Performance Evaluation of Adaptive Contention Slot Allocators for CATV Networks based on the European Cable Communications Protocol

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***Abstract*–Community Antenna Television (CATV) networks were originally designed for one-way analogue TV broadcasting (from the headend to residential areas). They are now being upgraded to provide a return path, referred as upstream channel, (from the home to the headend). New challenges arise in using the upstream channel due to the reduced bandwidth and the high levels of noise. The efficiency of the CATV MAC protocol depends highly on the bandwidth assigned to the contention access region. A high number of contention slots assigned to this region, reduces the bandwidth for data transmission. On the other hand, a small number of contention slots, results in an increased number of collisions with high traffic loads and degradation in system performance. In this paper, three adaptive contention slots allocators (CSA) are presented for the European Cable Communications Protocol: Digital Video Broadcasting (DVB)/ Digital Audio-Visual Council (DAVIC). These techniques dynamically adjust the number of contention slots needed to resolve collisions according to the traffic load, considerably improving overall system performance.**

I. INTRODUCTION

Initially, Community Antenna Television (CATV) networks were designed for analogue TV broadcasting (e.g. home entertainment). Evolving CATV networks into bi-directional broadband (digital data service) infrastructures requires the development and standardisation of new protocols. New technologies are now being researched which can be used to support high-speed digital interactive multimedia applications over CATV infrastructures. A mixture of coaxial cable and fibre optic cable - Hybrid Fibre Coax (HFC) is being chosen as the preferred topology.

Currently, there are few international CATV standards for cable communication protocols: Data Over Cable Service Interface Specification (DOCSIS) 1.0 [1], DOCSIS 1.1 [2], DOCSIS 1.2 [3], EuroDOCSIS [4], the Digital Video Broadcasting (DVB)/ Digital Audio-Visual Council (DAVIC) (ETSI ES 200 800) [5] and IEEE 802.14 [6]. In Europe the DVB/DAVIC standard has been adopted whereas within North America it is the DOCSIS, which have been widely adopted. However, the EuroDOCSIS specification is a serious alternative for the European market.

EuroCableLabs (ECL) and the European Cable Communication Associations (ECCA) have produced the EuroModem (European Cable Modem) specification [7], which is the technical description of an external cable modem based on the ETSI ES 200 800 standard [5]. Two different types of EuroModem devices have been defined: 1) The class "**A**" EuroModem which is the basic version and is used mainly for high-speed Internet access. This has functionality similar to that of a DOCSIS 1.0 compliant modem. (Using a class A EuroModem a secure data transmission is possible due to the defined encryption technique), and 2) The class "**B**" EuroModem is the enhanced version supporting some additional features. For instance it is possible to deliver high-quality telephony services or Integrated Services Digital Network (ISDN) connections and also IP telephony (VoIP) with a guaranteed Quality-of-Service (QoS) level. A common telephony interface allows connection of a telephone device directly to the class B EuroModem. This modem has functionality similar to that of the DOCSIS 1.1 compliant cable modem. For a further comparison of these three protocol specifications the readers are referred to [8] and [9].

According to [10], in cable networks the performance of a bi-directional communications system is strongly affected by its Media Access Control (MAC) protocol. Clearly, the more intelligently the control algorithms adapt the network to different load situations the better the overall performance of the communications system will be.

Although the first two versions of the DVB/DAVIC have been finalised (DVB/DAVIC 1.0 [11] and 1.2 [5]), further extensions to the protocol are constantly added. The most major revision came in the form of DVB/DAVIC 1.2, which addressed a *splitting tree algorithm* (used to resolve collisions), minislots (used to send shortened reservation requests) and particularly Quality of Service (QoS) support.

The protocol specification does not define any mechanism for bandwidth allocation and this task has been left open to implementation and vendor differentiation. Therefore, in this paper we study several techniques that will increase the system performance by adjusting the bandwidth to be allocated for contention access in each signalling frame.

The rest of this paper is structured as follows. Section II presents a general overview of the DVB/DAVIC MAC protocol. Then a description of the adaptive CSAs is presented in Section III. Finally, Section IV outlines the results in terms of performance analysis demonstrating the advantages and characteristics of the three mechanisms, followed by the conclusions.

II. DVB/DAVIC MAC PROTOCOL

The upstream channel uses Time Division Multiple Access (TDMA) for the transmission of data. This channel is divided into fixed slots of 64 bytes and its frame structure is based on the Asynchronous Transfer Mode (ATM) protocol. In the downstream two signalling methods are used: in-band and out-of-band. In the in-band signalling the downstream channel is embedded in the broadcast channel and is oriented for the EuroModem solution. The transmission of data packets and MAC messages is based on Motion Pictures Experts Group (MPEG-2) transport stream frames. In the out-of-band method the downstream is separated from the broadcast channel and is mainly oriented for the Set Top Box solution. This method uses a Signalling Link Extended Super frame (SL-ESF) framing structure based on ATM

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cells. Ten ATM cells are mapped into 24 sub-frames with additional signalling and error correction information. For a full description of the upstream and also downstream packet formats refer to [5].

In the MPEG-2 frame or SL-ESF structure (according to the solution adopted), a "signalling field" is used for synchronization of the upstream slots. Its main functionality is to co-ordinate the usage, assign access modes, and indicate if reception of contention-based slots was successful.

Each slot is assigned one of the four following classifications from the headend or Interactive Network Adaptor (INA): ranging (for synchronisation and calibration purposes), contention (for light traffic load and MAC messages transmissions), reservation (for bursty or high traffic loads) or fixed slots (for constant bit rate traffic). These frames are transmitted in the downstream channel (at least) every 3 ms when the upstream data rate is 6.176, 3.088 or 1.544 Mbps, and every 6 ms for 256 Kbps.

As introduced in [5] and [10], the authors have reported that several functions are performed by the MAC protocol for connection control and data transmissions. On power-on or reset the initialisation and provisioning procedure sees that a EuroModem or Network Interface Unit (NIU) is capable of tuning to the correct channel in the upstream and downstream directions and that it can receive the basic network parameters. Then, the sign-on and calibration are performed in order to adjust the internal clock and the transmission power of the NIU according to the specific transmission delay and cable attenuation. The initial connection is also established by default. The MAC protocol carries out the establishment and release of logical connections and allows for readjustment of transmission parameters as well as performing an exchange of encryption keys and the establishment of a secure connection. Here the Diffie-Hellman and Data Encryption Standard (DES) security techniques are used.

The DVB/DAVIC group has adopted two contention resolution algorithms (CRA): the *exponential backoff* and the *splitting tree algorithm*. The *splitting tree algorithm* takes advantage of the *exponential backoff algorithm* in the sense that feedback and allocation information allows a station, (with new incoming arrivals) to compete for contention-based slots without risk of collision with backlogged stations. This algorithm makes use of minislots, which decreases the risk of collisions, since one contention-based slot is divided into three minislots (of 21-bytes long transferring shortened reservation request messages), increasing the probability of successful request transmissions and consequently improving the system performance.

III. Adaptive Contention Slot Allocators

As introduced in [12] and [13], the authors have pointed out that the performance of a multi-access reservation protocol depends more on the overall framing structure and the capacity assigned to the reservation channel than the details of the CRA adopted. In this paper we focus on the performance impact when the reservation-capacity is dynamically adjusted by the use of a slot allocation mechanism. For a comprehensive performance evaluation of CRAs for the DVB/DAVIC protocol, the readers are referred to [14].

How many contention-slots per signalling frame?: After the INA has scheduled a number of reservation slots (*RSs*) to carry data packets, any number of contention slots (*CSs*) may then be allocated. When the load of the networks is low, very few *CSs* are required. On the other hand, since the load is low, there will be unused slots that could be used as *CSs* [12]. As the offered load increases, depending on the length of the packets, more slots will need to be allocated as *CSs*.

In this paper, three adaptive CSA for the DVB/DAVIC protocol are introduced, which adjust dynamically the number of *CSs* per signalling frame according to the current traffic load, mean packet size, mean requested slots and possible collisions. These mechanisms will improve the maximum system performance for the *exponential backoff algorithm* by sending more *CSs* when they are needed (and not when they are available) and by reducing the average number of contention slots needed per request to a value very close to the optimum '***e***(=2.718)' as suggested in [15].

We have called to these mechanisms as '***Simple-CSA'***, '***Forced-CSA***' and '***Variable-CSA***'. For the *splitting tree algorithm* a more efficient CSA is not necessary. We have found that the system performance is maximised by fixing the *Min. No. of CSs per signalling frame* to 1 *CSs*, regardless of the traffic type and when a medium size network has been considered (under 70 stations).

A. Simple-CSA

This mechanism allocates all slots that are not being used for data as *CSs*. At low traffic loads, many more *CSs* will be allocated than are required. The surplus of *CSs* significantly decreases the risk of collision (of reservation request) to a very low level, which in turn reduces the access delay for data packets.

This algorithm is a self-regulating mechanism, since should the number of *CSs* be too low, requests will not reach the INA and as a result more *CSs* will be automatically allocated. Conversely, if the number of *CSs* is too high, more successful requests will reach the INA and the number of empty slots that can be used as *CSs* will decrease to a minimum threshold value, which guarantees that at least few slots should be reserved for contention access. Thus, the performance of the network highly depends on the minimum number of CSs allocated in each signalling frame. In [12] and [13], the authors did not considered the minimum number of contention slots that should be allocated in each signalling frame, which would have led them to a low performance estimation.

B. Forced-CSA

This mechanism is based on the dynamics of the *splitting tree algorithm*. When a collision happens, the *splittting tree algorithm* automatically allocates one *CSs* in the next signalling frame, which is then split into 3MSs and used only between the stations involved in the collision. The *Forced-CSA* allocates a flexible number of *CSs* in the next signalling frame. We refer to these additional slots as Forced contention slots (*FSs*). With this new functionally, stations competing for contention access have more chance to transmit successfully, since more contention slots are allocated when they are needed, which reduces considerably the packet access delay.

The idea of allocating more contention slots, in additional to the unreserved slots that are also allocated to the contention-based access region, was first reported in [12]. Here the authors introduced a new contention slots allocator, referred to as '*Forced Mini-Slots CSA*' for the IEEE 802.14 protocol. The main difference between our *Forced-CSA* and the *Forced Mini-Slots CSA*, is that the latter allocates more *CSs* according to the maximum throughput of the Slotted Aloha System, defined in

$$h_{max} = N_a \cdot p \cdot (1 - p)^{N_a - 1} \quad (1)$$

where p is the retransmission probability and N_a is an estimation of the number of stations competing for a contention slot.

On high traffic loads, the *Forced Mini-Slots CSA* allocates $e(=1/\eta_{max})$ *CSs* for each data message to be transmitted. Conversely on low traffic loads, it allocates less than $e(=1/\eta_{max})$ *CSs*. However, our mechanism (*Forced-CSA*) allocates more *CSs* when a collision occurs and on high traffic loads, the average number of *CSs* required per requested data message approaches very close to $e(=1/\eta_{max})$.

For the *Forced Mini-Slots CSA* the authors found that by setting the number of forced mini-slots to 2 (instead of *e*), good results were yielded. In our analysis, we also found that by allocating 2 *FSs*, after a collision, an improvement in system performance was obtained.

C. Variable-CSA

This mechanism uses a variable slot regime in which the ratio of *CSs* to reservation data slots (*RSs*) is varied from signalling frame to signalling frame, based on the current traffic load, mean packet size and mean requested slots. Variable slot allocators have been used since 1998 in MAC protocols for HFC and wireless access networks. They were first introduced in [16]. Later on, reported in [17] for the IEEE 802.14 protocol. The mechanism presented here is similar to the technique presented in [17] with slight modifications for the DVB/DAVIC protocol. The variable number of *CSs* (*VCSs*) to be added in each signalling frame is dynamically adjusted as the headend converts the number of *RSs* into *CSs* (N_{RSs}), according to

$$N_{RSs} = \left\lfloor \frac{2 \cdot MAX\_DATA}{2 + Req\_Size} \right\rfloor \quad (2)$$

where *MAX_DATA* is the maximum number of data slots that can fit in a signalling frame (= 18 – '*Min. No. of CSs per signalling frame*', for a 3.088 Mbps upstream channel) and *Req_Size* is the average number of *RSs* that can be requested at once. *VCSs* can be determined as follows

$$VCSs = \begin{cases} 0 & \text{if MAX\_REQ} \geq a \cdot (\text{MAX\_DATA} - \text{N}_{\text{RSs}}) \\ \text{N}_{\text{RSs}} & \text{else} \end{cases} \quad (3)$$

where *MAX_REQ* is the total number of data slots requested but not yet allocated by the headend, α is a design parameter set to 2.5 as suggested in [17]. The total number of *CSs* to be included in the next signalling frame is then represented by the '*Min. No. of CSs per signaling frame*' (as proposed in the *Simple-CSA*), the variable number of *CSs* derived in (3) and the unused *RSs* converted to *CSs* as recommended in [12].

TABLE I
SIMULATION PARAMETERS

Parameter	Value
Upstream data rate	3.088 Mbps
Downstream rate (64-QAM, in-band)	42 Mbps
Min. and Max. backoff values	3 and 5
Minimum contention-based slots per signalling frame	3 slots
Transmission time of signalling frame	3 ms
Simulation time for each run	60s
Distance from nearest/farthest NIU to the headend	10-16 Km
Propagation delay (coax and fibre)	5 μs/Km

IV. PERFORMANCE ANALYSIS

In this section we present an overview of the performance analysis between the adaptive contention slot allocators for the *exponential backoff algorithm*. A detailed simulation model was implemented using the OPNET Package v.6.0 for the results. For a description of this model, the readers are referred to [14].

The parameters used for the simulations are given in Table I. In all simulations, one upstream channel with a capacity of 3.088 Mbps and one downstream channel with a capacity of 42 Mbps were used. A mixed traffic scenario was considered for the performace analysis, formed by 9.7 Kbps VoIP streams and 32 Kbps Internet traffic, as described bellow.

1) ***Internet traffic*** (IP): This traffic type emulates Internet traffic. The traffic distribution utilised is the one introduced by the IEEE 802.14 Working Group [18]. The message size distribution is as depicted in Fig 1. The inter-arrival times are set in such a way that the resulting mean offered load per active station is 32 Kbps.

2) ***Voice over IP*** (VoIP): This traffic type emulates a speech codec "G.723.1" [19], which according to the ITU, IETF and the VoIP Forum is the preferred speech codec for Internet telephony applications [21] and [22]. This codec generates a data rate of 5.3 Kbps or 6.3 Kbps depending on the modulation ACELP or MP-MLQ, respectively. In this paper codecs of 5.3 Kbps will be used. This codec generates and encodes a 20-byte every 30 ms. It is planned that in the near future, this codec will collect 4 data packets (every 120 ms) instead. Some H.323 terminals already use a value of 3 or 4 for the number of frames per audio packet in order to reduce protocol overhead latency and increase system throughput [20]. Thus, by adding the protocol headers 8-byte UDP, 12-byte RTP, 20-byte IP, 3-byte LLC, 5-byte SNAP, and 18-byte MAC, one obtains an improved VoIP stream of 9.7 Kbps. Therefore, in our performance analysis, VoIP streams at 9.7 Kbps will be considered. This is a novel and topical traffic type.

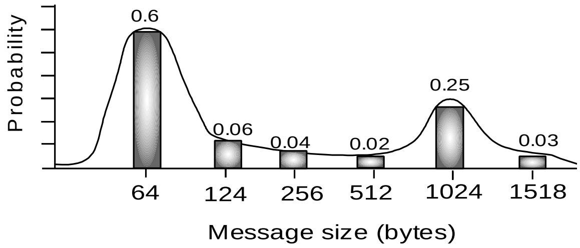

Fig. 1. IEEE 802.14 IP packet size distribution.

Figs. 2 to 5 present a performance comparison between the three adaptive CSA. In order to validate the results obtained by our simulation model, we first present the following formulation to obtain the maximum theoretical bound for the system throughput. By taking the mean packet sizes transmitted, the maximum system throughput can be obtained as

$$S_{\max} = \frac{\overline{Pk}}{\overline{Pk_{slots}} + e \cdot CSs} = \frac{Pb_{IP} \cdot Pk_{IP} + Pb_{VoIP} \cdot Pk_{VoIP}}{Pb_{IP} \cdot Pk_{slots\_IP} + Pb_{VoIP} \cdot Pk_{slots\_VoIP} + e \cdot CSs} \quad (4)$$

where Pk_{IP} *(=368.1)* and Pk_{slots_IP} *(=8.3)* are the average Internet packet sizes measured in bytes and data slots respectively. Pk_{VoIP} (=146) and Pk_{slots_VoIP} *(=4)* are the VoIP packet sizes measured in bytes and data slots respectively. Finally, Pb_{IP} and Pb_{VoIP} are the probability that an IP or VoIP packet will be generated, respectively. Equations (5) and (6) give these probabilities.

$$Pb_{IP} = \frac{Dr_{IP}}{Dr_{IP} + \frac{Pk_{IP}}{Pk_{VoIP}} \cdot Dr_{VoIP}} \quad (5)$$

$$Pb_{VoIP} = \frac{Dr_{VoIP}}{Dr_{VoIP} + \frac{Pk_{VoIP}}{Pk_{IP}} \cdot Dr_{IP}} \quad (6)$$

where Dr_{IP} *(= 32 Kbps)* and Dr_{VoIP} *(= 9.7 Kbps)* are the mean date rates of Internet and VoIP traffic generated per station, respectively. Therefore, the maximum theoretical system throughput that can be achieved for a mixed traffic situation is 46.4% of the channel capacity.

With reference to Fig. 2, we can appreciate that with an offered load over 46% of the channel capacity (produced by 34 stations or more), the highest system throughput was achieved by the *Forced-CSA*. This performance was obtained when 2 *FSs* were added after a collision and the *Min. No. of CSs per signalling frame* was set to 2 or 3 *CSs*, represented by the labels *Forced-CSA(FSs2, CSs2)* and *Forced-CSA(FSs2, CSs3)* in these figures, respectively. The sustainable throughput ranged form 45.5 to 46%, which is very close to the maximum theoretical bound, with a maximum deviation under 1%.

The *Forced-CSA* achieves a higher system performance because on high traffic loads, this mechanism only sends the contention slots needed to resolve the collisions. Fig.3 presents the average number of contention slots used per reservation request. From this figure we can appreciate that the *Forced-CSA* allocated on average from 2.821 to 2.783 per request, which approaches to the optimum value ***e***(≈2.718), as suggested in [14], optimising the bandwidth to be allocated to the reservation and contention-based access regions.

In terms of packet delays and number of streams supported, from Fig. 4 and 5, the lowest access delays were also gained using the *Forced-CSA*. For instance, at 45% of the channel capacity (produced by 33 stations), tolerable mean access delays for VoIP streams (under 50 ms) were seen only with the *Forced-CSA(FSs2, CSs3)*, supporting up to 33 stations. With a slight increase in offered load (e.g. 46% produced by 34 stations), only the *Simple-CSA (CSs5)*

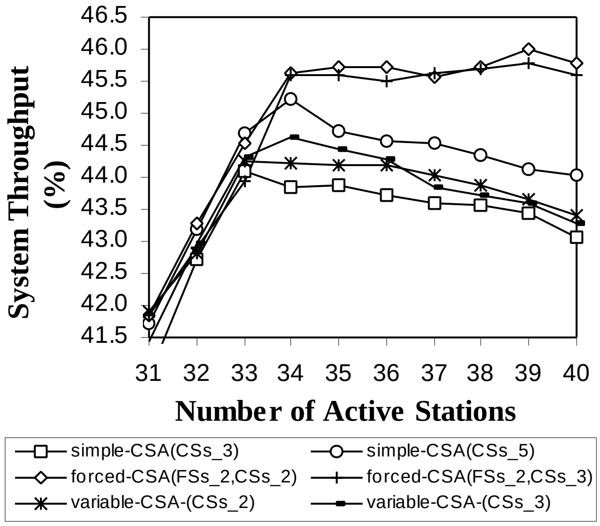

Fig. 2. System throughput vs. No. of active stations.

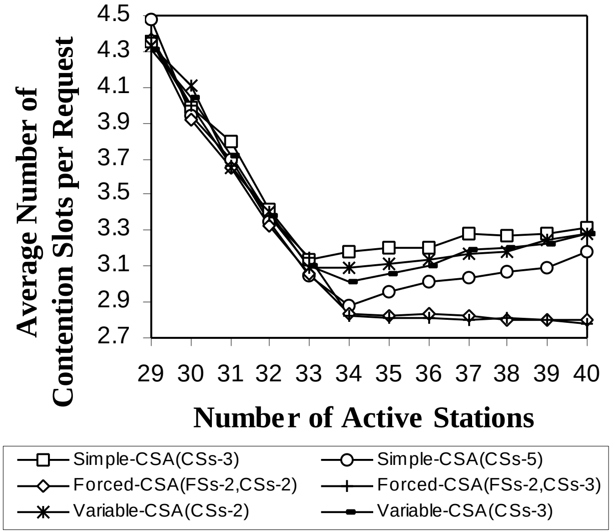

Fig. 3. Average No. of CSs per request vs. No of active stations.

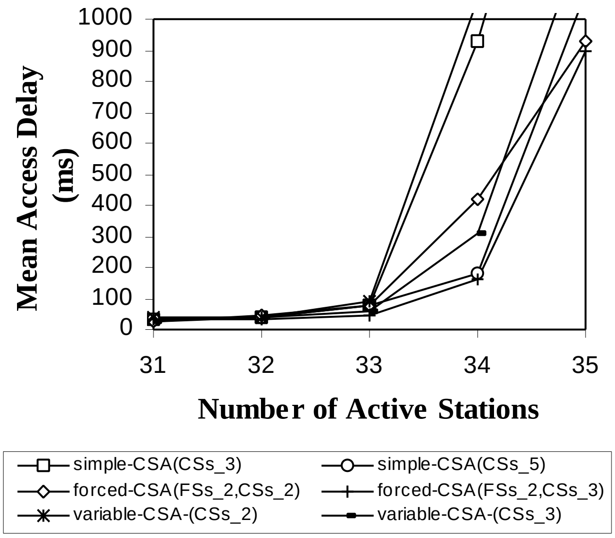

Fig. 4. Mean access delay vs. No. of active stations.

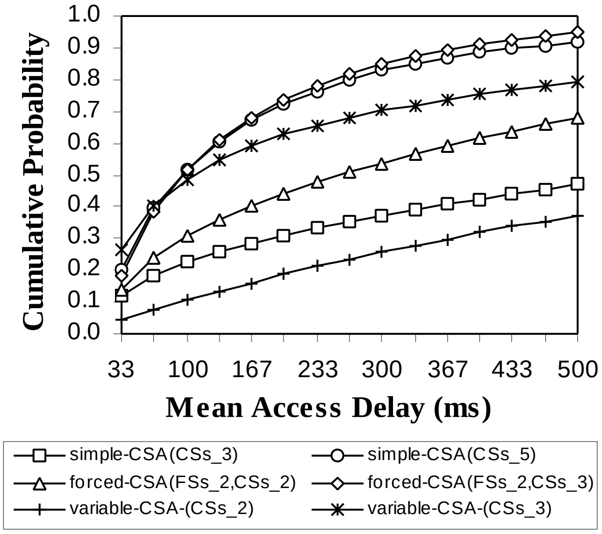

Fig. 5. Cumulative probability vs. mean access delay.
Offered load = 46% produced by 34 stations.

and the *Forced-CSA(FSs2, CSs3)* yielded tolerable low delays for the support of IP traffic. Approximately 73% of all data packets were transmitted in less than 200 ms as illustrated in Fig. 5. The *Simple-CSA (CSs_5)* produced relatively low packet access delays because it sent more *CSs* than currently needed to resolve faster collisions. On the other hand, it wasted many *CSs*, trying to minimise the packet access delays and therefore a slight decrease in system throughput was obtained.

CONCLUSIONS

In this paper a performance evaluation and analysis of three adaptive contention slots allocators has been presented for the European Cable Communication System "DVB/DAVIC". Simulations results revealed that the overall system performance of the DVB/DAVIC protocol could be significantly improved by adopting three novel Contention Slot Allocators (*Simple-CSA*, *Variable-CSA and Forced-CSA*), which distribute dynamically the number of contention slots that should be allocated in the next signalling frame, based on the current traffic load. Results presented in this paper pointed out that the *Forced-SCA* not only provides the highest system throughput, but also offers the lowest packet access delays for the *exponential backoff algorithm*. Simulation results were agreed well with results from theoretical analysis with deviation in the results not exceeding 1%.

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