User-Perceived QoS Performance Enhancement for VoIP Services in IEEE 802.16 Systems

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Abstract—This paper has evaluated the system performance for the voice over Internet protocol (VoIP) services supported by the IEEE 802.16 systems in terms of the packet loss and delay, and it has also calculated the mean opinion score (MOS) in order to estimate the user’s satisfaction for the VoIP services. By the simulation results, the MOS of the VoIP service supported by the unsolicited grant service (UGS) without auto repeat request (ARQ) has been decreased by 1.5 when the channel quality is poor. The reason of the MOS reduction is the increase of the packet loss due to the channel error. For this reason, this paper has considered the ARQ in order to compensate the packet loss when the channel quality is poor. The simulation results have shown that the extended real-time polling service (ertPS) and rtPS with ARQ can compensate the packet loss even if the channel quality is poor, however they can cause the reduction of the VoIP capacity because of the resource used for the retransmission. From the simulation results, the ertPS could save the resource by about 25% compared to the rtPS with guaranteeing a quality of service (QoS).

Keywords-VoIP, End-to-End QoS, IEEE 802.16 Systems, ARQ, Scheduling

I. INTRODUCTION

VoIP services have been considered as one of the most important service in the next generation wireless systems. For this reason, many researchers have investigated on the QoS provisioning technology for VoIP services. Particularly, most of the researches have focused on decreasing the delay because VoIP services are sensitive to the delay. In the IEEE 802.16 standards, the unsolicited grant service (UGS), the real time polling service (rtPS) and the extended real time polling service (ertPS) have been defined for providing a VoIP service [1]. In these scheduling types, a base station (BS) periodically allocates the grant to transmit a packet to a subscriber station (SS). The reason of this is to reduce the delay which is caused by the contention scheme for the bandwidth request. However, the uplink scheduling types have been designed without considering the retransmission scheme such as the auto repeat request (ARQ) and the hybrid ARQ (HARQ) for the VoIP services in the IEEE 802.16 systems. Unfortunately, the packet loss rate can be increased when an SS moves to the cell boundary. In [2], the authors have represented that the packet loss is more sensitive than the delay for the quality of VoIP services by the MOS. For this reason, the packet loss can have a serious impact on the QoS for the VoIP services. Therefore, this paper evaluates the uplink QoS performance of a VoIP service according to the channel condition considering the retransmission scheme for the VoIP service.

The IEEE 802.16 considers the ARQ and the HARQ in order to compensate the packet loss however these technologies have been defined as optional [1]. In general, the HARQ has been considered that it is more adequate for the delay-sensitive traffic than the ARQ, because the HARQ is designed in the physical (PHY) layer. However, the packet error can be detected after decoding the medium access control (MAC) protocol data unit (PDU) in the IEEE 802.16 systems, because the cyclic redundancy check (CRC) is merely attached to the MAC PDU. Thus, the HARQ cannot have a merit compared to the ARQ in the IEEE 802.16 systems. Therefore, this paper has investigated the ARQ in order to estimate the user’s satisfaction. The authors have found out the optimal ARQ parameters. In [3], the authors have proposed the fast transmission scheme for the ARQ feedback in the IEEE 802.16 systems because a SS has to send the ARQ feedback through the uplink by using the contention scheme. However, these papers have considered only the ARQ for the downlink transmission. For this reason, the performance of the uplink VoIP scheduling types for the ARQ could not be evaluated. In addition, the user’s satisfaction cannot be estimated because these papers have focused on the system performance. In [2], the MOS has been used to estimate the user’s satisfaction. The authors have found out the optimal MPDU size according to the channel condition in order to minimize the loss probability within the acceptable limit of the delay. However, the paper has not considered the uplink VoIP scheduling types for the ARQ. For this reason, the IEEE 802.16 systems can still have the following considerations:

- Can the uplink scheduling types for the VoIP services support the ARQ well?
- Which uplink scheduling type is the most adequate for the VoIP services supported by the ARQ with respect to...
the user-perceived QoS performance and the system performance?
Consequently, this paper evaluates the MOS and the system performance for the uplink VoIP scheduling types with the ARQ in order to solve the considerations mentioned above.

II. UPLINK SCHEDULING TYPES FOR VOIP SERVICES SUPPORTED BY THE ARQ IN IEEE 802.16 SYSTEMS

This section describes the uplink scheduling types for the VoIP services supported by the ARQ in the IEEE 802.16 systems. The uplink scheduling types for the VoIP services are mainly considered as the UGS, the rtPS, and the ertPS. In the UGS, a BS periodically allocates a fixed-size grant to a SS for providing a VoIP service with guaranteeing a QoS however it has a weakness. The UGS can waste the wireless resource when the size of the VoIP packet is variable. In the rtPS, a BS periodically allocates a grant to send a bandwidth-request header to a SS. For this reason, a SS can request a bandwidth for the variable-sized packet however the rtPS can waste a wireless resource and increase the delay [5]. In order to solve this problem, the ertPS has been proposed. In the ertPS, a BS periodically allocates the variable-sized grant to a SS to support a VoIP service with guaranteeing a QoS and improving the resource efficiency. However, these scheduling types do not consider a VoIP service supporting by the ARQ. For this reason, it is needed to investigate several considerations for the uplink scheduling types when the VoIP service is supported by the ARQ.

This paper assumes that the selective acknowledge (ACK) is applied to the system and the ARQ-BLOCK-SIZE is larger than a VoIP packet size. For this reason, a BS sends the ACK and negative-ACK (NACK) to a SS by transmitting the ARQ-feedback-IE included in the UL-MAP. Fig. 1 describes the ARQ transmit state of a SS. As shown in Fig. 1, if a SS has a transmitting packet, the state of the SS transits to the outstanding state after transmitting the packet. If the decoding of a packet is failure in a BS, the BS sends an ARQ-feedback-IE in order to deliver the decoding results. When the SS receives the ARQ-feedback-IE with NACK for the packet, the state of the SS transits from the outstanding state to the waiting state after retransmitting the packet. If the decoding of a packet is failure in a BS, the BS sends a new packet to the SS. Fig. 2 represents the operation for the uplink scheduling types when a packet is retransmitted by the ARQ. The considerations for each uplink scheduling types with the ARQ are as follows.

- In the case of the UGS, a packet is delayed up to next grant interval by the retransmitting packet because a BS periodically allocates a fixed-size grant as shown in Fig. 2. In addition, the piggyback scheme is not allowed for the UGS in the IEEE 802.16 standards. Thus, the delay is continuously cumulated whenever a packet is retransmitted by the ARQ. This can cause the serious deterioration of the QoS for the VoIP service.

- In the case of the ertPS, the SS can transmit the retransmitting packet at the next frame because the SS can request the additional bandwidth by sending the bandwidth-request header at the periodically allocated grant. At the next grant interval, the SS sends the grant-management subheader by piggybacking with the packet in order to save the wasted resource. For this reason, the ertPS can support the ARQ with guaranteeing the QoS and saving the wireless resource.

- In the case of the rtPS, the BS periodically allocates a grant for the bandwidth-request header to the SS. Thus, the SS can periodically request the required bandwidth for every polling interval. For this reason, the rtPS can also support the ARQ however it can reduce the system performance compared to the ertPS. The reason of this is that a wireless resource can be wasted and the access delay to transmit a packet can be occurred because of the periodically transmitted bandwidth-request header.

Figure 1. ARQ transmit state of a SS.

Figure 2. Operation for the uplink scheduling types with the ARQ.
III. USER-PERCEIVED QoS PERFORMANCE OF VOIP SERVICE

This paper evaluates the MOS for the uplink scheduling types with the ARQ in order to investigate the user-perceived QoS performance for the VoIP service supported by the ARQ. The MOS can be calculated as follows.

The ITU-T E-model defines an R-value that combines different aspects of voice quality impairment [6]. The R-value with default values is given by

$$R = 94.2 - I_e$$

where $I_e$ is an equipment impairment factor associated with the losses due to the codec and network and $I_d$ represents the impairment caused by the delay [2]. In (1), $I_e$ represents the effect of the packet loss rate as following:

$$I_e = \gamma_1 + \gamma_2 \ln(1 + \gamma_3 e)$$

where $\gamma_1$ is a constant related to the encoding, and $\gamma_2$ and $\gamma_3$ mean the impact of loss for a given codec [2].

$$e = e_{\text{network}} + (1 - e_{\text{network}}) e_{\text{playout}},$$

where $e_{\text{network}}$ is the loss probability in the network and $e_{\text{playout}}$ is the loss probability at the receiver side. $e_{\text{network}}$ is measured and $e_{\text{playout}}$ is given in this paper. In (1), $I_d$ means the effect of the delay as following:

$$I_d = 0.024d + 0.11(d - 177.3)H(d - 177.3),$$

where $H(x)$ is an indicator function [2]. $H(x) = 0$ if $x < 0$ otherwise, $H(x) = 1$. $d$ includes the delay in the codec, the playout, and the network as

$$d = d_{\text{codec}} + d_{\text{playout}} + d_{\text{network}},$$

where $d_{\text{network}}$ is measured, and $d_{\text{codec}}$ and $d_{\text{playout}}$ are given in this paper. The R-value is related to the MOS through the following nonlinear mapping:

$$\text{MOS} = 1 + 0.035R + 7 \times 10^{-6} R(R - 60)(100 - R),$$

for $0 \leq R \leq 100$ [2] and [6]. The MOS represents the user’s satisfaction from 1 to 5 as shown in Table 1.

<table>
<thead>
<tr>
<th>MOS Value</th>
<th>User Satisfaction</th>
</tr>
</thead>
<tbody>
<tr>
<td>4.3</td>
<td>Very satisfied</td>
</tr>
<tr>
<td>4.0</td>
<td>Satisfied</td>
</tr>
<tr>
<td>3.6</td>
<td>Some users dissatisfied</td>
</tr>
<tr>
<td>3.1</td>
<td>Many users dissatisfied</td>
</tr>
<tr>
<td>2.6</td>
<td>Nearly all users dissatisfied</td>
</tr>
<tr>
<td>1</td>
<td>Not recommended</td>
</tr>
</tbody>
</table>

IV. SIMULATION ENVIRONMENT

This paper has built the simulator using OPNET tool. In order to estimate the user-perceived QoS performance, this paper has designed the nodes (SS and BS) from physical layer to application layer as shown in Fig. 3. The IP, TCP/UDP, and application layer existed in the OPNET module have been used. The physical layer and MAC layer have been developed based on the IEEE 802.16 systems. Particularly, this paper has modeled the fragmentation/de-frAGMENTATION, scheduling (UGS, eTPS, rTPS, and BE), selective ACK, and resource allocation in the MAC layer.

<table>
<thead>
<tr>
<th>Attributes</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel Model Parameters [7]</td>
<td>Pathloss Model</td>
</tr>
<tr>
<td></td>
<td>Shadowing Model</td>
</tr>
<tr>
<td></td>
<td>Mobility and Channel Model</td>
</tr>
<tr>
<td></td>
<td>Thermal Noise Density</td>
</tr>
<tr>
<td>System Parameters [8]</td>
<td>Bandwidth</td>
</tr>
<tr>
<td></td>
<td>Frame Size</td>
</tr>
<tr>
<td></td>
<td>FFT</td>
</tr>
<tr>
<td></td>
<td>Number of Subcarriers per Subchannel</td>
</tr>
<tr>
<td></td>
<td>Number of Data Symbols per Frame</td>
</tr>
<tr>
<td></td>
<td>MCS Level</td>
</tr>
<tr>
<td>ARQ Parameters [1]</td>
<td>ARQ WINDOW SIZE</td>
</tr>
<tr>
<td></td>
<td>ARQ BLOCK LIFETIME</td>
</tr>
<tr>
<td></td>
<td>ARQ RETRY TIMEOUT</td>
</tr>
<tr>
<td></td>
<td>ARQ BLOCK SIZE</td>
</tr>
<tr>
<td>VoIP Codec Parameters [2]</td>
<td>R-Value Calculation</td>
</tr>
<tr>
<td>Network Parameters [9]</td>
<td>Backbone Propagation Delay</td>
</tr>
</tbody>
</table>
Fig. 4 represents the reference network architecture. This paper has evaluated the QoS and the system performance when the SS moves from cell boundary to cell center and returns. This paper has used the 100Base_T and PPP_SONET_OC12 link as shown in Fig. 4, because this paper has not considered the network congestion. This paper has also simulated with the G.711 without silence suppression for the VoIP speech codec. The simulation parameters are shown in Table 2.

V. SIMULATION RESULTS

This section describes the simulation results for the system performance and the MOS when the SS moves in the cell. Fig. 5 represents the signal to noise ratio (SNR) according to the simulation time. As shown in Fig. 5, the SNR varies within the range from -20 dB to 70 dB due to the pathloss, shadowing, and multipath fading. Under this channel condition according to the simulation time, the system performance can be evaluated as Fig. 6 and Fig. 7. The throughput means the number of the packet received in application layer at the CN for a second, and the delay indicates the time to transmit a VoIP packet from the application layer of the SS to the application layer of the CN.

Fig. 6 indicates the throughput according to the simulation results. As shown in Fig. 6, in the case of the non ARQ, the throughput is decreased by 50 % when the channel quality is poor. This means that the packet loss rate is 0.5. In the case of the non ARQ, the MOS is affected only by the packet loss because the delay is not increased according to the channel quality as shown in Fig. 7. Thus, this packet loss can cause the serious deterioration of the user-perceived QoS performance as shown in Fig. 8. When the channel quality is poor, the MOS is decreased by 1.5 in the case of the non ARQ. This indicates the nearly all users dissatisfied for the VoIP service in Table 1. For this reason, the ARQ is needed in order to guarantee the QoS of the VoIP service when the channel quality is poor.

As mentioned in section II, it is needed to evaluate the several considerations for the uplink scheduling types in order to apply the ARQ to the VoIP service. In the case of the UGS, the throughput is also decreased by 50 % as shown in Fig. 6 however this phenomenon is due to the increase of the delay as shown in Fig. 7. In the UGS, the delay is continuously cumulated whenever a packet is retransmitted because a BS periodically allocates a fixed-size grant as shown in Fig. 3. Therefore, since the MOS becomes 1 when the UGS is applied to the VoIP service supported by the ARQ, the UGS with the ARQ can cause the serious deterioration of the user-perceived QoS performance.

In the case of the rtPS and ertPS with the ARQ, the throughput can be maintained although the channel quality is poor as shown in Fig. 6. However, the delay is increased by 140 msec due to the retransmission of the packet experienced an error as shown in Fig. 7. The increased delay cannot be a problem in terms of the user-perceived QoS performance as shown in Fig. 8. In Table 1, most users satisfy the VoIP service when the MOS is larger than 3.5. Therefore, the rtPS and ertPS are adequate for the VoIP service supported by the ARQ in the view of the user-perceived QoS performance.

Here, it is needed to evaluate the system efficiency because the wireless resource is additionally used to improve the user-perceived QoS performance. This can cause the reduction of the VoIP capacity which is the maximum number of supportable VoIP users. For the system efficiency, this paper evaluates the number of the used resource for a second. The number of the used resource is increased when the channel quality is poor, because the number of the retransmission for the ARQ is increased. In Fig. 9, the ertPS can save the number of the used resource by 25 % compared to the rtPS because the rtPS has to consume the resource to request the bandwidth for every grant interval. This means that the ertPS with the ARQ can support more VoIP users than the rtPS with the ARQ. Consequently, the ertPS with the ARQ is the most adequate scheme for the VoIP service in terms of the user-perceived QoS performance and the system efficiency compared to the UGS and rtPS with the ARQ.
VI. Conclusion

This paper investigated the uplink scheduling types for the VoIP services supported by the ARQ. Particularly, this paper evaluated the user-perceived QoS performance and the system efficiency. From the simulation results, the ertPS can support the VoIP service with the ARQ with guaranteeing a QoS and saving the wireless resource.

The contributions of this paper are as follows:

- This paper represented the necessity of the retransmission scheme for the VoIP service in order to improve the user-perceived QoS performance when the channel quality is poor.
- This paper investigated the uplink scheduling types for the VoIP service with the ARQ and found out the serious problem for the UGS with the ARQ.
- This paper evaluated the user-perceived QoS performance for the uplink scheduling types with the ARQ in order to estimate the user’s satisfaction.

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