Module
10
MULTIMEDIA SYNCHRONIZATION

Lesson 37

Playback continuity

Instructional Objectives

At the end of this lesson, the students should be able to:

1. Explain why re-sequencing is necessary for display buffers.
2. State the essential condition of continuous playback in real decoders.
3. Explain how buffer underflow may result.
4. State how buffer underflow can be prevented.
5. Explain how buffer overflow may result.
6. State how buffer overflow may be prevented.
7. Explain how synchronization can be carried out using a master stream.

37.0 Introduction

In lesson-35 and 36, we have studied the architecture for packs and packets and the time-stampings necessary for inter-media synchronization. In this lesson, we are going to address another major issue for media storage and playback applications and that is intra-media continuity. To ensure that the media units are present before their decoding time stamps, it is necessary to buffer the incoming video bit stream at the decoder. In this lesson, we are going to derive the conditions for buffer overflow and underflow to ensure media playback continuity. Also, in a distributed environment, synchronization using one master stream will be discussed.

37.1 Requirement for re-sequencing

MPEG is mainly targeted towards stored media applications. In such applications, the video (and its audio) is captured, encoded and stored in a Digital Storage Medium (DSM) at the time of content creation, and later retrieved on demand for playback at which time, the DSM transmits the MPEG ISO 11172 pack stream to display sites.

Suppose, the original MPEG stream is:

I, B_2, B_3, B_4, P_5, B_6 B_7, B_8, P_9, B_{10}, B_{11}, B_{12}, I_{13} -------------------------------------- (1)

The corresponding encoded ISO 11172 stream would be:

I_1 P_5 B_2 B_3 B_4 P_9 B_6 B_7 B_8 I_{13} B_{10} B_{11} B_{12} …………………………………….. (2)

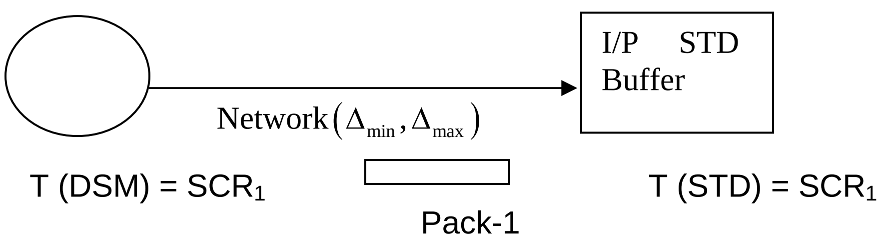
During decoding, the frame P_5 is decoded before B_2, B_3 and B_4; P_9 is decoded before B_6, B_7 and B_8 and so on.

The re-sequencing is done in the display reorder buffers. For I – picture or P-pictures, the decoding time and the presentation times differ by integral number of picture periods.

37.2 Conditions for continuous playback

For continuous playback of media streams in a real decoders, it is essential that media units be available at the decoder prior to their respective decoding times, and that the rate of consumption of packs at the System Target Decoder (STD) match the rate of transmission by the DSM. When the DSM transmits its first pack, DSM sets its clock to SCR_1. When the first pack reaches STD, STD also sets its clock to SCR_1

37.3 Buffer underflow and its conditions



$$T_{DSM} - T_{STD} = \Delta \in \left(\Delta_{\max}, \Delta_{\min}\right)$$

Figure-37.1 Lag between the DSM and the STD clocks.

If at any time, the STD buffer suffers an underflow, discontinuity in playback may occur.
The worst case scenario for buffer underflow is when an earlier pack arrives after min. delay and a later pack arrives after max delay.

Let us suppose that the first pack arrives after the minimum delay. Hence,

$T_{DSM} – T_{STD} = \Delta_{\min}$

Suppose, that a later pack p suffers max delay.

DSM's clock for pack p shows *SCR (p)*. At the same time, STD clock shows *SCR (p)* - $\Delta_{\min}$.

Adding a transmission delay of Δ_{max}, the arrival time of pack p is given by *SCR(p)* + $\Delta_{max} - \Delta_{min}$

To prevent underflow, the STD delays setting of its clock to the first pack's SCR by $\Delta_{max} - \Delta_{min}$ after the arrival of the first pack. With this modification, the problem of underflow gets solved. However, it compounds another problem, that of buffer overflow.

37.4 Buffer overflow and its conditions

Buffer overflow occurs if the first pack suffers maximum delay, and a later pack p suffers a minimum delay.

When the first pack arrives at the STD, and STD clock's effective time is
Arrival time of first pack = *SCR(1)* $- (\Delta_{max} - \Delta_{min})$.

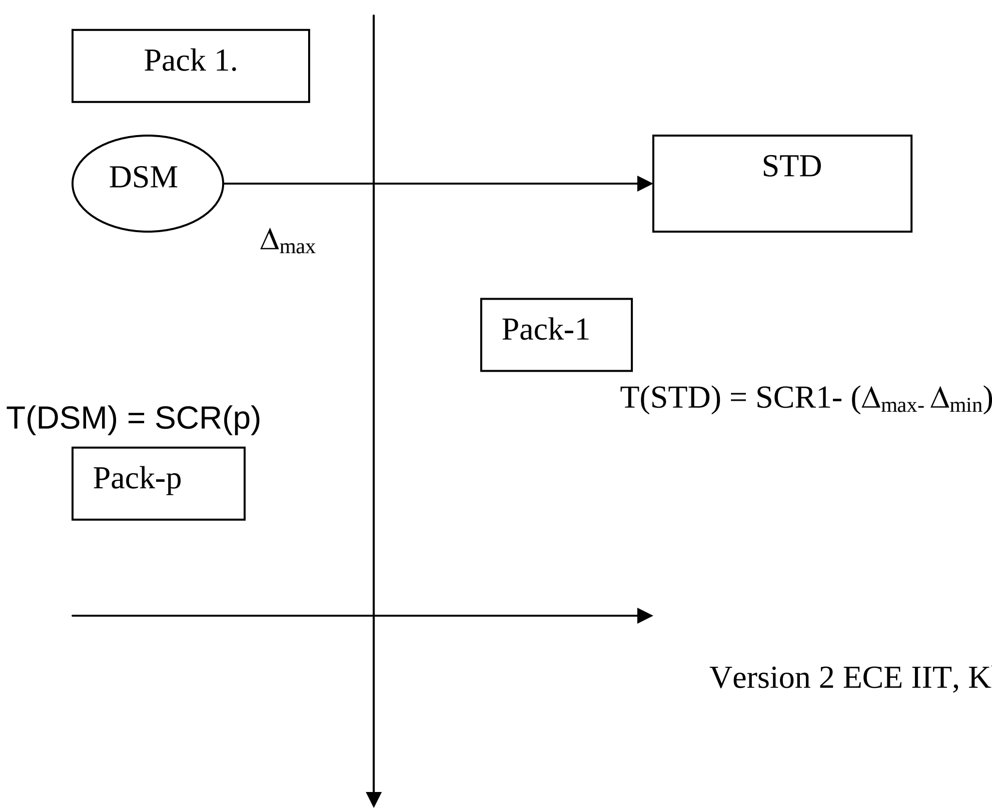
The later pack *p* suffers only Δ_{min} comparison with the maximum delay that the first pack suffers. Hence, its arrival time is

Arrival time of pack p

$= SCR(p) - (\Delta_{max} - \Delta_{min}) - (\Delta_{max} - \Delta_{min}).$

$= SCR(p) - 2 \times (\Delta_{max} - \Delta_{min})$

Early arrival of a pack, as given above, requires additional buffering , equal to the max. number of additional bytes that might arrive in an interval of $2 \times (\Delta_{max} - \Delta_{min})$



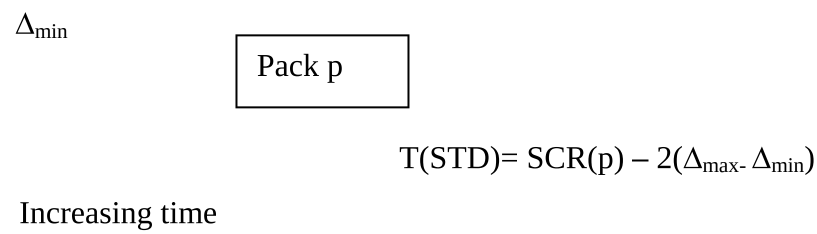


Figure-37.2 Delayed setting of STD clock to prevent playback discontinuity due to buffer overflow.

37.5 Synchronization at the packet layer

Synchronization in MPEG is handled at the packet layer, with the SCR, PTS and DTS fields serving as instruments. The decoders parse the presentation units to extract the PTS and the DTS fields. It must be remembered that the PTS and the DTS fields are not necessarily encoded for each video frame or audio presentation unit, but are only required to occur with intervals not exceeding 0.7 seconds for periodic updating of the decoders' clocks.

37.6 Synchronization using a master stream

All of the media streams being decoded and displayed must have exactly one independent master. Each of the individual media display units must solve the timing of their presentation to the master stream.

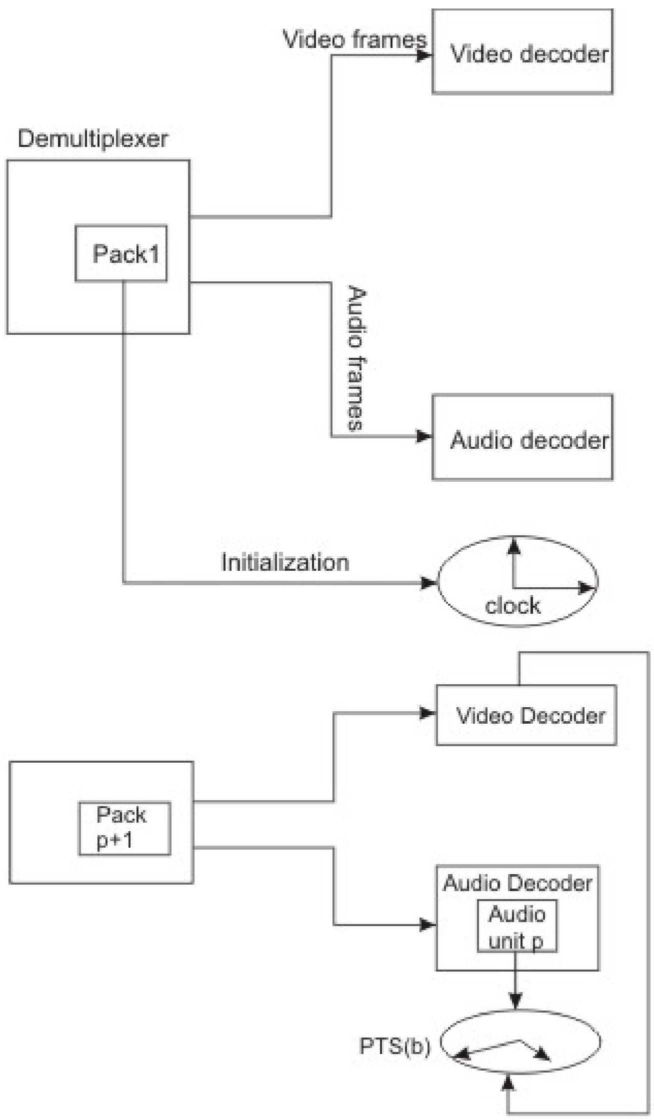


Fig 37.3 Synchronization using a master stream

The other decoders simply use the audio-controlled clock to determine the correct time to present their decoded data, at the times w~~hen their PTS field~~s are equal to the current values of the clock. Thus, video units are presented when the STD clock reaches their respective PTS values, but the clock is never derived from a video PTS value. Thus, if the video decoder lags for any reason, it may be forced to skip presentation of some video frames; on the other hand, if the video decoder leads, it may be forced to pause; but the audio is never skipped or paused.