A NEW METHOD FOR RECALCULATING THE PROGRAM CLOCK REFERENCE IN A PACKET-BASED TRANSMISSION NETWORK

M. ALEXANDRU[1] G.D.M. SNAE[2] M. FIORE[3]

Abstract: *This paper proposes and describes a novel method to be used in a video transmission network that synchronizes the video signal with the audio signal, replacing the program clock reference with a new value from an onboard high reliability source. The intended goal of recalculating the program clock reference values was a success and the test works perfectly on the DekTec decoder.*

Key words: *Program Clock Reference, DVB, MPEG-2.*

1. Introduction

Digital Video Broadcasting (DVB) is a set of standards that define digital broadcasting using existing satellite, cable, and terrestrial infrastructures (Figure 1). Today, the DVB Project consists of over 220 organizations in more than 29 countries worldwide. DVB-compliant digital broadcasting and equipment is widely available and is distinguished by the DVB logo. Numerous DVB broadcast services are available in Europe, North and South America, Africa, Asia, and Australia. The term *digital television* is sometimes used as a synonym for DVB. However, the Advanced Television System Committee (ATSC) standard is the digital broadcasting standard used in the U.S.

A fundamental decision of the DVB Project was the selection of MPEG-2, one of a series of MPEG standards for compression of audio and video signal. MPEG-2 reduces a single signal from 216 Mbit/s to 5 Mbit/s allowing broadcasters to transmit digital signal using existing cable, satellite, and terrestrial systems. MPEG-2 uses the loss compression method, which means that the digital signal sent is lost. This lost data does not affect how the human eye perceives the picture. Two digital television formats that use MPEG-2 compression are Standard Definition Television (SDTV) and High Definition Television (HDTV).

Just like HDTV, DVB has a problem with the synchronization of pictures and sound. This problem is resolved by recalculating the values of Program Clock Reference (PCR) [1], [3].

2. MPEG-2 and Transport Stream (TS) Characterization

Moving video images must be delivered in real time and with a consistent rate of

[1] Dept. of Electronics and Computers, *Transilvania* University of Braşov.
[2] Elettronika Research S.R.L. Braşov, Romania.
[3] Dept. Research & Development, Elettronika Italy.

presentation in order to preserve the illusion of motion. However, delays introduced by coding, multiplexing and transmission can cause a variable amount of delay for video packets arriving at the decoder. This delay wrecks havoc in the decoder process mandating buffers in the decoder [2].

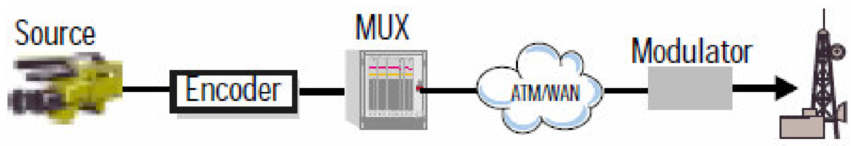

Fig. 1. *The transmission network from the studio to the transmitter*

The MPEG-2 standard provides an additional mechanism to ensure video frames can be decoded and delivered to the viewer with a consistent rate of display - that mechanism is called the Program Clock Reference (PCR).

PCR is fundamental for the timing recovery mechanism for MPEG-2 transport streams. PCR values are embedded into the adaptation field within the transport packets of defined packet identifiers (PID). PCR consists of two parts totalling 42 bits. The PCR values increment with a standard clock rate of 27 MHz, PCR values roll over roughly once a day.

As PCR is used by the Integrated Receiver Decoder (IRD) to derive the clock reference, any jitter or drift is the PCR clock can have damaging effect on the IRD's performance. The irregularities in the PCR can be broadly classified into jitter and offset. Jitter in the PCR is mainly attributed to two sources: systematic jitter (or PCR accuracy error, PCR_AC), and network jitter. Systematic jitter and network jitter are combined to get overall jitter (PCR_OJ).

The Transport Stream is a stream definition which is tailored for communicating or storing one or more programs of coded data according to ITU-T Rec. H.262|ISO/IEC 13818-2 and ISO/IEC 13818-3 and other data in environments in which significant errors may occur. Such errors may be manifested as bit value errors or loss of packets.

TS may be either fixed or variable rate. In either case the constituent elementary streams may either be fixed or variable rate. The syntax and semantic constrains on the stream are identical in each of these cases. The TS rate is defined by the values and locations of PCR fields, which generally are separate PCR fields for each program [4].

Figure 2 shows the MPEG-2 TS's management to maintain synchronization between the sender, which encodes the stream, and the receiver, which decodes it. As the elementary streams carrying video and audio content are packaged, their target Decoding Time Stamp (DTS) and Presentation Time Stamp (PTS) are determined based on the current sender clock and inserted into the packet header. For video streams, the access unit is a frame, and both DTS and PTS are given only for the first bit after the picture header of every frame, which are later used by the decoder to control the speed at which it starts to do decoding and presentation. For example, if at time 5.0 s an encoded frame comes to the multiplexing stage, and the encoder believes that the decoder should begin to decode that frame 0.5 s after it receives it and output the decoded frame 0.1 s thereafter, then the DTS should be set to 5.5 s and the PTS to 5.6 s. After that, as these entire packaged elementary stream packets are further multiplexed together, the final stream is time-stamped with Program Clock Reference (PCR), which is given by periodically sampling the sender clock.

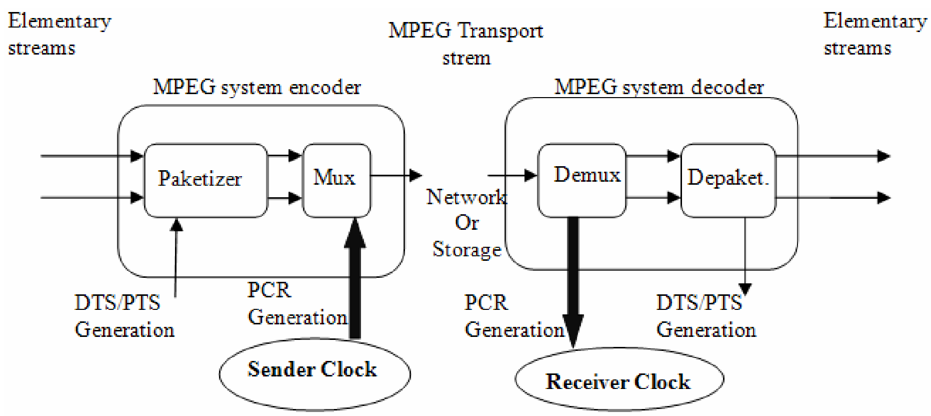

Fig. 2. *Generic coding of moving pictures and associated audio systems*

This resulted transport layer stream is then sent over the network to the receiver, or stored in storage devices for the decoder to read in the future. As long as the delay of the whole stream experiences remains constant from the receiver's point of view, the receiver should be able to reconstruct the sender's clock that has been used when the stream was encoded. The accuracy and stability of this recovered clock is very important, since the decoder will try to match the PTS and DTS against this clock to guide its decoding and displaying activities.

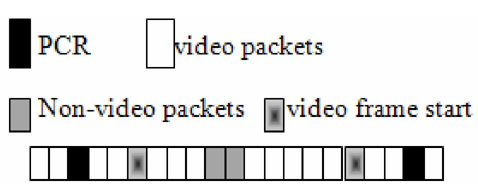

Fig. 3. *Generic coding of moving pictures and associated audio systems*

Knowing the general idea in timing, we introduce now an explanation about how the Transport Layer syntax works, as shown in the Figure 3. All sub streams (video, audio, data and timestamps) are segmented into small packets of constant size (188 bytes), and the Packet ID (PID) field in the 4-byte header of a packet tells which sub stream that packet belongs to. The PCR packets occur at constant intervals, and they form a running timeline along which all other packets are positioned at the right time point. Data packets arrive and they are read into the decoder buffer at constant rate; this rate can be calculated by dividing the number of bits between any 2 consecutive PCR packets by the time difference between their time stamps. In other words, if the number of packets between any 2 PCR packets remains constant, then the difference between their time stamps should also be constant. Considering an ideal state, packets are read into the decoder at the constant bit rate, and whenever a new PCR packet arrives, its time stamp should match exactly with the receiver clock, which confirms the decoder that so far it has successfully re-constructed the same clock as the encoder. However, since PCR packets may have experienced jitter in network transmission or storage device accessing before they arrive at the receiver, we cannot simply set the receiver's local clock to be the same as the time stamp carried by the next incoming PCR no matter when it comes. To smooth out the jitter and maintain a stable clock with a limited buffer size at the receiver, generally the receiver will resort to some smoothing technique like the Phase-Locked-Loop (PLL) to generate a stable clock from the jittered PCR packets. PLL is a feedback loop that uses an external signal (the incoming PCR packets in our case) to tune a local signal source (generated by a local oscillator) to generate a relatively more stable result signal (the receiver's reconstructed local clock). As long as the timing relation between PCR packets is correct, the jitter can be smoothed out with PLL [5].

After the brief introduction on the usage and importance of the PCR packets, now we are ready to discuss how the filtering operation may affect their validity and accuracy. First, even for the same type of filtering operation, e.g., low pass filtering, for different frames, the time required to do the calculation and processing can be quite different. Since the filtering is transparent to the decoder, it seems to the decoder that the jitter the stream has experienced is larger. This is not a big problem, since through longer buffering at the filter and the receiver and the use of jitter smoothing mechanisms, this stronger jitter will not greatly affect the decoding process. The second problem, however, is more intractable. As it's mentioned above, the packets for any access unit should be positioned within the stream and so arrive at the receiver at its supposed time point for the decoder to schedule where and how long to buffer it before decoding it. However, after the filter operations normally, a video frame becomes smaller or larger. It takes less or more packets to carry, and so its following frames are dragged earlier or pushed later along the timeline. In such circumstances, if we keep both the time stamp and the spacing of the PCR packets unchanged, then the receiver's clock can still be correctly reconstructed, but the arriving time of each frame will be skewed along the timeline. For example, if the stream is low pass filtered, and then every frame becomes shorter, the following frames are dragged forward to pack up the vacancy spared out. From the decoder's point of view, more and more future frames begin to come earlier and earlier, and to buffer them until their stamped time for decoding, the buffer will be overflowed in the long run no matter how large it is. The fundamental problem is that after the filtering, the actual bit rate becomes lower or higher, but the data is still read in by the decoder at the original rate. So if the new rate is lower, more and more future data is read in by the decoder, causing the decoder buffer to overflow eventually; on the other hand, if the new rate is higher, then at some point in the future, the data will be read in after its decoding time has already passed.

3. Our Solution

Because the length of the packets can be 188/204Bytes, our solution is designed to work with both types of packets.

The first step was to set-up a real-time packet length monitoring and to verify the nature of the packets, in order to set the system to work on 188/204Bytes.

Then, based on the PID of the packet, we checked if the header of the packet contains PCR values; if so, then those values have been removed from the packet.

For testing we used an external source for the PID, consisting of 3 jumpers. We also tested a software approach. In the final project we used a block that will calculate the PID.

Fig. 4. *Altera Cyclone III + Bitec HSMC BVDC & DekTec USB-2 ASI/SDI*

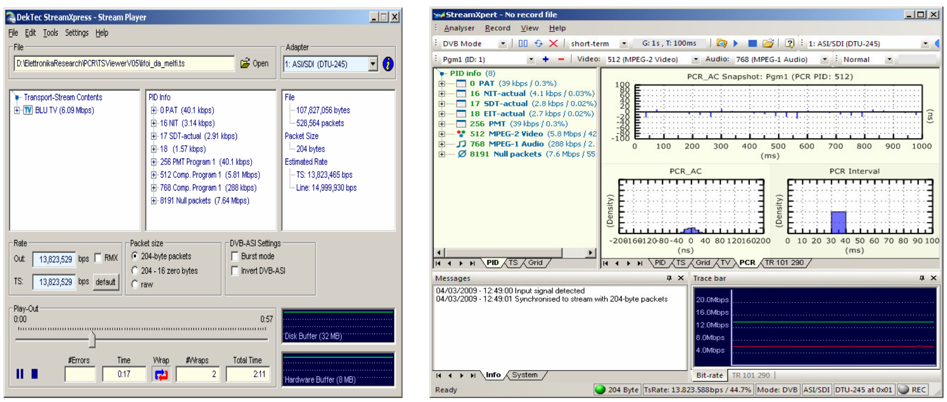

Fig. 5. *StreamXpress* (*right*)*; StreamXpert* (*left*)

Because the TS packets came in bursts, a dual-port RAM has been used to smooth the data throughput rate. The next step was to recalculate the new PCR values and to insert them into the packets.

To test our solution, the Altera Cyclone III starter kit, Bitec HSMC BVDC and a DekTec USB2 ASI/SDI player and analyzer have been used (Figure 4). TSViewer, designed by Elettronika - Italy, is the software used to view the data inside the transport stream packets.

Figure 5 shows screenshots taken from the DekTec applications which generate an ASI Transport Stream continuously feeding our system and analyze in real-time the processed Transport Stream generated by our restamping system.

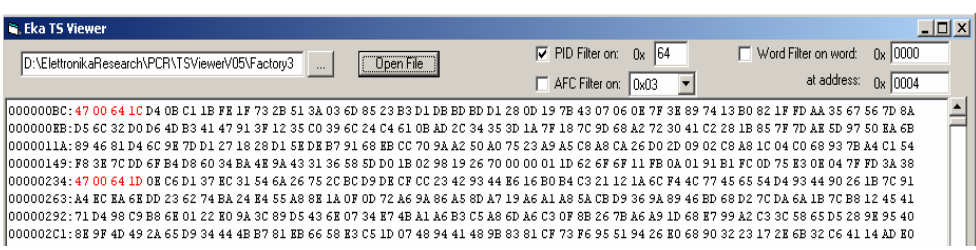

Fig. 6. *TSViewer*

The DekTec applications (named *StreamXpress* and *StreamXpert*) can operate with 188 and 204 packet lengths and run in combination with a USB module capable of bidirectional real-time conversion between the USB and DVB-ASI interfaces.

Figure 6 shows a screenshot of TSViewer. In this program we see the data from a transport stream. The TS is saved in a file which is then loaded in the program. The program can also select all the packets with a specified PID.

4. Conclusions and Future Work

The intended goal of recalculating the PCR values was a success. The test works perfectly on the DekTec decoder but on the Rohde&Schwarz DVMD decoder the video and audio are slightly discontinuous, which suggests that the PTS-DTS timestamps are not synchronized. For a better synchronization, further work on the PTS-DTS timestamps has to be carried out.

References

1. Allen, J.R.: *System and Method for Reducing Jitter in a Packet-Based Transmission Network.* U.S. Patent no. 5652627.
2. Navalpotro, J.: *Short Note on PCR Jitter.* Available at: www.abacantodigital.com/docs/ART-V_Short_Note_PCR-Jitter.pdf. Accessed: 15-04-2009.
3. Nemiroff, R., Liu, V., Wu, S.: *Regeneration of Program Clock Reference Data for MPEG Transport Stream.* U.S. Patent no. 6724825 B1.
4. Pandya, H.: *Application Note - AN101 Considerations for Measurement of a PCR Jitter.* Pixelmetrix Corporation.
5. ISO/IEC JTC1/SC29/WG11 N0801.