Satellite system performance assessment for in-flight entertainment and air traffic control

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Abstract. Concurrent satellite systems have been proposed for IFE (In-Flight Entertainment) communications, thus demonstrating the capability of satellites to provide multimedia access to users in aircraft cabin. At the same time, an increasing interest in the use of satellite communications for ATC (Air Traffic Control) has been motivated by the increasing load of traditional radio links mainly in the VHF band, and uses the extended capacities the satellite may provide. However, the development of a dedicated satellite system for ATS (Air Traffic Services) and AOC (Airline Operational Communications) seems to be a long-term perspective. The objective of the presented system design is to provide both passenger application traffic access (Internet, GSM) and a high-reliability channel for aeronautical applications using the same satellite links. Due to the constraints in capacity and radio bandwidth allocation, very high frequencies (above 20 GHz) are considered here. The corresponding design implications for the air interface are taken into account and access performances are derived using a dedicated simulation model. Some preliminary results are shown in this paper to demonstrate the technical feasibility of such system design with increased capacity. More details and the open issues will be studied in the future of this research work.

Keywords: DVB-RCS, resource management, ATC, IFE, Internet, GSM, FMT, QoS

1. Introduction

During the recent years, IFE (In-Flight Entertainment) has become a hot topic in the communications world and seems to be one of the winning factors for airlines and aircraft industry. Supplied IFE services for aircraft passengers can be VPN access, e-mail and web browsing. The two major aeronautical manufactures have shown a great interest in the IFE systems but the solutions proposed so far were technically limited hence economically non profitable.

At the same time, an increasing interest in the use of satellite systems for aircraft datalink communications, especially for ATS (Air Traffic Services), has been motivated by the increasing load of traditional radio links mainly in the VHF band, even new systems such as VDL (VHF DataLink) mode 2 or VDL mode 3 are currently deployed. However, the development of a dedicated satellite system for ATN (Aeronautical Telecommunication Network) seems to be a long-term perspective. In our design, the satellite link acts as a supplementary access network for aircraft datalink communications including ATC (Air Traffic Control, for the communications between pilots and controllers) but also AOC (Airline Operational Communications, for air/ground communications between airline teams).

The objective of this study focuses on the network system design using a single satellite link to provide Internet access and mobile telephony (GSM and UMTS) for passengers and a high-reliability channel for aeronautical applications such as CPDLC (Controller Pilot Datalink Communication) and ADS-C (Automatic Dependant Surveillance Contract). Section 2 provides an overview of the global system architecture. Section 3 focuses on the study of the resource management. Link radio design and fade mitigation techniques are discussed. Section 4 describes the on-board system de-

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sign. Firstly, access design is presented. Then, hypothesis on traffic characterization and QoS requirements are listed. Section 5 shows some preliminary results which demonstrate the technical feasibility. Finally, conclusion is provided in section 6 with proposes on the future work.

2. System definition

The system presented in this paper relies on the use of a multimedia geostationary satellite access network based on ETSI DVB-S2/DVB-RCS architecture [1,2].

The reference model for our DVB-S2/DVB-RCS network is shown in Fig. 1. Some present hypotheses of such architecture are mentioned as follows:

- Frequency band: the forward link (gateway to terminal) uses Ka (20/30 GHz) or Q/V (40/50 GHz) frequency and the return link (terminal to gateway) uses Ka frequency.
- Satellite broadband services: DVB-S2 standard is applied in the forward link and DVB-RCS standard for the return link.
- Topology: the network topology is a star network with the GW station as a Hub. Obviously, direct communications between aircrafts are not needed for the provided services.
- Satellite: a GEO bent pipe satellite is chosen for our system design. On-board processing and routing is not justified in the case of star networks. However, a regenerative payload could be considered for link budget enhancement and capacity increase.

DVB-S2 (Digital Video Broadcasting – Satellite 2) is the second-generation specification for satellite broadcasting. It offers a large capacity by using very high frequencies and can carry either unicast or broadcast traffic (like TV information programs). DVB-RCS (Digital Video Broadcasting – Return Channel by Satellite) is designed by adding a return channel to the DVB-S2 to provide interactive services via satellite.

3. Resource management in DVB-S2/DVB-RCS network

3.1. Fade mitigation techniques

The proposed system is designed using extremely high frequencies, namely Ka-band (20/30 GHz) between aircrafts and satellite. This choice is driven by the need of large bandwidth that can not be satisfied by L-band ($\sim$1.5 GHz) communications: the traffic from aircraft to hub is comparable to the traffic generated by a LAN (Local Area Network), the traffic from hub to aircraft is expected to be of tens of Mbit/s.

Due to the use of high frequencies, encountered fades should be very high. Figure 2 shows the encountered fades at Ka band. To compensate such deep fades, FMT (Fade Mitigation Techniques) must be considered. Obviously, the system can not be designed with a constant link margin; otherwise the waste of capacity would be excessively high.

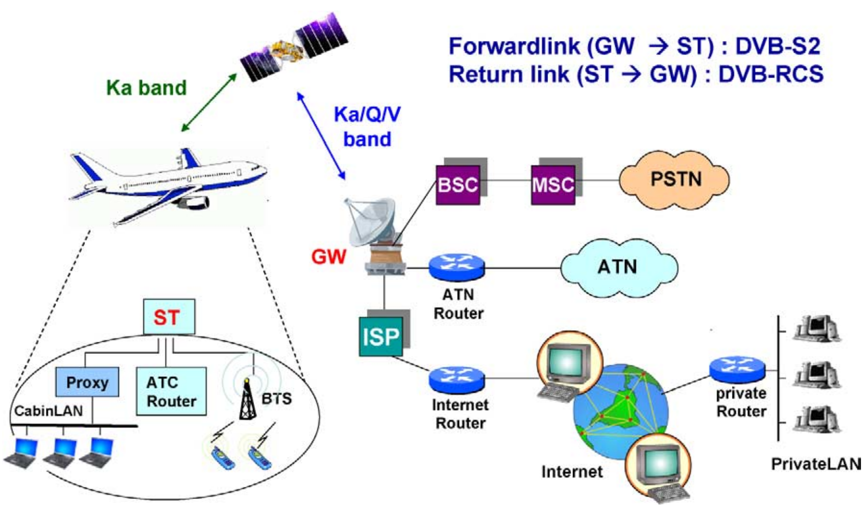

Fig. 1. Overall network architecture.

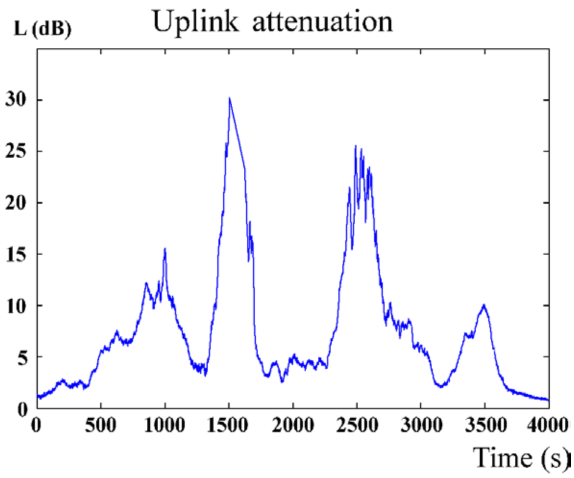

Fig. 2. Uplink fade event. Example of deep fade event (convective rain).

However, it should be noticed that the fade event presented in Fig. 2 corresponds to the attenuation as observed by a fixed terminal on ground. What is more, the probability of occurrence of such an event is low and directly related to the expected availability of the system (typically better than 99.9% in most systems). In the case of aeronautical communications, the observed attenuation profile can be expected to be strongly different due to the movement of the terminal (high attenuation rain cells cover a range of the order of few kilometers only) and the altitude of the aircraft (the terminal can be expected to be above clouds for a significant part of the flight). A more detailed discussion of these aspects is given later in the paper, two aspects need to be considered at this point. First, to the authors' knowledge, no precise model is available at present time for an aeronautical channel in Ka-band. It raises a difficulty in the determination of the system availability. Secondly, the availability constraint is highly different for ATS/AOC traffic and IFE traffic: ATS/AOC traffic should be handled during all the flight, possibly including on-ground phases; IFE service can be limited to the cruising phase. In order to take into account these aspects, the system relies on a highly flexible design where FMT are activated only when needed in order to obtain the highest resource use efficiency.

DVB-S2 access technique is TDMA (Time Division Multiple Access) with an embedded technique for fade mitigation based on FEC (Forward Error Correction) management. Data are sent in blocks of constant size (in number of coded bits) and modulation and coding are adapted to the actual propagation conditions observed on each individual link. It should be noticed that DVB-S2 relies on carrier to noise and interference ratio measurements (CNI) that are carried from terminal to gateway using the DVB-RCS return link. The CNI reports have been introduced in the last version of the DVB-RCS standard [1].

The resource management of DVB-RCS links is more challenging due to the MF-TDMA access. The standard does not explicitly introduce fade mitigation techniques; on the DVB-RCS link, the useable techniques are:

- UPC (Uplink Power Control): transmitter power is increased to counteract fade or decreased when more favorable propagation conditions are recovered to optimize satellite capacity.
- DRA (Data Rate Adaptation): nominal data rates are used under clear sky conditions (no degradation of the service quality with respect to the system margin), whereas reductions is introduced according to fade levels.
- ACM (Adaptive Coding and Modulation) [3]: using the different coding and modulation modes on the different carriers to match impairments due to propagation conditions.

In DVB-RCS, the resource management process DAMA (Demand Assignment Multiple Access) must take into account the required mode for a given terminal in order to choose the right carrier for resource allocation.

3.2. Return link radio design

On the return link, the considered modulations are QPSK (the DVB-RCS present definition), BPSK and 8-PSK (extension of available modes in the standard). Coding rates (using Turbo-coding) range from $1/3$ to $6/7$.

The radio design on the return link is proposed following three steps:

1. In the first step, calculate the classical link budget in the case of clear sky.
2. Then, analyze the link budget by considering the rain attenuation and calculate the overall available margin with FMT.
3. Finally choose the used modes.

In the considered design, the symbol rate is set to 683 ksymbol/s and the clear sky mode uses 8-PSK with a $1/2$ coding rate. The corresponding data rate is 1024 kbit/s. The clear sky link budget includes a 2 dB static margin that is needed mainly because of scintillation (this very fast fade variation is noticeable at Ka frequencies and can be considered as unpredictable. The variance of the phenomenon can however be modelled for margin determination). The most robust mode uses BPSK with a $1/3$ coding rate, the symbol rate is then 170 ksymbol/s (the reduction of symbol rate (DRA) gives an additional 6 dB margin). The FMT margin is summed up in Table 1.

The margin provided by FMT can be related to the availability of the link. Using the ITU model for long term rain fade prediction [4], the availability is 99.9% of the time.

DVB-RCS uses MF-TDMA (Multi-Frequency Time-Division Multiple Access) to allow a group of terminals to share bandwidth efficiently. Following DVB-RCS Guidelines [5] and using MPEG-2 Transport Stream (TS) time slots, the MF-TDMA structure is shown in Fig. 3. Five modes are selected as presented in Table 2. The modes have been chosen with a constant symbol rate at the exception of mode 0. This means that the carriers occupy a constant bandwidth; the mode management is hence easier, since the mode of a given carrier can be modified without any change of the frequency plan (DVB-RCS provides adequate signaling for frequency plan reconfiguration). Mode 0 should be considered as a rescue mode that has been introduced in order to provide high availability for ATC traffic, it should be noticed that the obtained data rate (56 kbit/s) is of the same order than using VDL Mode 2 link (31.5 kbit/s).

It should be noticed that the definition of 5 FMT modes does not mean that all modes are activated at a given time. The number of carriers associated to each mode can be dynamically modified according to the propagation conditions and the traffic load.

3.3. FMT decision loop

Obviously the efficiency of the system is conditioned by the performance of the FMT decision loop. For the return link, the signal is monitored by a Channel Estimator Block in the gateway and the modulation and coding (MODCOD) to be used is determined based on the channel quality estimate and the constraints imposed by the MAC layer (availability of the carriers pertaining to a modulation and coding for example). Estimation of the channel quality means determination of the Signal to Noise Ratio (SNR). The engine of the channel quality estimator block in the gate-

Table 1
FMT margin

Clear sky mode	8-PSK $1/2$	Req. Es/No. 8.7 dB
Most robust mode	BPSK $1/3$	Req. Es/No. -1.5 dB
ACM margin		10.2 dB
DRA margin		6 dB
Uplink PC margin		3 dB
Total FMT margin		19.2 dB

Table 2
FMT mode

Mode	Modulation and coding	Req. Es/No. (dB)	Symbol rate (ks/s)	Data rate (kbit/s)
0	BPSK $1/3$	-1.5	170	56
1	BPSK $1/3$	-1.5	683	228
2	QPSK $1/2$	4.5	683	683
3	QPSK $2/3$	6.9	683	910
4	8-PSK $1/2$	8.7	683	1024

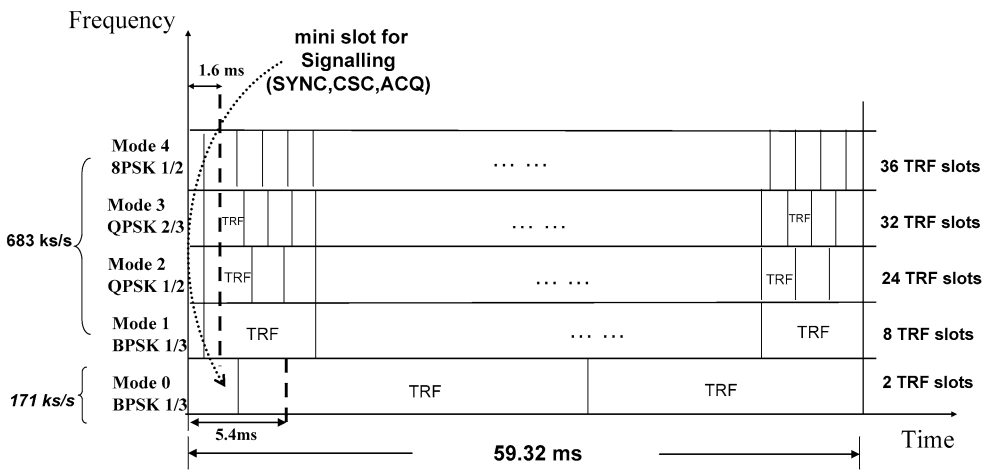

Fig. 3. Superframe structure.

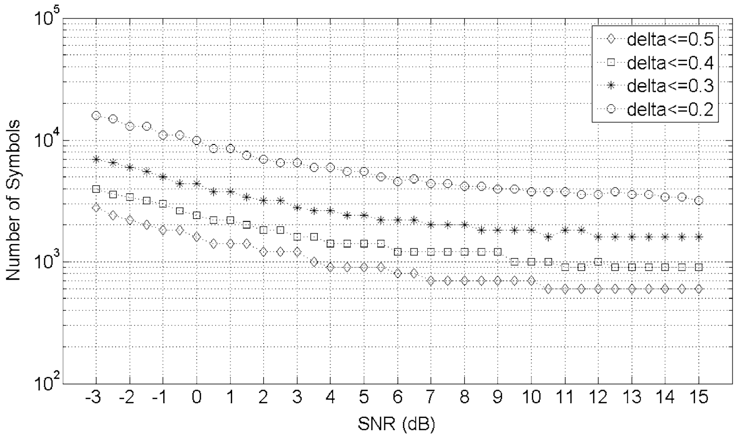

Fig. 4. Performance of the ML estimator for various error margins 'delta' as a function of SNR.

way is the SNR estimation algorithm which acts on the received signal to produce an SNR estimate. Of the algorithms published in the literature, the Maximum Likelihood (ML) estimator is the widely used one [6]. The performance of the SNR estimator is characterized by the number of symbols needed to produce an estimate with a given error margin. This error margin is defined as the magnitude of the difference between the true and the estimated SNR. For the ML estimator the error distribution is Gaussian. The performance of the ML estimator for various error margins is given in Fig. 4 [6].

As seen from Fig. 4, the error margin of the estimator depends on the number of available symbols for the estimation as well as the level of the SNR. For a fixed error margin, a lower SNR requires a considerably larger number of symbols compared to a higher SNR. Similarly, for a fixed number of symbols, a lower SNR has a higher error margin compared to a higher SNR. In the forward link as the symbols are plentiful the error margin due to the estimation process could be made small. However in the return link, the traffic is bursty with fewer symbols, so in general the estimation error margin is relatively high.

A specific attention has been put on the design of the FMT decision loop in order to minimize the errors in FMT mode determination. Margins are necessary in order to avoid the choice of a mode that is not robust enough for the observed propagation conditions: the higher the margin, the lower the probability of a false decision, but at the detriment of the global system capacity. In our design, a first margin DM (Detection Margin) is set in order to reduce the error in the selection of the appropriate FMT mode. A Hysteresis Margin (HM) is used in order to avoid repeated switches between adjacent FMT modes. Figure 5 illustrates the use of these two margins.

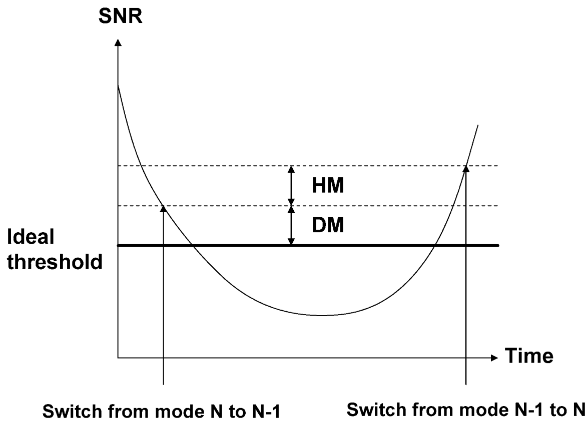

Fig. 5. Detection and Hysteresis Margins.

The chosen optimization criterion is the Packet Error Rate (PER) needed by the system. As a first approximation, the channel is considered as "error-free" if the FMT mode is compliant with the propagation conditions (actual PER in this case is better than 10^{-7}). On the contrary, frames are lost if the FMT mode is not robust enough. Simulations have been conducted on the basis of synthesized fade time series. Figure 6 illustrates an example of obtained result.

As presented in part 5, the system performances at layers 2 and above have been investigated using an OPNET model. In order to preserve the efficiency of the simulation, FMT simulation and the network model have been kept independent. The FMT control loop including MODCOD selection, margin optimiza-

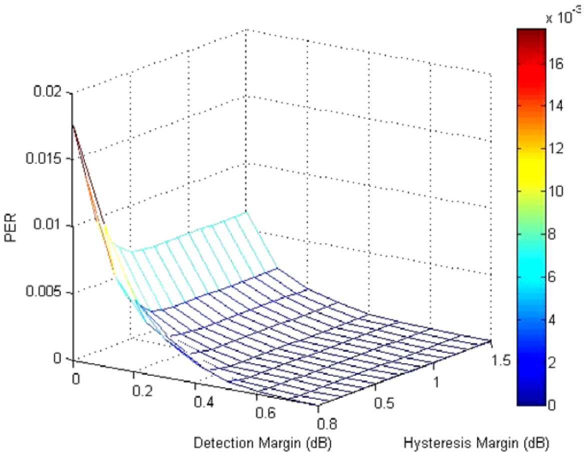

Fig. 6. PER versus DM and HM.

tion, channel estimation and decision has been optimized first and only the results of the FMT implementation are fed to the network model. The operation of the FMT for terminals in the service zone is emulated by providing the following data as inputs: a timeserie with used MODCOD and corresponding SNR values, a timeserie with MODCOD switching times.

4. On-board terminal design

The aspects presented in this part focus on the DVB-RCS return link. The terminal on board the aircraft has been designed in order to handle different traffic flows and manage access to the DVB-RCS link. The challenge is to multiplex traffic flows with highly different characteristics and quality of service expectations onto a single link.

4.1. Layer 2 packet format

DVB-RCS link can use either ATM or MPEG packet format. In this study, it has been chosen to use MPEG format and MPE segmentation and reassembly protocol.

Considering segmentation of SNDUs (Subnetwork Data Unit) for MPEG frame encapsulation, two solutions exist: MPE (Multi-Protocol Encapsulation) and ULE (Unidirectional Lightweight Encapsulation) proposed more recently by IETF [7]. It has been proved that ULE offers better efficiency than MPE mainly because of the decreased header overhead [8] and its use will be investigated.

4.2. Resource allocation process

In DVB-RCS system, the resource allocation process is shared between the user terminal and the NCC (Network Control Center). The on-board terminal must monitor the incoming traffic and generate capacity requests. The capacity requests are received by the NCC in the Hub and processed by the DAMA (Demand Assignment Multiple Access). Two alternatives can be considered:

- Centralized approach. Capacity requests can be associated to channel identifiers. In this case, the terminal considers each data flow independently and generates the associated capacity requests (either rate or volume based). The DAMA process in the NCC is then in charge to determine the TRF (traffic) slots associated to each channel id. As a result, each TRF burst allocation in the Time Burst Time Plan corresponds to a given terminal and a channel identifier.
- Distributed approach. Capacity request can be generated by monitoring all the traffic flows. Capacity requests are then associated to a single channel identifier (0 is the default value). This means the DAMA process allocates a global capacity to the terminal. The on-board terminal receives an allocation for a given number of TRF slots in a superframe, corresponding to an unmarked capacity. The terminal is then in charge of the service policy: the allocated capacity is shared between the traffic flows according to the respective QoS objectives.

The preferred approach in our study is the distributed one. It offers more flexibility for the differentiation of the traffic flows. This aspect is of tremendous importance here as the requirements for QoS are highly different from one traffic type to the other.

4.3. Core system design

We have considered two ways to design the structure of the on-board access terminal. The first one is compliant with the ISO/OSI philosophy; a strict separation between protocol layers is then respected and satellite dependant and independent layers can be defined as in [9]. A second design is proposed with a cross-layer signaling mechanism.

4.3.1. Classical design

Firstly, we can consider that each source (ATC, GSM or IFE) feeds an associated layer to encapsulation process. As shown in Fig. 7, this implies that QoS and capacity management are both achieved in DVB-RCS access layer in order to multiplex these different traffic flows on the single DVB link.

The terminal access layer accepts data on five SAPs (service access points). The first one is used by ATC traffic and the second one by GSM traffic; the remaining three SAPs are dedicated to IP traffic. The hypothesis is made that multimedia traffic is classified in the IP router with a DiffServ policy (3 QoS levels: EF, AF and BE); for each IP datagram, the corresponding SAP is determined on the basis of the DiffServ code point label. Then the incoming traffic is segmented in MPEG packets using MPE and stored in queues associated to the different priority levels. Hence, the satellite terminal determines the needed capacity on the basis of queue monitoring; this capacity is translated in capacity requests according to the DVB-RCS signaling formats, either rate based or volume based requests.

The resource management in the gateway has been adapted in order to take into account the requirements for the three main services. Slots allocation in the DVB-RCS MF-TDMA is made using three parameters: CRA (Constant Rate Allocation), which is a fixed allocation, available permanently to a satellite terminal whatever it is used; RBDC (Rate Based Dynamic Capacity) and VBDC (Volume-Based Dynamic Capacity) which are dynamically allocated on request of the ST. The allocation process uses a standard DAMA algorithm (allocation of CRA first, then RBDC requests and finally VDBC requests). The resulting allocation is sent to the terminals within the TBTP (Time Burst Time Plan).

Because of the long signaling loop (500 ms are needed between the capacity request send time and the corresponding TBTP reception), on-board the airplane, a server is needed in order to send data on the satellite link according to a priority-based service policy. A “super-priority” is given to ATC queue, and then the remaining queues are served with a WRR (Weighted Round Robin) service policy. It means that the ATC traffic “steals” capacity from multimedia traffic when needed in order to get the highest possible availability. This aspect is very important in the case the terminal needs to use FMT mode 0, since the available data rate is very low and does not permit to maintain a correct service for multimedia applications, the link is then used mainly for ATC traffic.

4.3.2. All-IP design

Figure 8 illustrates the second approach named all-IP approach. We now consider that, as IFE source, ATC and GSM sources will send their data via the IP router. Of course, it means we assume that ATC over IP and GSM over IP are both possible as shown respectively in [10] and [11].

Furthermore, as shown in more detailed Fig. 9, the capacity management and QoS management are no longer in the same layer; cross-layer interactions between DVB-RCS access terminal and the IP router should be considered. In other words, the capacity management in the satellite terminal will be based on

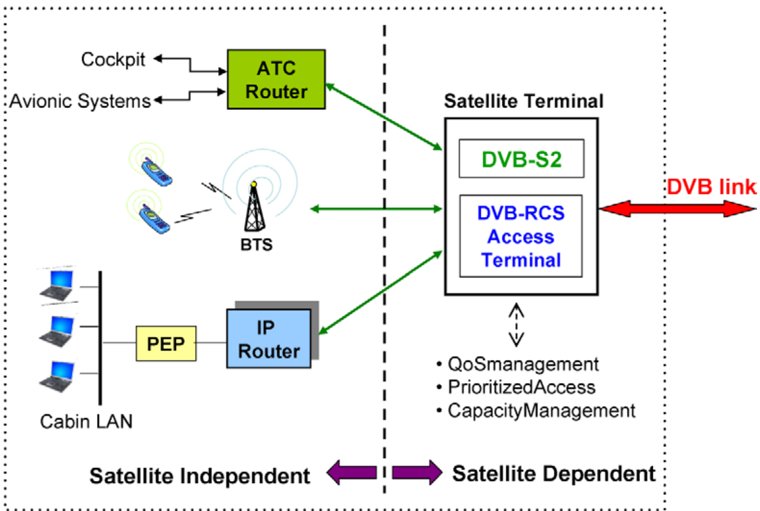

Fig. 7. Access design, first approach.

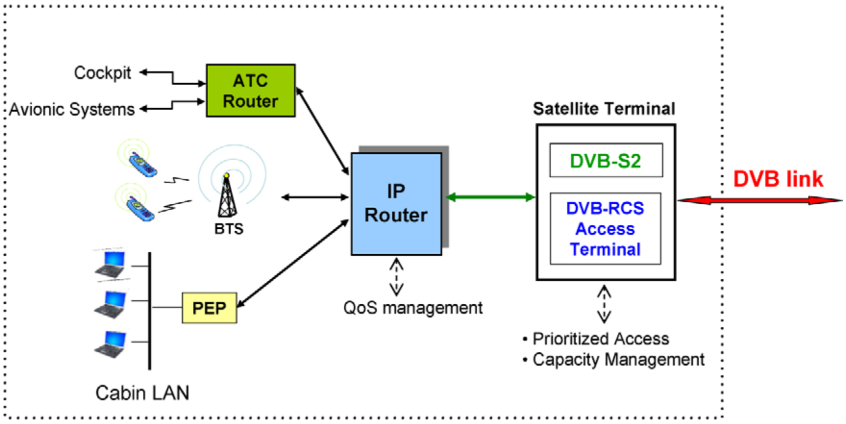

Fig. 8. Access design, second approach.

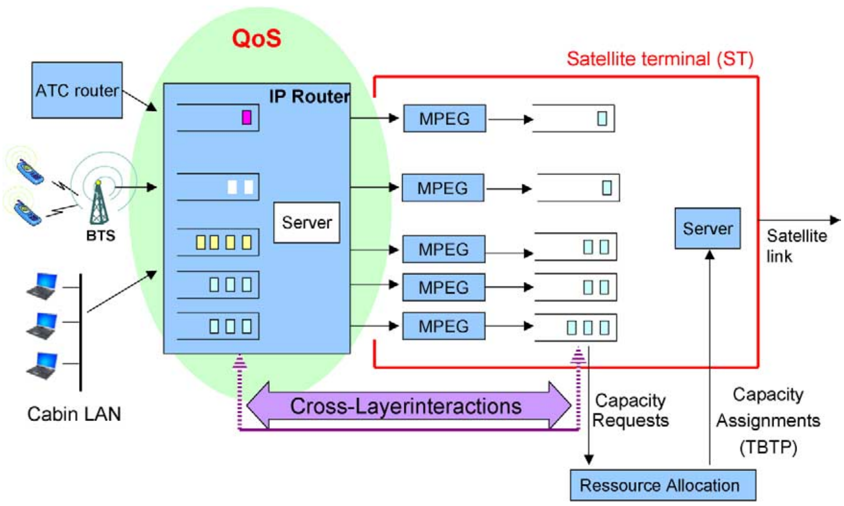

Fig. 9. Access design, second approach, detailed view.

the information from the IP router which classifies traffic with a DiffServ policy.

4.4. *Hypothesis, traffic characterization and QoS requirements*

The hypothesis in the considered system is that the performance objectives for the multimedia access and the aeronautical services have to be addressed separately. Very short outages (of the order of one superframe, e.g. 100 ms) are not detrimental for the Internet access, however some services need good performances in terms of transfer delay and delay variation (voice services for example). On the contrary, the link for aeronautical services must be compliant with the performance requirements issued by ICAO [12] and EUROCAE WG 53 [13]. These performances are given as time constraints (Maximum transaction duration ETRCP, duration for 95% of transactions TT95), availability (continuity CRCP, availability ARCP) and data transfer liability (integrity IRCP). In this case, priority must be set on availability and not capacity optimization, whereas time constraints are rather loose compared to Internet access (acceptable delays are of the order of 10 seconds). The return link access terminal will thus have to manage data flows with different requirements.

4.4.1. *ATC services*

The ATC router will generate traffic depending on the active applications. This traffic must be characterized according to the properties of these applications. At this point, two main applications have been identified:

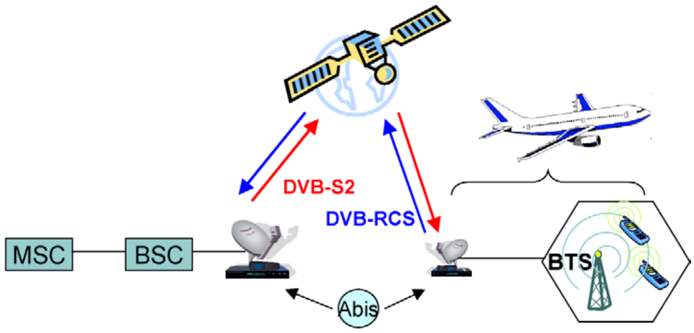

Fig. 10. GSM interface.

- ADS (Automatic Dependant Surveillance): this application is close to real time surveillance thanks to the use of periodical reports from the planes. An ADS report encompasses the position of the plane associated to its ICAO address.
- CPDLC (Controller-Pilot Data Link Communications): this application provides a message oriented link between pilots and air controllers. The gain expected from this data link is particularly high for transoceanic flights that rely so far on HF communications.

As first hypothesis on air–ground data link traffic characterization, we will use parameters values proposed by EUROCONTROL in ACTS simulator [14]. For both CPDLC and ADS-C services each aircraft generates messages conforming with:

- Packet size: 32 to 265 bytes,
- Mean arrival time 38.46 s (1.56 messages/minute).

Concerning QoS requirements, COCR [15] by EUROCONTROL explains that two phases have been identified for the definition of QoS parameters: phase 1 (from 2005 to 2030) and phase 2 (from 2020 to 2035). As our study lies within phase 1, we will use parameters and values that have been defined for this period:

- TD 95: 3.8 seconds,
- Integrity: 5.0E–6,
- Availability: 0.9965.

TD95 is defined as the end-to-end transaction delay observed for 95 percent of sent messages. Integrity is the acceptable rate of transactions that are completed with an undetected error. Availability represents the rate of undelivered messages.

4.4.2. Mobile telephony over DVB links

The study makes the hypothesis that a mobile telephony access can be deployed in the cabin. The main challenge is then to define the data format (encapsulation for transmission over DVB links) and service policies.

As shown in Fig. 10, we have considered that the interface between the aircraft and the gateway match with the one called Abis in the GSM default topology. This interface is located between the BTS (Base Transceiver Station) and the BSC (Base Station Controller). The feasibility of this topology by satellite has been proved by already existing solutions (Gilat, Cisco. . .). The main advantage of such approach is that voice is still uncompressed at this point; the needed capacity for one call is 16 kbit/s compared to 64 kbits/s at A interface. Furthermore, an equivalent interface named *Iub* is proposed in UMTS mobile telephony topology.

Abis interface is defined with a standard E1 multiplex. In our case, data must be encapsulated in MPEG packets. The solution is to recover TRAU (Transcoder/Rate Adapter Unit) frames at the output of the BTS. A signaling overhead is needed for resynchronization at the receiving end.

The hypothesis we made concerning traffic characterizations are as follows:

- Mean active period: 3 min,
- Mean idle period: 30 min,

and during a call [16]:

- Mean On period: 352 ms with 1 TRAU every 20 ms (16 kbit/s),
- Mean Off period: 650 ms with 1 TRAU every 480 ms.

The QoS requirements are those ones recommended by ITU-T [17]:

- Limits for one-way transmission: 150–400 ms (acceptable range),
- Delay variation (jitter): <1 ms,
- Frame erasure Rate (FER): $<3\%$.

Table 3
IFE traffic characterization

Service application	Application frequency	Mean holding time	Data rate return link (kbit/s)	Data rate forward link (kbit/s)	Burtiness
eMail	5/h	0.25 s	16	16	1.0
File transfer	5/h	4 s	144	144	20
WWW	2/h	30 min	16	144	20

Table 4
IFE QoS requirements

Application	E2E one way delay	Information loss
eMail	<4 s	0
File transfer	<10 s	0
WWW	<4 s/page	0

4.4.3. *IP traffic*

With the objective to provide an Internet access for passengers, it is of great importance to precisely define the TCP/IP architecture and the functionalities to be implemented in the different nodes of the access network:

- IP: definition of the addressing plane (NAT, DHCP...) and mobility management (re-routing of data in case of a change of earth station).
- TCP: choice of a connection management strategy (spoofing or splitting) and definition of the required functionalities of PEP (Performance Enhancement Proxy).
- Application: analysis of the offered quality of service depending on the activated options (persistent http, pre-fetching...).

In our study IFE traffic characterization and IFE QoS requirements are both based on results proposed by Wireless Cabin Project (Tables 3 and 4).

5. Preliminary simulation results

5.1. *Mobile telephony dimensioning*

A simulation model has been developed for the purpose of mobile telephony dimensioning. The chosen encapsulation process for the Abis interface presented in Section 4.4.2 implies that the DVB-RCS link should provide a variable capacity large enough to handle the on-going voice calls. Simulation results are presented in Fig. 11. The considered frame formats encompasses 36 TRF slots (FMT mode 4). An average traffic load of 50 Erlang uses a mean of 35% of the carrier capacity and a maximum of 70%.

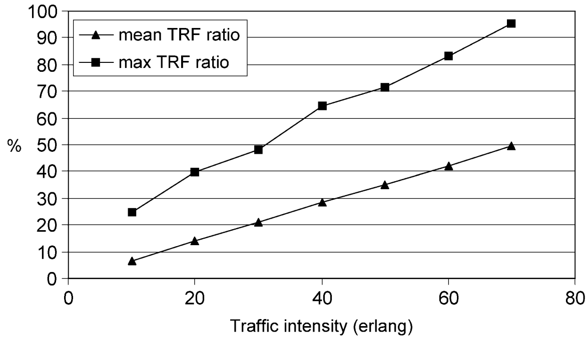

Fig. 11. GSM traffic frame occupancy.

5.2. *General feasibility demonstration*

A simulation platform has been developed according to the presented design using the OPNET Modeler software (OPNET Technologies Inc.).

The simulation model which is shown in Fig. 12 encompasses 50 terminals and one gateway corresponding to the traffic load expected in one satellite beam and a given superframe ID group (the simulated carriers represents a subset of the carriers in a 25 MHz repeater). Within each terminal, 4 traffic sources are activated with different characteristics for each priority level. The ATC traffic is made of packets with a size uniformly distributed between 32 and 265 bytes and a mean arrival time of 38.46 s. The Internet traffic is sorted with a DiffServ policy, so 3 traffic sources have been inserted corresponding to EF, AF and BE qualities of service. The EF traffic source corresponds to G.729 VoIP codecs. The AF traffic source represents an rt-VBR service (packets are also IP datagram). The BE traffic source models bulk data transfer with both exponential laws for packet sizes and inter-arrival times, as for example FTP sessions. Each airplane can be considered as a LAN (Local Area Network) and the aggregate traffic obtained from the sources should be representative of a typical LAN. It should be noted that in this first version of the model, the mobile telephony sources have not been inserted yet.

When terminal is concerned by very deep fades, outages occur when Es/No (resp. Eb/No) becomes lower

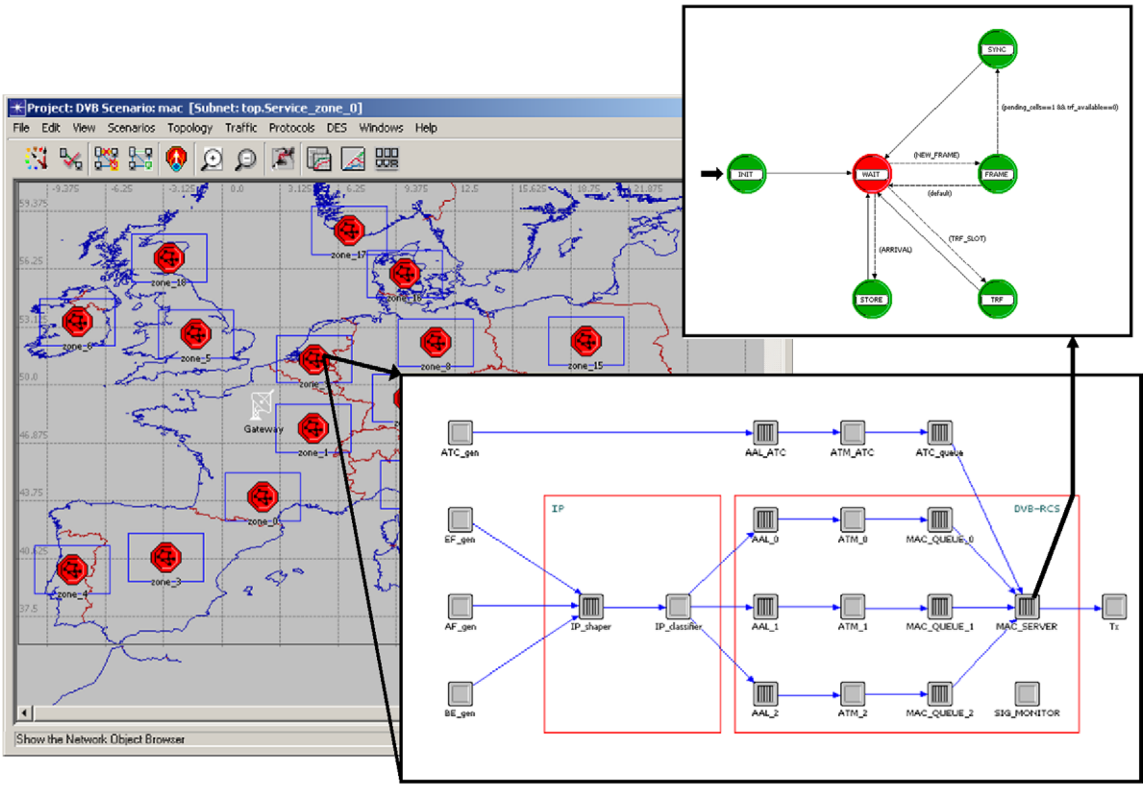

Fig. 12. OPNET simulation model.

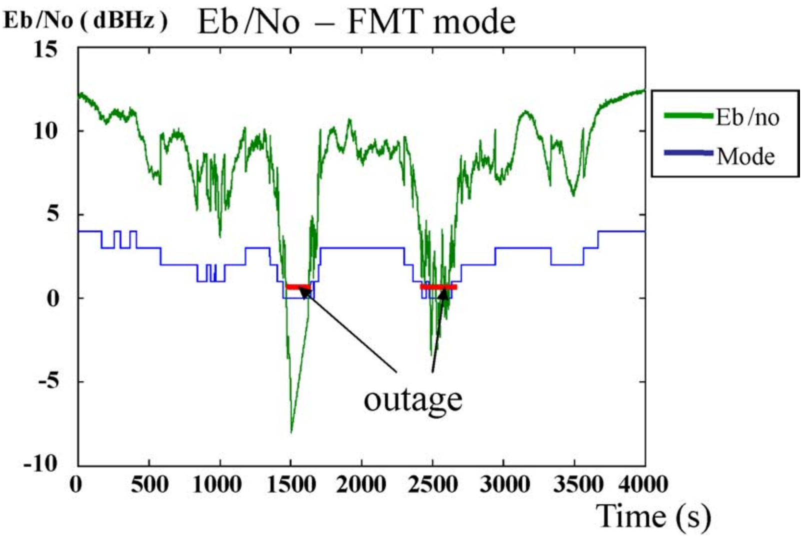

Fig. 13. System in deep fade, ACM activation.

than the minimum required level. Figure 13 illustrates the behavior of such system in deep fade. ACM modes are chosen in order to maintain the link quality to the detriment of the useable data rate.

As presented in part 2, the choice of the ACM mode relies on measurements made by the gateway and a FMT decision loop. A compromise must be found between the following parameters:

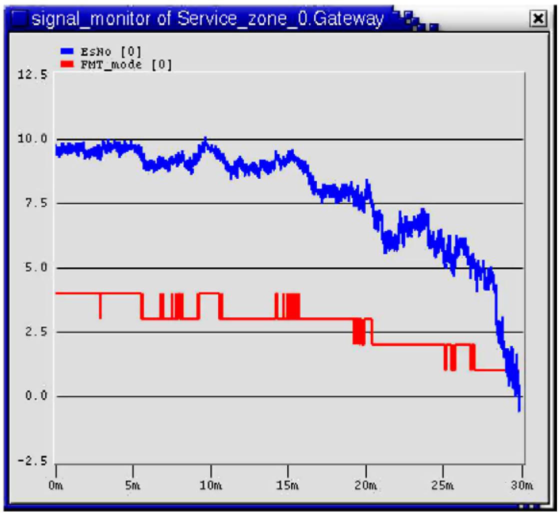

Fig. 14. FMT mode activation in satellite access terminal.

- Probability of false detection (ACM mode not fitted to actual propagation conditions),
- Desired link quality (Packet Error Rate PER),
- Optimization of the resource utilization (chosen ACM mode should provide the best spectral efficiency).

In order to simulate the propagation conditions and the FMT loop behavior, terminals and gateway make use of input files that provide either the current FMT mode or the signal to noise ratio. These files are obtained using synthesized attenuation time series [18] and a Matlab simulator of the decision loop. Figure 14 shows the SNR timeserie and the transitions between FMT modes for one terminal as interpreted by the OPNET simulator.

In the satellite access terminal, queuing delays have been monitored to check the correct behavior of the priority-oriented service policy. The probability density functions for queuing delays are plotted in Fig. 15 (outage periods are discarded).

Observed queuing delays are in conformance with the expected behavior. Priority 0 (corresponding to the EF traffic) gets shorter service times on average than the other priorities; the generated EF traffic is sent within a maximum of two superframes. Delays are spread over a wider range for priorities 1 and 2. Because of the particular traffic shape and service policy, the ATC traffic gets the better service.

It is also of interest to investigate the behavior of higher layer protocols. For example, a long-lived TCP/Reno connection has been monitored in terminal 0. The traces presented in Fig. 16 are the transmission delay in seconds, the congestion window in bytes (not to be confused with the effective transmission window) and the measured RTT in seconds (used for RTO determination). As the connection uses Best Effort quality of service, the Round Trip Time (RTT) can vary very quickly from one superframe to the next one, depending on the evolution of the global network traffic load and the DAMA reaction in the Hub. Timeouts are observed with slow start phases. The use of more sophisticated TCP flavors could presumably solve this problem.

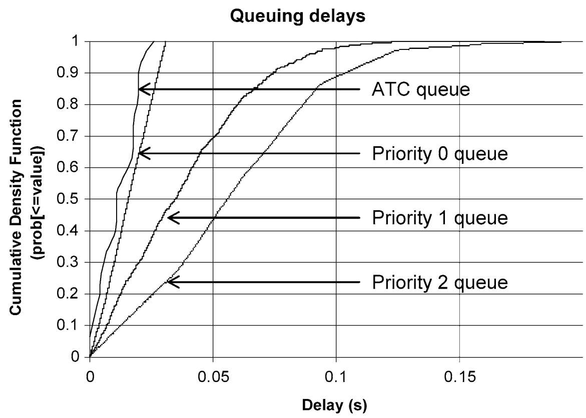

Fig. 15. Queuing delays in satellite access terminal.

6. Conclusion

In this paper, we have discussed an interactive system design for in-cabin services (Internet, GSM) and ATS/AOC traffic. The preliminary design bases on ETSI DVB-S2/DVB-RCS architecture using a GEO bent pipe satellite. The requirements for the different traffic types are different: multimedia traffic needs capacity and bounds on delay performance parameters; voice traffic must comply with tight time constraints; ATS/AOC traffic needs low capacity but high reliability.

The fundamental hypothesis of the presented system design is the use of extremely high frequency. The use of Ka-band (20/30 GHz) on the return link is favorable since bandwidth has been reserved at these frequencies for mobile services (on the contrary to Ku-band (12/14 GHz) where derogatory dispositions are needed). However, FMT (Fade Mitigation Techniques) must be activated with a major impact on resource management and eventually system performance. A specific effort has been made to obtain a

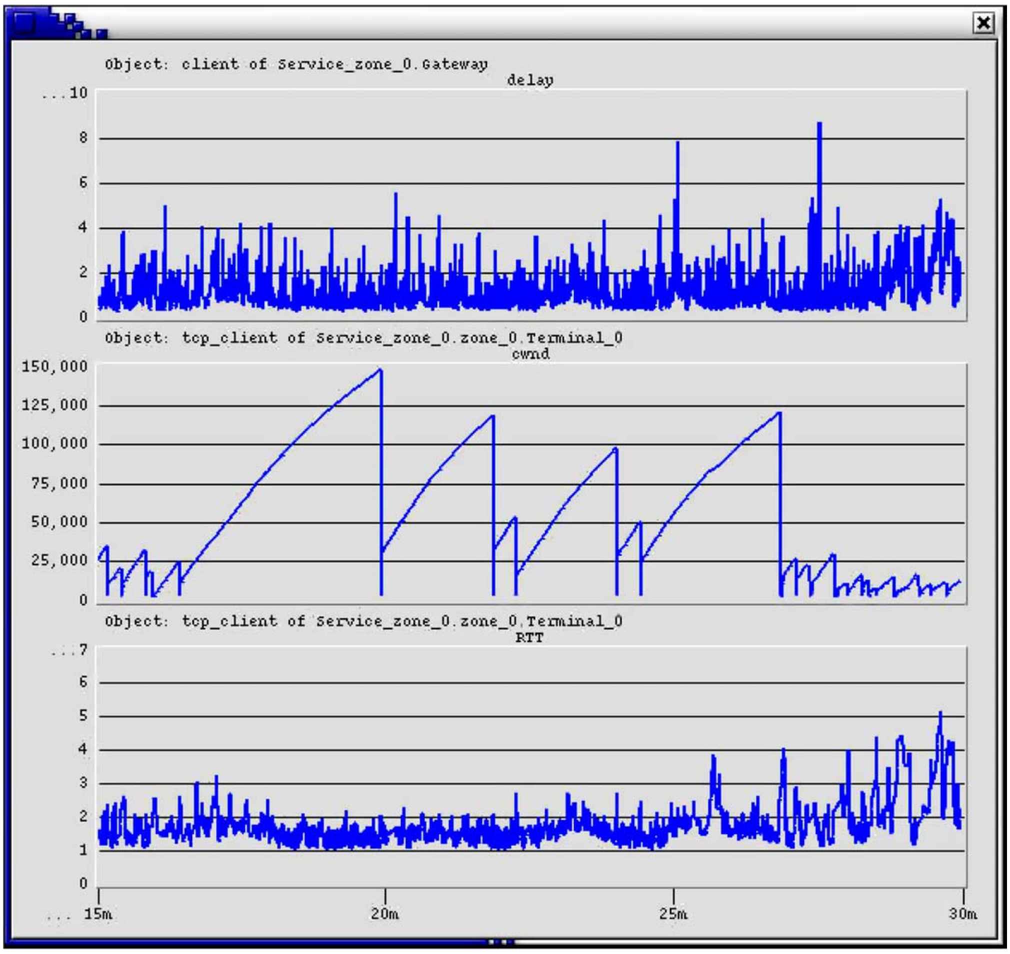

Fig. 16. TCP/Reno monitoring (delay, congestion window, RTT).

highly flexible radio link design. FMT decision loop has been investigated and design parameters have been optimized. As the result, a realistic behavior of the radio channel can be simulated for physical layer characterization but also for upper layers study.

The access layer design must face the challenge of converging highly different traffic flows on the same link. The study has been focused on the DVB-RCS return link where the more challenging problems are raised by the resource allocation process. The proposed design relies on a share of the allocation process between the on-board terminal and the NCC (Network Control Center) in the Hub. Two approaches can be proposed within the terminal, either with strict protocol layer separation or with cross-layer signaling. A network-level simulation model has been developed using OPNET and preliminary results demonstrate the technique feasibility of such system with increased capacity. Thanks to the chosen prioritizing mechanisms, all traffics can be transmitted on the same link with acceptable performances. However, the classical approach design faces a problem in QoS management; performances can be guarantied at MAC level, but not at the IP level. For this reason, the on-going work concentrates on the development of the model for the all-IP approach and will then focus on the comparison of the two proposed designs with the following objectives:

- Determination of the best suited service policies in the on-board terminal,
- Adaptability to the channel evolution (change in available capacity due to FMT or resource allocation process),
- Transport layer protocol friendly network behavior.

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