Graceful Degradation FEC Layer for Multimedia Broadcast/Multicast Service in LTE Mobile Systems

Seok Ho Won

This paper proposes an additional forward error correction (FEC) layer to compensate for the defectiveness inherent in the conventional FEC layer in the Long Term Evolution specifications. The proposed additional layer is called a graceful degradation (GD)-FEC layer and maintains desirable service quality even under burst data loss conditions of a few seconds. This paper also proposes a non-delayed decoding (NDD)-GD-FEC layer that is inherent in the decoding process. Computer simulations and device-based tests show a better loss recovery performance with a negligible increase in CPU utilization and occupied memory size.

Keywords: Forward error correction, unequal error protection, graceful degradation.

Manuscript received May 17, 2013; accepted Aug. 16, 2013.

This work was supported by the MSIP (Ministry of Science, ICT & Future Planning), Korea in the ICT R&D Program 2013.

Seok Ho Won (phone: +82 42 860 5449, shown@etri.re.kr) is with the Communications & Internet Research Laboratory, ETRI, Daejeon, Rep. of Korea.

I. Introduction

There is increasing demand for multimedia services delivered not only through wired networks but also through wireless networks in which the terminals are constantly moving, which results in the network conditions varying as the time and location of the terminals move and change. The variation of the network conditions may also result in severe packet error rates; hence, to combat these error conditions, a minimum level of acceptable service quality should be guaranteed by providing a graceful degradation. As an example of this, when a video stream stalls, the audio should be supplied at a minimum. Therefore, this paper proposes a new sublayer called a graceful degradation forward error correction (GD-FEC) sublayer (see Fig. 1).

The GD-FEC sublayer is implemented during the first process in the transport layer and is directly applied to the data received from the media layer, as shown in Fig. 1. The GD-FEC mechanism provides an unequal error protection (UEP) method that protects important parts of the media (for example, audio) bitstream(s) more strongly than others.

Since the UEP method was first introduced in 1967 by Masnick and Wolf [1], it has become well known and has been implemented in such practical systems as digital audio broadcasting and MediaFLO™ [2]. Owing to video data becoming scalable, more advanced UEP methods were proposed for video data, and the authors in [3]-[5] proposed noteworthy UEP methods using the unequal importance of frames in a group of pictures (GoP) and that of macroblocks in video frames (for example, I or IDR frames are more important than other frames, such as P and B frames) [3], [4]. Furthermore, a UEP method with a cross layer operation for SVC streaming was proposed [4], [5].

http://dx.doi.org/10.4218/etrij.13.0113.0486

The latest UEP methods with FEC coding include the layer aware (LA)-FEC method and inter-layer (IL)-FEC method, as proposed in [6]-[9]. The LA-FEC precedes the hard decision bit operation in the binary erasure channel of the transport layer (TL) [7]-[9]. The LA-FEC also uses the unequal interleaving (UI) size method for fast zapping with a long time interleaving or an adaptive time interleaving [8], [9]. In contrast, IL-FEC precedes the soft decision bit operation in the mutual information channel of the link layer or the physical layer [6]. These methods are innovative in terms of their performance improvement, but they have significant drawbacks in their decoding complexity as well as an impossible implementation capability in the Long Term Evolution (LTE) mobile system structure, as shown in Fig. 1 (for example, in the FEC layer, all data is interleaved and enciphered, so it is impossible to use a layer-aware method). The method proposed in this paper is simple, requires low hardware resource and battery consumption, and allows the terminal CPU to simultaneously operate other application programs.

Regarding the related studies in the standards, LA-FEC has been reviewed in the MPEG media transport standard group, and the method proposed in this paper is found in the 3GPP official document, that is, Technical Report 26.947 [10]; however, the method proposed in this paper is still under review regarding the standardization check points, such as a multi-vender implementation problem, control message overhead, and overall delay aspects.

II. MBMS with Proposed GD-FEC Overview

In multimedia broadcast/multicast service (MBMS) in LTE systems, the FEC layer is located in the lower layer in the real-time transport protocol (RTP) or secure RTP (SRTP) sections to fully protect the packets and headers generated in the upper sublayers (see Fig. 1). The GD-FEC sublayer may also be located above the File Delivery over Unidirectional Transport (FLUTE) protocol layer when it is used for DASH-based streaming services over FLUTE [10].

A general method, such as the GD-FEC layer with a retransmission, has already been implemented; in this implemented method, the more important bitstreams can be packetized with playback order indices and transmitted first to reserve time for a retransmission when a packet is lost [11]. However, broadcast systems do not allow for a retransmission. Therefore, the proposed GD-FEC layer sends repair packets first, before the target source packet is sent, the process of which is described in detail in the next section.

Many GD-FEC layer algorithms that provide more redundancies to more important media data were investigated [12], [13]. The foremost difference between these GD-FEC layers and the proposed layer is that they are implemented in the media layer or application layer instead of the transport layer. The GD-FEC layer in the application layer could be inefficient because it violates the layered protocol rules [14]. Moreover, the proposed GD-FEC layer can be implemented simultaneously with a conventional FEC layer, and it borrows a small number of redundancy bits from the conventional FEC layer block, allowing the total number of redundancy bits to be constant in the FEC layer without consuming additional channel bandwidth. In addition to the bandwidth increment, another drawback of the general GD-FEC is the decoding delay inherent in the decoding process that protects the burst data losses that are a few seconds long. However, the decoding delay results in increases in the start time and switching time. To resolve these issues, this paper proposes a non-delayed decoding (NDD)-GD-FEC layer.

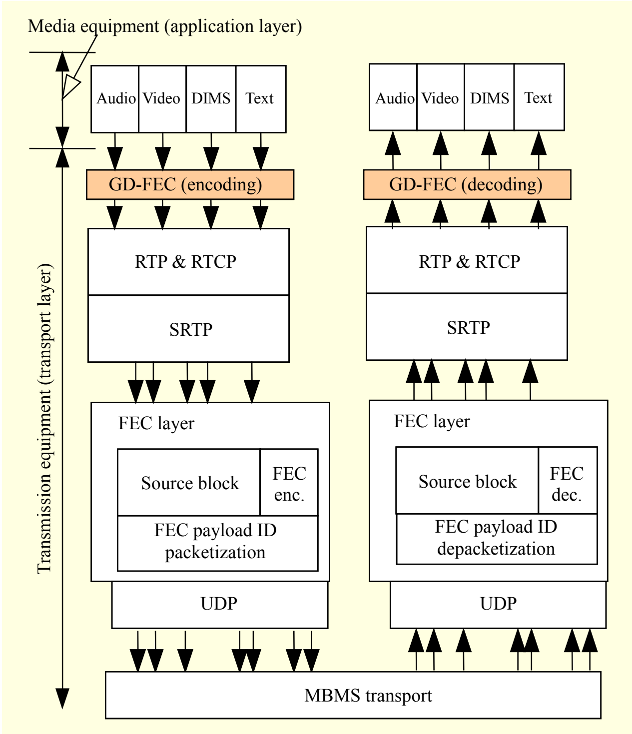

Fig. 1. MBMS system structure with proposed GD-FEC.

III. Proposed GD-FEC Method

To describe the proposed GD-FEC method, the audio packets are selected as the target source data for GD-FEC protection. There are two important terminologies in the GD-FEC, that is, the encoded multimedia data group (EMDG) and the GD-FEC encoding group (GDEG), as shown in Fig. 2. The EMDG is a packet group that contains a group of media data providing a certain amount of information (for example, all

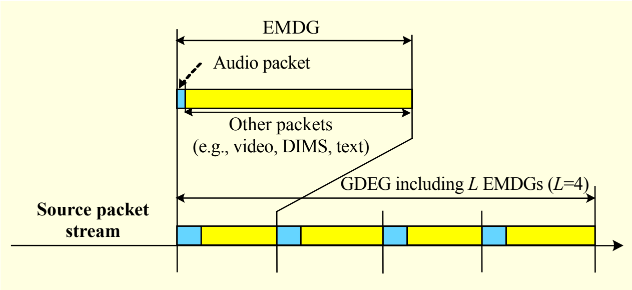

Fig. 2. EMDG and GDEG.

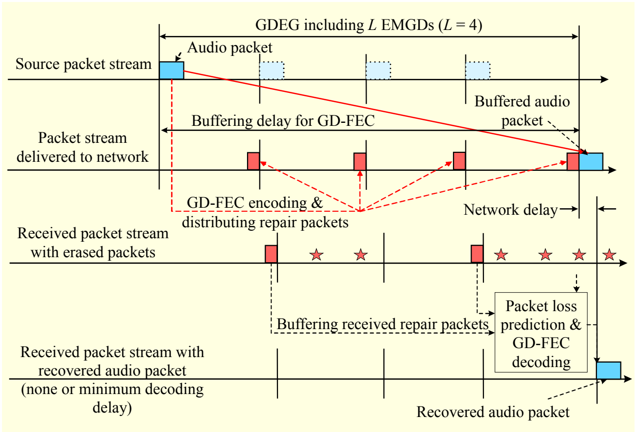

Fig. 3. No delay or minimum delay GD-FEC decoding with sufficient delay yielded by the GD-FEC encoding (for example, $L = 4$).

media data in one picture frame unit). The GDEG is defined as a group of L EMDGs, where $L = 1, 2, \ldots$ (for example, $L = 4$ in Fig. 2).

As described above, the important requirement for the GD-FEC is the decoding delay. Figure 3 shows no delay or minimum delay for the GD-FEC decoding, whereas sufficient delay is yielded in the GD-FEC encoding (in the case of $L = 4$). In the figure, the packet located on the left of the highest line, labeled "source packet stream," is assumed to be the target source packet or frame (for example, audio packet or Mfr). More packets can be used from the stream, but one audio packet is selected for a simpler explanation. This audio packet (blue box) is encoded; the encoded audio packet can then produce a group of repair packets or redundancies (Pfr) using a general application layer (AL)-FEC (for example, Reed-Solomon [RS]) [10], [15]. Next, this Pfr is divided into four DPfrs (L = 4; that is, four red boxes) and distributed to each EMDG in the GDEG. The Mfr (blue box) is located in the last of these four packets, that is, all packets form the "packet stream delivered to network," as labeled in the second line of the figure. The third line, labeled "received packet stream with erased packets," shows the lost packets, including the Mfr that should appear in the right but was assumed lost. Finally, the last line, labeled "received packet stream with recovered audio packet," demonstrates that the lost Mfr is repaired through the AL-FEC decoding process using the repair packets received before the position of Mfr; hence, the decoding delay can be negligible.

IV. Algorithm Evaluation

1. Evaluation Model

The recovering capability of the FEC code in the FEC layer is proportional to the number of redundancies because the amount of received data after the message loss and redundancies combined in the mobile channel must be greater than that of the original message data. The most frequently used FEC codes in the FEC layer are the RS and RaptorQ codes, for which the decoding failure probabilities are given in (1) and (2), respectively [15], [16].

- RS code:

$$Pf_{\mathrm{RS}}(n,k) = \begin{cases} 1, & n \le k, \\ 0, & n > k, \end{cases} \tag{1}$$

- RaptorQ code:

$$Pf_{\mathrm{RQ}}(n,k) = \begin{cases} 1, & n \le k, \\ 0.01 \times 0.01^{n-k}, & n > k, \end{cases} \tag{2}$$

where n denotes the number of symbols received and k denotes the number of original message symbols. With these equations, it is known that the sender needs a sufficient number of redundancies; thus, the amount of remaining data received must be greater than that of the original message. Therefore, suppose that there are α portions of loss in k bit messages and that, when a suitable FEC layer code is used, the degree of parity can approach that of the lost bits for a perfect repair. For a small positive value (ϵ), the degree of parity (m) can then be expressed by $m = \alpha k + \epsilon$. Here, α can be set as the target loss recovery rate (TLRR).

Suppose that the τ fraction of the message in the FEC layer is protected in the GD-FEC layer. With the total number of redundancies between the conventional FEC (CV; FEC layer only) and proposed GD-FEC (PR; FEC layer + GD-FEC layer) being the same, the effective TLRR of the CV method can then be expressed using that of the PR method, as follows:

$$\frac{\alpha_{\mathrm{CV}}}{1-\alpha_{\mathrm{CV}}} = \frac{\alpha_{\mathrm{PR}}}{1-\alpha_{\mathrm{PR}}} + \frac{\beta}{1-\beta}\tau, \tag{3}$$

where α_{CV} and α_{PR} denote the TLRRs at the FEC layer for the CV method and PR method, respectively. Furthermore, β denotes the TLRR at the GD-FEC layer for the PR method.

The denominators in each term in (3) come from the fact that loss can occur in the redundancy parts. That is, as for the term "effective," data loss can also occur in the redundancy packets; thus, the "effective TLRR" refers to the TLRR that considers all of the required redundancies.

Finally, the decoding failure rate ($Pf_{\text{GD-FEC,DLY}}$) in the decoding delayed version of the GD-FEC layer (that is, the DD-GD-FEC layer) is given as follows:

$$Pf_{\text{GD-FEC,DLY}} = \Pr\{\frac{\delta}{1-\delta} > \beta_{\text{eff,DLY}}\}, \tag{4}$$

where δ is the FEC layer coded BLR or residual BLR. Through (4), the effective TLRR in the NDD-GD-FEC layer ($\beta_{\text{eff,DLY}}$) can be given as follows. The remaining amount of data in the current EMDG and L previous consecutive EMDGs is $(1-\beta_0)k$ and $\beta_0 k/(1-\sum_{i=-L}^{-1}\beta_i/L)$, respectively. The sum of these two values must be greater than the amount of the original message, k. Finally, (5) can be obtained as follows:

$$\beta_{\text{eff,NDLY}} = \frac{\beta_0}{1-\sum_{i=-L}^{-1}\beta_i/L}. \tag{5}$$

The decoding failure rate ($Pf_{\text{GD-FEC,DLY}}$) in the NDD-GD-FEC layer is given as follows:

$$Pf_{\text{GD-FEC,NDLY}} = \Pr\{\frac{\delta_0}{1-\sum_{i=-L}^{-1}\delta_i/L} > \beta_{\text{eff, NDLY}}\}, \tag{6}$$

where δ_i is the FEC layer coded BLR at index i. Finally, with the decoding failure rate, the coded BLR or residual BLR can be easily obtained.

2. Computer Simulation Setup and Method

In the simulation model, the burst loss bearer process is used with a two-state Markov model to verify the performance of the GD-FEC layer [17], [18]. Furthermore, for the simulation of LTE RLC-PDU losses, the 3GPP SA4 standardization group recommends using the parameter values of the two-state Markov model as the working assumptions; thus, this paper follows these conventions [10], [17]-[20]. Under this condition, the simulation is performed using (1), (4), and (6) to obtain the coded BLR (residual BLR).

3. Device-Based Evaluation Setup and Method

Based on the Android operating system for mobile devices (for example, Samsung's Nexus-S™ smartphone), several device-based tests are performed to compare the performance in terms of complexity and power consumption between the two cases: with and without adopting the GD-FEC [20]. The detailed conditions and procedures were given in [10], [19]-

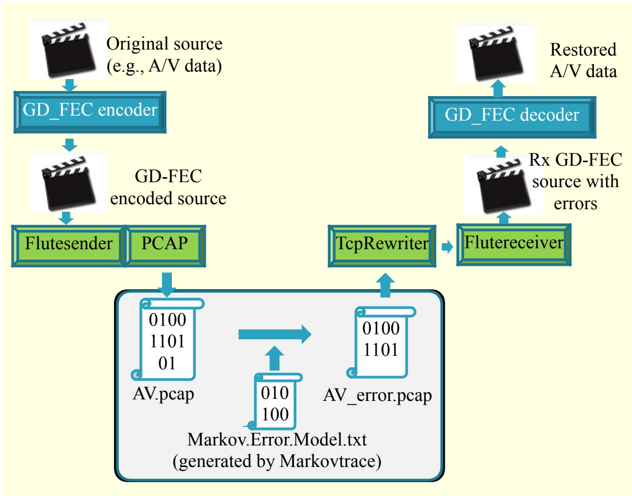

Fig. 4. Device-based GD-FEC procedure.

Table 1. Parameter assumptions for device-based test.

Parameter	Assumption/explanation
Loss model	Markov 3 km/h with 5%, 10%, and 20% BLR losses
FLUTE file segment duration	2 sec
No. of segments (time = 30 sec)	15
FLUTE FEC	RS with redundancy ratio of 20%
Segmented source size	2 Mbytes (total source file size: 30 Mbytes)
EMDG	1 GoP (group of pictures of a video sequence)
GDEG length	L = 4, 5
Total media data size in an EMDG	536 Kbytes (30 Mbytes/56 GoPs)
GD-FEC target source data size in an EMDG	10 Kbytes
GD-FEC redundancy amount	400% (four times that of original source)

[21], and a brief overview of the procedure is given in Fig. 4 [22]-[24]. The other test conditions and parameter values used in the device-based test are given in Table 1.

4. Evaluation Results and Discussion

Figure 5 presents an illustrative simulation result. In the figure, DLY and NDLY indicate the DD-GD-FEC and NDD-GD-FEC, respectively; Vm = 3 km/h, τ = 0.01, and TLRR = 10%; (x, y, z) indicates the (BLR, α, β); and L = 5 is used for both the DLY GD-FEC layer and the NDLY GD-FEC layer.

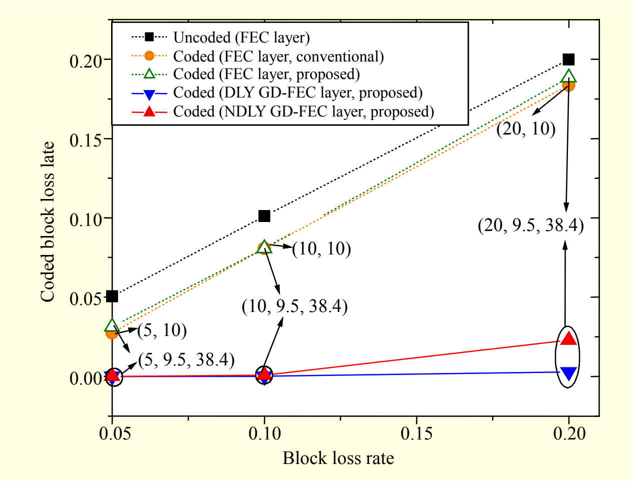

Fig. 5. Coded BLR (residual BLR) results for FEC layer and GD-FEC layer in MBMS system: TLRR = 10%.

Table 2. Device-based example test results for GD-FEC in terms of CPU utilization and occupied memory size.

	Test item	Without GD-FEC	With GD-FEC
Case 1 BLR = 5%	CPU	31%	31%
	Memory	31,420 Kbytes	31,494 Kbytes
Case 2 BLR = 10%	CPU	60%	60%
	Memory	31,405 Kbytes	30,426 Kbytes
Case 3 BLR = 20%	CPU	61%	62%
	Memory	31,468 Kbytes	31,703 Kbytes

As shown in the figure, the GD-FEC results indicate a low residual loss rate (the two lowest curves are below 0.025, and more test results can be found in the Appendix and [21]).

Tables 2 and 3 show the device-based test results for the GD-FEC in terms of CPU utilization and occupied memory size and in terms of the successful restoration rates of the GD-FEC target source, respectively. These test results demonstrate that there are no significant differences in the CPU utilization and UE's occupied memory between the two cases: with and without adopting GD-FEC. The rationale for these results can be summarized as follows.

(i) For the CPU utilization: The amount of GD-FEC decoding sources is relatively small. Thus, the decoding burden of the GD-FEC is also small.

(ii) For the UE's occupied memory: Additional buffering for the GD-FEC decoding is not required but can be performed naturally using the display buffer of the streaming system. Because the display buffer of the streaming system naturally covers the GD-FEC buffer, no additional physical memory for the GD-FEC decoding is needed. Note that the display buffer is required in this case because the speed of the network delivery is much faster than that of the media display.

Table 3. Device-based example test results for GD-FEC in terms of successful GD-FEC target source restoration rates.

	Without GD-FEC no. of pkts (success rate)	With GD-FEC no. of pkts (success rate)
Case 1 BLR = 5%	55 (98%)	56 (100%)
Case 2 BLR = 10%	53 (94%)	56 (100%)
Case 3 BLR = 20%	42 (75%)	56 (100%)

As shown in Table 3, the total number of GD-FEC target sources is 56. When the GD-FEC is adopted, the restoration rate is perfect in this test because the number of redundancies is sufficient (for example, 400%).

V. Conclusion

This paper proposed new GD-FEC and NDD-GD-FEC methods to guarantee minimum service quality for users in severe burst loss conditions. The computer simulation results demonstrated that the proposed GD-FEC suppresses coded BLR or residual BLR for all given test cases, while the device-based test results demonstrated a slight increase in the terminal power consumption and complexity for the GD-FEC operation. As a result, compared with the conventional FEC method (single FEC layer method), the proposed method exhibits better residual packet loss rates with negligible increases in complexity in terms of CPU utilization and occupied memory size.

Appendix

This appendix provides more coded BLR (residual BLR) results for the FEC layer and GD-FEC layer in an MBMS system (see Fig. A1): $Vm = 3$ km/h, $\tau = 0.01$, and TLRR = 20% and 30% for (a) and (b), respectively; (x, y, z) indicates (BLR, α, β), and $L = 5$ is used for both the DLY GD-FEC layer and the NDLY GD-FEC layer. It also provides the results of Vm = 120 km/h, $\tau = 0.01$, and TLRR = 20% and 30% for (c) and (d), respectively. In the figures, "conventional" indicates all normal FEC methods used in the LTE system, and other state-of-the-art methods, such as LA-FEC and IL-FEC, cannot be evaluated because they cannot be implemented in LTE structures.

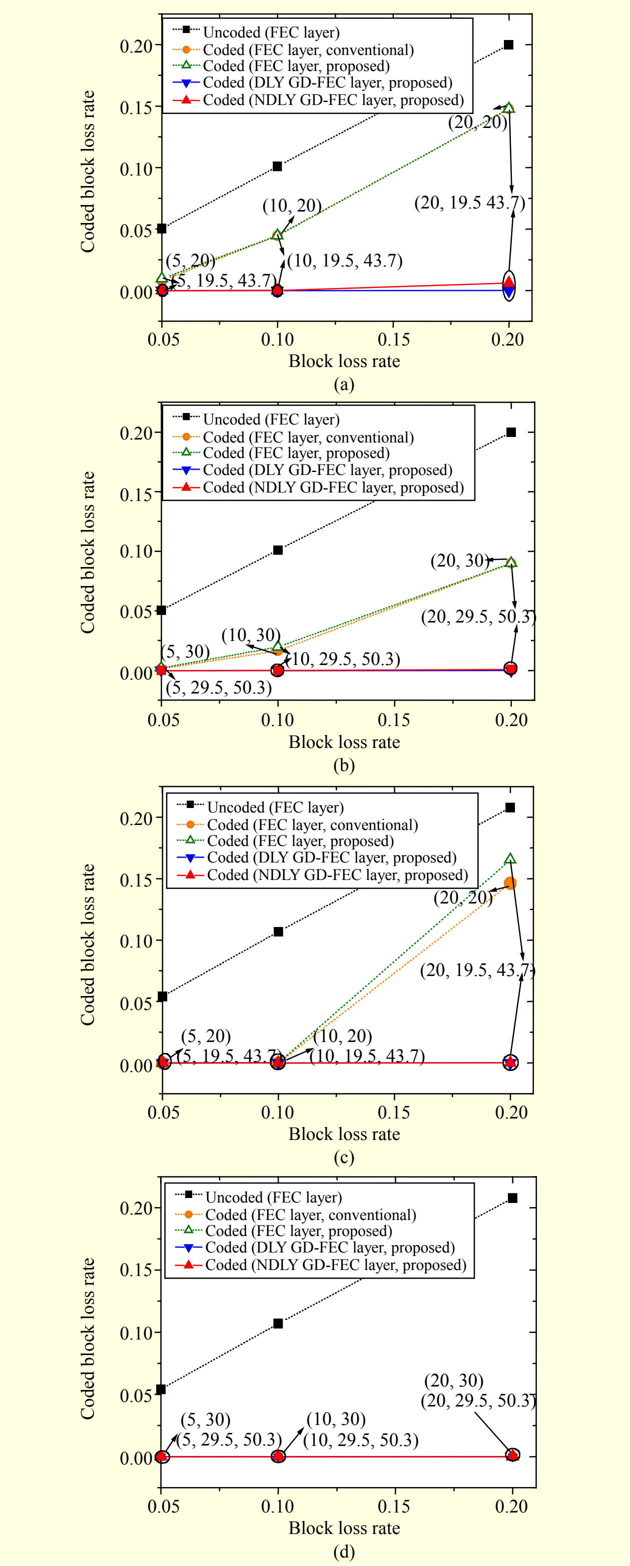

Fig. A1. Coded BLR (residual BLR) results for FEC layer and GD-FEC layer in MBMS system: (a) Vm = 3 km/h, TLRR = 20%; (b) Vm = 3 km/h, TLRR = 30%; (c) Vm = 120 km/h, TLRR = 20%; (d) Vm =120 km/h, TLRR = 30%. (τ = 0.01; (x, y, z) = (BLR, α, β); and L = 5 for DLY and NDLY GD-FEC layers.)

References

[1] B. Masnick and J. Wolf, "On Linear Unequal Error Protection Codes," *IEEE Trans. Inf. Theory*, vol. 13, Oct. 1967, pp. 600-607.

[2] W. Wu, J. Boyce, and A. Stein, "An Unequal Error Protection Framework for DVB-H and Its Application to Video Streaming," *IEEE Globecom*, New Orleans, LA, USA, Dec. 2008, pp. 1-6.

[3] Y. Chang, S. Lee, and R. Komyia, "A Fast Forward Error Correction Allocation Algorithm for Unequal Error Protection of Video Transmission over Wireless Channels," *IEEE Trans. Consum. Electron.*, vol. 54, Aug. 2008, pp. 1066-1073.

[4] Y.C. Chang, S.W. Lee, and R. Komiya, "A Low Complexity Hierarchical QAM Symbol Bits Allocation Algorithm for Unequal Error Protection of Wireless Video Transmission," *IEEE Trans. Consum. Electron.*, vol. 55, Aug. 2009, pp. 1089-1097.

[5] E. Maani and A. Katsaggelos, "Unequal Error Protection for Robust Streaming of Scalable Video over Packet Lossy Networks," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 20, Mar. 2010, pp. 407-416.

[6] Y. Huo, M. El-Hajjar, and L. Hanzo, "Inter-Layer FEC Aided Unequal Error Protection for Multilayer Video Transmission in Mobile TV," *IEEE Trans. Circuit Syst. Video Technol.*, vol. 23, no. 9, Sept. 2013, pp. 1622-1634.

[7] C. Hellge et al., "Layer-Aware Forward Error Correction for Mobile Broadcast of Layered Media," *IEEE Trans. Multimedia*, vol. 13, June 2011, pp. 551-562.

[8] C. Hellge et al., "Mobile TV with Long Time Interleaving and Fast Zapping," *IEEE Int. Conf. Multimedia Expo Workshops*, July 2012, pp. 623-628.

[9] V. Pullano et al., "Application Layer FEC with Long Time Interleaver and Fast Tune-in for Mobile Satellite TV Services," *IEEE Int. Conf. Comput., Netw. Commun.*, Jan. 2013, pp. 108-112.

[10] 3GPP, "Technical Specification Group Services and System Aspects; Multimedia Broadcast/Multicast Service (MBMS); Selection and Characterization of Application Layer Forward Error Correction (FEC) (Release 11)," 3GPP Technical Report (TR) 26.947, Mar. 2013. http://www.3gpp.org/ftp/Specs/archive/26_series/26.947/26947-b00.zip

[11] ITU, "Transport of AVC Video Data over ITU-T Rec. H.222.0|ISO/IEC 13818-1 Streams," ITU-T Rec. H.222.0 (2000)/Amd.3|ISO/IEC 13818-1: 2000/Amd.3 Systems AMENDMENT 3, 2004.

[12] J. Lee et al., "Loss-Aware Rate-Distortion Optimization for Redundant Picture Allocation in H.264/AVC," *Opt. Eng.*, vol. 49, no. 2, 027005, Feb. 2010.

[13] D. Vukobratovic and V. Stankovic, "On Unequal Error Protection Random Linear Coding for Scalable Video Broadcasting," *IEEE 18th Workshop Packet Video*, Hong Kong, China, Dec. 13-14, 2010, pp. 48-55.

[14] 3GPP, "Technical Specification Group Services and System

Aspects; Multimedia Broadcast/Multicast Service (MBMS); Protocols and Codecs (Release 10)," 3GPP Technical Specification (TS) 26.346, Sept. 2011. http://www.3gpp.org/ftp/Specs/archive/26_series/26.346/26346-a10.zip

[15] S. Lin and D.J. Costello, Jr., *Error Control Coding*, 2nd ed., Upper Saddle River, NJ: Prentice Hall, 2004.

[16] S.B. Wicker, *Error Control Systems for Digital Communication and Storage*, Upper Saddle River, NJ: Prentice Hall, 1995.

[17] E. Gilbert, "Capacity of a Burst-Noise Channel," *Bell Syst. Technical J.*, vol. 39, 1960, pp. 1253-1265.

[18] Fraunhofer HHI, Samsung, and ETRI, "MMT D1-FEC: Proposal Evaluation Model for AL-FEC Schemes," Doc. Number M22840, ISO/IEC JTC1/SC29/WG11 MPEG 98th Meeting, Geneva, Switzerland, Nov. 2011.

[19] Qualcomm, "Rationale for MBMS AL-FEC Enhancements," Tdoc S4-110449, 3GPP TSG-SA4#64, San Diego, CA, USA, Apr. 2011.

[20] ETRI, "Mobile Device-based Test Results for GD-FEC in DASH+FLUTE Protocol for MBMS Streaming Delivery Service," Tdoc S4-121412, 3GPP TSG-SA4#71, Bratislava, Slovakia, Nov. 2012.

[21] ETRI, "Evaluation Results for Graceful Degradation-FEC for MBMS Streaming Delivery," Tdoc S4-121060, 3GPP TSG-SA4#70, Chicago, IL, USA, Aug. 13-17, 2012.

[22] S. Kim et al., "A Subthreshold CMOS RF Front-End Design for Low-Power Band-III T-DMB/DAB Receivers," *ETRI J.*, vol. 33, no. 6, Dec. 2011, pp. 969-972.

[23] Y. Cho et al., "Multihop Rate Adaptive Wireless Scalable Video Using Syndrome-Based Partial Decoding," *ETRI J.*, vol. 32, no. 2, Apr. 2010, pp. 273-280.

[24] Y. Kim, J. Shin, and J. Park, "Design and Implementation of a Network-Adaptive Mechanism for HTTP Video Streaming," *ETRI J.*, vol. 35, no. 1, Feb. 2013, pp. 27-34.

Seok Ho Won received his B.S. degree in clinical pathology and electrical engineering from Kwangwoon University, Seoul, Rep. of Korea, in 1985 and 1990, respectively, and his Ph.D. degree in electrical engineering from Chungnam National University, Daejeon, Rep. of Korea, in 2002. Since 1985, he has been a clinical pathologist at Sin-Chon General Hospital, Gyeonggi-do, Rep. of Korea. Since 1990, he has been a principal engineer at ETRI, Daejeon, Rep. of Korea. He was a research faculty member at Virginia Tech, USA, in 2005. His research interests include information theory, error correction coding, MIMO, and beamforming with an emphasis on mobile communications and broadcasting.

http://dx.doi.org/10.4218/etrij.13.0113.0486