Quality of Service in High Altitude Platform Networks: Study of a DVB-S2 based Architecture

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Abstract—The purpose of this paper is the presentation of the wide range of applications available, that High Altitude Platform networks (HAP networks) can provide. HAP Networks have many advantages against satellites and terrestrial systems and there are also many supporting technologies/standards for the data transmission. We focus on the analysis of the new standard DVB-S2 and the possible ways that DVB-S2 can support IP services, beyond the classical Broadcast services. A crucial evaluation factor of these advantages is the key performance indicators of Quality of Service applying to possible traffic-mix scenarios. The central control unit is an adaptive radio resource management system which supports Quality of Service (QoS) and contains three main mechanisms: Buffer Management - Queueing System, Call Admission Control and Congestion Control. For the inquiry of these means, a discrete event simulator based on DVB-S2 standard was developed by using OPNET Modeler. Emphasis was given to packet and frame scheduling. Thanks to the results analysis, useful conclusions were exported.

Index Terms—High Altitude Platforms (HAP), Digital Video Broadcasting (DVB-S2), Adaptive Coding Modulation (ACM) Quality of Service (QoS).

I. INTRODUCTION

One of the future crucial roles of HAP Networks is to provide bandwidth and quality demanding services to mobile and fixed users [1],[2]. A lot of promising broadband wireless access technology is the Digital Video Broadcasting, especially the DVB-S2 standard [3]. DVB-S2 is an evolution of the DVB-S and there is compatibility maintenance achieving increased capacity by 30% compared to its predecessor. It is based on optimal performance transmission, absolute flexibility and the least possible receiver’s complexity.

The competitive advantage of the new technology is High Definition TV, MPEG Transport Streams, IP and ATM packet transmission due to advanced encapsulation techniques. So, broadband internet access and multimedia content delivery can be supported among video and radio broadcasting. Analytically, there is a need to follow up a categorization of the provided services and take into consideration the demand for quality aspects. The crucial unit for this purpose is an Adaptive Radio Resource Management system.

The rest of the paper is organized as follows: SECTION II outlines the Services and QoS Foundations. SECTION III presents a description of a DVB-S2 Adaptive Radio Resource Management and the relative evaluation simulation study. Finally, SECTION IV presents our conclusions and possible future work.

II. SERVICES AND QoS FOUNDATIONS

A. Services Categorization

The services can be categorized by delay or/and jitter sensitivity and real time data transmission criteria. As it is well known by traffic engineering we can determine conversational, streaming, interactive and background types of services. Such characteristics may be Constant Rate (CR), Variable Rate (VR) and Best Effort (BE), delay sensitivity, jitter sensitivity, tolerance in packet loss.

B. Supported Services

Following the above categorization, we can mention various broadcasting services like Constant Coding and Modulation Digital TV, Standard Definition TV and High Definition TV, Distribution of Multiple TS Multiplexes to DTT Transmitters, Digital Satellite News Gathering (DSNG) over Adaptive Coding and Modulation (ACM).

Supplementary services to the residual bandwidth may be IP unicasting and multicasting. Consequently, potential users or groups can enjoy web browsing, video streaming (video on demand - VoD), electronic mail, IP telephony (voice over IP - VoIP), file transferring [4].

C. QoS Foundations

It is important to declare three layers resources management so as to support quality aspects [5]. Control Layer is consisted of admission control and QoS routing. Data Layer handles processes such as Recognition and Marking, Congestion Control and Buffer-Queueing Management. Management Layer hands out policing and traffic analysis by network traffic.

It is worthwhile making a general observation regarding quality indicators. So, we should mention parameters like
transfer delay, delay variation, packet loss ratio, service/network availability, call failure ratio, call setup time, voice connection quality for VoIP sessions.

III. ADAPTIVE RADIO RESOURCE MANAGEMENT SYSTEM - SIMULATION STUDY

A. Simulation Scenario

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IV. ADAPTIVE RADIO RESOURCE MANAGEMENT SYSTEM - SIMULATION STUDY

A. Simulation Scenario

The scenario that was tested was that of a Content Distributor, where possible target groups are home users as well as SOHOs. In the technologies that are used currently (e.g. DVB-S through satellite), the end user has access to traditional non-interactive services, like television and radio.

By taking advantage of the, already mentioned, increased bandwidth of DVB-S2 and the considerably low propagation delay times of a HAP architecture, new possibilities appear. These two important factors make it viable that new interactive services are introduced to the user, by exploiting the bandwidth left unused by bandwidth services. Such services could presumably be VoIP, Video on Demand and Web Browsing capabilities.

In our case, simulation was focused on the downlink of a HAP system, and mainly on finding ways to take advantage of those already mentioned characteristics in order to provide QoS guarantees. As far as the uplink is concerned, a protocol like DVB-RCS could be used, but this was not examined in this simulation.

Figure 1 shows a possible system design, with the substantial difference over a satellite system being the point where the data is transmitted from the transponder antenna to the HAP. From this point on, the network differentiates in the way that Unicast services are provided along the usual Broadcast services. So, end-user terminals connected through decoders include Television sets, normal PCs or special VoIP phones.

B. Simulator Description

The core of our simulation consists of the parts shown in figure 2. These are:

1) Broadcast Traffic Generators

The Broadcast services taken into consideration are SDTV, HDTV and Radio. Radio was simulated as a simple Constant Bit Rate service, while for the TV services trace files were used to represent the data packets to be generated. These traces were produced out of realistic H.264 video files. All those services run during the whole simulation.

2) IP Traffic Generator

The Unicast services simulated are VoIP, as a real-time service, VoD, as a streaming multimedia service, and WWW as a best effort service.

Several sessions were created during each simulation, with the time of arrival between sessions of each service following an exponential distribution. The traffic models used for WWW and VoIP were those of an on-off process, while for VoD were again used trace files, of H.263 coding this time.

Central part of the unicast traffic simulation was the implementation of Call Admission and Congestion Control functionalities [6], [7]. Those applied to VoIP and VoD services only, since WWW is a Best Effort service.

Call Admission Control (CAC) is responsible for allowing or denying a new session each time it is generated, depending on the current network traffic. Since modulation and coding rate of the FEC frames to be sent can change due to varying channel conditions, we cannot consider the available bandwidth in bits/sec as a constant value. Instead, the measure to be used as a threshold for deciding whether to accept or not a new session is the available Symbol Rate. A typical value for a HAP transponder with bandwidth of 36 MHz and roll-off factor $a=0.20$ is 30 MBaud. This will be the value used for CAC calculations.
In the case of Congestion Control (CC), we are interested in situations, where an already accepted session changes modulation due to varying channel conditions. In that case it has to be reexamined whether this session can continue sending data, without encumbering the rest of the sessions. The criterion used is the same as in CAC, but now services may decrease their bit rate instead of being completely denied. In the case of channel conditions becoming better, it is possible for a service to increase its bit rate.

3) DVB-S2 Modulator

The DVB-S2 Modulator represents the gateway part of our architecture and its main purpose is to provide time scheduling and modulation of the data packets into DVB-S2 datagrams.

In order to do that, the modulator separates Unicast from Broadcast packets, and then uses 3 separate buffers-queues
for each priority of the Unicast packets. Then in order to fill
the FEC frames to be sent with data, we select packets from
the broadcast queue with statistical multiplexing. In the
case of Unicast, we select packets from the first priority
queue (VoIP packets) and then, if the FEC frame is not full,
we proceed to the second queue (VoD packets) and finally
to the third. In any case we have to fill the FEC with
packets corresponding to only one modulation scheme. In
order to choose the modulation to use for each FEC frame,
we use the first data packet available in the highest priority
queue each time [6].

Finally, the FECs are further modulated into physical
frames and are sent consecutively, with a round-robin
algorithm, one Unicast and one Broadcast frame each time.
In the case where no FECs of the one type are ready, we
send again a FEC from the same queue as before.

The modulations used in the simulation are specified in
the following table:

<table>
<thead>
<tr>
<th>Service</th>
<th>Modulation</th>
<th>Code Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>HDTV</td>
<td>8PSK</td>
<td>5/6</td>
</tr>
<tr>
<td>SDTV</td>
<td>QPSK</td>
<td>1/2</td>
</tr>
<tr>
<td>Radio</td>
<td>QPSK</td>
<td>1/2</td>
</tr>
<tr>
<td>VoD</td>
<td>QPSK, 8PSK, 16PSK</td>
<td>1/2, 5/6, 8/9</td>
</tr>
<tr>
<td>VoIP</td>
<td>QPSK, 8PSK, 16PSK</td>
<td>1/2, 5/6, 8/9</td>
</tr>
<tr>
<td>WWW</td>
<td>16PSK</td>
<td>8/9</td>
</tr>
</tbody>
</table>

4) End User Equipment

The role of the user terminals in the simulation was to
merely keep statistics (including the delay time from the
HAP link) and to send updates of the channel condition, in
order to choose the right modulation and code rate for
Unicast services.

C. Summary Of Results

We ran the simulation with the following parameters:

<table>
<thead>
<tr>
<th>Service</th>
<th>Percentage</th>
<th>Load</th>
<th>SymbolRate</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>15%</td>
<td>0.8</td>
<td>3.6</td>
</tr>
<tr>
<td>VoD</td>
<td>25%</td>
<td></td>
<td>6</td>
</tr>
<tr>
<td>WWW</td>
<td>5%</td>
<td></td>
<td>1.2</td>
</tr>
<tr>
<td>Total</td>
<td></td>
<td></td>
<td>10.8</td>
</tr>
</tbody>
</table>

As we can see, the theoretical use of the available
bandwidth (30Mbaud) by the broadcast services is 45%,
while the percent occupied by Unicast is 36%.

Following is a sample of the results obtained after
running the simulation for 8 minutes.
V. CONCLUSIONS

This work has examined the possibilities of providing broadband access and multimedia content distribution over DVB-S2 oriented HAP Network. For this purpose, it was a great necessity to determine a radio resource management schema adapted to QoS demands. A discrete event simulator indicated that an enterprise of this kind is feasible under conditions. More specifically, broadcast services use more efficiently the given bandwidth. Furthermore, there is good support for real time services with an accepted delay. Also, it was clear that VoIP services waste FEC because of their small packet size.

There are a few steps for further research. Open issues are development of efficient algorithms in RRM unit and support user mobility (micro and macro-mobility).

REFERENCES