I
Introduction

The IP technology was not primarily designed for the transport of real-time multimedia content and, as such, it presents challenges when optimizing network capacity, while meeting high quality standards for the audio and video contents being transported. The present work concerns channel coding for distribution of audio and video (A/V) services over IP networks, more specifically, services in the form of MPEG-2 Transport Streams protected with *Fountain Codes*.

The motivation for this work lies in the growing demand for video over IP in many current applications. To mention a few examples, in the form of distribution of multimedia content over the Internet, as contribution links inside traditional broadcasters' networks or as standard and high definition contents transported in IPTV networks to the end customer.

Let us start the introduction of the scenario with a brief description of the overall topology of an IPTV network. This is shown in fig. I.1 and is used to introduce the most important concepts and evaluation mechanisms employed in the simulation scenarios herein.

The network presented in our example comprises a National Hub, which receives contribution signals, mostly over satellite, that will compose the end programming, Local Head-Ends, that receive the Main program over a core network and adds local contents to the same and, finally, Multiplexing locations where the streams are multiplexed together with other services such as broadband access and voice, for distribution to the end user.

In the National Hub, in the left-most portion of the figure, dozens of satellite feeds are received. These are typically modulated according to *Digital Video Broadcasting – Satellite* (DVB-S or DVB-S2) standards and transport multiple MPEG-2 Transport Streams each — such streams and its main characteristics will be explained throughout this reading.

The received Transport Streams carry the audio and video (A/V) services of interest, which are re-multiplexed into new streams and encapsulated into IP packets, employing network protocols, typically *Real Time transport Protocol* (RTP) into *User Datagram Protocol* (UDP).

The resulting IP string is transported to the Regional Remote Head-Ends (RRHE) through the service provider's Core Network, typically an IP/MPLS network. In the RRHE, the Transport Stream is de-encapsulated from the network protocols and re-multiplexed with new services, which are local to the region where the RRHE is found. The string is packetized back into the appropriate network protocols and transported through the service provider's Access Network to local distribution facilities, where the IPTV services are finally multiplexed at IP level with other services, such as broadband, typically employing a *Digital Subscriber Line Access Multiplexer* (DSLAM), in order to reach the end user set-top-box (STB) through the DSL lines.

Such topology might face some issues, which generate quality impairments to the A/V services reaching the end user that justify the deployment of protection schemes for the A/V content. Commonly observed issues are listed hereunder:

- The Core network is usually high speed, high capacity and robust, but the access network connected to it has less capacity and might be overloaded with IPTV and broadband traffic and discard packets carrying IPTV content. Jitter might also be present in this hop;
- The last mile DSL circuit, feeding the end customer STB, suffers more often from degradation, which also generates packet drops;
- The services that arrive at the Main Hub can be compressed at rates that affect the signal quality, a problem that is originated at the source of the information;

The aforementioned factors can result in visual impairments in the decoded video, such as the presence of pixelization and artifacts. These issues can be translated into high rates of customer dissatisfaction and thus less programming attendance, affecting the service providers' revenues.

In order to detect such problems as soon as possible, particular points in the network can be continuously monitored, as indicated in the figure. The set of measurements listed hereunder can be employed with this purpose:

- Transport Stream Measurements;
- Objective Blocking Artifact measurements;

– IP packet drops and jitter monitoring.

Transport Stream measurements are typically employed for monitoring the incoming feeds at the main Hub, for verifying its integrity prior to re-multiplexing it as a new Transport Stream and at the DSLAM facilities, for verifying the integrity of the stream after its transport through the Access Network, subject to packet loss. A subset of these measurements will serve as criteria for the analysis performed in the simulations presented in this work and the same are described in [23] and [2].

The second item, Blocking Artifacts measurements, can be performed at the main hub, for quality assurance of the incoming services, i.e. that these were not over compressed at the source. Blocking Artifact measurements are also commonly employed at the output of a few samples of customer grade STB's, for monitoring the end user Quality of Experience (QoE). The objective measurement employed in the simulations herein, for comparison between a Fountain code and Reed-Solomon schemes, can be found in [12].

Finally, the third item, IP packet drops and jitter monitoring, explained in [5] and [22] , entitled *Media delivery Index* (MDI), which consider inter-arrival IP packets jitter and IP packet losses across the network. This type of monitoring is commonly employed for monitoring of Access Network receptions in the DSLAM facilities, where packet drops and jitter are more likely to be observed.

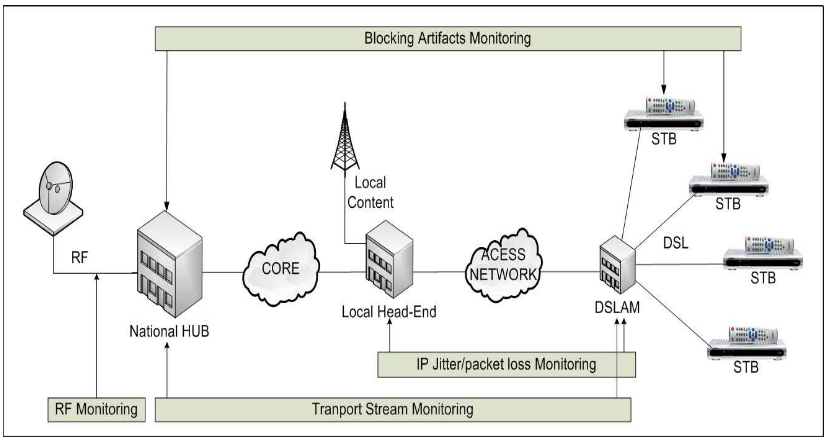

Figure I.1: Typical IPTV architecture

At this point, important impairments which result from transport of the content through a real network with issues and methods of quantifying these were cited. It is now possible to employ these quantification methods for evaluation of different channel coding schemes.

The traditional retransmission of dropped packets, as in *Automatic Repeat Request* (ARQ) schemes, is not applicable to real-time multi-media, due to the low-latency requirements of such contents. Channel coding with provision to recover from packet loss throughout the transport over IP is the right choice for improving the overall customer QoE. Techniques involving well known block codes, such as Reed-Solomon, are widely adopted by equipment vendors and specified in RFC's and standards.

Fountain Codes are sparse graph codes, with properties that make it attractive for transport of multimedia over IP. These codes have finite dimension and infinite block length and so a rateless scheme can be implemented, for scenarios where the channel conditions are unknown to the sender prior to initiation of the communication. The overhead requirements are also low in Fountain coding schemes.

This document is organized as follows: chapter II presents a description of Transport Streams, which is the content to be encapsulated into IP protocols, focusing in the parameters monitored in the simulations scenarios of chapter IV.

Chapter III, presents a model for the IP channel and the methods for overcoming packet drops and jitter adopted up to date.

Once the scenario is mapped, the simulations and results are presented in Chapters IV. and V respectively.

Theoretical aspects on Fountain codes can be found in Appendix A.

This work is concluded with the proposal of a *Transport Stream rate adaptive channel encoder*, that is presented in B and which rate may adapt to the importance of the content and fields in the Transport Stream subject to transmission.