are located in one-hop distance far from PoC Clients so that it is possible to forwarding media packets efficiently, which still has the voice capacity problem if it is a wireless network [6, 7].By reducing the outbound voice traffic from the Controlling PoC Server as small as possible, the interference from other traffic sources can be avoided significantly.

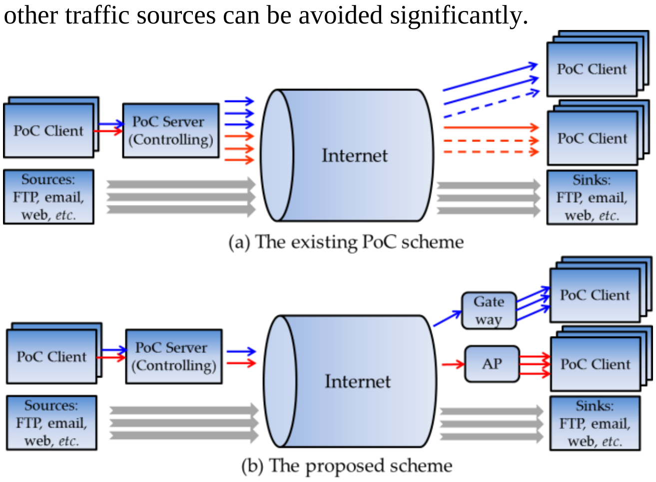

Figure 5. The basic concept of the proposed scheme

3.2 Algorithms

First we define the notation used for the operations used in the proposed scheme.

- G : the set of groups, $G = \{G_i \mid 0 \le i < k\}$
- G_i : the set of group members in the i-th group,
  $G_i = \{m_j^i \mid 0 \le j < n\}$ for $n \ge 0$
- m_j^i : the j-th member of the i-th group
- V : the set of LRPF agent nodes,
  $V = \{v_i \mid 0 \le i < l\}$
- $M_{v_i}^{G_x}$: the set of active group members of LRPF agent node v_i for the given group G_x
- C_{v_i} : the set of PoC Clients which are located with LRPF agent node v_i
- F_{sid} : the forwarding set of the current session, *sid*, for received RTP media packets
- M_{sid} : the set of active group member sets for v_i's of the current session *sid*

Now we explain the operations for the LRPF agent functionality in Fig.6, which can be divided into three operations for PoC Client management, session establishment and RTP packet management. These operations are related to those operations in the location management server, which maintains the location information of PoC Clients and the group information. In

Operation I, the LRPF agent collects the information of PoC Clients in the same network by packet sniffing SIP REGISTER requests to Participating PoC Servers and then sends the information to the location management server. Operation II will be performed while the PoC Session is established. The Controlling PoC Server will send the information of a forwarding set for the new PoC session, which will be used for RTP packet forwarding in Operation III. We call the node which performs the LRPF agent functionality the LRPF agent node.

Operation I. PoC Client Management

1. do {
2. get a packet copy from every upward packets;
3. if the packet type is the SIP REGISTER request with the PoC feature tag {
4. send the tuple (v_i, m_k^j) to the location management server;
5. }
6. } while (true);

Operation II. Session Establishment

1. do {
2. wait the tuple message (sid, $M_{v_i}^{G_x}$) from the Controlling PoC Server;
3. make a forwarding set $F_{sid} = \{ m_k^x \mid m_k^x \in M_{v_i}^{G_x} \}$;
4. } while (true);

Operation III. RTP Packet Management

1. do {
2. wait a RTP packet;
3. get the session information, *sid*, from the RTP packet;
4. for every PoC Client $m_k^x \in F_{sid}$ {
5. make a copy of the received RTP packet;
6. send the copy to the PoC Clients m_k^x ;
7. }
8. } while (true);

Figure 6. Operations for the LRPF agent functionality

Operation I. LRPF Agent Node Management

1. do {
2. wait a tuple message (v_i, m_k^j) from any LRPF agent node;
3. add the PoC Client m_k^j to the PoC Client set of v_i, i.e., $C_{v_i} \leftarrow C_{v_i} \cup \{ m_k^j \}$
4. } while (true);

Operation II. Session Establishment

1. do {
2. wait an INFO request from the Controlling PoC Server;
3. get the set of group members $G_x = \{ m_k^x \}$ from the request;
4. prepare a forwarding set $F_{sid} = \{v_i | m_k^x \in C_{v_i} \}$ for every $m_k^x \in G_x$;
5. prepare the set of active group member sets for v_i's of the current session $M_{sid} = \{ (v_i, M_{v_i}^{G_x}) \}$, where $M_{v_i}^{G_x} = \{m_k^x | m_k^x \in G_x \bigwedge m_k^x \in C_{v_i} \}$ for every $v_i \in F_{sid}$;
6. send an INFO response with the tuple (G_x, F_{sid}, M_{sid}) to the Controlling PoC Server;
7. } while (true);

Figure 7. Operations for the location management server

Fig.7 shows the operations for the location management server (LMS), of which the functionality is introduced to manage the relationship between LRPF agent nodes and PoC Clients. In Operation I, LMS maintains the information of PoC Clients for each LRPF agent node while it receives a message from a certain LRPF agent node. In Operation II, LMS computes the forwarding set, F_{sid}, and the set of active group member sets, M_{sid}, for LRPF agent nodes for a given set of group members G_x whenever it receives an INFO request from the Controlling PoC Server. This information will be sent to the Controlling PoC Server and then sent to LRPF agent nodes from the Controlling PoC Server for RTP packet forwarding. This LMS functionality should be introduced since we cannot determine which PoC Server performs the Controlling PoC Function until the session establishment by the SIP INVITE request is started. One of PoC Servers in the system may perform this LMS functionality. Two messages of type INFO request and INFO response are introduced in the proposed scheme to exchange the required information between the Controlling PoC Server and LMS.

Operations for the Controlling PoC Server are described in Fig.8. Operation I is for the session establishment. We assume that the on-demand signaling is used for the session establishment, which enables the session initiation by looking up the SIP INVITE request. Whenever the Controlling PoC Server receives an SIP INVITE request, it retrieves the forwarding set, F_{sid}, and the set of active group member sets, M_{sid}, for LRPF agent nodes from LMS, which will be done by exchanging the INFO request and the INFO response. The information in the INFO response will be sent to all the necessary LRPF agent nodes for RTP packet transmission. Operation II is involved in the RTP packet management. When the Controlling PoC Server receives a RTP packet from a PoC Client, it forwards its copies to LRPF agent nodes in the forwarding set, F_{sid}, not to all PoC Clients. This makes the main difference between the existing PoC scheme and the proposed scheme. The outbound voice traffic is reduced to the amount of difference between the number of LRPF agent nodes and the number of PoC Clients. In general, the number of PoC Clients may be larger than the number of LRPF agent nodes because LRPF agent nodes are either gateways or access points and PoC Clients belong to one of them. Thus we can find easily that the outbound voice traffic will be reduced significantly if most PoC Clients are under the management of a few LRPF agent nodes.

Operation I. Session Establishment

1. do {
2. wait an SIP INVITE request with the PoC feature tag;
3. get the set of group members $G_x = \{ m_k^x \}$ in the SIP INVITE request;

4. set the session id, sid, for the SIP INVITE request;
5. send an INFO request to the LMS with the group information G_x ;
6. receive an INFO response to get a tuple message (G_x, F_{sid}, M_{sid}) from the LMS;
7. send the tuple (sid, $M_{v_i}^{G_x}$) to the corresponding LRPF agent node v_i for every$(v_i, M_{v_i}^{G_x}) \in M_{sid}$;
8. } while (true);

Operation II. RTP Packet Management

1. do {
2. wait a RTP packet;
3. get the session information, sid, from the RTP packet;
4. for every LRPF agent node $v_i \in F_{sid}$ {
5. make a copy of the received RTP;
6. send the copy to the LRPF agent node v_i ;
7. }
8. } while (true);

Figure 8. Operations for the Controlling PoC Server

Fig.9 shows an example of the proposed scheme. For simplicity, we assume that two groups exist: $G_0 = \{m_0^0, m_1^0, m_2^0, m_3^0\}andG_1 = \{m_0^1, m_1^1, m_2^1\}$. We also assume that PoC Clients $m_j^i$are located as shown in Fig.9 and thus they can be accessed through LRPF agent nodes v_0, v_1, and v_3. While PoC Clients register to their PoC Server, LRPF agent nodes collect and send the information to LMS, which is shown as dotted lines in Fig.9. Thus LMS has the information for PoC Clients and this information is used for processing the INFO request of the Controlling PoC Server. The information in the INFO response is used in Operation I and II of the Controlling PoC Server. The set of active group members of an LRPF agent node is sent to each LRPF agent node involved in the new PoC Session, which is used as forwarding set in an LRPF agent node to guarantee that invited PoC Clients will receive RTP media packets correctly.

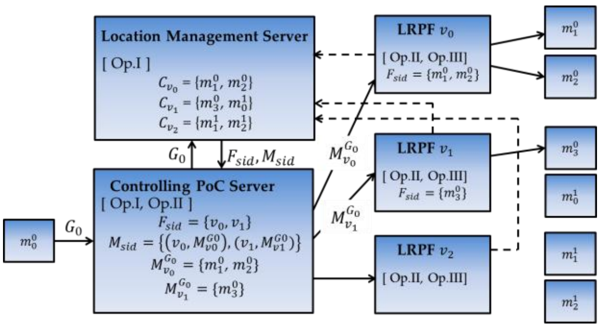

Figure 9. An example of the proposed scheme

IV. Performance Evaluation

In this section, we perform the simulation study to see the effect of the proposed scheme by reducing the outbound voice traffic of the Controlling PoC Server.

4.1 Assumptions

Fig.10 depicts the network configuration used for the simulation. The PoC Server is placed in node 0 and a number of FTP sources are placed in node 3 and node 4 is used as FTP sinks to see the effect of network congestion. In the shaded area of Fig.10, RTP packets for PoC compete with TCP packets for FTP for network resources to be transmitted to their destinations. src(i) and dest(i,j) nodes belong to the i-th group, where $0 \le i < nGr$ and $0 \le j < nGm$. nGr and nGm are the number of groups and the number of group members per group, respectively. RTP packets generated by src(i) are transferred to the PoC Server in node 0 and then they are copied and forwarded to dest(i,j) nodes for group communication. In the proposed scheme, we assume that both the LMS functionality and the PoC Server functionality are located in node 0 and the LRPF agent functionality is in every gateway, *i.e.*, node 6, of destination networks.

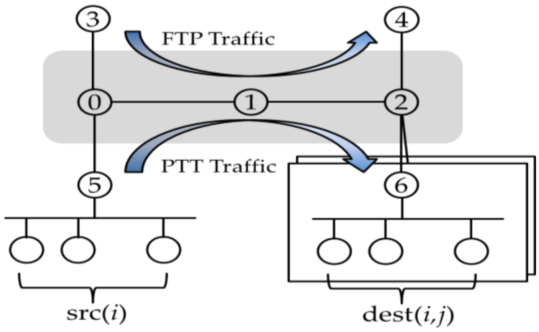

Figure 10. Network configuration for the simulation

We use the discrete-event network simulator *ns-3* [6] for the simulation study, which has been introduced as a replacement of the existing network simulator *ns-2*. To connect nodes, the class *PointToPointChannel* is used with attributes *DataRate* of 100Mbps and *Delay* of 1msec. src(i)'s and node 5 are connected using the class *CsmaChannel* with attributes *DataRate* of 100Mbps and *Delay* of 0.1msec to model the Ethernet although ns-3 does not support the collision detect functionality. Destination networks are also connected using *CsmaChannel*. The queue size of *DropTailQueue* for both *PointToPointChannel* and *CsmaChannel* are fixed to 200. By this configuration, we can focus on the effect of network congestion for the outbound voice traffic of the PoC Server. Start times of src(i) and FTP sessions are uniformly distributed in the time interval (1,3). G.729A is used as a voice codec since its bandwidth is low enough to be used in VoIP applications through the Internet [15]. We first assume that there is no bit error in transmission and then we perform the simulation to see the effect of the bit error.

4.2 Simulation Results

We compare the performance of the proposed scheme with that of the existing PoC or PTT scheme in terms of MOS (mean opinion score) which is measured by using the E-model [16]. In legends of the following figures, 'ptt' and 'prp' represent the existing PoC/PTT scheme and the proposed scheme, respectively. We assume that the number of groups is 30, 40 and 50 and the number of group members is 10.

Fig.11 shows the MOS values of the existing PoC/PTT scheme and the proposed scheme. As the number of FTP sessions increases, we can find that the proposed scheme outperforms the existing PoC/PTT scheme up to 95, 76 and 49% for 30x10, 40x10 and 50x10 group-members, respectively. MOS values of the proposed scheme are greater than 3.5 on the average, which means that some users are dissatisfied but most users are satisfied for the voice quality. But MOS values of the existing PoC/PTT scheme go down under 3.5 as the number of FTP session becomes greater than 30. Thus, we can find that the network congestion has a great effect on the performance of the existing PoC/PTT scheme. But the packet transmission of the proposed scheme was much less affected by the network congestion since the outbound voice traffic is significantly reduced by the number of group members compared to the existing PoC/PTT scheme.

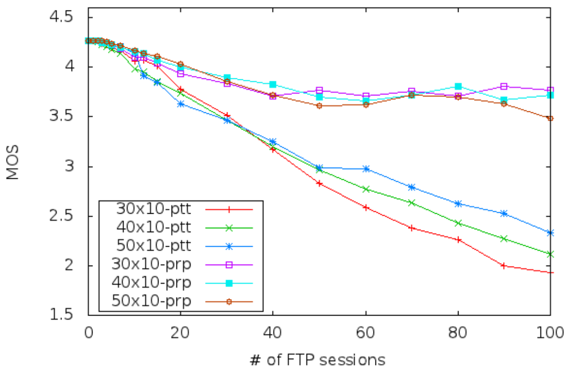

Figure 11. Comparison of the mean opinion score

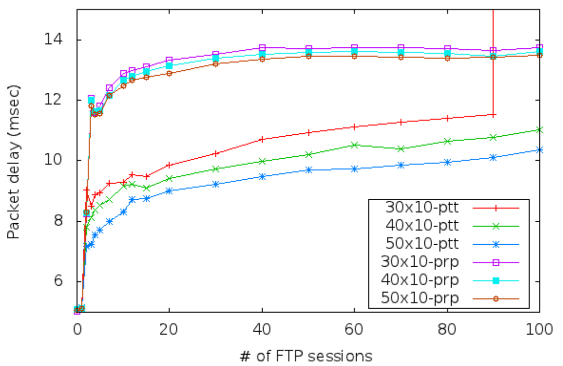

Figure 12. Comparison of the packet delay

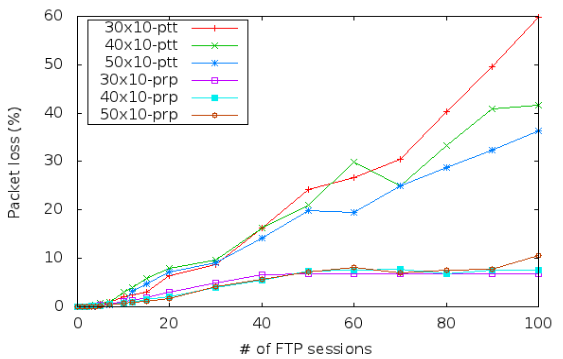

Figure 13. Comparison of the packet loss

Fig.12 and Fig.13 show the packet delay and the packet loss of the existing PoC/PTT scheme and the proposed scheme, which are two main components for the E-model. At first glance, the packet delay of the proposed scheme is larger than that of the existing PoC/PTT scheme. But the packet loss of the existing PoC/PTT scheme becomes very larger than that of the proposed scheme as the number of FTP sessions increases. When network congestion happens, we can find that lost RTP media packets are the significant factor of losing MOS values, which degrades the overall end-to-end voice quality.

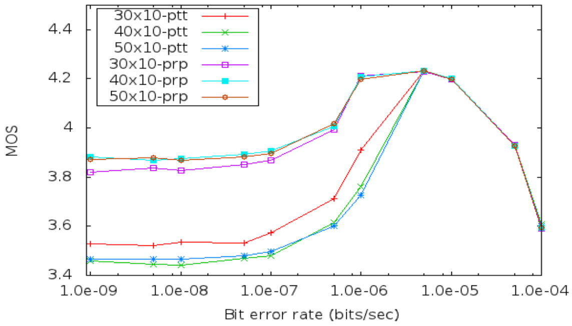

Figure 14. The effect of bit errors: MOS values (FTP=30)

Fig.14 shows MOS values when the bit error rate changes in case of 30 FTP sessions. As the bit error rate increases, the overall performance in terms of MOS values tends to be improved. When the bit error rate reaches 5e-6, all the simulation results show the best performance. After then, the overall performance degrades as the bit error rate increases. While the bit error rate is smaller than 5e-6, the FTP traffic dominantly uses the network. However, the FTP traffic makes no more influence to the PTT voice traffic while the bit error rate is greater than or equal to 5e-6 due to the congestion control of TCP which carries the FTP traffic, which will be explained in the simulation result of the packet loss in Fig.15.

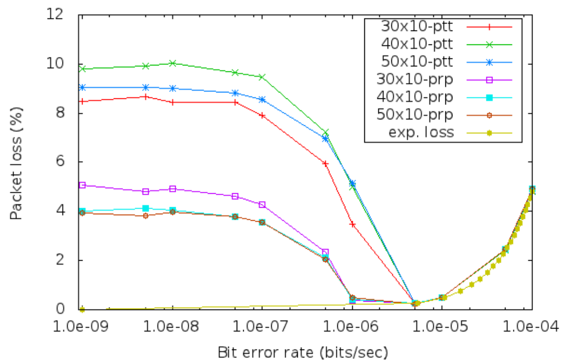

Figure 15. The effect of bit errors: packet loss (FTP=30)

The packet loss in Fig.15 shows also the same behavior as the simulation result in Fig.14. The expected packet loss, which is shown in Fig.15 as the line 'exp. loss,' can be obtained numerically. The RTP/UDP/IP header is of size 40 and G.729A generates 20 bytes per every 20 msec. Thus we can obtain the expected packet loss percentage in Eq.1 if we multiply the RTP media packet size by the bit error rate *ber*.

$$\text{Packet loss (\%)} = ber * (40 + 20) * 8 * 100 \qquad (1)$$

The difference between the actual packet loss and the expected packet loss explains the effect of network congestion which results from TCP packet flows of FTP sessions. While the bit error rate is less than 5e-6, the packet loss is mainly affected by the FTP traffic. In this case, the number of FTP sessions was assumed to be 30 so that there may be many TCP packet flows through the network, which prevent the outbound voice traffic from being forwarded to their destination PoC Clients. While the bit error rate is greater than or equal to 5e-6,the packet loss shows the same result as the expected packet loss, which means that FTP sessions has no more effect on the outbound voice traffic.

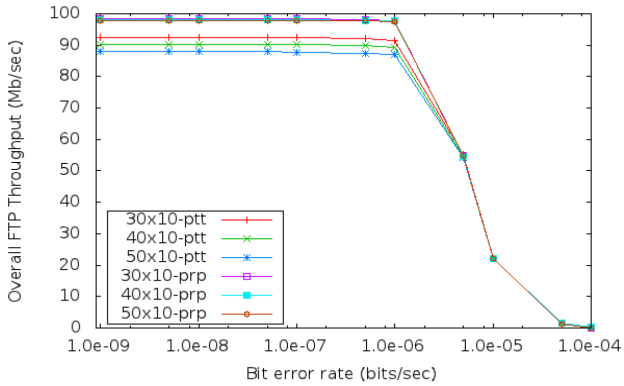

Figure 16. The effect of a bit error on the overall FTP throughput (FTP=30)

Fig.16 shows the overall FTP throughput for 30 FTP sessions as the bit error rate changes. We obtain this throughput by adding each throughput of 30 FTP sessions. While the bit error rate is less than or equal to 1e-7, there is no significant performance degradation in terms of the overall FTP throughput. While the bit error rate is greater than 1e-7 and less than 1e-6, the overall FTP throughput declines a little bit but its effect on the RTP media packet has been diminished dramatically as shown in Fig.15, which results in the increase of MOS values in Fig.14. When the bit error rate is 5e-6, the overall FTP throughput is about 55 Mbps and the outbound voice traffic is affected barely by the FTP traffic, which can be seen by the smallest packet loss in Fig.15 and the largest MOS value in Fig.14. As the bit error rate becomes greater than 5e-6, the overall FTP throughput degrades dramatically to near 150 kbps. FromFig.15 and Fig.16, we can find that the network with the bit error rate greater than 1e-7 has a significant effect on the TCP packet transmission as the bit error rate becomes larger but the small UDP packets such as RTP media packets carrying the voice data may be affected barely until the bit error rate becomes larger than 5e-6.

The packet delay in Fig.17 also shows the same behavior at the center of the bit error rate 5e-6. While the bit error rate is greater than or equal to 5e-6, the packet delay seems very low. However, this low packet delay is meaningless because the packet loss is too high to get good end-to-end voice quality. Although the packet delay of the proposed scheme is greater than the one of the existing PoC/PTT scheme while the bit error rate is less than 5e-6, the overall performance of the proposed scheme is much better than the one of the existing scheme due to the packet loss as shown in Fig.15.

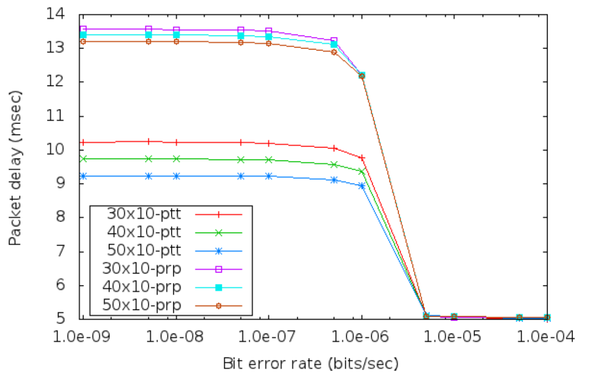

Figure 17. The effect of bit errors: packet delay (FTP=30)

V. Conclusion

In this paper, we proposed the efficient outbound voice traffic reduction scheme to improve the VoIP performance in the push-to-talk environment. The OMA PoC architecture has been standardized since 2005 to support the push-to-talk environment for the cellular network. In this paper, we used the OMA PoC architecture as the base network. We focused on the network performance in case of the network congestion since it has a great effect on the packet transmission. We have minimized the effect of the network congestion by reducing the outbound voice traffic of the Controlling PoC Server. We have presented three functionalities for the proposed scheme. The first one is the location reporting and packet forwarding (LRPF) agent functionality to obtain the location information of PoC Clients. The LRPF agent functionality can be located to either gateways or wireless access points because these nodes can easily see the packet transmission between PoC Clients in their network and the PoC Server. The second one is the location management server (LMS) functionality which is introduced to manage the location information of PoC Clients because the Controlling PoC Server can be determined after the SIP INVITE request is issued for the group communication and it is necessary to store the location information of PoC Clients somewhere. The last one is for the Controlling PoC Server where RTP media packets including voice data are copied and forwarded to their destination PoC Clients. The LMS functionality can be merged into one of PoC Servers for simplicity.

To see the performance of the proposed scheme, the discrete-event network simulator *ns-3* was used. We performed the simulation study in case of the network congestion which was caused by the FTP traffic using the same network resource. By reducing the outbound voice traffic of the Controlling PoC Server, the overall performance in terms of MOS has been greatly improved. We have shown the effect of TCP packets flows of FTP sessions on the packet delay and the packet loss, which are two main components affecting the E-model to obtain MOS values. Also we have shown the effect of the bit error rate on the overall performance of the proposed scheme. The bit error rate has a significant effect on TCP packets due to the TCP congestion control, but it has little effect on the RTP media packet until the bit error rate becomes larger than 5e-6. Thus the proposed scheme outperforms the existing PoC/PTT scheme significantly when network congestion happens. Other network congestion scenarios will be considered as a further research in case that either the UDP traffic such as media streaming or mixed traffic of FTP and UDP packets are used.

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