Dynamic Rate Adaptation (DRA) and Adaptive Coding and Modulation (ACM) efficiency comparison for a DVB-RCS system

[1]M.A. Vázquez Castro, [2]L.S. Ronga, [3]M. Werner,

[1]Dept. Telecomunicació i d'Enginyería de Sistemes
Universitat Autònoma de BarcelonaCampus Universitari, s/n, 08193 Bellaterra Barcelona – SPAIN
e-mail:angeles.vazquez@uab.es, FAX: (+34) 93 581 4031, PHONE(+34) 93 581 4246
[2]CNIT,University of Florence Unit, Via di Santa Marta 3, 50139, Florence, Italy
E-mail:luca.ronga@cnit.it , FAX: (+39) 0554796485 , PHONE(+39) 0554796485
[3]DLR, Institute of Communications and Navigation, Oberpfaffenhofen, 82234 Weßling, Germany
E-mail: markus.werner@dlr.de, Phone: (+49) 8153 282826

***Keywords*—DVB-RCS, ACM (Adaptive Coding and Modulation), DRA (Dynamic Rate Adaptation), DVB-S2**

I. INTRODUCTION

Broadband communications are deemed increasingly more central in the telecommunications realm as it will ease service convergence merging broadcasting services with provision of IP-based services. The DVB-S standard is widely used today by satellite systems to broadcast TV and radio programs, but is gradually more used for unicast communication and Internet transmissions.

The new generation DVB-S2 will allow boosting the use of interactive services by efficiently increasing the capacity through the use of Adaptive Coding and Modulation (ACM) for unicast services. ACM allows different modulation formats and error protection levels (i.e. coding rates) to be used upon a constant symbol rate carrier. It is able to cope with a large variety of spectrum efficiencies (from 0.5 to 4.5 bit/s per unit bandwidth) and associated CNIR requirements (from –2 dB to 16 dB). Moreover DVB-S2 allows a large variety of input data formats, from MPE/MPEG to IP datagrams [1].

The interactive DVB-RCS system was developed by adding a return channel to the DVB-S system. Standardization is right now under way to adopt the return channel (RCS) to the new DVB-S2 standard providing higher throughput and hence reduced cost even for high bands like Ka. These new bands show severe atmospheric attenuation, which call for transmission technology and framing design enhancements, the main one being adaptation.

Current DVB-RCS systems already implement slow adaptation. Several carriers of different bandwidth are possible and depending on traffic and channel needs users are moved from one carrier to another. This technique is called Dynamic Rate Adaptation (DRA) and may not be optimal if the bandwidth segmentation is fixed and different carriers are not efficiently used. In order to provide sufficient flexibility to the uplink of DVB-S2 DRA must be optimized and compared with other solutions.

In this paper we present an analysis of DRA performance implementing two improvements as regards current solutions. The first one is that it is assumed that bandwidth can be optimally segmented on a superframe-by-superframe basis according to channel conditions and traffic needs. The second one is that coding rate can be adapted to the channel conditions also on a superframe-by-superframe basis. These solutions are allowed by the standard. We compare the performance against a different solution based on ACM using one single high rate carrier, equal to the highest possible DRA carrier. This latter solution is not DVB-RCS compliant. Performance is analyzed primarily at physical layer but considerations regarding bandwidth efficiency are also discussed. It is shown the trade-offs arising from the different solutions and the good performance of the solution based on DRA with AC, which is compliant with the DVB-RCS standard.

This paper is organized as follows. In Section II we describe the possible physical layer configurations defined by DRA, DRA and AC and ACM. In Section III, performances are calculated analytically by identifying measures of power efficiency for a maximum available bandwidth. Section IV presents additional performance analysis in terms of both framing and bandwidth efficiency. Section V gives numerical results for realistic system parameters. Finally, conclusions are drawn by discussing the trade-offs arising from the different solutions.

II. DESCRIPTION OF THE ADAPTIVE PHYSICAL LAYER CONFIGURATIONS

We assume a DVB-RCS multibeam transparent scenario using dynamic framing, i.e. frames are of different sizes and terminals can hop from one carrier to another. At system level, we focus on the uplink of a single beam or group of beams controlled by the same Gateway.

The first solution we want to analyze is DRA using fixed modulation format and fixed coding rate, i.e. the spectral efficiency is η^{DRA}. We assume that N^W types of carriers are possible and we also assume the following commonly used relationship between carriers

$$R_s^{DRA}(i) = R_1 \cdot 2^i \qquad i = 0,1,\ldots,N^W - 1 \tag{1}$$

R_1 is the slowest carrier and $R_s^{DRA}(i)$ is the symbol rate that will be used by a user u at time t as long as the signal-to-noise plus interference ratio, $\gamma(t,u)$, lies between the thresholds

$$\rho_i^{DRA} < \gamma(t,u) < \rho_{i+1}^{DRA} \qquad i = 0,1,\ldots,N^W - 1 \tag{2}$$

where ρ_i^{DRA} stands for the symbol energy-to-noise plus interference spectral density ratio threshold that must be reached by user u at time t to hop to $R_s^{DRA}(i)$. Possible bit rates are

$$R_b^{DRA}(i) = \eta^{DRA} \cdot R_s^{DRA}(i) \qquad i = 0,1,\ldots,N^W - 1 \tag{3}$$

The second solution we want to analyze is DRA used in conjunction with AC, which we call DAC. DVB-RCS standard [3,5], includes a convolutional and an optional TURBO encoding process. The convolutional option allows for a range of punctured convolutional codes, based on a rate 1/2 mother convolutional code with constraint length K = 7 corresponding to 64 trellis states.

This choice will allow selection of the most appropriate level of error correction for a given service or data rate. Code rates of 1/2, 2/3, 3/4, 5/6 and 7/8 shall be supported. The inner code can be bypassed. In that case, the MSB bit is affected to the I channel, the next bit to the Q channel and so on. The convolutional inner code is always by-passed when using Turbo codes. In the encoding process, a block of "0 bits is added to the transmitted message in order to reset to the initial state the decoder. This block is called the "Postamble". The output shall be continued until the encoder is in its all-zero state. If the inner code is bypassed, then the postamble is also omitted. Due to puncturing, faster rates present a degradation of performances with respect to un-puncturted equal rate convolutional codes.

Let us assume that N^{AC} coding rates are possible and therefore (1) still holds but now more bit rates are achievable as follows

$$R_b^{DAC}(i,j) = \eta^{DAC}(j) \cdot R_s^{DAC}(i) \qquad \begin{matrix} i = 0,1,\ldots,N^W - 1 \\ j = 1,2,\ldots,N^{AC} \end{matrix} \tag{4}$$

$\eta^{DAC}(j)$ is the j-th spectral efficiency defined by the fixed modulation used (QPSK in DVB-RCS) and the j-th coding rate. $R_b^{DAC}(i,j)$ is the bit rate that can be used as long as the signal-to-noise plus interference ratio, $\gamma(t,x)$, lies between the thresholds

$$\rho_{i,j}^{DAC} < \gamma(t,u) < \rho_{i,j+1}^{DAC} \qquad \begin{matrix} i = 0,1,\ldots,N^W - 1 \\ j = 1,\ldots,N^{AC} \end{matrix} \tag{5}$$

Note that the following condition must hold

$$\rho_{i,j+1}^{DAC} < \rho_{i+1,1}^{DAC} \tag{6}$$

The previous two solutions are allowed by the DVB-RCS standard. The third solution we want to analyze is ACM, which is not. We assume that N^{ACM} possible combinations of modulation formats and coding rates are possible. It is assumed that the first modulation format is the one used by DAC and that the coding rates are the same as used by DAC. We assume that the symbol rate is the highest one allowed by DRA

$$R_s^{ACM} = R_1 \cdot 2^{N^W - 1} \tag{7}$$

in this case the possible bit rates are

$$R_b^{ACM}(k) = \eta(k) R_s^{ACM} \qquad k = 1,2,\ldots,N^{ACM} \tag{8}$$

where N^{ACM} is the total number of possible spectral efficiencies. $R_b^{DRA}(i)$ is the symbol rate that will be used as long as the signal-to-noise plus interference ratio, $\gamma(t,u)$, lies between the thresholds

$$\rho_k^{ACM} < \gamma(t,u) < \rho_{k+1}^{ACM} \qquad k = 1,\ldots,N^{ACM} - 1 \tag{9}$$

Fig. 1 represents one superframe showing the differences among the different solutions.

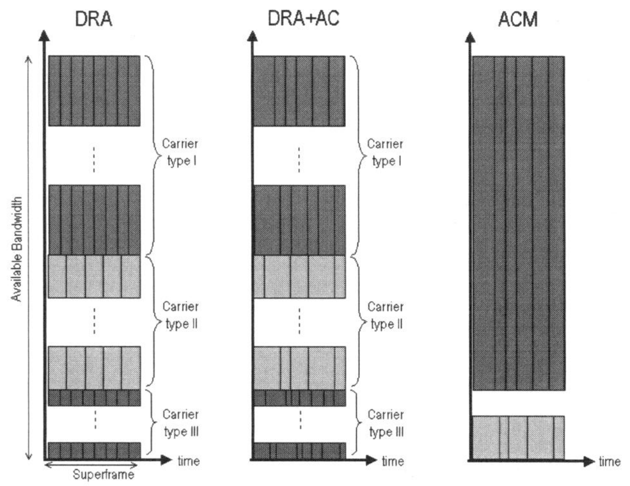

Fig. 1. Superframe structure with multirate frames and different carriers

III. POWER EFFICIENCY ASSUMING BANDWIDTH LIMITATION

It is not evident how the different solutions defined in the previous section can be compared one to each other.

Figure 1 shows a general graphical representation of (3). It can be observed that the losses due to the chosen granularity are apparent from the staircase shape due to the discrete nature of the physical layer adaptation.

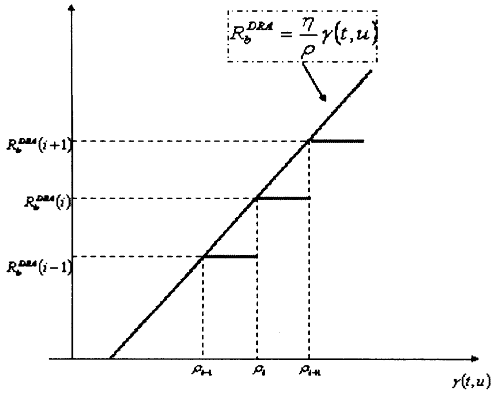

Figure 1. Bit rate quantization error.

These losses can be seen as quantization losses and to some extent they give a measure of power waste. As a matter of fact, when $\gamma(t,u)$ is above the threshold of one physical layer but below the following threshold, power is effectively wasted since it could be used to achieve a higher bit rate. Consequently, in this paper we measure the power efficiency of each solution under analysis precisely by computing the corresponding quantization error.

Following, we derive the analytical expressions of the quantization errors for every considered solution.

From Fig. 1 the bit rate quantization error for the DRA can be expressed as

$$Q^{DRA} \cong \sum_{i=1}^{N^W-1}\left[R_b^{DRA}(i)-R_b^{DRA}(i-1)\right]\left[\rho^{DRA}(i)-\rho^{DRA}(i-1)\right] \quad (10)$$

according to (3) and Fig. 1 (10) can be expressed as

$$Q^{DRA} \cong \eta^{DRA}\sum_{i=1}^{N^W-1}2^{i-1}\Delta\rho^{DRA}(i) \quad (11)$$

where Q^{DRA} has been normalized to R_1 and

$$\Delta\rho^{DRA}(i)=\rho^{DRA}(i)-\rho^{DRA}(i-1) \quad (12)$$

note that bandwidth limitation is implicitly considered by N^W. Note also that it obviously depends on which spectral efficiency is chosen. It also depends on the granularity chosen in the design, i.e, N^W.

The bit rate quantization error for the DAC case can be expressed as

$$Q^{DAC} \cong \sum_{i=1}^{N^W-1}\sum_{j=1}^{N^{AC}-1}\left[R_b^{DAC}(i,j)-R_b^{DAC}(i,j-1)\right]\left[\rho^{DAC}(i,j)-\rho^{DAC}(i,j-1)\right]$$

(13)

introducing (4) into (13) we get

$$Q^{DAC} \cong \sum_{i=1}^{N^W-1}\sum_{j=1}^{N^{AC}-1}R_s^{DAC}(i)\Delta\eta^{DAC}(i,j)\Delta\rho^{DAC}(i,j)= \quad (14)$$

and normalizing by R_1 we get

$$Q^{DAC} \cong \sum_{i=1}^{N^W-1}\sum_{j=1}^{N^{AC}-1}2^{i-1}\Delta\eta^{DAC}(i,\Delta j)\Delta\rho^{DAC}(i,\Delta j) \quad (15)$$

where

$$\begin{aligned}\Delta\rho^{DAC}(i,\Delta j)&=\rho^{DAC}(i,j)-\rho^{DAC}(i,j-1)\\ \Delta\eta^{DAC}(i,\Delta j)&=\eta^{DAC}(i,j)-\eta^{DAC}(i,j-1)\end{aligned} \quad (16)$$

Finally, the bit rate quantization error for the ACM case is given by

$$Q^{ACM}=\sum_{i=1}^{N^{ACM}-1}\left[R_b^{ACM}(i,j)-R_b^{ACM}(i,j-1)\right]\left[\rho^{DAC}(i,j)-\rho^{DAC}(i,j-1)\right] (17)$$

introducing (8) into (16) and normalizing by R_1 we get

$$Q^{ACM} \cong \sum_{k=1}^{N^{ACM}-1}2^{N^W}\Delta\eta^{ACM}(k)\Delta\rho^{ACM}(k) \quad (18)$$

where

$$\begin{aligned}\Delta\eta^{ACM}(j)&=\eta^{ACM}(i,j-1)-\eta^{ACM}(i,j-1)\\ \Delta\rho^{ACM}(i,j)&=\rho^{ACM}(i,j)-\rho^{ACM}(i,j-1)\end{aligned} \quad (19)$$

From the above expressions it can be noticed that power losses depend on the following values:

- Increments of spectral efficiencies (except DRA, which depends directly of spectral efficiencies)
- Increments of symbol energy to noise plus interference spectral density ratio threshold
- Step chosen to increase the bandwidth of sequenctial carriers (chosen to be 2 above)

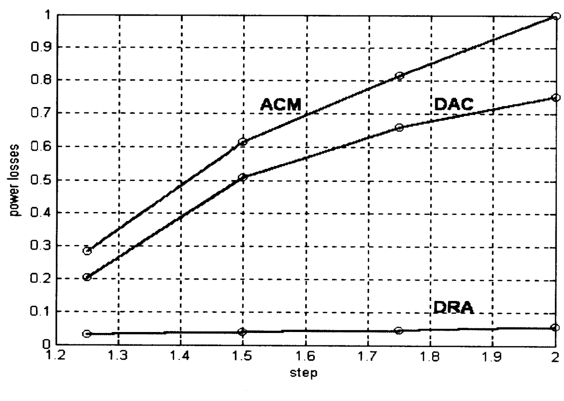

Fig. 2. Power losses normalized to the maximum losses, as a function of the step between bandwidths

Fig. 2 shows some numerical values. Power losses are computed assuming realistic values of all parameters involved: spectral efficiency between 1/3 and 9/10 and realistic thresholds. Values are normalized to the maximum value to ease comparison. It can be observed that we obtain that ACM presents the highest error while DRA the lowest. However, it should be clarified that DRA achieves such an efficient power use at expenses of offering a lower range of bit rates. The increasing trend with the step shows that the more different bandwidths the carriers have the more losses are experienced. ACM presents more losses than DAC but it offers a higher use of the bandwidth by offering both lower and higher bit rates for the same maximum available bandwidth:

$$\Delta R_b^{DRA} < \Delta R_b^{DAC} < \Delta R_b^{ACM} \tag{20}$$

The estimation of power efficiency based only on the quantization error implicitly assumes that all physical layers are evenly used. A more realistic measure might take into account the probability of occurrence of every physical layer. However, given that expressions (15) and (18) depend on the increments of spectral efficiencies and thresholds, such average does not really affects the trend shown in Fig. 2.

IV. BANDWIDTH EFFICIENCY

Bandwidth efficiency is more difficult to quantify since the final usage depends on the MAC layer and the packet scheduling. In the system we are considering, scheduling is not not only at packet level but two scheduling can be distinguished: bandwidth segmentation into carriers and packet scheduling. Efficiency due to packet scheduling is out of the scope of this paper since it would require investigating the efficiency of different scheduling policies, which moreover may be optimized in a number of ways. However, the choice on how to optimize the bandwidth segmentation determines the efficiency of the packet scheduling.

In [2] an algorithm of bandwidth segmentation is presented for DRA only, based on the calculation of traffic demands during one superframe. The optimal segmentation is obtained by first selecting those terminals with more demands and then a fairness ratio is applied. Differently to [2], we look for an algorithm not only simpler that but valid also for DRA+AC. Bandwidth segmentation for ACM is not considered since since we consider that most of the ACM traffic will be absorbed by only one high rate bandwidth, although a small portion of the overall bandwidth might be allocated to users with very low SNIR. Thus bandwidth efficiency of ACM will be primarily caused by packet scheduling.

The difficulty when adaptive coding and or adaptation is considered is that slots are of different size in times since the number of symbols is constant, either ATM or MPEG. Our algorithm circumvents this issue.

Let R_i, $i=1,2,\ \ ,N^W$ be the bandwidth of the i-th type of carriers out of the N^w that are possible. We formulate the following Knapsack (KNP) problem, to be solved on a superframe-to-superframe (scheduling cycle) basis:

$$\begin{aligned} &\max \textstyle\sum_{i=1}^{N^W} c_i x_i \\ &\textstyle\sum_{i=1}^{N^W} x_i R_i \le B_T \\ &x_i \ge 0, \text{integer}, \mathrm{i} = 1,2,\ldots,\mathrm{N}^{\mathrm{W}} \end{aligned} \tag{21}$$

where c_i is the weight given to the i-th type of carrier, B_T is the maximum available bandwidth and x_i is the integer number of carriers of type i-th that are needed. The KNP problem can be solved by any general Interger Linear Programming (ILP) algorithm, but, because it has only one constraint, more efficient algorithms are possible. With this algorithm, the N^W-dimensional hyper surface defined by the number of carriers characterizing the segmentation is optimized according to the weights.

The efficiency will depend on the weights we choose. We propose that the highest priority demands drive the segmentation. This means that remaining capacity will be allocated to the successively lower priorities until filling up the entire available bandwidth. It should be realized that any type of service will be allocated a minimum of traffic. Such a minimum allocation can be actually regarded as traffic with the highest priority. Therefore our algorithm inherently satisfies the minimum requirements for all classes of service. It would be part of the dimensioning of the system to decide which amount of each type of traffic will be regarded as of highest priority. Once this is decided, a-priori estimation of the demand traffic over the multibeam coverage will yield the correct system dimensioning, including maximum required bandwidth.

Let us assume N^u active users, N^{QoS} classes of Service and N^w types of carrier. The demands (in number of slots) for the highest priority traffic class are given by:

$$D_i^{DRA} = \sum_{u=1}^{N^u} \sum_{q=1}^{N^{QoS}} d_u^{i,q}, i = 1,2,\ldots,N^W \tag{21}$$

$$D_i^{DAC} = \sum_{u=1}^{N^u} \sum_{j=1}^{N^{AC}} \sum_{q=1}^{N^{QoS}} d_u^{i,j,q}, i = 1,2,\ldots,N^W \tag{22}$$

The computation of the highest priority demands is possible since user terminals send different types of requests [3] depending on the traffic they send, which can be appropriately mapped to the amount of slots given by (21) and (22). We propose

$$c_i = \frac{D_i}{\sum_{i=1}^{N^W} D_i} \tag{23}$$

For the sake of clarity, Table I shows a simple example of bandwidth segmentation applying (21) when the total bandwidth is 500 MHz and N^w=2, 50 MHz and 100 MHz.

Table 1. Example of bandwidth segmentation, B_T=500MHz.

(c_1,c_2)	Carriers Type 1 (50 MHz)	Carriers Type 2 (100 MHz)
(0.2, 0.8)	2	4
(0,8, 0.2)	8	1

It can be observed that with this algorithm, bandwidth efficiency is guaranteed given an appropriate system dimensioning for any distribution of SNIR throughout the multibeam coverage.

Recall also that traffic priority rather than QoS required by services drive the bandwidth segmentation of our algorithm since different QoS will have a minimum guaranteed allocation via corresponding prioritizing, which can be designed according to different operator policies.

V. FRAMING EFFICIENCY

Another measure is naturally given by the framing efficiency defined as follows

$$E = \frac{S_I}{S_T} \tag{15}$$

where S_I is the number of information symbols and S_T is the total number of symbols sent in a frame of duration T_f msec. In general, it can be assumed that a number of extra symbols, N^E are added to the burst for synchronization issues. Then, the framing efficiency can be computed as follows

$$E(i,j) = \frac{R_s(i)}{T_f}\left(\frac{N^{DVB}}{\eta_j} + N^E\right) \quad \begin{array}{l} i = 0,1,\ldots,N^W - 1 \\ j = 1,\ldots,N^{AC} \end{array} \tag{16}$$

where N^{DVB} is the number of bits of the packet to be sent, which in DVB-RCS can be either MPEG or ATM. In Fig. 3 it is shown the probability of having a certain framing efficiency for DRA+AC assuming uniform distribution of codes for ATM packets (left) and MPEG packets (right). It is apparent that in this case use of ATM packets would render a more efficient framing.

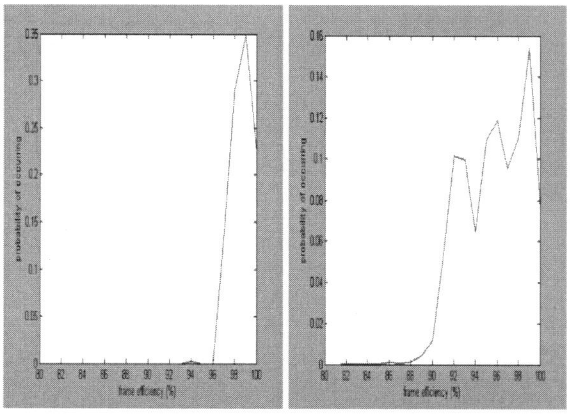

Figure 2. Example of frame efficiency probability for DRA+AC assuming uniform distribution of codes for ATM packets (left) and MPEG packets (right).

In [4] framing efficiency is comprehensively analyzed for a given packet scheduling policy.

VI. CONCLUSION

In this paper we present an analysis of DRA performance implementing two improvements as regards current solutions. The first one is that it is assumed that bandwidth can be optimally segmented on a superframe-by-superframe basis according to channel conditions and traffic needs. The second one is that coding rate can be adapted to the channel conditions also on a superframe-by-superframe basis. Both of them are compared against a different solution based on ACM, which is not DVB-RCS compliant.

A methodology of computing efficiency in terms of power usage by deriving analytical expressions has been presented. The solutions based on DRA+AC and ACM are shown to be close in performance for the given system assumptions and trade-offs have been pointed out. Complexity has not been taken into account.

Bandwidth efficiency has also been analyzed considering bandwidth segmentation only in order to exclude out the effect of packet scheduling policies. We have proposed a simple algorithm for bandwidth segmentation based on a Knapsack (KNP) problem. This algorithm computes the optimal number of carriers of each available type driven by traffic priorities, thus guaranteeing minimum traffic allocation requirements. The remaining capacity will be allocated to the successively lower priorities until filling up the entire available bandwidth. However, the algorithm is flexible enough to allow weights calculation according to different strategies. Finally, framing efficiency has been also presented showing the effect of different packet lengths.

VII. ACKNOWLEDGEMENT

This work was supported in part by the IST-507052 SatNEx Network of Excellence.

REFERENCES

[1] R. Rinaldo, M.A. Vázquez Castro, A. Morello, "DVB-S2 ACM modes or IP and MPEG unicast applications", International Journal of Satellite Communications, Vol 22(3), , May-June 2004, pp. 367-399.

[2] K.D. Lee, K. N. Chang, 'A Real-Time Algorithm for TimeslotAssignment in Multirate Return Channels of Interactive Satellite Multimedia Netwroks', IEEE JSAC, Vol. 22, No 3, April 2004.

[3] Digital Video Broadcasting (DVB); Interaction Channel for Satellite Disttibution Systems, ETSI EN 301 790, 2000

[4] M.A. Vázquez Castro, M. Ruggiano, L. Simone Ronga, M. Werner, 'Uplink Capacity Limits for DVB-RCS Systems with Dynamic Framing and Adaptive Coding', Joint 23rd AIAA and Ka-Band Conference 2005 September 25-28, 2005

[5] Digital Video Broadcasting (DVB); Interaction Channel for Satellite Disttibution Systems ETSI EN 101 790, 2000, Guidelines for the use of EN 301 790.