# CONTENTION RESOLUTION ALGORITHMS FOR CATV NETWORKS BASED ON THE DVB/DAVIC PROTOCOL

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## ABSTRACT

In a bi-directional Community Antenna Television (CATV) Network the use of a Contention Resolutions Algorithm (CRA) is essential in order to allocate the bandwidth of the multiaccess medium in the upstream direction between the active stations. Two of the CRAs known as *exponential backoff algorithm* and s*plitting tree algorithm*, have recently been adopted by the Digital Video Broadcasting (DVB)/Digital Audio-Visual Council (DAVIC) Cable Television Protocol. In this paper we present a performance analysis and a comparison of both algorithms in terms of mean access delay and system throughput for different traffic source scenarios (i.e. Internet traffic, Voice over IP, and mixed traffic). The analysis focuses on changes in performance when selecting different *backoff* bounds for the e*xponential backoff algorithm* and different values for the *Entry-Spreading* factor of the s*plitting tree algorithm*. The results presented here show that an increase over 10% in system performance can be obtained by selecting the *splitting tree algorithm* for different traffic sources.

## INTRODUCTION

Emerging technologies for residential CATV networks with transmission speeds up to 40 Mbps -more than 700 times faster than a typical 56k-modem- will most likely provide the next-generation data communications services, including Internet access and timing critical interactive services (such as VoIP and Videoconferencing, among others). Within Europe and USA, Cable Companies are now exploring new technologies which can be used to support digital interactive applications. Currently, there are few international CATV standards for digital video: DOCSIS 1.0 (1), DOCSIS 1.1 (2), EuroDOCSIS, Barton (3) DVB/DAVIC (ETS EN 200 800) (4) and IEEE 802.14 (5). In Europe the Digital Video Broadcasting (DVB)/ Digital Audio-Visual Council (DAVIC) - cable modem specification has been adopted whereas within North America it is the Data Over Cable Service Interface Specification (DOCSIS 1.0 and 1.1) which has been widely used. However, the EuroDOCSIS specification is a serious alternative for the European market. This research focuses on performance issues of the DVB/DAVIC European Communications Protocol.

In a bandwidth-limited system, the available capacity is divided in two access regions. The first is a contention-based region, (with random access) used to transmit station's request for bandwidth. The second region is a reservation-based region (with reserved access) used to transmit stations' information in a collision-free manner. Since more than one station can transmit a request for bandwidth at the same time, resulting in a collision, a Contention Resolution Algorithm (CRA) must be implemented. Therefore, the performance of the multiaccess system highly depends on the CRA selected.

The theory of CRAs for medium access in computer networks has been studied since the early 1970's. Many algorithms already exist and have been thoroughly examined in Golmie

(6) and Sala (7). There are two major candidates: ALOHA-based algorithms like binary exponential backoff and p-persistence, and splitting tree algorithms. In this paper, we present a performance evaluation and optimisation of the CRAs (*exponential backoff* and *splitting tree algorithm*) adopted by the DVB/DAVIC protocol for different traffic scenarios.

## DVB/DAVIC MAC PROTOCOL AND CONTENTION RESOLUTION ALGORITMS

### MAC issues

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 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 (4).

In the MPEG-2 frame or SL-ESF structure (according to the solution adopted: in-band or out-of-band), a "*signalling information field*" is used for synchronisation of the upstream slots. The main functionality of this field is to co-ordinate the usage, assign access modes, and indicate if reception of contention-based slots has been successful. Each slot is assigned one of the four following classifications from the headend (which is also referenced as Interactive Network Adaptor -INA): ranging (for synchronisation and calibration purposes), contention (for light traffic load and MAC messages transmission), reservation (for bursty or high traffic load) 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 (4) and Schunke (8), the authors have reported that several functions are performed by the MAC protocol for connection control and data transmission. On power-on or reset the initialisation and provisioning procedure sees that a Network Interface Unit (NIU) or EuroModem, 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, *Diffie-Hellman* and *Data Encryption System* security techniques are used.

### Contention Resolution Algorithms

The DVB/DAVIC group has adopted two contention resolution algorithms, which are used to resolve collisions: the exponential backoff and splitting tree algorithms. The splitting tree algorithm takes advantage of the binary exponential backoff 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. In addition, 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. What follows is a description of the operation of each algorithm.

***Exponential Backoff Algorithm***: In case of collision, this algorithm defines how many cells a station (NIU) needs to let pass before it can transmit. This number of cells is computed as a uniform random variable in the range of [0-$2^{backoff}$]. The *backoff* value is first initialised with an *initial backoff* ('Minimum Backoff Exponent') value and is updated according to the reception indicator received from transmission. The *truncated backoff* ('Maximum Backoff Exponent') value is the maximum allowable *backoff*. If a collision occurs the *backoff* value is incremented by one. Once the *backoff* value reaches the maximum number determined by the *truncated backoff*, the *backoff* remains at this value regardless of the number of subsequent collisions. After every successful transmission the *backoff* is reset to the *initial* value. The values of *initial* and *truncated backoff* are sent by the INA to the stations at the beginning of the connection and remain the same until the NIUs stops the connection with the INA. These two values are equal for all the NIUs as opposed to the *backoff* value that is different at each station.

***Splitting Tree Algortihm***: As defined above, minislots are only used to transmit reservation request when using this algorithm. After a NIU sends a reservation request in a contention-based minislot, it waits for feedback information from the INA. In the case of collision, the NIU enters the contention resolution cycle and activates its resolution state machine, according to (4). The resolution is carried out according to an INA controlled ternary splitting algorithm as shown in Figure 1. All necessary information to resolve collisions is provided in the Reservation Grant Message, which contains minislot feedback (such as 'Feedback Offset' and 'Feedback Collision Number' 1, 2 and 3) and minislot allocation information (Such as 'Stack Entry', 'Entry Spreading', 'Number of Allocations', 'Allocation Offset' and 'Allocation Collision Number').

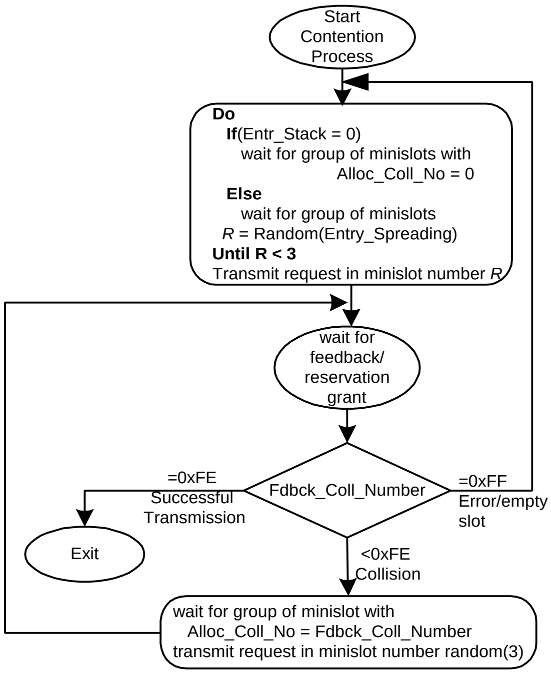


Figure 1 – Ternary Splitting Algorithm.

In the current Reservation Grant Message, If 'Stack Entry' is not set, a NIU may enter the contention process only when the 'Allocation Collision Number' is equal to zero. If 'Stack Entry' is set, the NIU may enter the contention resolution in any of the contention-based minislots, independent of the value of 'Allocation Collision Number'. In both cases the random number for the minislot selection in the range between 0 and 'Entry Spreading' should be in the window from 0 to 2 before sending the request. The 'Feedback Collision Number' equals to 0xFF and 0xFE for idle and successful transmission, respectively. All other values of the 'Collision Number' are numbered as collisions and used to select the retransmission minislots: the NIU should retransmit in a minislot having an 'Allocation Collision Number' equal to 'Collision Number'. The retransmission of the collided request takes place in a minislot that is randomly selected among the group of three minislots with the corresponding 'Allocation Collision Number'.

## SIMULATION MODEL

A simulation model was implemented using the OPNET Package v.6.0. A hierarchical design as shown in Figure 2, was used. At the top level of the network topology, the network components, for example the INA and NIUs, along with their connectivity are shown in Figure 2a. The next level, Figure 2b, defines the functionality of a node in terms of components such as traffic sources, Medium Access Control (MAC), and their interfaces, etc. The operation of each component is defined by a state machine (an example of which is shown in Figure 2c). The actions of a component when it is at a particular state is defined in Proto-C code such as that in Figure 2d. This approach allows modifications to be applied to the operation of the modelled protocol and different optimisations to be tested.

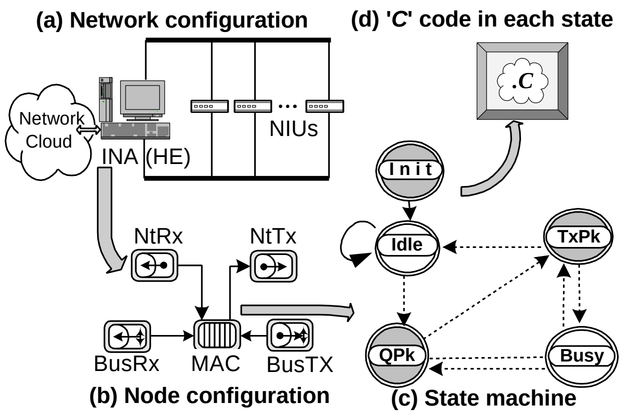


Figure 2 - OPNET simulation model.

## PERFORMANCE ANALYSIS

The parameters that were used for the simulation are given in Table 1. The values chosen for the simulation parameters are typical values and are selected either from those used in actual implementation (manufacturer specific) and/or default values proposed in the specification.

| Parameter | Value |
|---|---|
| Upstream data rate | 6 Mbps |
| Downstream rate (in-band) | 42 Mbps |
| Min. and Max. backoff values | [2–10] |
| Entry spreading values | [5-15] |
| Minimum contention-based slots per signalling frame | 3 slots |
| signalling frame cycle | 3 ms |
| Simulation time for each run | 60 s |
| Number of Stations | 10-220 |

Table 1 - Simulation parameters.

### Traffic Types

Three traffic types will we considered in this paper and their impact on system performance will be analysed: Internet, Voice over IP and Mixed traffic, as defined below.

*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 (9). The message size distribution is as depicted in Figure 3. The inter-arrival times are set in such a way that the resulting mean offered load per active station is 64 kbps.

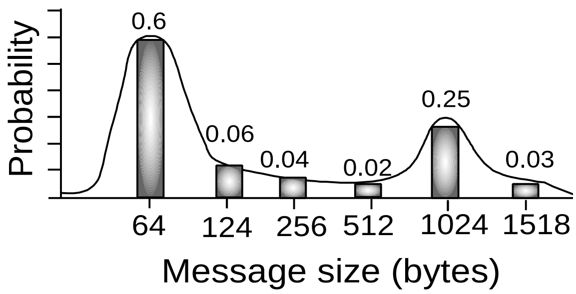


Figure 3 – Packet size distribution.

*2) Voice over IP (**VoIP**)*: This traffic type emulates a speech codec "G.723.1", which according to the ITU, IETF and the VoIP Forum is the preferred codec for Internet telephony applications. This codec generates a data rate of 5.3 kbps or 6.3 kbps depending on the mode, where 20-byte data packets are generated and encoded every 30 ms. In the near future, this codec will collect 4 data packets (every 120 ms) instead. Thus, by adding the complete headers one obtains an improved VoIP stream of 9.6 kbps. Therefore, in our analysis, VoIP streams of 9.6 kbps will be considered.

*3) Mixed traffic*: This traffic type emulates a combined traffic situation, where each station is generating a VoIP data stream as introduced in 2). Additionally, some Internet traffic as presented in 1) is multiplexed into the data stream and is transmitted via the upstream channel. The mean data rate per active station is set to 32 kbps, consisting of 9.6 kbps of VoIP traffic and 22.6 kbps of Internet traffic.

### Exponential Backoff Algorithm Performance

As introduced above, an NIU with a new incoming packet will transmit its request in any of the available contention slots with the current contention based access region, and a NIU already involved in a collision (also termed as *backlogged* NIU), must let pass a number of contention slots before it can transmit. This number is computed according to the *initial* and *truncated* values defined.

In a CATV Network, which makes use of the *exponential backoff algorithm*, the performance of such Network is determined by mainly three factors: 1) the number of active stations; 2) the packet distribution and inter-arrival times; and 3) the *initial* and *truncated* values. Figures from 4 to 7 show the performance impact of these factors, for two different traffic scenarios.

**a) Internet traffic.**

The first scenario deals with Internet traffic only, where a small network size (up to 52 active stations) is considered. Figures 4 and 5 present the mean packet access delay and the normalised system throughput as the number of active stations (offered load) increases for different *initial* and *truncated* backoff values. By defining short values for these parameters (e.g. *initial* = 2, and *truncated* = 4, represented by '**i2-t4**' in the figures) the *backlogged* NIUs, (which have suffered only one collision) are forced to transmit in any of the next 4 contention slots which may result in another collision with NIUs having newly incoming packets, duplicating the number of *backlogged* stations as the network begins to be crowded. Thus, having a large packet delay leads to a drastic reduction in system throughput. This can be appreciated from Figure 4 and Figure 5.

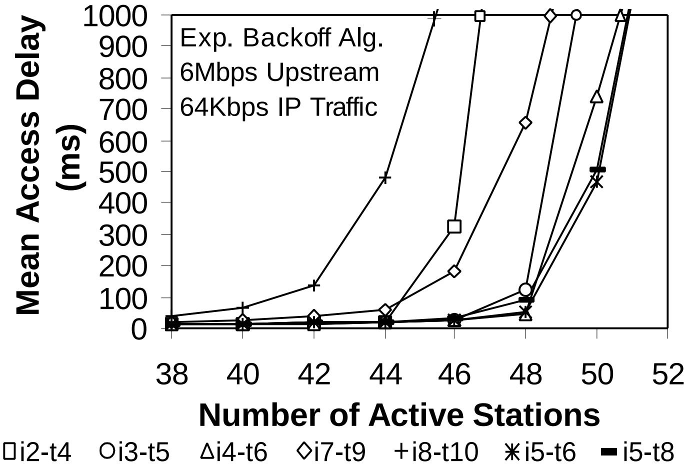

Figure 4 – Mean Delay for IP traffic, Backoff Alg.

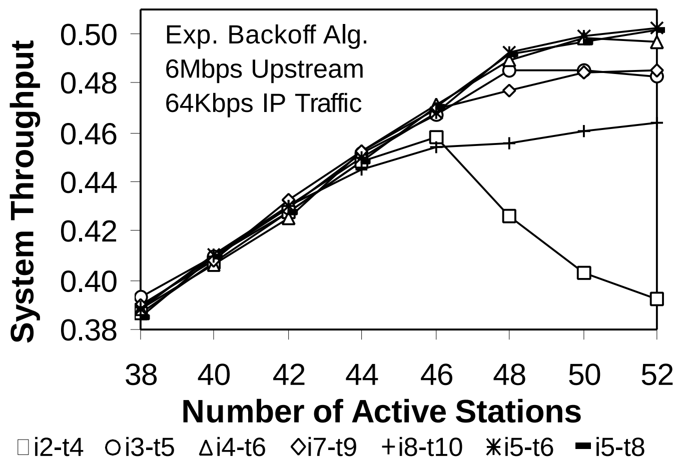

Figure 5 – Throughput for IP traffic, Backoff Alg.

Similar effects (perhaps even worst) are obtained by defining large values for these parameters (e.g. i7-t9 or i8-t10). The consequence of this being to cause *backlogged* NIUs to wait for a relatively long period before they can transmit a request, which results in letting many contention slots pass without use. Therefore, very large mean packet access delays and a decrease in system throughput performance results.

Conversely, by defining intermediate values (e.g.: i3-t5, i4-t6, i5-t6 and i5-t8), almost the same system performance can be obtained when the offered load goes below 47% of **c**hannel **c**apacity (***cc***), which is produced by 46 active stations. Similar results (not showed in the figures) were also obtained with i5-t7 i5-t9, i5-t10 and i6-t8. In such circumstances even a slight offered load increase will result in system instability. The backoff values that offered best system performance were i5-t6, (i5-t7-not showed in figures) and i5-t8. When the offered load reached 51% of the link capacity, tolerable mean access delays of under 500 ms for Internet traffic were seen.

**b) VoIP traffic.**

This second scenario emulated pure VoIP traffic where a large network size (up to 220 active stations) is considered. Figure 6 and Figure 7 show the mean packet access delay and the normalised system throughput as the number of stations increase, respectively. In general, VoIP streams require bounded end-to-end delays under 200 ms, (including delays such as: propagation, compression, decompression, congestion and jitter buffers).

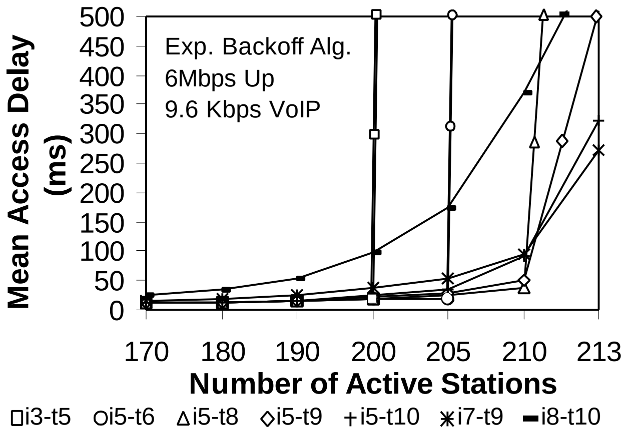


Figure 6 – Mean Delay for VoIP, Backoff Alg.

In these figures, we can see now that when the network load becomes busy at ≈32% of its capacity, the backoff values that offer mean packet access delays under 50 ms are i5-t8, (i5-t7, i6-t8 not showed in figures) and i5-t9, leaving and extra 150 ms for its final destination, (from the headend to the end-station destination). We can also observe from Figures 5 and Figure 7, that the maximum system throughput supported by the network has been reduced from 50% down to 32% of the ***cc***. This is due to the extended number of stations supported which produce a larger number of collisions when competing for contention-based slots, and also because the mean packet size has been decreased from 368 bytes, which is the average packet size used for Internet traffic in accordance with (9), to 144 byes (used for VoIP traffic), leading to a higher protocol overhead and a reduction in system throughput.

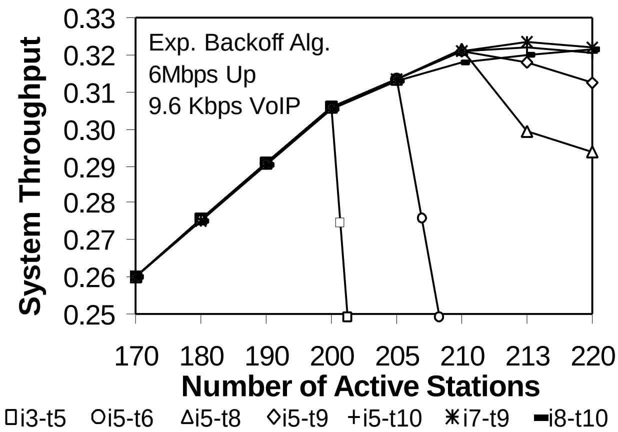


Figure 7 – Throughput for VoIP, Backoff Alg.

**Splitting Tree Algorithm Performance**

In CATV networks, INAs and NIUs that support a splitting tree algorithm, achieve a better system performance by reducing the collision risk when transmitting reservation request messages at the cost of a higher processing times and complexity. This increase in performance is obtained by i) dividing one upstream contention-based slot into three independent minislots that carry shortened request messages and ii) providing two regions of contention-based slots; the first region is normally used to resolve collisions of backlogged stations, where NIUs with new incoming packets are not allowed to transmit reservation requests in these contention-based slots; and the second region is used only to transmit request of new incoming packets.

When selecting the splitting tree algorithm, a factor that determines the performance of the network (apart from the number of active stations, packet distribution and inter-arrival times) is the '*Entry Spreading*' (***Es***) value. This parameter can be either computed or fixed at the INA. For general purposes, the value of the ***Es*** factor will be fixed and ranged from 5 to 15. Figures 8 to 11 inclusive show the performance impact of the ***Es*** factor, using the same traffic scenarios as for the previous algorithm for Internet and VoIP traffic.

**a) Internet traffic.**

Figure 8 and Figure 9 present the mean access delay and system throughput when Internet traffic is being delivered. From Figure 8 we can observe that for all values of the ***Es*** factor, with the exception of ***Es***-5, produce delays under 300 ms, when the offered load reaches 60 % of the ***cc*** (generated by 59 stations). By increasing the offered load by just one station (to 60 NIUs, 61% ***cc***), very large delays are evident. At this point, the ***Es*** values that still offer tolerable delays under 500 ms are ***Es***-14 and ***Es***-8. In terms of system throughput (see Figure 9), the difference of using distinct ***Es*** values is much less significant; with ***Es***-14 and ***Es***-8 giving best system throughput performance.

**b) VoIP traffic.**

In Figure 10, we can see that delays under 25 ms can be obtained for all values of the ***Es*** factor, (except ***Es***-5) given an offered load up to 44% ***cc*** (290 active stations). Defining small values for ***Es*** (e.g.: ***Es***-5), NIUs with new arrival are being forced to transmit in one of the next three available contention-based minislots, reserved for new incoming packets, (with probability of 3/5 = 0.6), providing a higher risk of collision when more than two NIUs are competing for request transmission. However, in Figure 11, the throughput increases linearly as the number of active stations goes up, regardless of the value of the ***Es*** up to 44% of the link capacity (290 stations). After this point, the system can no longer provide tolerably low delay for VoIP streams and the

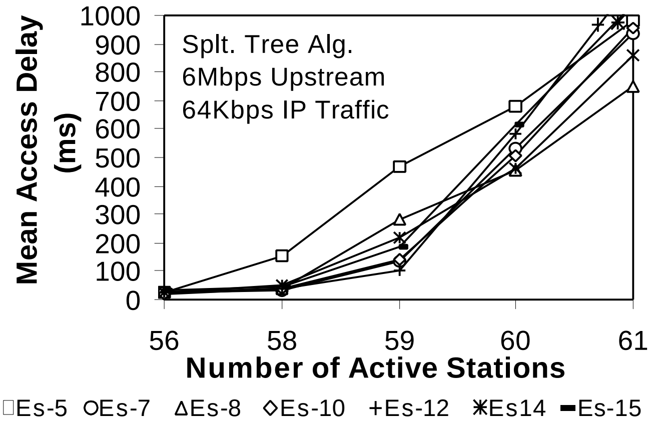

Figure 8 – Mean Delay for IP Traffic, Split. Alg.

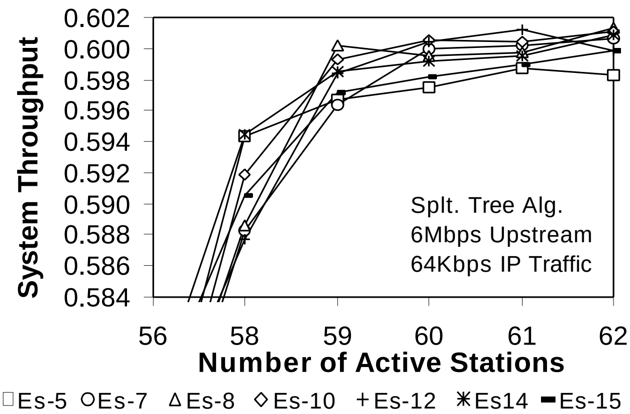

Figure 9 – Throughput for IP, Split. Alg.

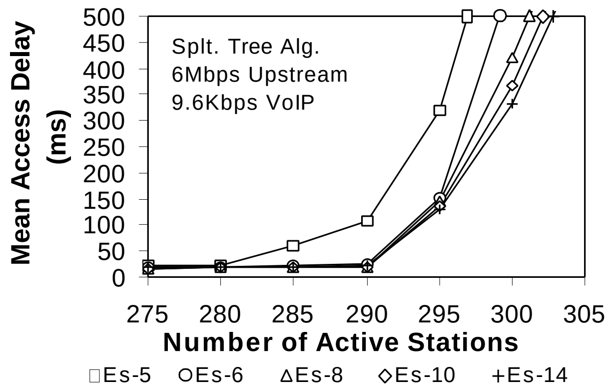

Figure 10 – Mean Delay for VoIP, Split. Alg.

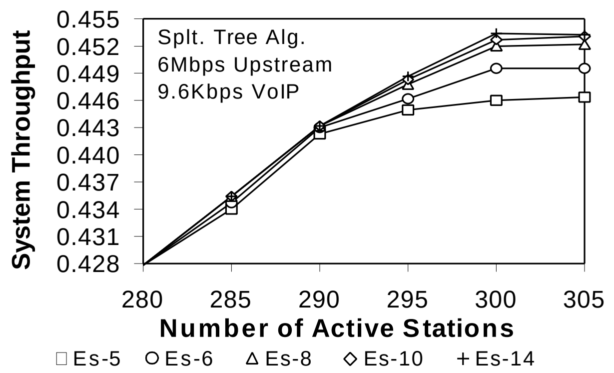

Figure 11 – Throughput for VoIP, Split. Alg.

system throughput starts to stabilise, with ***Es***-8, ***Es***-10 and ***Es***-14 yielding the highest system throughput.

### Comparison

Figure 12 to Figure 14 show a performance comparison of the backoff and splitting three algorithms, for three different traffic situations. Figure 13 indicate that the increase on system throughput performance when using a splitting tree algorithm is of 11%, 13% and 13% of the cannel capacity for Internet, VoIP and mixed traffic, respectively. Figure 13 also shows that the maximum throughput achieved varies from 32% to 60% of the ***cc***. From Figure 12, by taking the limits before the network gets unstable, the increase in terms of supported streams corresponds to 12, 85 and 25 stations for Internet, VoIP and mixed traffic, respectively.

In terms of efficiency, Figure 14 shows that the maximum link utilisation varies from 86 to 89% of the ***cc*** for the exponential backoff algorithm and from 90 to 92% of the ***cc*** for the splitting tree algorithm. The difference between utilisation and system throughput is attributed to protocol overheads (produced by AAL5 encapsulation, ATM header and DVB physical layer overhead) and successful request transmissions, retransmissions and collisions, which varies according to the traffic type, and the contention resolution algorithm utilised.

Finally, with the use of advanced reservation request mechanisms (such as piggyback requests, continuous piggyback requests or reserved requests, which will be also incorporated in the last version of the DVB/DAVIC protocol) will further increase considerably the system performance as introduced in Rangel (10) and (8).

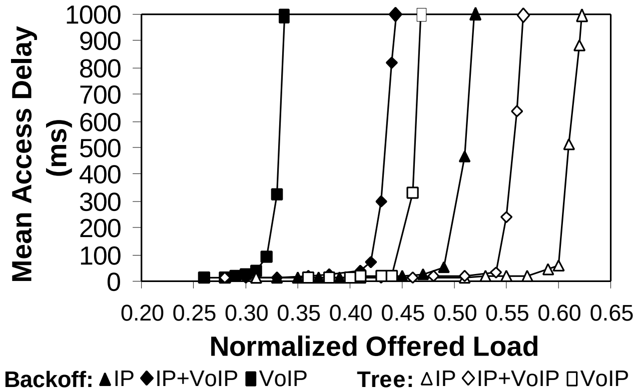


Figure 12 – Mean Delay, Backoff vs. Split. Alg.

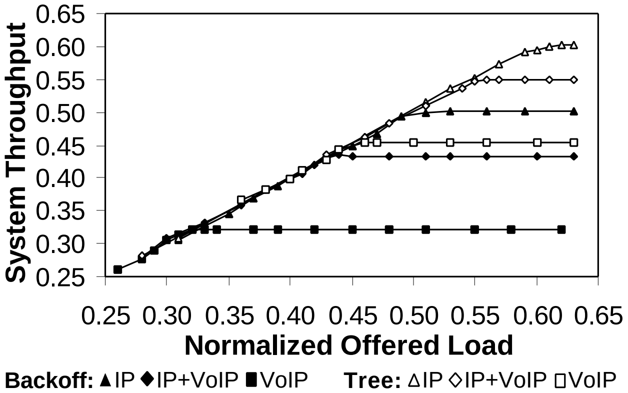


Figure 13 – Throughput, Backoff vs. Split. Alg.

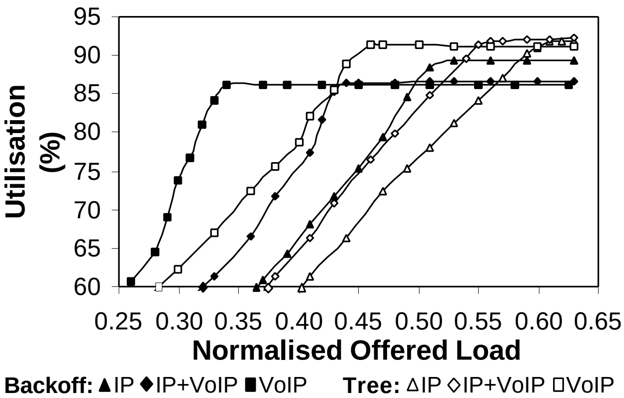


Figure 14 – Utilisation, Backoff vs. Split. Alg.

### CONCLUSIONS

In this paper we have presented a performance analysis of contention resolution algorithms adopted by the DVB/DAVIC protocol. The splitting tree algorithm takes advantage of the binary exponential backoff in the sense that feedback and allocation information allows a station, (with new incoming arrivals) to compete for contention-based slots without the risk of collision with backlogged stations. One more advantage is that the use of minislots for reservation requests further decreases the risk of collisions, since one contention-based slot is divided into three minislots, increasing the probability of successful request transmissions. However, some of the drawbacks of the ternary splitting algorithm are higher complexity at the headend, increased processing times of the feedback and allocation information at the station, and higher bandwidth control information at the downstream channel, since every contention slots should be acknowledged regardless of whether it is used or not. Results presented by both contention resolution algorithms, showed that the system performance is a trade-off between mean access delay/system throughput and the values selected for the *initial/truncated* backoff values and the *Entry_Spreading* factor, respectively.

In general results presented here show that an increase over 10% on system performance, can be obtained by the use of a splitting tree algorithm when backoff values (*initial/truncated*) and the *Entry_Spreading* factor have been optimised, for different traffic types (such as: Internet, VoIP and mixed traffic).

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