Adaptive Rate Control and Quality of Service Provisioning in Direct Broadcast Satellite Networks

Fatih Alagöz, David H. Walters, Amina AlRustamani, Branimir R. Vojcic and Raymond L. Pickholtz

Department of Electrical and Computer Engineering
The George Washington University
Washington, DC, 20052
{fatih1,vojcic,amina,pickholt}@seas.gwu.edu

Orbital Sciences Corp.
20301 Century Boulevard
Germantown, MD, 20874
walters.david@oscsystems.com

Abstract

Adaptive rate control, if properly employed, is an effective mechanism to sustain acceptable levels of Quality of Service (QoS) in wireless networks where channel and traffic conditions vary over time. In this paper we present an adaptive rate (source and channel) control mechanism, developed as part of an Adaptive Resource Allocation and Management (ARAM) algorithm, for use in Direct Broadcast Satellite (DBS) networks. The algorithm performs admission control and dynamically adjusts traffic source rate and Forward Error Correction (FEC) rate in a co-ordinated fashion to satisfy QoS requirements. To analyze its performance, we have simulated the adaptive algorithm with varying traffic flows and channel conditions. The traffic flow is based on a variable bit rate (VBR) source model that represents Motion Picture Expert Group (MPEG) traffic fluctuations while the DBS channel model is based on a two-state Additive White Gaussian Noise (AWGN) channel. For measures of performance, the simulator quantifies throughput, frame loss due to congestion during transmission as well as QoS variations due to channel (FEC) and source (MPEG compression and data transmission) rate changes. To show the advantage of the adaptive FEC mechanism, we also present the performance results when fixed FEC rates are employed. The results indicate significant throughput and/or quality gains are possible when the FEC/source pairs are adjusted properly in co-ordination with source rate changes.

1. Introduction

In this paper, we consider a Direct Broadcast Satellite (DBS) system supporting multimedia traffic (video, voice and data). In a DBS system the channel quality is continuously changing over time due to fading, propagation anomalies, intentional jamming, or other user interference. Moreover, a satellite beam coverage may include several regions, which may be subject to different channel conditions. Thus in contrast to wireline systems, the channel bit error
rate may range from having no impact on quality of service (QoS) to dramatically degrading QoS diversely for every end-receiver in every region. In the wireless environment, if the propagation delay is small and latency requirements are not stringent, automatic repeat request (ARQ) techniques may become a feasible solution for the error recovery as compared to forward error correction (FEC) mechanisms [Pej96, Zor98]. However, for multimedia satellite applications, the high propagation delay and stringent latency suggest that FEC mechanism should be employed. Since the channel quality is time-variant, the system should be designed either to combat error based on the worst channel conditions by employing the powerful FEC coding rates\(^1\) or to adaptively adjust the required FEC coding rate accordingly. The latter is adapted in our system since it provides another degree of freedom to maximize the utilization of system resources.

In addition, offered traffic rates in multimedia applications fluctuate significantly over time based on user needs and application characteristics. For example, video coding techniques are employed to reduce traffic rates while maintaining constant distortion level; but the bit rates generated by such encoders are highly variable. In our work, we consider the International Standard Organization's (ISO's) Motion Picture Expert Group (MPEG) coded video traffic which is targeted for video storage and transmission applications [Leg91, Iso92, Iso94, Has97]. This results in a high peak-to-mean rate ratio (burstiness) for these variable-bit-rate (VBR) sources. If capacity allocations were based on the peak-source-rates, these networks would have low utilization. Alternatively, if these networks rely on statistical multiplexing, which is employed in our work, then congestion will occur during peak periods.

Within the context of traffic characterization, admission control, resource allocation and management, the QoS provisioning of multimedia traffic has been extensively studied for wireline networks [Kur93, Bie93, Mar94, Gib95, Kan95, Wan96, Yea96, Gop96, Cho98, Kru99, Kni99, Mcd00]. The control schemes addressed for such networks may be useful in wireless networks; however, it is insufficient to guarantee the QoS due to both user mobility and channel dynamics [Nag99]. Recent studies for next generation wireless/mobile networks supporting multimedia traffic focus on mobility issues in a cellular environment [Aca94, Nag96, Per96, Atm97, Nag97, Lag97, Oli98, Nag99, Ren99, Ier99]. Specifically, these studies focus on the variability in the network traffic conditions due to different loads in every micro-cell, in which,

\(^1\) Throughout the paper, channel rate and FEC (coding) rate are used interchangeably.
depending on the loss, delay and jitter requirements, the ARQ or FEC or their appropriate combination may be selected by the error control mechanisms at the base and mobile stations [Yun96, Aya96]. However, in our work we consider a centralized system that not only faces variable load but variable channel conditions with stringent propagation delay and latency as well.

This variability in channel conditions and traffic rates suggests the need for adaptive network management algorithms to maintain QoS. Traditionally, the channel variability was primarily accounted for, and combated within the context of the physical layer and separately from the data link layer. However, it has been recognized recently that, because of the channel variability, there is a potential in coordinated design and operation of multiple layers, such as physical, data link, transport, network and even application layer depending on specific network architectures. In our work, we coordinate the design and operation of the physical (FEC Rate) and application (Source Rate, Admission Control) layers. Specifically, we have developed the Adaptive Resource Allocation and Management (ARAM) system that attempts to maximize system throughput and utilization under varying channel and traffic conditions by adjusting the source (MPEG compression and data transmission) and channel (FEC) rates during periods of congestion as well as severe channel conditions [Ala99a]. One major contribution of this research is the integration of admission control and adaptive source/FEC rate control in the design of wireless network system. In order to utilize the adaptive control of channel and source rates, we use transmit queue monitoring and end-receiver's QoS reports as a closed loop feedback mechanism. Since measures of congestion, QoS and channel variations involve observations over different time periods, the adaptive control mechanisms are activated on different time scales as a response to these variations. In particular, short-term time scales relate to transmit queue management issues, moderate-term time scales relate to statistical traffic fluctuations and QoS variations due to channel problems, and long-term time scales relate to the maximization of throughput and persistent QoS violations.

This paper presents the adaptive FEC feature, in conjunction with adaptive source rate control, of the ARAM system for use in DBS networks supporting real-time multimedia services. It consists of a heuristic assignment algorithm to determine a suitable source/FEC rate pair that satisfies QoS constraints with minimal degradation in system utilization. Other aspects of the

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2 MPEG compression may indicate all the aspects of video stream scaling, i.e. quantization, frame size and rate.
ARAM system where presented and analyzed in [Ala99b, Ala00]. The rest of the paper is organized as follows: The general concept of the target system for ARAM deployment and the ARAM algorithm concepts are described in Section 2. Then, Section 3 presents DBS channel model and the source/FEC rate allocation mechanism both for fixed FEC and adaptive FEC concepts. Section 4 presents the simulation results while Section 5 summarizes the conclusions of this work. Glossary includes the acronyms that we could not avoid using.

2. ARAM Overview

This section provides an overview of the target system architecture for ARAM deployment, the ARAM algorithm concept, and the individual algorithm components.

2.1 System Architecture

The target system for deployment of the ARAM system, depicted in Figure 1, has an asymmetric communications architecture. In the forward direction, Source Host systems transmit large volumes of information over the DBS to DBS Field Terminals (DFTs) located near the intended information receivers. In the return direction DFTs forward control messages or service request messages over a low bandwidth satellite or terrestrial channel. At the source site, the ARAM Host performs the scheduling and control of the forward link transmission; the ARAM Host may also be a Source Host. With the DBS satellite complemented with a return link, the primary focus is on Internet Protocol (IP) multicast applications. Both MPEG coded VBR and available-bit-rate (ABR), lower priority data, traffic types are supported.
A DFT may serve several end-receivers either connected locally via local area network or connected by wide area terrestrial wireless networks. As shown in Figure 1, a DBS satellite beam coverage may include several regions, which may be subject to different atmospheric conditions. A multicast group will typically correspond to one or more VBR or ABR applications. Users in the multicast group may have different local connectivity to the DFT and be operating under different atmospheric channel conditions.

2.2 ARAM Concept

Figure 2 depicts the time phased control strategies utilized by ARAM. These control strategies are intended to respond to events in the network such as increases in the offered traffic, statistical fluctuations in individual services, or changes in the DBS channel conditions. The symptoms used to identify problems in network performance include Requests Waiting for Service, Queue Sizes, Error Rates, and Jitter Rates.
In response to problems identified by analyzing these symptoms, ARAM uses four time phased control strategies with different time scales: short (STC), short-to-medium (SMTC), medium (MTC) and long (LTC) term controls. As shown in Figure 2, each control strategy may utilize one or more of the following adjustments: video compression rates, data transmission rates, FEC rates and Resource Allocation.

After resources are assigned to services, short-term control actions are limited to queue management that implements rules for prioritizing packets. For example, VBR packets may be given priority over ABR packets due to the delay sensitivity of real-time VBR traffic, or MPEG I frame may be given priority over MPEG P or B frames due to the MPEG coding scheme [Yea96].

For intermediate time periods, ARAM adapts source rate adjustments for both VBR and ABR services and adjusts the FEC coding rate to accommodate small changes in conditions. Long term control is intended to accommodate bigger changes in network conditions. In this case, ARAM may have to re-allocate resources if network performance degrades below the desired QoS, i.e., drop (terminate) services.

2.3 Algorithm Architecture

Figure 3 graphically illustrates the ARAM algorithm architecture in terms of its major processing components. As shown in the figure, ARAM is driven by:

- Service Requests that originate from data sources (data push) or from user requests (data pull),
- Quality of Service Reports periodically transmitted by the DFT and end-receivers.
The majority of the ARAM algorithm software components (Adaptive Resource Allocation, QoS Analysis, and Control Mechanism) are resident in the ARAM host depicted in Figure 1, while the Transmit Queue Manager is resident in the Uplink Router. The ARAM algorithm components are described below.

1. The **Adaptive Resource Allocation** component admits/rejects services based on available resources and estimated statistical multiplexing gain. It is performed when a User Service Request arrives or a service in progress completes or as part of Medium and Long Term Controls when the Control Mechanism determines that resources must be re-allocated due to degradation in the QoS. ARA is based on the greedy knapsack algorithm that utilizes a cost function which represents the ratio of profit (priority and flexibility of the service request) to weight (measure of resources consumed) [Ala99b]. The admission control algorithm in ARAM is given in the next section.

2. The **Transmit Queue Manager (TQM)** component, resident in the Uplink Router, controls the queue management for Short Term Control. In ARAM, priority queue management is employed with the following levels of priority (lowest to highest):
   - ABR packets,
   - Packets containing MPEG B frame data,
   - Packets containing MPEG P frame data,
   - Packets containing MPEG I frame data.
As a practical matter, only ABR and MPEG B frame data gets dropped in a properly managed system. Also, the TQM continuously monitors the frame dropping rate (FDR) and delay variations of ongoing services and periodically forwards this information to the ARAM host for processing by the QoS Report Analysis (QRA) and Control Mechanism (CM) components.

3. The **QoS Report Analysis (QRA)** component examines the QoS of ongoing services to detect problems and determine their causes as part of Medium and Long Term Control. As shown in Figure 3, QRA component gathers information both from the TQM and end-receivers. The TQM component continuously sends the frame dropping rate (FDR) which is measured at the uplink while DFT and end-receiver transmit periodically frame error rates (FER) and optionally Average Frame Delay (AFD) and Frame Delay Jitter (FDJ) in return link messages.

The ARAM system utilizes these reports to assess the current QoS and to take corrective actions such that the QoS is maintained within QoS boundaries. Upon completion of analysis, it informs the CM of necessary changes to maintain the end-receiver’s QoS. In the following paragraphs, we briefly present the QoS assessment strategy that is implemented in the QRA component.

For a given multicast group depicted in Figure 1, let the indices i, j and k correspond to DFT, region and multicast group, respectively. With this notation, \( \text{FER}_{ijk} \) is the frame error rate for DFT i, in region j, with multicast group k and \( \text{FDR}_k \) is the fraction of dropped frames for the multicast group k at the uplink router. Then FER due to only the channel can be estimated as:

\[
\text{FER}_{ij} = \text{FER}_{ijk} - \text{FDR}_k
\]  

The frame failure rate, \( \text{FFR}_{jk} \), is the ratio of the number of DFTs in k-th multicast group having unacceptable FER to total number of DFTs in k-th multicast group. Also, let \( \text{FER}_k^{\max} \) denote the maximum allowable FER of the k-th multicast group, then \( \text{ARFER}_k \) can be defined as average ratio of \( \text{FER}_{ij} \) for DFTs experiencing \( \text{FER}_{ij} > \text{FER}_k^{\max} \):

\[
\text{ARFER}_k = \frac{1}{N_k \text{FER}_k^{\max}} \sum_{i,j} \text{FER}_{ijk} \quad \text{if } \text{FER}_{ijk} > \text{FER}_k^{\max}
\]

where \( N_k \) is the total number of DFTs experiencing \( \text{FER}_{ij} > \text{FER}_k^{\max} \). This metric is used to
quantify the frame failure rate-error level for the k-th multicast group in service.

Part of the report analysis is to determine whether there is a problem common to several members of the multicast group or if it is a user specific problem. For example, if all reporting DFTs in a multicast group are experiencing performance problems, then the problem is most likely a congestion problem. However, if only one DFT is experiencing a problem, it is most likely a local problem, such as antenna out of alignment. Similarly, if only DFTs from a given region report performance problems, a channel problem is the most likely cause.

Problems in the wide area terrestrial network must also be identified. Delays introduced in the terrestrial network can only be identified at the receiver. Therefore, the receiver should report AFD and FDJ. By comparing the locally measured delay and delay jitter to those reported by the receiver, ARAM can detect if network nodes beyond the DFT are degrading performance.

4. The Control Mechanism (CM) component selects the actual source/FEC rate pair to be used as a response to change in system conditions informed by QRA and TQM components. The CM component also coordinates control and adaptation actions, handles adjustment of QoS parameters and service attributes in the system, and supports necessary command transfers within the ARAM system.

CM is designed to operate in a time phased control approach that can respond to changes in the system in a timely manner yet make these changes more targeted as more information (status reports) become available. That is, in SMTC, CM makes small rate adjustments relatively frequently immediately upon the receipt of indications that something is not going well. As more information becomes available, the CM tries to make a better adjustment of rates, and in larger steps, with MTC. Finally, the LTC makes even larger adjustments based on more detailed statistics of QoS reports and may include even termination or admission of services. The time scale of these control mechanisms depends, in principle, on system dynamics (traffic and channel variations) and return link capacity (limiting frequency of the QoS status reports).
2.4 Admission Control in ARAM

As discussed above, the adaptive resource allocation (ARA) algorithm, a feature of the ARAM system, is responsible for assigning service requests to a DBS transponder, thereby admits, queues, or rejects a service request based on QoS parameters. In ARAM, we employ an admission control algorithm based on the aggregate overall rates (Source rate)/(FEC rate) of the services, and we adapt to channel and/or traffic rate variations. New assignments of FEC rates are based on the best available knowledge of the channel quality for region(s) of interest for pending requests. For services in progress the FEC rates are continually adjusted in Medium and Long Term Control. Changes in rates trigger resource reallocation to admit new services or drop some services in progress. On the other hand, since the traffic variations are, in general, non-stationary, statistical multiplexing gain may vary accordingly. Thus the admission control must account for these variations based on the measured statistics in the system.

We formulate the admission control algorithm on the basis of a bandwidth assignment problem as given in [Blo92]. N sources, each of which generates bit rates with a finite mean and variance, will share a transmission link with a finite transmission capacity of C bps. Let $\alpha_i$ denote the bandwidth expansion factor (BEF) of the i-th source. The $\{\alpha_i\}$ are measure of excess bandwidth (relative to the average) that must be assigned to the i-th incoming traffic so that the probability of the aggregate instantaneous rate exceeding the fraction of the capacity assigned to the admitted VBR services is smaller than some specified value. Specifically, $\{\alpha_i\}$ are determined according to:

$$\Pr\left\{ \sum_{i=1}^{N} R_i \geq C \right\} = \int_{-\infty}^{\infty} f_x(x) dx \leq \gamma$$

(3)

$$C = \sum_{i=1}^{N} \alpha_i R_i^m$$

where, $R_i$ is the instantaneous overall rate of the i-th VBR service, $R_i^m$ is the average overall rate (average source rate divided by FEC rate) of the i-th VBR service, and $f_x(x)$ is the probability density function of the aggregate overall rate.

For simplicity, the aggregate traffic resulting from N statistically multiplexed VBR sources is estimated based on Gaussian approximations relying on the central limit theorem.
The accuracy of this approximation profoundly depends on the value of the QoS parameter $\gamma$ and the number of sources $N$. To examine the accuracy of the Gaussian approximation, Monte Carlo simulation was used to obtain the probability of exceeding the assigned bandwidth curves, Equation (3). We use the VBR source model developed in [Ala99a], which is a mixture of two first-order autoregressive AR(1) processes driven with lognormally distributed noise sequences. One AR(1) process attempts to capture short term and the other one to capture long term dependencies of VBR traffic. The choice of such model arises from our experimental observations that the marginal probability distribution, the autocorrelation and the first-in-first-out queuing statistics of the model closely fit the empirical statistics [Ala99a]. We also assume that the generated VBR sources are independent identically distributed (i.i.d) sources, and thus they all have the same BEF. When the results were plotted against the bandwidth expansion factor and the number of admitted services, the Gaussian approximation was accurate for high probabilities ($\geq 10^{-2}$). For low probabilities ($<< 10^{-2}$), the Gaussian approximation significantly (by several orders of magnitude) mis-estimates statistical multiplexing gain, and thus sub-exponential or hybrid approximations may yield more accurate results at low probabilities [Wal99]. However, for aggressive statistical multiplexing strategy ($\gamma \geq 10^{-2}$) taken in our approach, the Gaussian approximation is acceptable.

Our approach recognizes that the admission control can only approximately estimate the statistical multiplexing gain and attempts to use the characteristics of past traffic streams to better estimate the gain that can be achieved. In addition, we consider the effect of changing the FEC rates due to channel dynamics, since FEC rate assignments affect the number of services that can be admitted, and consequently the statistical multiplexing gain. Therefore, we argue that an optimal bandwidth management scheme should employ an adaptive admission control algorithm that incorporates both traffic and the channel characteristics of the system. Adaptive admission control adjusts the BEF such that the actual value of $\gamma$ is close to the desired value. Nevertheless, when a low utilization of system resources is detected, adaptive admission control adjusts BEF to correct for underestimation of statistical multiplexing gain by admitting pending requests without violating the QoS limits of services in progress.
3. Adaptive FEC Algorithm

This section describes a typical DBS channel model, the adaptive source/FEC assignment algorithm and the fixed FEC assignment approach that is used for comparison purposes.

3.1 Channel Model

In this paper we use a simple two-state model that can describe rain fading over a satellite channel, mispointed antennae or similar slow fading phenomena. To obtain specific results, the system parameters are taken from the Digital Video Broadcasting (DVB) standards [DVB97] because of its widespread use in the DBS industry. It is assumed that a convolutional inner code of rate 1/2 and constraint length 7, punctured to obtain higher code rates (2/3, 3/4, 5/6 and 7/8) when desired, while the outer code is a shortened (204,188) Reed-Solomon code. The inner code and outer codes are decoded by soft and hard decision decoding, respectively. Typical performance curves for Additive White Gaussian Noise (AWGN), in terms of frame error rate, are shown in Figure 4 for frame sizes of 1kbits and 1Mbits, respectively [Ala99b].

It is important to notice from Figure 4 that there are multiple choices of FEC rates based on the available energy per bit to noise power spectral density rate (Eb/No), performance requirements, and available capacity for a specific service. For a given FEC rate the transmission of a service may be completely degraded when Eb/No falls below certain value, but can be restored by employing a lower code rate (down to rate 1/2). The threshold can be adjusted by varying the FEC and/or symbol rate and/or signal power. It should be noticed that, in general, one may choose another set of codes and different channel assumptions to better match the application of interest. This may result in a change of Eb/No levels and may involve a multi-state channel model. However, regardless of the choice of code set and channel scenario, ARAM is general enough to illustrate the potential of adaptive rate control under varying channel conditions.
3.2 Adaptive Source/FEC Rate Assignments

The problem of optimally selecting joint source and FEC rate assignments with lossy encoding is still an open research problem, primarily because of the complexity of computing an appropriate metric to quantify their impact on perceptual QoS. Consequently, in ARAM we utilize a heuristic approach, using a set of qualitative rules, to choose good pairs of rates for services of interest. ARAM operates with a discrete two-dimensional set of rate pairs and attempts to find a rate pair that meets the QoS objectives/constraints without introducing unnecessary FEC overhead. The decision is based on the knowledge about the channel and congestion conditions which, in turn, is based on the QoS reports obtained from DFTs and TQM. Also, when increasing FEC rate, ARAM considers the past QoS reports to determine if the chosen FEC rate provided acceptable FER in the past for all the services belonging to that region. Moreover, if the source rate needs to be significantly reduced to relieve congestion, then it may be necessary to reduce the FEC rate to compensate for the incurred error sensitivity of the compressed video streams [Riley97]. The detailed algorithm to find good source/FEC rate pairs in an adaptive manner under changing channel and traffic conditions is given in the Appendix.

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3 Source (MPEG compression and data transmission) rate and channel (FEC coding) rate
Figure 5 illustrates several potential new pair assignments for an initial pair \((P_{ij})\). Note that the x-axis in the figure represents the FEC coding rate and y-axis represents the overall rate (source rate divided by FEC rate) for a single source. Let \(P_{ij}(SR,FEC)\) denote the current source and FEC rate pair and \(\Delta B\) denote the estimated excessive overall rate, i.e., difference between the overall assigned bandwidth for VBR services and actual traffic rate. Depending on the problem identified by the QoS analysis (indicated in bold letters in the figure), ARAM calculates the best suitable pair (the number given in parenthesis in the figure) using the several discrete thresholds for the FFR and ARFER measures (Section 2.3). For example, if QRA only identifies a poor channel problem given that the estimated excessive overall rate, \(\Delta B=0\), the new pair can be selected as (1) or (2) depending on the ARFER level. Another identified problem may be excessive frame dropping (FDR) at the uplink router, then the new rate pair assignment can be selected as (3) if ARFER level is at its nominal rate or (4) if ARFER level is very small. Pairs (6) or (7) may be selected as new rate pairs depending on the combination of these problems and thresholds.

When the QRA indicates that QoS levels of services in progress are satisfied and \(\Delta B<<0\), implying under utilization of system resources, then two possible actions may be taken. First, admission control is triggered to admit a new pending service request without violating the QoS levels. It is clear from Figure 4 that when the channel gets bad (lower Eb/No) then to maintain the same Frame Error rate, the FEC coding rate must be reduced.
limits of services in progress. Second, if admitting a new request violates the QoS limits, then the source and/or FEC rates of services in progress are improved to provide higher QoS.

The complexity of the above algorithm depends on the number of services in progress, the number of distinct regions, and the sizes of multicast groups (number of DFTs and end-receivers). Because of the low capacity of the return link, the QoS reports may be generated statistically, i.e., in a random manner as in the real-time transport protocol (RTCP) [Tho96]. The system may be designed to have the same statistical frequency of the QoS reports per multicast group, regardless of group sizes, or to have more frequent reports for larger or higher priority multicast groups.

It is worthy to mention that the above algorithm accounts for several basic scenarios and can be extended further to incorporate more specific and complex situations subject to the availability of more detailed QoS reports and traffic negotiations [Ala99b].

3.3 Fixed FEC Rate Assignments

The fixed FEC rate is used as a baseline approach for comparison purposes to demonstrate advantages of adaptive rate control. In this approach only changes in source rate are allowed, i.e., the FEC rate is fixed regardless of channel and congestion conditions. This approach (fixed FEC rate) is used in most existing wireless applications, following the logic of a layered network architecture and design of physical layer for either predominant or worst channel conditions.

The lower FEC rates are essential to combat channel problems, especially in error prone conditions where throughput may go to zero at higher rates. However, the use of the lower rate with its increased overhead may result in frame loss during congestion at the transmit queue and reduced throughput. The qualitative impact of FEC rate on system performance is summarized in Table 1.

<table>
<thead>
<tr>
<th>Employed FEC rate</th>
<th>Low (1/2)</th>
<th>Moderate (3/4)</th>
<th>High (7/8)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Error recovery in bad channel state</td>
<td>High</td>
<td>Low</td>
<td>Very low</td>
</tr>
<tr>
<td>Frame loss in network due to overcoding</td>
<td>High</td>
<td>Moderate</td>
<td>Low</td>
</tr>
<tr>
<td>Multiplexing efficiency</td>
<td>Low</td>
<td>Moderate</td>
<td>High</td>
</tr>
</tbody>
</table>

Table 1. Fixed FEC algorithm
4. ARAM Simulations

This section presents the parameters, metrics, and results of the simulations.

4.1 Parameters

To compare the fixed and adaptive FEC mechanisms, we have conducted a set of simulation experiments, on the frame basis, using a discrete event simulator [Ala99b]. The simulator performs all the control actions discussed above and includes both VBR traffic (Section 2) and channel state (Section 3) generators. In addition, to simplify the analysis and get a better understanding of the performance results, we have chosen the same statistical characteristics for all VBR traffic streams while ABR services are not considered. The VBR sources are generated based on MPEG coded "Starwars" movie statistics [Gar93]. Table 2 and Table 3 present the traffic and channel characteristics used to demonstrate ARAM potential for DBS application, respectively. Also, a DBS link capacity of 22Mbps and maximum allowable Frame Error Rate (FER) of $10^{-4}$ are used as simulation parameters.

<table>
<thead>
<tr>
<th>Traffic</th>
<th>MPEG coded VBR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Generated traffic:</td>
<td>GOP pattern: IBBPBBBPBBPBB</td>
</tr>
<tr>
<td>“Starwars” movie</td>
<td>Frame duration=1/24 sec.</td>
</tr>
<tr>
<td></td>
<td>Mean =187kbits/GOP; Std =72kbits/GOP</td>
</tr>
<tr>
<td></td>
<td>Maximum source reduction rate = 47kbits/GOP</td>
</tr>
<tr>
<td>VBR service time</td>
<td>1800 seconds (43200 frames/service request)</td>
</tr>
<tr>
<td>Request time epoch</td>
<td>1800 seconds</td>
</tr>
</tbody>
</table>

Table 2. Traffic characteristics

| Channel State:                | (Eb/No)$_{GOOD}$ = 5.3 dB  |
| Semi-markov process          | P$_{GOOD/GOOD}$ = 0.90      |
|                               | (Eb/No)$_{BAD}$ = 3.6 dB    |
|                               | P$_{BAD/BAD}$ = 0.85        |
|                               | Total time in good channel states = 1201 seconds |
|                               | Total time in bad channel states = 2399 seconds |

DVB FEC rate: 1/2, 2/3, 3/4, 5/6, 7/8

Table 3. Channel characteristics

Based on the discussions in Section 2, the Gaussian approximation is used to model the aggregate VBR traffic. For admission control, the QoS threshold, $\gamma$ in Equation (3), is set to 0.01. The time scales of STC, SMTC, MTC and LTC are 1, 10, 30, and 90 seconds respectively. The sensitivity analysis of the ARAM system for different values of the parameters and different traffic mixes is presented in [Ala99b, Ala00].
4.2 Metrics

In this section, we provide a description of the performance metrics that are measured during simulation experiments. These metrics, which are summarized in Table 4, include link utilization and throughput measured in terms of completed services.

Quality assessment of digital video sequences is a very important issue and has received considerable attention from the networking community. Several metrics are used to specify QoS at the network layer. However, at the application layer, the choice of parameters has not been adequately addressed. This is mostly due to a wide diversity in the types of applications and their communication requirements [Gop96]. The QoS metric given in [Rei92], is based on the peak signal-to-noise (PSNR) evaluations. A quantitative video quality measure is proposed by the Institute for Telecommunication Science (ITS) that agrees closely with quality judgements made by a large number of viewers. Alternative approaches measure perceptual video quality based on the human visual system [Lam98]. Most common opinion scores (MCOS) can give only discrete outputs such as imperceptible, perceptible but annoying, slightly annoying, annoying and very annoying [Tob96]. However, the quality ratings done this way can not be reliable since they do not rely on an analytical foundation but subjective observations [Lam98].

<table>
<thead>
<tr>
<th>Metric</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Average DBS link utilization</td>
<td>Average transmitted information (bit/sec) divided by the link capacity (bit/sec).</td>
</tr>
<tr>
<td>Average number of active services</td>
<td>Average of number of services in progress per second</td>
</tr>
<tr>
<td>Number of completed services</td>
<td>Total number of completed services at the end of the simulation</td>
</tr>
<tr>
<td>Number of terminated services</td>
<td>Number of services dropped due to unsatisfied QoS limits</td>
</tr>
<tr>
<td>Number of rejected services</td>
<td>Number of services rejected due to time epoch expiration before they are admitted</td>
</tr>
<tr>
<td>Frame dropping rate</td>
<td>Ratio of the number of dropped frames to the total number of frames assigned to be served.</td>
</tr>
</tbody>
</table>

Table 4. Summary of simulation metrics

In ARAM, we introduced the QoS measure presented in Equation (4), which is purely quantitative. Berkeley's MPEG codec [Gon94] is used to examine the perceptual degradation in QoS. An objective evaluation of these cognitive related issues is out of the scope of this paper, however, based on our experimental observations, we believe that the defined measure should

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1 When the simulation is ended, the services in progresses are aggregated based on their completion percentage and integer part of this number is included to the number of completed services.
correlate with subjective QoS. Therefore, QoS is measured as the rate loss seen by the end-
receiver and is defined as:

\[
QoS = 1 - \frac{\Delta R_s + \Delta R_d + \Delta R_c}{R_n}
\]  \quad (4)

where \(\Delta R_s\) accounts for changes in source rate due to congestion and/or FEC rate changes to keep
overall rate constant, \(\Delta R_d\) accounts for rate reduction due to dropped frames, \(\Delta R_c\) accounts for
sum of rates of all frames that are received with error or lost in the channel and \(R_n\) is the nominal
rate. Note that \(\Delta R_s\), \(\Delta R_d\), and \(\Delta R_c\) are parameters dependent on the traffic rate, channel state, and
control actions.

This metric is calculated for each service in every second. If a service is terminated the
QoS is set to zero for the remaining service time since no frames are delivered for that service.
Thus, given that control strategies maintain the delay and frame error rate within acceptable
ranges, this metric captures the overall rate loss. This can be considered as a measure of the
respective subjective video quality with more freedom of scaling. Relative to the original coded
video sequence, QoS below 50% may be considered very annoying, 60% is annoying, 70% is
slightly annoying, 80% is perceptible but annoying, 90% is imperceptible.

### 4.3 Results and Discussion

The simulation was performed six times for the same channel states and VBR service
requests, for total of 3600 seconds. While VBR request arrival times are generated following a
uniform distribution, there were 50 VBR service requests at the beginning of the simulations,
summing to 147 requests. The simulation starts with a “good” channel, but it switches twice to a
“bad” channel such that on average the channel was in the bad state for, one-third of the total
time. While the initial FEC rate in adaptive FEC simulation is set to 3/4, for the five fixed FEC
rate simulations, it is 1/2, 2/3, 3/4, 5/6 and 7/8, respectively. The corresponding performance
results for adaptive and the examined fixed FEC rates are presented in Table 5 and Figures 6 and
7.

We first consider the performance of fixed FEC rate assignments. For the assumed
channel and traffic models, the fixed FEC with rate = 2/3 outperforms other fixed rate options in

---

2 QoS is unsatisfied when the nominal source rate is reduced more than 25%. This measure is chosen somewhat arbitrarily and can be relaxed
further to accommodate more services at the expense of lower quality, or vice versa.
that it achieves the best tradeoff between throughput and link utilization on one hand and QoS\textsubscript{T} on the other hand. This result clearly suggests that available capacity should be carefully apportioned between channel redundancy and information throughput for a given traffic and channel scenario. For example, the performance of rate = 2/3 FEC code is less robust with respect to channel impairments than rate = 1/2 code, but allows admission of 27% more services on the average, as can be observed from Table 5 and Figure 6. Likewise, rates higher than 2/3 provide even more raw information throughput, but the actual throughput is now penalized by excessive frame errors due to bad channel conditions.

These observations indicate that in situations of slow channel variations it should be possible to further optimize the throughput/QoS\textsubscript{T} tradeoff by attempting to match the traffic (source) and channel rate pairs and admission control. Indeed, as can be seen from Table 5, the adaptive strategy further enhances the overall throughput but at the expense of termination of some services during congestion or bad channel conditions. However, the impact of terminations on the actual throughput is minimized by terminating services that had the smallest percentage of completion.

Traces of measured QoS\textsubscript{T} during simulation time of one hour, for the considered FEC schemes, are shown in Figure 7. It can be seen that the time profiles of QoS\textsubscript{T} for adaptive and lower rate (1/2 and 2/3) fixed FEC schemes are comparable, but the adaptive scheme achieves larger throughput. Higher rate fixed schemes (3/4, 5/6 and 7/8) exhibit unacceptable behavior of QoS\textsubscript{T} during bad channel states so they are all inferior to the adaptive approach.

It is possible to further improve the performance of the adaptive strategy by increasing the rate on the return channel to provide more frequent QoS reports. For example, by providing signal-to-noise ratio (SNR) measurements via the return channel, it is possible to further smooth out variations in QoS\textsubscript{T} and to enhance throughput with the adaptive FEC scheme [Ala99a]. Further enhancements are possible by providing additional degrees of freedom to adjust transmission attributes. For example, in slow channel scenarios it would be possible to incorporate adaptive modulation format as well so that by adaptively adjusting source and FEC rates and modulation format jointly, one would be able to achieve better performance tradeoffs by having more degrees of freedom. In addition, adaptive power control can be used to augment the control strategies for further enhancements in throughput and QoS, but it would normally require larger throughput on the return channel.
<table>
<thead>
<tr>
<th></th>
<th>FEC Adaptive</th>
<th>FEC 1/2</th>
<th>FEC 2/3</th>
<th>FEC 3/4</th>
<th>FEC 5/6</th>
<th>FEC 7/8</th>
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<tr>
<td>The number of VBR services in progress</td>
<td></td>
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<tr>
<td>Mean</td>
<td>47.33</td>
<td>32.84</td>
<td>41.57</td>
<td>45.39</td>
<td>49.03</td>
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<td>3.56</td>
<td>3.44</td>
<td>3.19</td>
<td>3.23</td>
<td>2.58</td>
<td>2.88</td>
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<tr>
<td>Total number of completed VBR services</td>
<td>92</td>
<td>65</td>
<td>83</td>
<td>90</td>
<td>98</td>
<td>102</td>
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<tr>
<td>Total # of terminated + rejected VBR requests</td>
<td>15</td>
<td>20</td>
<td>1</td>
<td>0</td>
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<td>0</td>
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<tr>
<td>Fraction of dropped frames</td>
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<td>Mean</td>
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<tr>
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<td>0.096</td>
<td>0.076</td>
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<tr>
<td>Mean</td>
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<td>3.66</td>
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<td>Percentage of time FEC codes used (%)</td>
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<tr>
<td>1/2</td>
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<td>2/3</td>
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<td>3/4</td>
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<tr>
<td>5/6</td>
<td>6.4</td>
<td>0</td>
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<td>0</td>
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<tr>
<td>7/8</td>
<td>45.9</td>
<td>0</td>
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<td>QoS1 Statistics (including terminated svc)</td>
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<tr>
<td>Mean</td>
<td>0.8147</td>
<td>0.7865</td>
<td>0.8653</td>
<td>0.6489</td>
<td>0.6153</td>
<td>0.6170</td>
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<td>0.0906</td>
<td>0.0719</td>
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<td>0.4372</td>
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<td>Mean (during Bad Channel state)</td>
<td>0.6905</td>
<td>0.7903</td>
<td>0.8775</td>
<td>0.1793</td>
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<td>Mean (during Good Channel state)</td>
<td>0.8619</td>
<td>0.7845</td>
<td>0.8593</td>
<td>0.8842</td>
<td>0.9237</td>
<td>0.9263</td>
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<tr>
<td>QoS1 Statistics (only completed services)</td>
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<td></td>
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<td>Mean</td>
<td>0.8844</td>
<td>0.8208</td>
<td>0.8756</td>
<td>0.6489</td>
<td>0.6153</td>
<td>0.6170</td>
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<tr>
<td>Standard Deviation</td>
<td>0.0945</td>
<td>0.0893</td>
<td>0.0721</td>
<td>0.3368</td>
<td>0.4372</td>
<td>0.4390</td>
</tr>
<tr>
<td>Mean (during Bad Channel state)</td>
<td>0.7874</td>
<td>0.8310</td>
<td>0.8891</td>
<td>0.1793</td>
<td>0.0000</td>
<td>0.0000</td>
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<tr>
<td>Mean (during Good Channel state)</td>
<td>0.9329</td>
<td>0.8157</td>
<td>0.8688</td>
<td>0.8842</td>
<td>0.9237</td>
<td>0.9263</td>
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</table>

Table 5. Summary of the simulation statistics

Figure 6. Number of services in progress
5. Conclusions

In this paper we described the rate control subsystem of an Adaptive Resource Allocation and Management (ARAM) system developed to manage a Direct Broadcast Satellite (DBS) system supporting multimedia traffic and operating under dynamic channel conditions. The key features of ARAM are a time phased control strategy and a selective use of coarse and fine adjustments and prioritization, based on the co-ordinated management of the physical and application layers. With these features, ARAM provides efficient and fair operation with graceful degradation under network congestion and/or poor channel conditions. The efficient use is reflected in a very high link utilization and actual throughput. The fair operation and graceful degradation are achieved based on the design premise that as many users as possible should be accepted into the system to maximize statistical multiplexing and rate averaging over channel conditions. However, when congestion occurs, all services are degraded by the same, small amount of rate (in the absence of priorities). That is, all accepted services draw benefits from statistical multiplexing and correspondingly they are all penalized equally during congestion.

Figure 7. Average QoS for completed VBR services
We described the interaction of the rate control subsystem with other elements of ARAM and studied the impact of adaptive source/FEC rate control on the resource allocation, throughput and QoS. In the absence of solid theory that would enable quantification of the impact of source and channel rate changes on QoS, a heuristic algorithm was proposed and investigated. The proposed algorithm attempts to simultaneously minimize frame losses due to congestion and channel impairments yet taking into account the relative impact of source and FEC rates on QoS at the application layer.

Extensive simulation experiments were conducted to quantify the potential of integrated network management and to assess the impact of adaptive rate control. The performance was measured using several metrics such as throughput, link utilization and QoS (measured in relative rate reduction). It was shown that by optimizing the fixed FEC rate, in conjunction with adaptive source rate, it is possible to optimize the tradeoff between throughput and QoS for given average traffic and channel conditions. Likewise, by employing an adaptive FEC strategy it was shown that throughput and/or QoS can further be increased profoundly.

Acknowledgements

This work was performed as part of the DARPA Global Mobile Information Systems Program under contract number DABT-95-C-0103 to the U.S. Army Fort Huachuca, AZ.

Glossary

<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Definition</th>
</tr>
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<tbody>
<tr>
<td>2LAR</td>
<td>Mixture of two first-order autoregressive processes each with lognormally distributed residuals</td>
</tr>
<tr>
<td>ABR</td>
<td>Available Bit Rate</td>
</tr>
<tr>
<td>ACF</td>
<td>Autocorrelation Function</td>
</tr>
<tr>
<td>AFD</td>
<td>Average Frame Delay</td>
</tr>
<tr>
<td>ARA</td>
<td>Adaptive Resource Allocation</td>
</tr>
<tr>
<td>ARAM</td>
<td>Adaptive Resource Allocation and Management</td>
</tr>
<tr>
<td>ARFER</td>
<td>Average Ratio of $FEC_{ijk}/FER_k^{\text{max}}$ for DFTs experiencing $FEC_{ijk}&gt;FER_k^{\text{max}}$</td>
</tr>
<tr>
<td>ARQ</td>
<td>Automatic Repeat reQuest</td>
</tr>
<tr>
<td>ATM</td>
<td>Asynchronous Transfer Mode</td>
</tr>
<tr>
<td>AWGN</td>
<td>Additive White Gaussian Noise</td>
</tr>
<tr>
<td>BER</td>
<td>Bit Error Rate</td>
</tr>
<tr>
<td>BEF</td>
<td>Bandwidth Expansion Factor ($\alpha$)</td>
</tr>
<tr>
<td>$B_p$</td>
<td>Bandwidth required for the pending requests</td>
</tr>
<tr>
<td>C</td>
<td>Forward link Capacity (bps)</td>
</tr>
<tr>
<td>CM</td>
<td>Control Mechanism</td>
</tr>
<tr>
<td>DBS</td>
<td>Direct Broadcast Satellite</td>
</tr>
<tr>
<td>DFT</td>
<td>Direct Broadcast Satellite Field Terminal</td>
</tr>
<tr>
<td>DVB</td>
<td>Digital Video Broadcast</td>
</tr>
<tr>
<td>Eb/No</td>
<td>Energy per bit to Noise power spectral density rate</td>
</tr>
<tr>
<td>FDJ</td>
<td>Frame Delay Jitter</td>
</tr>
<tr>
<td>FDR</td>
<td>Frame Dropping Rate</td>
</tr>
<tr>
<td>FEC</td>
<td>Forward Error Correction</td>
</tr>
</tbody>
</table>
FER Frame Error Rate
FERC Frame Error Rate due to Channel problems
FER\(_\text{max}\) Maximum allowable Frame Error Rate
FFR Frame Failure Rate (the ratio of the number of DFTs in a multicast group having unacceptable FER to total number of DFTs in that multicast group)
GloMo Global Mobile Information Systems Program
GOP Group of Pictures
IP Internet Protocol
LEO Low Earth Orbit
LTC Long Term Control
MCOS Most Common Opinion Score
MPEG Motion Picture Expert Group
MTC Medium Term Control
N Numbers of VBR source
NC No Control
P\(_e\) Probability of Bit Error
P\(_{fer}\) Probability of Frame Error
P\(_i\)\(_{SR,FEC}\) Source Rate and FEC coding rate pair
PDF Probability Density Function
R\(_i\)\(_m\) Average rate of the i-th VBR service
R\(_i\) Instantaneous rate of the i-th VBR service
RTCP Real Time Control Protocol
RTP Real Time Protocol
SCR Source and Channel Rate Decision
SMP Semi-Markov Process
SMTC Short-to-Medium Control
SNR Signal to Noise Ratio
STC Short Term Control
TCP Transmission Control Protocol
TQC Transmit Queue Controller
TQM Transmit Queue Monitoring
Tx Transmitter
UDP User Datagram Protocol
QoS Quality of Service
QoS\(_T\) Quality of Service Metric (for ABR and VBR services)
QoS\(_D\) Quality of Service Metric (for ABR services)
QRA Quality of Service Report Analysis
VBR Variable Bit Rate
WAN Wide Area Network
WLAN Wireless Local Area Network
W\(_i\)\(_{\text{new (old)}}\) New (old) overall rates of the i-th service
\(\alpha_i\) Bandwidth expansion factor for the i-th VBR service
\(\gamma\) QoS threshold for the prob. of the aggregate instantaneous rate exceeding the capacity assigned to the admitted VBR services
\(\mu_{\text{agg}}\) Average rate of the aggregate traffic
\(\sigma_{\text{agg}}\) Standard deviation of the aggregate traffic
\(\Delta R_i\) Sum of the frames that are received with error or lost in the channel
\(\Delta R_d\) Rate reduction due to dropped frames
\(\Delta \hat{w}\) Changes in source rate
\(\hat{w}\) Average excessive rate
Appendix

Start QoS Reports Analysis
for multicast group k

if (FFRk = 0) // FER is acceptable for all DFTs
  if (FDRk > γ0) // excess. frame drop due to queue buildup. e.g. γ0=8/24
    ⇒ Excessive Frame Dropping(FFRk, ARFERk)
  else
    ⇒ Resume Normal Operation(FFRk, ARFERk)
  end
elseif (Fx % or more of FFRjk >=Py %) // e.g. Px , Py >= 50 majority of DFTs are complaining
  if (FDRk > γ0) // excess. frame drop due to queue buildup
    ⇒ Service Specific Error & Excessive Frame Dropping(FFRk, ARFERk)
  else
    ⇒ Service Specific Error(FFRk, ARFERk)
  end
else // examine every region
  if (FDRk > γ0) // excess. frame drop due to queue buildup
    Excessive Frame Dropping
  end
  for each j // check for every region
    if (FFRjk >=Pz % and Nj >1) // more than one DFT in region j, e.g. Pz>=50
      Poor Channel // channel problem in region j
    elseif (Nj=1 & FFRjk=100%) // only one DFT in region j
      Compare with other multicasts and look at previous reports
      if (other services reported poor channel in region j)
        Poor Channel
      else
        Look at previous Nw QoS Reports // e.g. Nw >=2
        if (Previous QRA shows Potential receiver error)
          DFT Error
        else
          Wait for next report (Potential DFT error)
        end
      end
    else
      Look at previous Nw QoS Reports // e.g. Nw >=2
      if (Previous QRA shows receiver error)
        DFT Error
      else
        Wait for next report (Potential DFT error)
      end
    end
  end
else // examine every region
  Look at previous Nw QoS Reports // e.g. Nw >=2
  if (Previous QRA shows Potential receiver error)
    DFT Error
  else
    Wait for next report (Potential DFT error)
  end
end

Start Source/FEC Assignments

\[ F E C = \begin{bmatrix} 1 & 2 & 3 & 4 & 5 & 7 \\ \frac{1}{2} & \frac{3}{4} & \frac{5}{6} & \frac{7}{8} \end{bmatrix} \]

// SRp: Previous Source rate;  FECp: Previous FEC rate;
  // (FECp,i): i=±1, ±2, ±3, ±4 i.e., change previous FEC rate by i steps
  // flag=0; // Indicate if FEC changes failed
for each multicast k
  if (Poor Channel) // Decrease FEC rate
if (X_h < ARFER_k < X_hh) // Decrease FEC rate by one step
    if (FEC^p rate > 1/2)
        FEC^new = FEC^p-1
    else
        FEC^new = FEC^p; flag=1;
    end
elseif (ARFER_k > X_hh) // Decrease FEC rate by two steps
    if (FEC^p rate > 2/3)
        FEC^new = FEC^p-2
    elseif (FEC^p rate = 2/3)
        FEC^new = FEC^p-1; flag=1;
    else
        FEC^new = FEC^p; flag=1;
    end
    end
elseif (Last N reports Poor channel & X_o < ARFER_k < X_1) //Keep FEC rate
    FEC^new = FEC^p
elseif ((ΔB>0 & FEC^p<7/8 & ARFER_k < X_o) //Increase FEC
    FEC^new = FEC^p+1
end
SR^new = FEC^new ((SR^p/FEC^p) - ΔB)
If (flag=0 & SR^new > SR_min)
    ⇒ Change Rate Accepted(SR^new, FEC^new)
else
    SR^new = max( SR_min, SR^new)
    ⇒ Change Rate Failed(SR^new, FEC^new)
end
References


