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Transmission Algorithm with QoS Considerations for a Sustainable MPEG Streaming Service

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Abstract: With the proliferation of heterogeneous networks, there is a need to provide multimedia stream services in a sustainable manner. It is especially critical to maintain the Quality of Service (QoS) standards. Existing multimedia streaming services have been studied to guarantee QoS on the receiving side. QoS has not been ensured due to the fact that the loss of streaming data to be transmitted has not been considered in network conditions. With an algorithm that considers the QoS and can reduce the overhead of the network, it will be possible to reduce the transmission error and wastage of communication network resources. In this paper, we propose a scheme that improves the reliability of multimedia transmissions by using an adaptive algorithm that switches between UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) based on the size of the data. In addition, we present a method that retransmits essential portions of the multimedia data, thus improving transmission efficiency. We simulate an MPEG (Moving Picture Experts Group) stream service and evaluate the performance of the proposed adaptive MPEG stream service.

Keywords: sustainability; MPEG; scalability; layered coding; Quality of Service; Forward Error Correction

1. Introduction

The simultaneous advances in computing and information communication technologies have enabled the processing of massive volumes of data using personal computers. Several studies analyze multimedia data services [1–8] for a variety of communication media and devices. The process of providing real-time multimedia services over mobile and broadband Internet connection has different requirements in terms of the Quality of Service (QoS) [9–27]. This prompts the division of multimedia data into two layers—the base layer and the enhancement layer. This modularization helps in the customization of encoding and decoding operations as per the user connection capabilities. The layers can be divided based on spatial and temporal aspects; the fundamental characteristics and scalability of these have been previously analyzed [28–41]. Here, scalability refers to the adaptability of the multimedia data service to a dynamic network environment with the aim of reducing data loss and providing best QoS.

Several measures have been proposed to improve the efficiency of multimedia data services. However, these are not suitable for deployment in resource constrained environments, where the users experience poor QoS. A multi-layered methodology has been proposed to implement adaptive multimedia transmission to improve efficiency [42–46]. In this, depending on the streaming bandwidth, either TCP or UDP sessions are chosen for transmission through the heterogeneous network [47–56]. To prevent the loss of packets during the transfer, important data segments are selected using the Forward Error Correction (FEC) methodology [57–65] and these are retransmitted.

The rest of the paper is organized as follows. In Section 2, the layered coding methodology, a data division method for adaptive MPEG system, is described. A description of the adaptive QoS algorithm based on session selection and FEC is provided in Section 3. The evaluation of the proposed algorithm is provided in Section 4. Section 5 concludes the paper by providing an overall summary and describing further research tasks.

2. Related Work

2.1. Layered Coding Methodology

In order to adapt to changing network environments, multimedia data must be divided. The QoS is improved by adding decoding data. When the network condition is good, the multimedia data is transmitted as a single unit without any interference. However, when the network is relatively slow, we perform basic data encoding to improve the QoS.

2.1.1. Spatial Division Method

The type of division method to be implemented depends on the data resolution. The encoding of the basic layers is performed first and the extension area is encoded based on the in between differential. Figure 1 illustrates the process for an object using the minimum value. There is an expansion of the layers with an increase in the value of the expression layer. The left upper area is designated as the low frequency area while the right lower area has higher frequency. The area information is used to determine the spatial resolution.

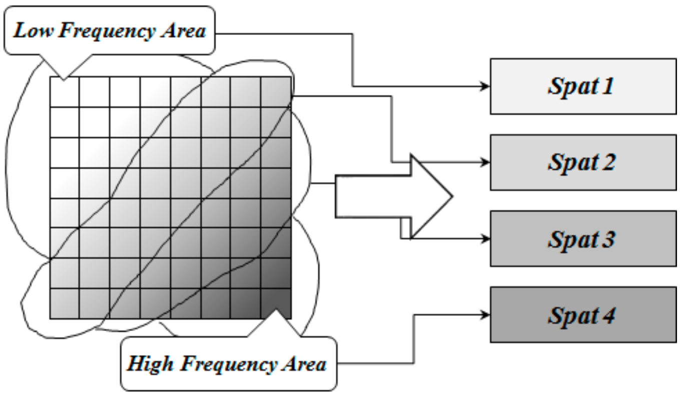


Figure 1. Spatial division method.

2.1.2. Time Division Method

In the time division method, the process of dividing the base layer and extended layer is the same as that in the spatial division method. However, the time division method utilizes the time difference between the base layer and the extended layer. This is done by identifying different streams in the frame expression and segregating them. When the base layer is subject to continuous replay, the result is an expression of wide movement. A smooth streaming replay can be acquired by adding the extended layers to the wide movement frame.

The time division method uses the time axis and is shown in Figure 2. The redundancy is removed for coding movements and can be categorized by its application methods.

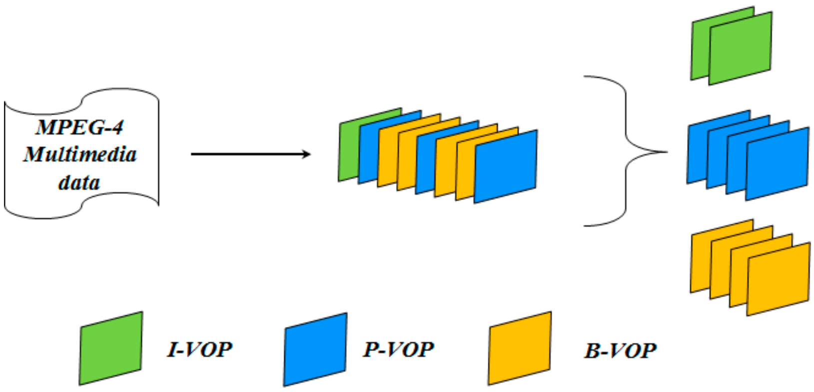


Figure 2. Time division method.

2.1.3. Relationship between Layers

Theoretically, an MPEG stream can be divided into 192 different layers by applying time and other division methods. The time division method produces up to 15 video layers that exhibit inter-dependencies as shown in Figure 3. Here, "*n*" represents the number of layers generated using spatial division. The base layer T_layer1 is a time division layer used to decode T-layer2, which in turn is used to decode T_layer3.

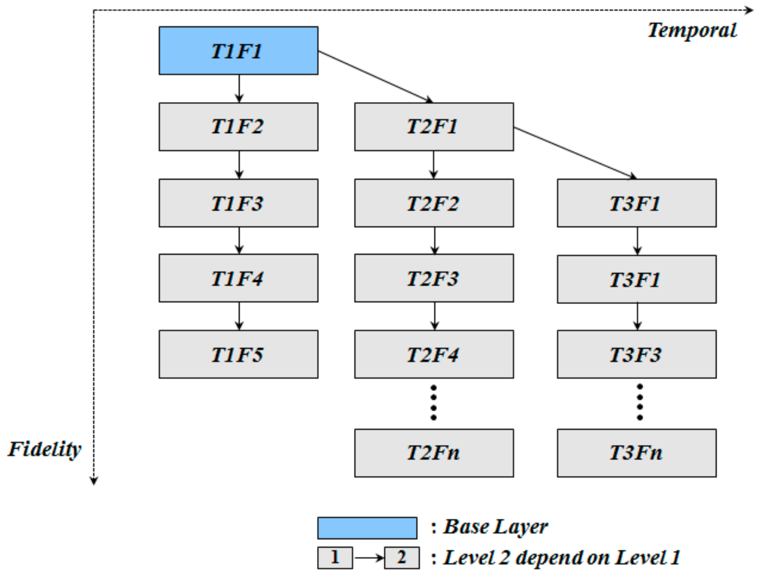


Figure 3. Relationship between the layers.

Layers that are divided based on the definition of an image also have dependencies. Each layer exists independently, but is dependent on other layers for replay. For decoding a lower layer, the higher layer should have already been decoded. This results in the existence of a base layer and arbitrary lower layers for encoding and decoding operations.

2.2. *Adaptive MPEG System*

An optimum stream transfer takes into consideration the real-time network bandwidth measurement. Figure 4 represents the proposed MPEG system structure that consists of an algorithm that splits a multimedia stream into many streams and transfers each of them to the client. The transferred streams are merged into a single multimedia stream and are sent to the client through the MPEG player. This process can provide a better service by using the real-time network bandwidth information to estimate the network load and minimize data loss. The proposed system uses an autonomous real-time bandwidth calibration method to monitor the changing stream information. It first sends metadata to the client and initiates stream service. This helps to adapt to bandwidth changes and provide better stream service.

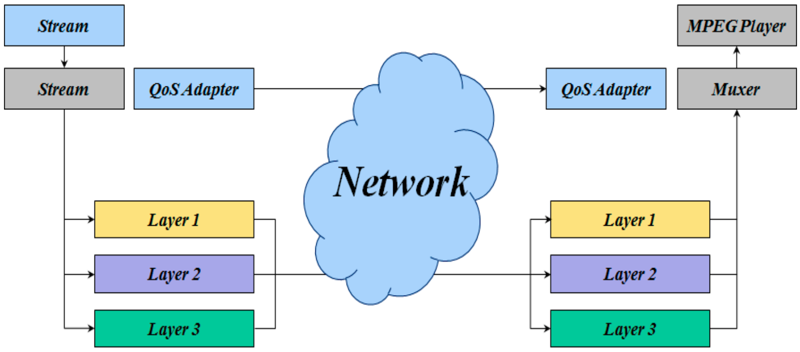


Figure 4. Adaptive MPEG system transfer module.

3. Adaptive QoS Transfer Algorithm

Figure 5 illustrates the temporal-fidelity scalability method used in transferring data from a server to the client. Steps 1–4 show the initial server and client states. The server sends a requested multimedia file to the client first and the actual multimedia data later as shown in steps 5–7. The received file is merged into a single file using the merging module. As shown in steps 8–11, this is played back to the client via a player. The QoS Monitor obtains the current session information based on the multimedia data (steps 10–11). This information is resent to the server QoS Adapter, which determines the data to be transferred next (steps 12–13).

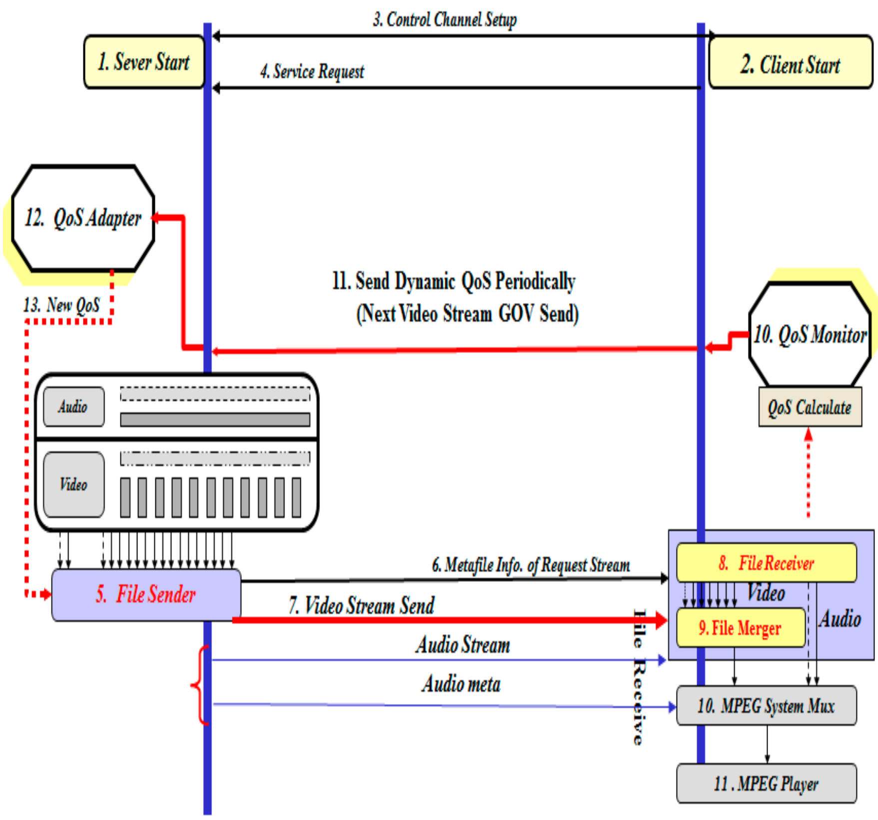


Figure 5. Adaptive QoS algorithm.

3.1. Session-Based Adaptive QoS Algorithm

Existing streaming services run over UDP, which enables them to meet real-time requirements. However, there is a possibility of losing important data when using UDP, which could result in a loss of integrity. On the other hand, the TCP ensures reliable delivery but at the cost of increased time latency. The proposed session transfer algorithm exploits the advantages of both the protocols. It uses UDP to transfer real-time data and TCP to transfer critical high-priority data. The categorization of the data is determined by client feedback and based on this, the session group is increased or decreased.

Figure 6 illustrates the session size (group) calculation at time instances A and B. The client side counts the number of packets n for the final transfer and the playback time from packet n to packet $n + 1$, where $n > 0$, to determine whether the receiving operation is complete. When the time of completion is smaller, it indicates that there is still time to receive more packets. The difference value can be used to determine the time required to send more packets and the next session packet number. The steps in the session size (group) calculation algorithm are shown in Algorithm 1.

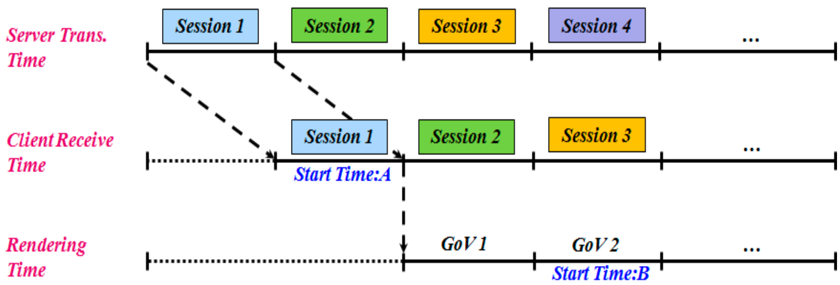


Figure 6. Session group calculation.

Algorithm 1. Session size calculation

NRT : the time it takes to play the next GoV
CRT : the time it takes to complete the current session
CSPN : the number of packets received in the current session
iLayerPackNo : the number of packets in a particular spatial and temporal layer obtained from the metafile
NSS : the size of the next session to be transmitted
PN : the number of extra packets that can be sent during the free time

```
T = NRT-CRT;
PN = ROUND((T * CSPN)/CRT);
NSS = 0;
 for (procedure of Spatial layer) {
   for (procedure of Temporal layer) {
     if ((NSS + iLayerPackNo) > CSPN + PN)
       break;
     else NSS += iLayerPackNo;
   }
}
```

3.2. FEC-Based Adaptive QoS Algorithm

The existing FEC method does not use client feedback information. The server calculation is performed for transmission operations only and places a heavy load on the network bandwidth. It is not suited for a variable network environment as it requires that the transmitted data segments have the same size. To address this issue, we propose that client feedback information be used to obtain knowledge of the network condition. This information is used to implement data redundancy and minimize data loss. Figure 7 illustrates the indirect loss situation that arises during network transfer after termination assuming that there is no higher layer.

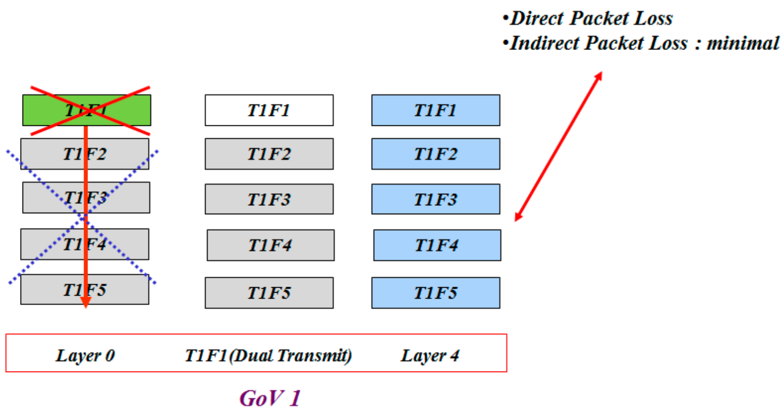


Figure 7. Indirect packet loss.

Consider a five-layered data (*T1F1~T1F5*) scenario, where the layers decrease in importance from top to bottom, i.e., *T1F1* > *T1F2* > *T1F3* > *T1F4* > *T1F5*. Here, *T1F2* needs *T1F1* for replay and so on. Therefore, when transferring data, *T1F1* is retransmitted to prevent indirect loss of other levels. The proposed algorithm uses feedback information from the client side to improve the QoS. The amount of data transmitted by the server and received by the client is calculated based on the feedback information and error rate of the current network. The metrics used to quantify this is

$$E_{rate} = 1 - \sum_{i=0}^{Gi} SRL_i / \sum_{i=0}^{Gi} SSL_i \tag{1}$$

where G_i is the number of transmitted GoV, SSL is the size of the transmitted data in GoV units, and SRL is the size of the received data in GoV units.

To reduce the network error rate, data can be retransmitted, but this is a waste of network bandwidth. The proposed algorithm uses client feedback information to determine an adaptive quantity of data for retransmission. In this mechanism, the key challenges are related to the determination of the desired QoS and the associated redundancy. The QoS is determined by using feedback information from the client. We assume that the same amount of data is sent again and therefore the size of the next data packet is known. The data quantity decision algorithm is shown in Algorithm 2.

Algorithm 2. Data quantity decision

```
iSizeofNet : the predictable network bandwidth
for (procedure of Temporal layer) {
  for (procedure of Spatial layer) {
    iSizeofLayer = size of layer;
    iSizeofSend = 0;
    if (expansion layer)
      iSizeofSend += iSizeofLayer;
    else (base layer)
      iSizeofSend += (iSizeofLayer * amount of redundancy)
    if (iSizeofSend > iSizeofNet) break;
  }
  if (iSizeofSend > iSizeofNet) break;
}
```

The error-rate calculation, transfer determination algorithm, transfer data error rate, and data redundancy should be determined. For meaningful MPEG streaming, higher layer data associated with the current layer should be transmitted first. The proposed algorithm guarantees that this higher layer data is transferred with a higher reliability. The probability of redundant transfer can be derived from Equations (2) and (3).

$$(T_1S_1) = P_{111} \times P_{112} \times \cdots \times P_{11N_{11}} = \prod_{k=1}^{N_{11}} P_{11k} \tag{2}$$

where P_{ijn} is the probability that the nth packet of the T_iS_j *layer* will be transmitted, N_{ij} is the number of packets in T_iS_j *layer*, and ΔP is the size of the packet.

$$\left(\prod_{k=1}^{N_{11}} P_{11k}\right) \times Dup > ProBotLmt \tag{3}$$

where *ProBotLmt* is the minimum probability that a layer should be transmitted and *DuP* is the amount of redundant transmission.

4. Experimental Evaluation and Performance Analysis

The performance of the proposed algorithm was analyzed by simulation using a video stream. The adaptive QoS algorithm's performance was compared with a transfer technique that did not consider the QoS. The direct and indirect packet losses were computed in terms of the GoV packet size by processing the merged data at the client side. The result proved that the proposed adaptive QoS algorithm is better than the current methods.

Figure 8 illustrates a simulation environment for the performance evaluation of the proposed algorithms. The streaming server was implemented in Visual C++ under Windows 2000, and the client experimented with a laptop using an Intel Core (TM) i3 CPU. The TS (Transport Stream) Divider is responsible for dividing; the TS Sender is in charge of sending the divided stream; the TS Writer is in charge of data receiving; and the TS Merger decodes the received data. The Error Rate Controller determines the loss of the packet sent and the Error Inspector measures the indirect data loss between the higher layer and lower layer in accordance with the presented error rate.

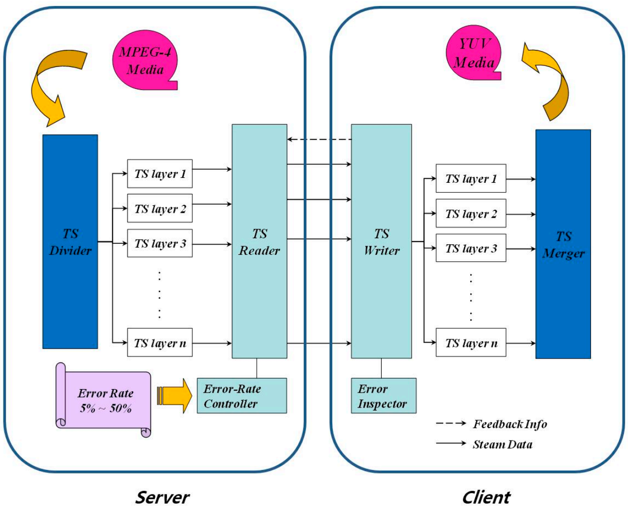
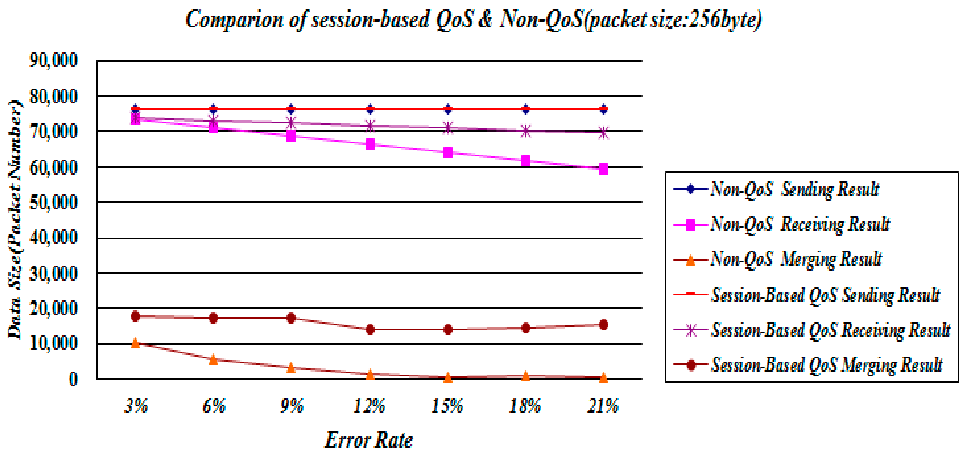


Figure 8. Simulation diagram.

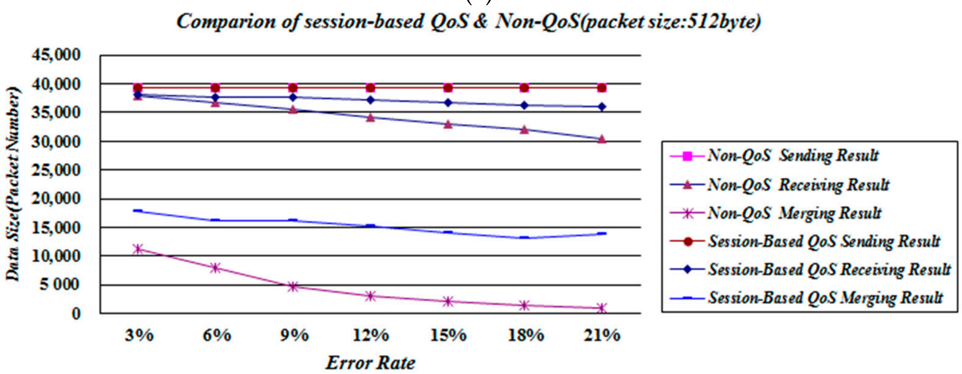
The streaming data used in the experiment is an MPEG-4 stream called Foreman, which is 176×144 pixels in size. The total number of frames is 400 frames, and the number of frames per second is 30 frames. We divide the data into 12 layers through the divider of the server and assume that the data that can be transmitted in each time unit is the same. We controlled the error of the network in between 5% and 50% for comparison of send efficiency.

4.1. Performance of the Session-Based Adaptive QoS and Non-QoS Techniques

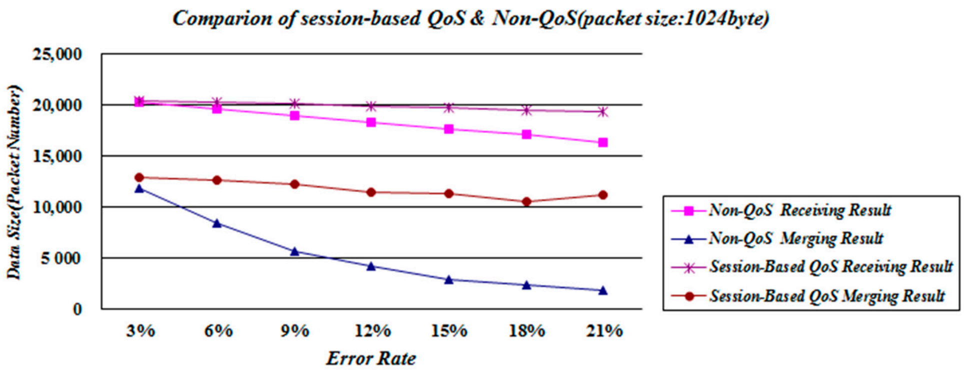
The data transmitted using the non-QoS and session-based adaptive QoS techniques was not large in size. However, the final merged multimedia data showed big differences because the relationship between the layers was different. Figure 9a–c show the comparison results. This demonstrates that the lower layer can only be decoded by using higher layer data and that the loss of higher layer data can result in loss of lower layer data. To minimize this, the session-based method transmits this layer using TCP, resulting in a more reliable transfer as compared to UDP. The transfer time is reduced, however, reliable transfer is acquired. This difference widens as the packet size increases. As the data transfer rate decreases, the data transfer failure possibility and the indirect loss size increase.



(**a**)



(**b**)

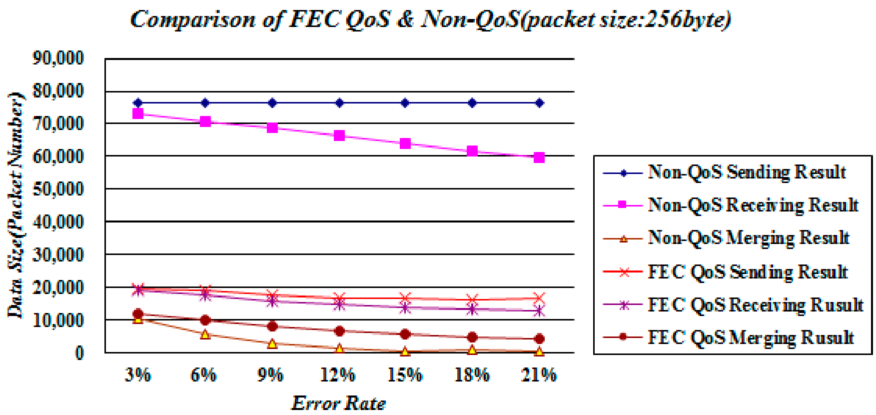


(**c**)

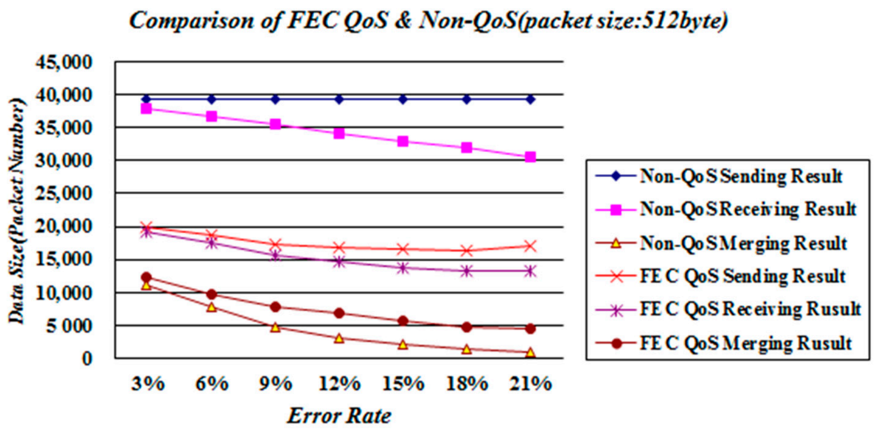
Figure 9. (**a**) Non-QoS and Session-based adaptive QoS-1; (**b**) Non-QoS and Session-based adaptive QoS-2; (**c**) Non-QoS and Session-based adaptive QoS-3.

4.2. FEC-Based Adaptive QoS and Non-QoS Performance Compare

Figure 10a–c show the comparison of results obtained using the FEC-based adaptive QoS algorithm and the non-QoS technique. The FEC-based adaptive QoS algorithm transfer result is lower in comparison with non-QoS. This result is a consequence of the adaptive QoS algorithm's flow control.



(**a**)



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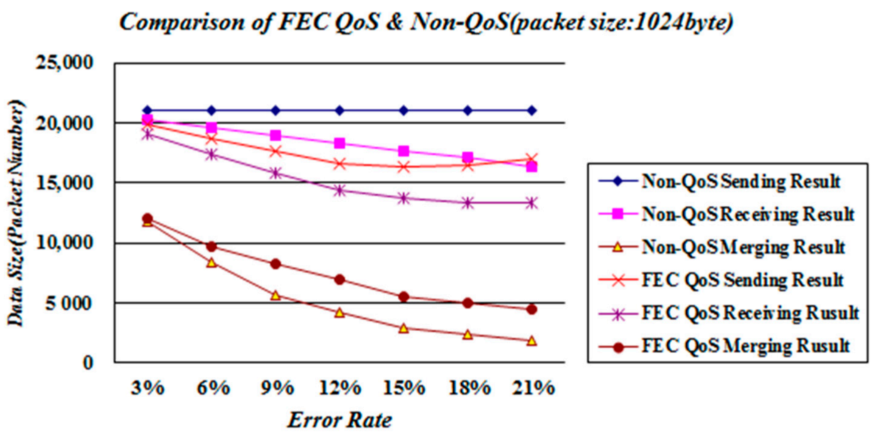

(**c**)

Figure 10. (**a**) FEC method adaptive QoS and non-QoS-1; (**b**) FEC method adaptive QoS and non-QoS-2; (**c**) FEC method adaptive QoS and non-QoS-3.

To compare the two techniques, the same amount of data is transferred using both. As the sizes of all the layers are not equal, we observe, from Figure 10a–c, that the transferred data is not continuous. The amount of data received using the FEC-based adaptive QoS method is smaller than that received from the non-QoS technique. Nevertheless, the whole data transfer loss rates are similar because of the direct loss factor. In the real-time data quantity comparison, the FEC-based adaptive QoS system shows better results. This is caused by FEC-based redundant retransmission of data. The higher level data is repeatedly sent, resulting in a lower transfer failure rate as compared to the non-QoS system. The FEC-adaptive QoS transfer method reduces the indirect loss of the transfer.

4.3. Performance Analysis Result

The performance of the adaptive QoS transfer algorithm and the existing transfer algorithm are compared. The experimental results show that the proposed method exhibits better data transfer results. During the transfer of a large multimedia file, the indirect loss of data caused by network limitations is comparable to the direct loss. It is almost impossible to compensate for direct loss at the client side. Therefore, data recovery techniques based on the transfer data specification that reduce indirect losses play a vital role. The proposed method is based on the principle that the higher layer transmission must be reliable in order to improve overall transmission efficiency. The server determines the data transfer rate by using the real-time bandwidth information. This results in an efficient data transfer system. In order to quantify the efficiency of the system, we measured the error rate in data. The results indicate that the adaptive QoS algorithm system performs better than the non-QoS technique.

5. Conclusions

Multimedia transmissions have gained research interest due to advances in networking. The emergence of mobile communication technologies and improved device performance are the key factors that have motivated studies on multimedia data transfer. This paper proposed two adaptive QoS transfer algorithms. The first method involved the selection of important data based on relationships and the other advocated setting of data transmission rates in accordance with the network bandwidth. When applied to a network that involves multimedia transmission, this adaptive QoS algorithm can be used to solve different problems. The proposed adaptive QoS data transfer algorithm can be adapted for use in wired and wireless networks. It can be considered as a service for server/client-based network structure that guarantees reliable service. In our future research, we will be analyzing new QoS algorithms and transfer-error prevention methods to address various heterogeneous network quality issues.

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References

1. Tian, G.; Liu, Y. Towards Agile and Smooth Video Adaptation in HTTP Adaptive Streaming. *IEEE/ACM Trans. Netw.* **2016**, *24*, 2386–2399. [CrossRef]
2. Liu, J.; Li, B.; Hou, Y.T.; Chlamtac, I. On optimal layering and bandwidth allocation for multisession video broadcasting. *IEEE Trans. Wirel. Commun.* **2004**, *3*, 656–667. [CrossRef]
3. Lee, J.; Lee, K.; Han, C.; Kim, T.; Chong, S. Resource-Efficient Mobile Multimedia Streaming with Adaptive Network Selection. *IEEE Trans. Multimed.* **2016**, *18*, 2517–2527. [CrossRef]

4. Oyman, O.; Singh, S. Quality of experience for HTTP adaptive streaming services. *IEEE Commun. Mag.* **2012**, *50*, 20–27. [CrossRef]
5. Mok, R.K.P.; Chan, E.W.W.; Chang, R.K.C. Measuring the quality of experience of HTTP video streaming. In Proceedings of the 12th IFIP/IEEE International Symposium on Integrated Network Management, Dublin, Ireland, 23–27 May 2011; pp. 485–492.
6. Chen, C.; Choi, L.K.; Veciana, G.D.; Caramanis, C.; Heath, R.W., Jr.; Bovik, A.C. A dynamic system model of time-varying subjective quality of video streams over HTTP. In Proceedings of the 2013 International Conference on Acoustics, Speech, and Signal Processing, Vancouver, BC, Canada, 26–31 May 2013; pp. 3602–3606.
7. Bhargava, B.; Wang, S.Y.; Khan, M.; Habib, A. Multimedia data transmission and control using active networks. *Comput. Commun.* **2016**, *36*, 632–639. [CrossRef]
8. Yang, H.S.; Sun, J.H. A Study on Stable Data Transmission Using Hierarchical Share Group in Mobile Ad Hoc Network. *Wirel. Pers. Commun.* **2016**, *86*, 333–349. [CrossRef]
9. Ji, W.; Li, Z. Heterogeneous QoS video broadcasting with optimal joint layered video and digital fountain coding. In Proceedings of the 2011 IEEE International Conference on Communications, Kyoto, Japan, 5–9 June 2011; pp. 1–6.
10. Ahn, S.H.; Kang, M.G.; Kim, D.H.; Kim, H.C. QoS adaptive MPEG-2 streaming based on scalable Media Object Framework. In Proceedings of the International Conference on Information Networking, Beppu City, Oita, Japan, 31 January–2 February 2001; pp. 683–688.
11. Kumar, S.; Sarkar, M.; Gurajala, S.; Matyjas, J.D. MMMP: A MAC Protocol to Ensure QoS for Multimedia Traffic over Multi-hop Ad Hoc Networks. *J. Inf. Process. Syst.* **2008**, *4*, 41–52. [CrossRef]
12. Rodriguez, D.Z.; Rosa, R.L.; Alfaia, E.C.; Abrahão, J.I.; Bressan, G. Video Quality Metric for Streaming Service Using DASH Standard. *IEEE Trans. Broadcast.* **2016**, *62*, 628–639. [CrossRef]
13. Luo, H.; Shyu, M.L. Quality of service provision in mobile multimedia—A survey. *Hum.-Centric Comput. Inf. Sci.* **2011**, *1*, 5. [CrossRef]
14. Li, Q. Providing Adaptive QoS to Layered Video Over Wireless Local Area Networks through Real-Time Retry Limit Adaptation. *IEEE Trans. Multimed.* **2004**, *6*, 278–290. [CrossRef]
15. Zivizni, A.; Wolfinger, B.E.; de Rezende, J.F.; Duarte, O.C.M.B.; Fdida, S. Joint Adoption of QoS Schemes for MPEG Streams. *Multimed. Tools Appl.* **2005**, *26*, 59–80. [CrossRef]
16. Verscheure, O.; Garcia, X.; Karlsson, G.; Hubaux, J.P. User-Oriented QoS in Packet Video Delivery. *IEEE Netw.* **1998**, *12*, 12–21. [CrossRef]
17. Wolfinger, B.E.; Zaddach, M. Techniques to Improve Quality-of-Service in Video Communications via Best Effort Networks. In Proceedings of the International Conference on Networking, Colmar, France, 9–13 July 2001; pp. 754–765.
18. Shin, J.; Kim, J.W.; Lee, D.C.; Kuo, C.C.J. Adaptive Packet Forwarding for Relative Differentiated Services and Categorized Packet Video. In Proceedings of the IEEE International Conference on Communications, Helsinki, Finland, 11–14 June 2001; pp. 763–767.
19. Shin, J.; Kim, J.; Kuo, C.C.J. Quality of service mapping mechanism for packet video in differentiated services network. *IEEE Trans. Multimed.* **2001**, *3*, 219–231. [CrossRef]
20. Da Silva Gonçalves, P.A.; Rezende, J.F.; Duarte, O.C.M.B.; Pujolle, G. Optimal Feedback for Quality Source-Adaptive Schemes in Multicast Multi-layered Video Environments. In Proceedings of the Second International IFIP-TC6 Networking Conference, Pisa, Italy, 19–24 May 2002; pp. 563–574.
21. Cavusoglu, B.; Schonfeld, D.; Ansari, R.; Kumar Bal, D. Real-Time Low-Complexity Adaptive Approach for Enhanced QoS and Error Resilience in MPEG-2 Video Transport Over RTP Networks. *IEEE Trans. Circuits Syst. Video Technol.* **2005**, *15*, 1604–1614. [CrossRef]
22. Pliakas, T.; Kormentzas, G.; Skianis, C. End-to-end QoS issues of MPEG-4 FGS video streaming traffic delivery in an IP/DVB/UMTS network. *Eur. J. Oper. Res.* **2008**, *191*, 1089–1100. [CrossRef]
23. Cranley, N.; Davis, M. Performance Analysis of Network-level QoS with Encoding Configurations for Unicast Video Streaming over IEEE 802.11 WLAN Networks. In Proceedings of the 2005 International Conference on Wireless Networks, Communications and Mobile Computing, Maui, HI, USA, 13–16 June 2005; pp. 510–515.
24. Ng, J.K.Y.; Leung, K.R.P.H.; Hui, C.K.C. A QoS-Enabled Transmission Scheme for MPEG Video Streaming. *Real-Time Syst.* **2005**, *30*, 217–256. [CrossRef]

25. Malgi, M.A.; Gaikwad, G.N. A Study on QoS Enhancement of MPEG-4 Video Transmission over Wireless Mesh Network. In Proceedings of the 2005 International Conference on Pervasive Computing, Pune, India, 8–10 January 2015; pp. 1–5.
26. Song, D.W.; Chen, C.W. QoS Guaranteed SVC-based Video Transmission over MIMO Wireless Systems with Channel State Information. In Proceedings of the 2006 IEEE International Conference on Image Processing, Atlanta, GA, USA, 8–11 October 2006; pp. 3057–3060.
27. Fan, Y.; Su, F.; Li, Y.; Xu, H. Netwvork-aware Adaptive QoS Archiitecture or Video Delivery over Differentiated Service Network. In Proceedings of the 2006 International Conference on ITS Telecommunications, Chengdu, China, 21–23 June 2006; pp. 1330–1333.
28. Ruijin, F.; Busung, L.; Gupta, A. Scalable Layered MPEG-2 video Multicast Architecture. *IEEE Trans. Consum. Electron.* **2001**, *47*, 55–62.
29. Arvind, R.; Civanlar, R.; Reibman, R. Packet Loss Resilience of MPEG-2 Scalable Video Coding Algorithms. *IEEE Trans. Circuits Syst. Video Technol.* **1996**, *6*, 426–435. [CrossRef]
30. Hsio, Y.-M.; Chen, C.-H.; Lee, J.-F.; Chu, Y.-S. Designing and implementing a scalable video-streaming system using an adaptive control scheme. *IEEE Trans. Consum. Electron.* **2012**, *58*, 1314–1322. [CrossRef]
31. Van Der Schaar, M.; Radha, H. A Hybrid Temporal-SNR Fine-Granular Scalability for Internet Video. *IEEE Trans. Circuits Syst. Video Technol.* **2001**, *11*, 318–331. [CrossRef]
32. Kim, T.J.; Kim, B.G.; Park, C.S.; Jang, K.S. Efficient Block Mode Determination Algorithm Using Adaptive Search Direction Information for Scalable Video Coding (SVC). *J. Converg.* **2014**, *5*, 14–19.
33. Wien, M.; Schwarz, H.; Oelbaum, T. Performance analysis of SVC. *IEEE Trans. Circuits Syst. Video Technol.* **2007**, *17*, 1194–1203. [CrossRef]
34. Zhang, Q.; Zhu, W.; Zhang, Y.Q. Network-Adaptive Rate Control and Unequal Loss Protection with TCP-Friendly Protocol for Scalable Video over Internet. *J. VLSI Signal Process. Syst. Signal Image Video Technol.* **2003**, *34*, 67–81. [CrossRef]
35. Schierl, T.; Stockhammer, T.; Wiegand, T. Mobile Video Transmission Using Scalable Video Coding. *IEEE Trans. Circuits Syst. Video Technol.* **2007**, *17*, 1204–1217. [CrossRef]
36. Schwarz, H.; Marpe, D.; Wiegand, T. Overview of the scalable video coding extension of H.264/AVC. *IEEE Trans. Circuits Syst. Video Technol.* **2007**, *17*, 1103–1120. [CrossRef]
37. Wiegand, T.; Sullivan, G.J.; Bjontegaard, G.; Luthra, A. Overview of the H.264/AVC video coding standard. *IEEE Trans. Circuits Syst. Video Technol.* **2003**, *13*, 560–576. [CrossRef]
38. Radha, H.; van der Schaar, M.; Chen, Y. The MPEG-4 fine-grained scalable video coding method for multimedia streaming over IP. *IEEE Trans. Multimed.* **2001**, *3*, 53–68. [CrossRef]
39. Song, D.W.; Chen, C.W. Scalable H.264/AVC Video Transmission over MIMO Wireless Systems with Adaptive Channel Selection Based on Partial Channel Information. *IEEE Trans. Circuits Syst. Video Technol.* **2007**, *17*, 1218–1226. [CrossRef]
40. Van der Auwera, G.; David, P.T.; Reisslein, M. Traffic and Quality Characterization of Single-Layer Video Streams Encoded with the H.264/MPEG-4 Advanced Video Coding Standard and Scalable Video Coding Extension. *IEEE Trans. Broadcast.* **2008**, *54*, 698–718. [CrossRef]
41. Xu, C.; Fallon, E.; Qiao, Y.; Zhong, L.; Muntean, G.M. Performance Evaluation of Multimedia Content Distribution over Multi-Homed Wireless Networks. *IEEE Trans. Broadcast.* **2011**, *57*, 204–215.
42. Wu, D.; Hou, Y.T.; Zhu, W.; Lee, H.J.; Chiang, T.; Zhang, Y.-Q. On End-to-End Transport Architecture for MPEG-4 Video Streaming over the Internet. *IEEE Trans. Circuits Syst. Video Technol.* **2000**, *10*, 923–941.
43. Chang, K.-D.; Chen, C.-Y.; Chen, J.-L.; Chao, H.-C. Challenges to Next Generation Services in IP Multimedia Subsystem. *J. Inf. Process. Syst.* **2010**, *6*, 129–146. [CrossRef]
44. Tan, W.; Zakhor, A. Real-Time Internet Video Using Error Resilient Scalable Compression and TCP-Friendly Transport Protocol. *IEEE Trans. Multimed.* **1999**, *1*, 172–186. [CrossRef]
45. El Essaili, A.; Schroeder, D.; Staehle, D.; Shehada, M.; Kellerer, W.; Steinbach, E. Quality-of-experience driven adaptive HTTP media delivery. In Proceedings of the 2011 IEEE International Conference on Communications, Budapest, Hungary, 9–13 June 2013; pp. 2480–2485.
46. Park, S.H.; Lee, S.J.; Kim, J.W. Network-Adaptive High Definition MPEG-2 Streaming over IEEE 802.11a WLAN using Frame-based Prioritized Packetization. In Proceedings of the Third ACM International Workshop on Wireless Mobile Applications and Services on WLAN Hotspots, Cologne, Germany, 2 September 2005; pp. 84–87.

47. Setton, E.; Yoo, T.; Zhu, X.; Goldsmith, A.; Girod, B. Cross-layer design of *ad-hoc* networks for real-time video streaming. *IEEE Wirel. Commun. Mag.* **2005**, *12*, 59–65. [CrossRef]
48. Taubman, D.; Thie, J. Optimal erasure protection for scalably compressed video streams with limited retransmission. *IEEE Trans. Image Process.* **2005**, *14*, 1006–1019. [CrossRef] [PubMed]
49. Kim, S.H.; Lee, S.I.; Yoo, W.K.; Yoo, K.J. A Study of Session-based Algorithm for Optimal Streaming Data Transmission. In Proceedings of the 4th Asia-Pacific International Symposium on Information Technology, Gold Coast, Australia, 26–27 January 2005; pp. 519–522.
50. Vakili, A.; Grégoire, J.-C. Modelling the Impact of the Position of Frame Loss on Transmitted Video Quality. *J. Converg.* **2011**, *2*, 43–48.
51. Van der Schaar, M.; Turaga, D.S. Cross-Layer Packetization and Retransmission Strategies for Delay-Sensitive Wireless Multimedia Transmission. *IEEE Trans. Multimed.* **2007**, *9*, 185–197. [CrossRef]
52. Rhee, I.; Joshi, S.R. Error Recovery for Interactive Video Transmission over the Internet. *IEEE J. Sel. Area Commun.* **2000**, *18*, 1033–1049. [CrossRef]
53. Lindeberg, M.; Kristiansen, S.; Plagemann, T.; Goebel, V. Challenges and techniques for video streaming over mobile ad hoc networks. *Multimed. Syst.* **2011**, *17*, 51–82. [CrossRef]
54. Majumdar, A.; Sachs, D.G.; Kozintsev, I.V.; Ramchandran, K. Multicast and unicast real-time video streaming over wireless LANs. *IEEE Trans. Circuits Syst. Video Technol.* **2002**, *12*, 524–534. [CrossRef]
55. Liu, Q.; Wang, X.; Giannakis, G.B. A Cross-Layer Scheduling Algorithm with QoS Support in Wireless Networks. *IEEE Trans. Veh. Technol.* **2006**, *55*, 839–847. [CrossRef]
56. Xu, C.; Fallon, E.; Qiao, Y.; Muntean, G.-M.; Li, X.; Hanley, A. Performance Evaluation of Distributing Real-time Video over Concurrent Multipath. In Proceedings of the Wireless Communications and Networking Conference, Budapest, Hungary, 5–8 April 2009; pp. 1–6.
57. Pinzon, M.H.; Choi, L.K.; Bovick, A.C. Temporal video quality model accounting for variable frame delay distortions. *IEEE Trans. Broadcast.* **2014**, *60*, 637–649. [CrossRef]
58. Kim, S.H.; Yoo, K.J. An FEC-based error control scheme for Layered MPEG media delivery. In Proceedings of the 2009 International Conference on Multimedia, Information Technology and Its Applications, Osaka, Japan, 19–21 August 2009; pp. 307–310.
59. Marcotte, R.J.; Olson, E. Adaptive forward error correction with adjustable-latency QoS for robotic network. In Proceedings of the 2016 IEEE International Conference on Robotics and Automation, Stockholm, Sweden, 16–21 May 2016; pp. 5283–5288.
60. Wu, H.; Claypool, M.; Kinicki, R. Adjusting Forward Error Correction with Temporal Scaling for TCP-Friendly Streaming MPEG. *ACM Trans. Multimed. Comput. Commun. Applicat.* **2005**, *1*, 315–337. [CrossRef]
61. Puri, R.; Ramchandran, K.; Lee, K.W.; Bharghavan, V. Forward error correction(FEC) codes based multiple description coding for internet video streaming and multicast. *Signal Process. Image Commun.* **2001**, *16*, 745–762. [CrossRef]
62. Frossard, P.; Verscheure, O. Joint Source/FEC Rate Selection for Quality-Optimal MPEG-2 Video Delivery. *IEEE Trans. Image Process.* **2001**, *10*, 1815–1825. [CrossRef] [PubMed]
63. Mohr, A.E.; Riskin, E.A.; Ladner, R.E. Unequal Loss Protection: Graceful Degradation of Image Quality over Packet Erasure Channels through Forward Error Correction. *IEEE J. Sel. Areas Commun.* **2000**, *18*, 819–828. [CrossRef]
64. Cosman, P.C.; Rogers, J.K.; Sherwood, P.G.; Zeger, K. Combined forward error control and packetized zerotree wavelet encoding for transmission of images over varying channels. *IEEE Trans. Image Process.* **2000**, *9*, 982–993. [CrossRef] [PubMed]
65. Sherwood, P.G.; Zeger, K. Error protection for progressive image transmission over memoryless and fading channels. *IEEE Trans. Commun.* **1998**, *46*, 1555–1559. [CrossRef]

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