# Video broadcast using cloud computing with metadata

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Abstract- In this paper the model of broadcast digital video signal with embedded audio (SDI) using cloud computing is detailed in each transcoding process of the signal. The SDI signal is coding and multiplexing in ASI signal, Multidecriptor transcoding the signal in Transport Stream, with GT-3 we can change the program in a new TS. Then with CPC develop chuncks with HLS (Http live streaming), with diferents profile. Anevia recive this chunck and delivery the signal to cloud. Testing bandwidth and bit rate for each signal process is performed, without losing the quality control standards and without losing the services of metadata (V-chip, closed caption, cue tone, DPI, GPI, Watermarket, SCTE 35 y104, etc.). How the stream is received by the end user it is shown.

Keywords – DPI, GPI, V-chip, Watermarker, Profedata2015

I. INTRODUCTION

The Television broadcast in different ways, satellite, fiber, microwave or cable, the trouble with these broadcast is that can only receive users within the satellite footprint and coverage area. Broadcasters, cable and satellite television broadcasting signals sent all at once and the user decides that signal (channel) requires seeing. This means that all signals are needed at the same time and wasted bandwidth [1].

For that reason we need a broadcast model in which it can have the greatest worldwide coverage without lose quality control, easy, secure and in real time. The signal includes ancillary data embedded video and audio, for use by television stations, repeaters and users when required. This services are embedded and don't affect the signal quality because they are in video lines reserved for signaling, aren't visible to user. Video broadcast IP has a more efficient process. All video signals are on a central server and only the signal (channel) that user chooses is going to sending. This means that it uses less bandwidth with better quality signals or the option to add other applications (interactive TV). Then we design a model in which digital video signal SDI is processed, either HD or SD in real time, it's encoded and transcoded by different devices to broadcast by the cloud (internet). We show each transcoded signal, the presence of ancillary data is verified at the end of the model.

#### II. . SDI VIDEO SIGNAL

Serial Digital Interface (SDI) is a standardized digital video interface. Mainly used for transmission of uncompressed video signal (RGB video) without encryption (optionally including audio). It is also used for transmitting data packets. This format specifications are contained in ITU-R BT-656 recommendation for video signal transmission in digital components, using a flow of 270 Mbits (Figure 2). This figure is derived from [1]:



•pictures (or frames) per second \* lines of each frame

samples each line \* bits of each sample.

Figure 1. Bit rate of the video signal.

In the HD / SD SDI signal, lines are embedded in video, closed caption have been inserted in line 9 and 21 respectively, V - chip, DPI, GPI, AFD, audio has cue tone and watermarked. We used standard SCTE 35 and SCTE. The Digital SDI video signal with all active services we called PROFEDATA2015 video signal. The following figure shows the area where these services have shown.



Figure 2 Lines Video.

To understand how IPTV need to understand each of the processes carried out in the original signal. The digital video

signal uncompressed SD and HD have a bit rate of 270Mbps and 1.485Gbps respectively, so that these signals are transmitted by internet has to encode and compress the signal to reduce its bit rate [3].

#### III.- COMPRESSION MPEG 2 VS H.264

MPEG-2 has excellent image quality, but historically has been that requires too much bandwidth for WAN applications. Outside broadcast, only the most demanding users in distance education, business collaboration, government / military, and medical markets have implemented MPEG-2 transport across the WAN. MPEG uses the following frames: I-Frames (Intra Pictures) are independent compressed frames, are as independent JPEG images. P-frames are predicted from the closest I or P frames available. B frames are a second level of prediction (adjacent images on eachside).

A group of pictures (GOP) is the number of frames of an I frame to another. Thus, in a GOP, there is a comprehensive framework, and a number of frames to predict the movement associated with the full frame. The highest levels of compression (less bandwidth for a particular clip) are achieved through larger GOP and prediction deeper box (that is, using a compression structure IBP instead of a structure of IP), see Figure 4. However, the deeper compressions require more time to encode, resulting in higher latencies [4].



Figure 3. Estructure MPEG

The need to decrease the bandwidth and the wide availability of broadband connections, have introduced the MPEG-4 AVC compression standard for video in high quality network distributed everywhere [5].

"MPEG-4 AVC", also called "H.264" and "MPEG-4 Part 10 "is being rapidly adopted in all segments of the network video industry, as it saves about 60% of the bandwidth [6]. It is now the accepted standard for communications, broadcast, and streaming applications. It is used in Flash, Silverlight, QuickTime, iPhones, iPod, PlayStation, Nero, HD DVD, and technologies of Blu-ray discs, see figure 6.



Figure 4. Compression standar.

#### IV.- VIDEO BROADCAST USING CLOUD COMPUTING

SD/HD video signal is encoded in MPEG2/H2.64 The respectively and multiplexing, we have a new signal called ASI (Asynchronous Serial Interface), is a streaming data format which often carries an MPEG Transport Stream (MPEG-TS). An ASI signal can carry one or multiple SD, HD with audio programs that are already compressed. The stream in MPEG 2 has 2 Mbits/s of bandwidth and H.264 has 4Mbits/s (these parameters were obtained with different test of optimization bandwidth), as opposed to an uncompressed SD-SDI (270 Mbit/s) or HD-SDI (1.485 Gbit/s). In this process the signal maintained all the services ancillary data, we can check it in MTM (monitor transport stream), see figure 5.



Figure 5. MTM (Monitor Transport Stream).

PROFEDATA2015 signal have been used in the top of this model, each step we explain. The following figure is flow chart model.



Figure 6. Video broadcast using cloud computing.

#### V.- MULTIDECRIPTOR

The satellite decoder multidecriptor DSR- MD is designed for cable operators and other commercial satellite operations. After we configured properly, we may receive the authorization and control of information of the video signal. You can have as input RF satellite signal or ASI. You have to configure the IP address port 10/100 MD: port of the bottom of the back panel. Gateway and configure network subnet mask. We have to realize the same procedure for GigE port IP address: port plane upper back. You have to choose in menu mode to enable MPTS GigE output transport stream over Gigabit Ethernet using MPEG Transport Stream Protocol Multiprogram. In the signal output we obtained MPTS Multiple Program Transport Stream. The next figure shows the device (figure 7).

	Table 1 VLAN 2					
EQUIPO	IP 10/100	IP GigaE	<b>Puerto Multicast</b>			
MD 1	10.1.2.121	234.1.2.121	1234			

Table 1 shows the IP address of MD.

#### VI.- MÚLTIPLE BIT-RATE (MBR) TRANSCODIFICADOR

The GT -3 Multiple Bit -Rate (MBR) transcoder offers three gigapixels/seg MPEG-4 AVC capacity transcoding for video delivery multi-screen, using silicon technology for higher quality video, density and energy efficiency.

In VLAN 2 (Multicast sources) it is input 2 to GT3 each device with 2 cards, the device can have until 4 cards. The card has the capacity for 12 signals SD MPEG2 or 6 signals

HD H.264/MPEG 4, you can perform a combination of both.

VLAN 3 is management for computers; the IP that was assigned to GT3-1 is 172.16.10.10 and 172.16.10.20 to GT3-2. The IP and UDP port address is taken from the table. When we created the group, the programs were automatically displayed as shown in Figure 8.



Figure 8. In MPTS

Each program is a video signal in real time, if you select a program; you can rename and edited to obtain a new output group. Each program is a television channel in real time, in the input program you select PID's of the program to be used for the new output group. When the transport stream of the new group is activated, the output is for port 2, which was previously set up as output port.

Transport Stream Name	Primary	IP Address	Mode
ESTLN_320X240	Eth2	235.1.2.57	HLS
ESTLN_480X360	Eth2	235.1.2.58	HLS
ESTLN_640x480_1	Eth2	235.1.2.59	HLS
ESTLN_640x480_2	Eth2	235.1.2.60	HLS

Table 2 shows IP's address profiles.

## VII.- ADAPTIVE BIT RATE (ABR)

It has a new transport stream with different profiles. Adaptive bit rate (ABR) streaming is an essential technology for delivering video over IP network. With ABR streaming, we can made multiple versions of bit rate with the same video content; available to customers via streaming servers. A customer can dynamically change a bit rate to another, based on network conditions.

The CPC allows a client dynamically choose the correct bit rate. The input of CPC can be:

• A live (MPEG2 transport streams).

• An active video file, as (VOD) movies video on demand.

The CPC takes this input and converts packet flow or chunks, according to one of these protocols: HTTP Live Streaming (HLS version 3) or IIS Smooth Streaming and then delivers the files to server. Select HLS to format chunks. Enter the length of the segments. It is the size in video second of each packet or chunks (figure 9).

Job Detail					
General	Input Streams	Publishing Points	Client Access Points		
				Name	BANDAMAX_Output
			Chur	nking Format	HLS *
			Play	list Filename	BANDAMAX_Output
			Segment / Fragment Dura	tion (second)	10
			Retain	All Segments	
			Window Size	Per Manifest	15
			Segm	ents To Hold	31
			Allow CI	ient Caching	
				Output DRM	None
			Verimatri	ix Content ID	
				Submit	Cancel

Figure 9. Configuration chunks.

In the page packager jobs configuration showed all chunks created, one by one is activated with start button.

#### VIII.- ANEVIA

ViaMotion Streamer is a video server designed to implement OTT Live, Video -on- Demand, Catchup TV. The ViaMotion Streamer server is able to manage Smooth Streaming and multi - bitrate MPEG -TS, live video input streams with H.264 and AAC encoding, SD or HD resolutions. Use streams that are within protocols ABR (HLS, Smooth Streaming, HDS and MPEG-DASH).

To active a service in real- time; live on page, we can add the channels live to repackage. To add a new channel, several fields must be specified in the input configuration:

- Input Type: depending on the encoder output: MS Smooth Streaming or Apple HLS.
- Disk Storage: Disk declared where OTT record the live stream.

Recording length: Maximum size fragment /package on the disk for buffer extraction.

HLS input: When you choose this protocol, the encoder must create the following FTP path of the WebDAV protocol. The publishing point is:

- ftp://<ip\_address>/live/<disk>/<channel\_name> / for FTP
- http://<ip\_address>/live/<disk>/<channel\_name >/ for WebDAV

They are discharged each service as shown in Figure 10.

Input				
🔿 All input streams are OK				
Video tracks				
<ul> <li>700 Kbps - 640 x 360 - AVC1 - 2 s - archive 100%</li> <li>1 Mbps - 852 x 480 - AVC1 - 2 s - archive 100%</li> <li>2.4 Mbps - 1280 x 720 - AVC1 - 2 s - archive 100%</li> </ul>				
Audio tracks				
<ul> <li>128 Kbps - German - AACL - 48 Khz - archive 100%</li> <li>128 Kbps - French - AACL - 48 Khz - archive 100%</li> <li>128 Kbps - Language not specified - AACL - 48 Khz - archive 100%</li> <li>128 Kbps - Language not specified - AACL - 48 Khz - archive 100%</li> </ul>				

Figure 10. PID's of video and audio.

By having the output Anevia with http protocol, we only need one public IP (Internet output) and redirect public IP to output Anevia, you can have a conditional access; username and password for each user by enabling and creating a website. The final streaming can be played from any fixed or mobile device (Tablet, Smartphone, Smart TV) with internet access. With a waveform analyzer can be verified the ancillary data in any process of the signal video (Figure 11).



Figure 11. Ancillary data status.

## IX.CONCLUSION

In the present, the TV stations and service providers worldwide are limiting coverage either satellite, fiber optic, microwave or coaxial cable, offering only a few services of metadata (V -Chip, Watermarker, Cue tone, GPI, Closed caption, AFD, DPI, Dolby and recommendations as SCTE 104 and 35). Therefore, the interest of this work is to find a model to transmit all services in the same signal with worldwide coverage, with standards of quality video and without use or implement any additional infrastructure, any television station, repeater and users can receive the video signal on any computer or device with internet access, using cloud computing (cloud).

A signal able to have all services of ancillary data for television stations, repeaters, fixed and mobile users with optimum quality we have called Profedata2015 (Professional data), this name is designated because the signal use professional technologies (Profe), include digital processing (data) and all the services for ancillary data (V - Chip, Watermarker, Cue tone, GPI, Closed Caption, AFD, DPI, Dolby and recommendations as SCTE 104 and 35) in

this 2015. It should be noted that the model depending on the needs of broadcasters and users, they can use some or all services that your device can receive and support, if the user should not require any; the video signal with quality standards continue because the signaling is embedded. All this is achieved with optimal bandwidth management.

In conclusion, a model is designed to transmit worldwide (cloud) Profedata2015 video signal to all devices with internet access, optimizing bandwidth. However we are working in two ways, the first drafting improvements in the standards of video signal so that users can enjoy the best quality of video that can be offered in the present and in second term interactivity television and mobile applications.

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