Adaptive Error Control Scheme for Multimedia Applications in Integrated Terrestrial-Satellite Wireless Networks

Sungrae Cho

Broadband and Wireless Networking Laboratory
School of Electrical and Computer Engineering
Georgia Institute of Technology, Atlanta, GA 30332

Email: srcho@ece.gatech.edu

Abstract— **This paper presents an adaptive error control (AEC) scheme for multimedia applications in integrated terrestrial-satellite wireless networks. The AEC protocol supports both real-time and non-real-time applications. In the AEC protocol, we propose new adaptive FEC (AFEC) and hybrid ARQ (HARQ) schemes for real-time and non-real-time traffic, respectively. Throughput performance for non-real-time application shows that the proposed AEC protocol outperforms hybrid ARQ (HARQ) protocols with the same code used. Under real-time application, the AEC protocol outperforms the static FEC (SFEC) protocols with respect to packet miss probability.**

Keywords— **Error Control, ARQ, FEC, and Terrestrial-Satellite Hybrid Networks**

I. Introduction

Terrestrial-satellite hybrid networks have become increasingly popular for multimedia services due to several advantages [5]. Many multimedia applications, particularly in case of file transfer, video multicast, and database services, are asymmetric in nature, i.e., the bandwidth requirement from a transmitter is different from that of a receiver. Although two-way satellite channels can be used for such asymmetric applications, it is also possible to combine a one-way satellite channel for information flow with a parallel terrestrial channel for control data flow, e.g., retransmission request [7]. This can remarkably reduce the cost of relatively expensive and scarce satellite feedback channel. Furthermore, satellites are able to offer efficient bandwidth services to a large geographical area and easy to add new users to the system by simply installing the stations, i.e., possible network expansions will be a simple task. Hence, the satellite network is an excellent infrastructure for multimedia multicast services [1]. Possible applications [5] of the integrated terrestrial-satellite wireless network include:

This work is supported by NASA-Ames under Contract #NAG2-1262.

- Data transfer to large user populations,
- Continuous sensor data feed to large user populations,
- Group tele-conferencing,
- Dissemination of video and/or audio streams to large user populations,
- Distance learning, and
- Distributed interactive simulation

However, satellite link is not a perfect communication channel as it has both non-homogeneous and dynamic characteristics. This time-varying channel condition would create bursty errors. As in all communication systems, an error control protocol is required for multimedia services in the terrestrial-satellite hybrid networks [5]. In terms of error control, multimedia application can be roughly divided into two traffic types: *non-real-time* application such as data and image, and *real-time* application such as audio and video [3]. Real-time applications have lower quality of service (QoS) commitments regarding packet error rate (PER) than non-real-time applications because of the limitation of human ears or eyes. However, they are less tolerant to delays than non-real-time applications which require higher reliability than real-time applications [3]. In general, automatic repeat request (ARQ) protocols are used for non-real-time applications; forward error correction (FEC) protocols are used for real-time applications.

This paper presents an adaptive error control (AEC) system for multimedia applications in the integrated terrestrial-satellite wireless networks. In our error control system, original data packet is transmitted via satellite, while the retransmissions of the packet are carried out over terrestrial link. All corresponding control packets are transmitted through terrestrial links. The proposed protocol supports both real-time and non-real-time applications. In the AEC protocol, we propose a new adaptive

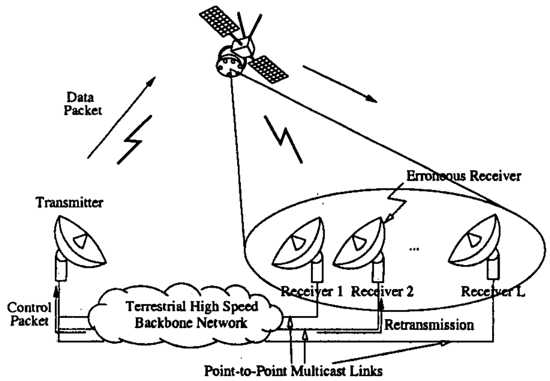

Fig. 1. Satellite-Terrestrial Hybrid Network Architecture

FEC (AFEC) scheme for real-time traffic and a new hybrid ARQ (HARQ) for non-real-time traffic.

The presentation of our error-control scheme is organized as follows: the AEC protocol is presented in section II; channel model is described in section III, followed by the performance evaluation in section IV; then, conclusions are drawn in section V.

II. Protocol Description

Consider a pure-satellite multicast network in which retransmission as well as transmission of a packet take place through the satellite link. If only one receiver gets an erroneous packet and requests a retransmission, then other good receivers are forced to receive redundant (futile) packets. This can breach the fairness of all other good users in the multicast group. If the ARQ scheme takes secondary route such as terrestrial link, the retransmission could be sent only to the receiver with an erroneous packet. In our error control system as shown in Fig. 1, original data packet is transmitted via satellite, while the retransmissions of the packet are carried out over terrestrial link. All corresponding control packets are transmitted through terrestrial links. In this way, we show that the integrated terrestrial-satellite wireless networks substantially increase the throughput efficiency [5].

Most error control protocols are typically designed with fixed values for link layer parameters such as coding rate and error combating capability for the worst channel condition. This can provide error probability to be below a predefined value. However, throughput performance becomes smaller than the achievable performance using optimum code parameters. A more efficient approach is to use an adaptive error control scheme that responds to the actual channel error condition by selecting the optimum code rate [9]. In this section, the proposed adaptive error control (AEC) protocol is presented. For non-real-time traffic, we use a hybrid ARQ (HARQ); for real-time traffic, we use an adaptive FEC (AFEC), as shown in Fig. 2.

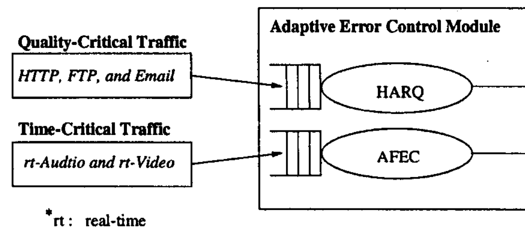

Fig. 2. Adaptive Error Control Module.

In section II-A, a proposed HARQ protocol for non-real-time traffic is presented, followed by a proposed AFEC protocol for real-time traffic in section II-B.

A. Hybrid ARQ

Let define the code set $C = \{c_1, c_2, \ldots, c_N\}$, where the code rate of code c_i is greater than that of code c_{i-1} for all $i \in \{2, 3, \ldots, N\}$, and N is the size of the code set. Then, a transmitter can use any code of a given set C, in each of its transmissions. Hence, the proposed HARQ is a family of type II HARQ protocol. In our proposed hybrid ARQ (HARQ), a receiver feeds back positive or negative acknowledgment (ACK/NAK) for each data packet, and also provides the code used in the data packet.

Then, the transmitter computes the packet error rate (PER) ξ_i for each code c_i based on the number of ACKs and NAKs fed from the receiver. The PER is reset and then updated whenever the measurement period T_m is expired.

Then, the throughput efficiency η_i for code c_i is given by

$$\eta_i = (1 - \xi_i)\gamma_i, \qquad \forall i \in C \tag{1}$$

where γ_i is the code rate when code c_i is used.

When there is a data packet to be sent at a transmitter, the transmitter computes throughput efficiency for each code based on (1) and finds the code c with maximum throughput efficiency, η_c given by

$$c \quad = \quad \arg\max_{i \in C} \eta_i \tag{2}$$

B. Adaptive FEC

For real-time applications which has deadline constraints, retransmission of a packet whose deadline is expired, is futile and discarded by the application. This results in deterioration in QoS perceived by the application. Hence, in the AEC protocol, a transmitter decides to retransmit a packet or not based on the its deadline. Moreover, our scheme chooses the code rate providing maximum throughput efficiency based on the channel state as in the proposed Hybrid ARQ scheme.

In our scheme, a transmitter does not retransmit a packet, if the probability that a packet cannot be transmitted correctly in a deadline D is greater than a predefined packet miss probability, P_m. In other words, a

AEC ALGORITHM AT TRANSMITTER

```
if ((Application = Real-Time) and (Retransmission))
  // Real-Time Application //
  η_max ← 0;
  Opt_Code ← 1;
  for (i = 1; i ≤ N; i++)
    Calculate P_c using (3) for Code c_i;
    if (P_c < 1 − P_m)
      Calculate η_i using (1) for Code c_i;
      if (η_i > η_max)
        η_max ← η_i;
        Opt_Code ← i;
      end;
    end;
  end;
  if (η_max = 0)
    Do not retransmit the packet;
  else
    Select the Code with Opt_Code;
  end;
else // Non-Real-Time Application
  or First Transmission of Real-Time Traffic //
  η_max ← 0;
  Opt_Code ← 1;
  for (i = 1; i ≤ N; i++)
    Calculate η_i using (1) for Code c_i;
    if (η_i > η_max)
      η_max ← η_i;
      Opt_Code ← i;
    end;
  end;
  Select the Code with Opt_Code;
end;
end.
```

Fig. 3. AEC Algorithm at Transmitter

transmitter retransmits the packet only if the probability that a packet can be transmitted correctly within D, P_c, is less than $1 - P_m$, where P_c, is given by

$$\begin{aligned} P_c &= P[\text{No. of retransmission} \leq \lfloor \frac{D-t}{RTD} \rfloor] \\ &= P_s(1-P_t) \sum_{i=1}^{\lfloor (D-t)/RTD \rfloor} P_t^{i-1} \qquad (3) \\ &= P_s[1 - P_t^{\lfloor (D-t)/RTD \rfloor}] \end{aligned}$$

where P_s and P_t is packet error probability for satellite and terrestrial link respectively. t is the current time at the instance of retransmission and RTD is the average round-trip delay.

At the receiver, our scheme decides whether to request a retransmission or not, based on the deadline requirement. Suppose an erroneous packet having a deadline of D arrives at the receiver at time t. Only for $\lfloor \frac{D-t}{RTD} \rfloor \geq 1$, the receiver requests a retransmission.

The detail of the AEC algorithm at transmitter is given in Fig. 3.

C. Packet Format

In order to support the schemes described in sections II-A and II-B, data and control packets should have adequate fields. All of the packets in this algorithm assume that the transmitter uses a framing mechanism that accommodates variable packet sizes, as in the case for most link layer framing mechanisms used in practice [2]. Also, packet fields and corresponding sizes are subject to network design implementation. Fig. 4 shows our suggestion for data and control packets.

TI	TOS	SN	LN	Rate	Data	FEC
(1bit)	(1bit)	(8bits)	(16bits)	(2bits)	(variable)	(variable)

(a) Data Packet Format

TI	TOS	SN	ACK/NAK	CODE	FEC
(1bit)	(1bit)	(8bits)	(1bit)	(2bits)	(19bits)

(b) Control Packet Format

Fig. 4. Packet Formats.

There are four types of fields in common for data and control packets: TI, TOS, SN, and FEC. TI is *type indication* which is used to indicate whether the packet is data or control. TOS is *type of service* which distinguishes if the application is real-time or non-real-time. If it indicates real-time, the proposed AFEC scheme will be performed; otherwise, the proposed HARQ protocol will be used. *sequence number* SN is to detect packet loss at the receiver. This field is intended to measure the round-trip delay (RTD). FEC for data packet is a AEC code. Its size is determined by *Rate* field. FEC for control packet uses static FEC (SFEC) with powerful error correcting capability.

Rate indicates which code is used and it is used to decode the receiving packets. LN field is the *length* of the data in octet since the data size is variable. *Data* is user information with variable size. ACK/NAK field indicates whether the message is a positive acknowledgment (ACK) or negative acknowledgment (NAK). $CODE$ is used to identify the code used for the data packet.

D. Channel Coding

For channel coding, we use a concatenated FEC code, which incorporates Reed-Solomon (RS) code and rate-compatible punctured convolutional (RCPC) code [10]. Concatenation is a scheme in which two codes, an *inner code* and an *outer code*, are used in tandem. The inner code should be designed to produce a moderate bit error rate with modest complexity, whereas the outer code can be more complex and should be designed to correct almost all the residual errors from the inner decoder [6]. When the outer and inner codes work in tandem, the inner code corrects most errors and spreads out burst errors, then the outer code corrects the small block errors that remain. In our concatenated FEC, the inner coding scheme is a rate-compatible convolutional code (RCPC), which is a family of convolutional codes. RCPC codes generate different code rates from an original rate–1/2 convolutional code. Higher or lower code rates can be obtained with rate-tables by puncturing or repetition [10]. The outer coding scheme in our concatenated FEC is the

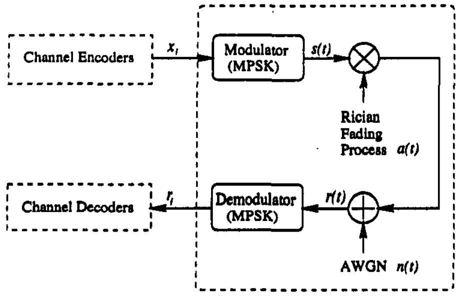

Fig. 5. Rician Channel Model.

Reed-Solomon (RS) code, which is particularly effective at correcting short bursts of errors in a data stream. we also have interleaver/deinterleaver pairs in order to break up burst errors introduced by the channel, and to spread them across several code words. The symbol interleaver disperses bursts errors out of the inner decoder at the symbol level, while the channel interleaver randomizes channel burst errors at the bit level [10].

III. Channel Model

In this paper, the channel model is assumed to be a frequency non-selective slow Rician fading, which is typical in satellite communication channels [9]. Slow fading causes Rician envelope to be constant during one signal interval, T_s. In this model, we assume no shadowing and coherence detection. Hence, the phase changes of the channel are tracked by the receiver. Accordingly, only the amplitude changes are appeared in the channel model.

The sequence from the channel encoder, $\mathbf{x} = (\cdots, x_{i-1}, x_i, x_{i+1}, \cdots)$ is transformed into signal, $s(t) = \mathrm{Re}\{\sqrt{2E_s}\sum_i x_i s_T(t - iT_s)e^{j\omega_0 t}\}$ by M-ary phase shift keying (MPSK) modulator where $s_T(t)$ is the envelope of the transmitted signal with duration T_s and unit energy, ω_0 is the carrier frequency, and E_s is the energy per symbol. Then, the received signal, $r(t)$ can be represented by $r(t) = a(t)s(t) + n(t)$ where $n(t)$ is additive white Gaussian noise (AWGN) process. The probability density function of the envelope, Γ of the Rician fading process, $a(t)$ is given by [8]

$$f_\Gamma(\gamma) = \frac{\gamma}{\sigma^2} e^{-(\gamma^2+s^2)/2\sigma^2} I_0(\frac{\gamma s}{\sigma^2}) \tag{4}$$

where the parameter s denotes the peak amplitude of the dominant signal or non-centrality parameter of the distribution and $I_0(\cdot)$ is zero-order modified Bessel function of the first kind. The Rician distribution is often described in terms of a parameter $\mathcal{K}$ which is defined as the ratio of powers in the direct and diffuse components. It is given by $\mathcal{K}$=$s^2/2\sigma^2$ or in terms of dB, $\mathcal{K}$ (dB) $= 10\log\ (s^2/2\sigma^2)$ dB.

The sequence of $\mathbf{r} = (\cdots, r_{i-1}, r_i, r_{i+1}, \cdots)$ is obtained after demodulation, where r_i can be represented by $r_i = \sqrt{E_s/N_0}a_i x_i + n_i$ where a_i and n_i are discrete sample of $a(t)$ and $n(t)$, respectively.

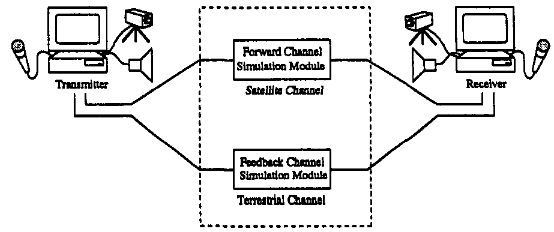

Fig. 6. Simulation Environment.

TABLE I

Code Set in the Transmitter

	Concatenated Code	Code Rate
Code 1	RS(255,68) x CC(Rate=3/4)	0.2
Code 2	RS(255,128) x CC(Rate=4/5)	0.4
Code 3	RS(255,184) x CC(Rate=5/6)	0.6
Code 4	RS(255,239) x CC(Rate=6/7)	0.8

IV. Performance Evaluation

In this section, simulation model and performance evaluation are described. The simulation environment is shown in Fig. 6. In this environment, two hosts are used as transmitter and receiver, whereas two other hosts are used for the forward and feedback channel model. These hosts communicate through UDP sockets. The proposed AEC protocol is implemented in transmitter and receiver hosts, while the two other hosts incorporate time-varying wireless channel model described in section III.

The performance metrics measured in the simulation are the following:

- Throughput: a ratio of the number of bits in successful packet to the total number of bits transmitted
- Packet miss probability: a fraction of the erroneous or tardy packets to the total packets transmitted

For the channel model, $\mathcal{K}$ is assumed to be 10, and signal-to-noise ratio (SNR) is assumed to be 15 dB. Moreover, the mean round trip delay is assumed to be 300 msec. In our simulation, the code set size N is assumed to be 4, and the codes in the code set is given in Table I.

A. Non-Real-Time Application

In this example, file transfer application is used for input traffic. Traffic is generated with mean rate 8 kbps. For performance comparison, we evaluate the following schemes:

- HARQ functionality of the proposed AEC scheme,
- HARQ 1 scheme: RS(255,68) x CC(Rate=3/4),
- HARQ 2 scheme: RS(255,128) x CC(Rate=4/5),
- HARQ 3 scheme: RS(255,184) x CC(Rate=5/6), and

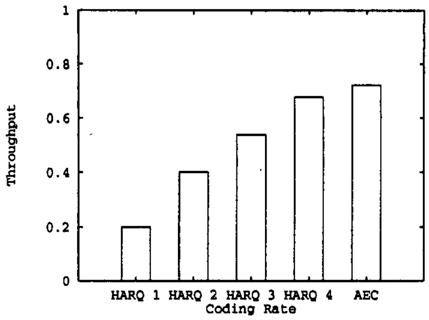

Fig. 7. Throughput Efficiency under Non-real-time Applications.

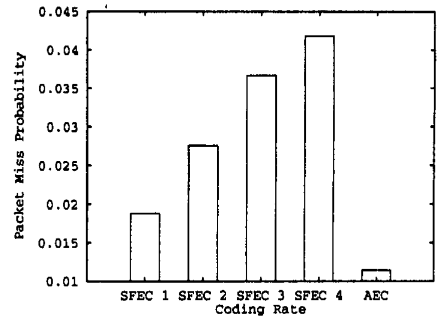

Fig. 8. Packet Miss Probability under Real-time Applications.

- HARQ 4 scheme: RS(255,239) x CC(Rate=6/7)

Note that the each of the static HARQ schemes is a family of type I hybrid ARQ protocols, i.e., if an uncorrectable error pattern is detected, then the receiver asks for a retransmission with the same code rate. Fig. 7 shows the throughput performance of the proposed AEC protocol with HARQ protocols. The throughput performance shows that the proposed AEC protocol outperforms the other HARQ protocols. For HARQ 1 and HARQ 2, the throughput is approximated to their coding rate; for the other HARQ schemes, the throughput decreases due to the packet errors.

B. *Real-Time Application*

In this example, audio application is used for input traffic. The data rate used in the simulation is 64 kbps. This service is considered as constant bit rate (CBR) traffic. For performance comparison, we evaluate the following schemes:

- AFEC functionality of the proposed AEC scheme,
- SFEC 1 scheme: RS(255,68) x CC(Rate=3/4),
- SFEC 2 scheme: RS(255,128) x CC(Rate=4/5),
- SFEC 3 scheme: RS(255,184) x CC(Rate=5/6), and
- SFEC 4 scheme: RS(255,239) x CC(Rate=6/7)

In this example, we show how our scheme behaves under real-time application. We assume the playout delay as 5 sec, which is a time interval between the first packet arrival and starting time of the playout. We set predefined packet miss probability as 0.015. Fig. 8 shows the packet miss probability the proposed AEC protocol with other static FEC (SFEC) schemes. Although the code with higher coding rate achieves the higher throughput efficiency, it deteriorates the packet miss probability. Higher packet miss probability in the higher rate code is due to longer packetization and de-packetization delay. However, our AEC scheme which has a technique to cope with real-time application, provides the lower packet miss probability in the application.

V. Conclusions

In the AEC protocol, we propose a new adaptive FEC (AFEC) scheme for real-time traffic and a new hybrid ARQ (HARQ) for non-real-time traffic. Throughput performance for non-real-time application shows that the proposed AEC protocol outperforms other HARQ schemes. Under real-time application, the AEC protocol outperforms the static FEC protocols with respect to packet miss probability.

Acknowledgments

I would like to give my particular thanks to my advisor Dr. Ian F. Akyildiz for his insightful comments and constructive suggestions.

References

[1] I. F. Akyildiz, and S. Jeong, "Satellite ATM Networks: A Survey," *IEEE Comm. Mag.*, vol. 35, no. 7, pp. 30–44, July 1997.

[2] D. Bertsekas and R. Gallager, *Data Networks*, Prenctice-Hall, Englewood Cliffs, NJ, 1987.

[3] J. B. Cain and D. N. McGregor, "A Recommended Error Control Architecture for ATM Networks with Wireless Links," *IEEE J. on Selec. Areas in Comm.*, vol. 15, no. 1, pp. 16–28, January 1997.

[4] S. Cho, "Adaptive Error Control for Hybrid (Satellite-Terrestrial) Networks," *in Proc. of IEEE WCNC'99*, New Orleans, USA, pp. 1013–1017, September 1999.

[5] S. Cho, "Rate-Adaptive Error Control for Multimedia Multicast Services in Satellite-Terrestrial Hybrid Networks," *in Proc. of IEEE ICC 2000,* New Orleans, USA, pp. 446–450, June, 2000.

[6] D. J. Costello Jr., J. Hagenauer, H. Imai, and S. B. Wicker, "Applications of Error-Control Coding," *IEEE Trans. on Inf. Theory,* vol. 44, no. 6, pp. 2531–2560, October 1998.

[7] D. Friedman and A. Ephremides, "A Scheme to Improve Throughput for ARQ-Protected Satellite Communication," *in Proc. of Int. Mobile Satellite Conf.,* Pasadena, USA, June 1997.

[8] J. G. Proakis, *Digital Communications,* McGraw-Hill Inc., 1995.

[9] B. Vucetic, "An Adaptive Coding Scheme for Time-Varying Channels," *IEEE Trans. on Comm.*, vol. 39, no. 5, pp. 653–663, May 1991.

[10] S. B. Wicker, *Error Control Systems for Digital Communication and Storage,* Prentice-Hall Inc., New Jersey, 1995.