

# UBR Congestion controlled Video Transmission over ATM

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## ABSTRACT

The ATM unspecified bit rate (UBR) class of service –some times referred to as best effort service- was originally thought of as a transport class for bursty traffic requiring no bandwidth, cell loss, and delay guarantees to get the data from source to destination. Ideally this would suit IP traffic for transporting data. However, this article will try to discuss various aspects of using this same service class to transport video (with all the reservations and requirements necessary to do that) through controlling the network congestion and the prospects and possibilities that same service could be applied in the existing Saudi telecom (STC) ATM network.

## 1. INTRODUCTION

ATM is known to be a platform for all services, Data, voice and Video, however, depending on the nature of each service a specific set of parameters named QOS parameters were introduced to meet the requirements of each service based on the sensitivity to delay, cell loss and cell delay variation (jitter). Data is tolerant to these parameters, unlike the real time applications, voice and video, which are normally transported via constant bit rate (CBR) and to some extent variable bit rate (VBR) class of services, since these classes provide guarantees and control over traffic parameters.

Cell loss, delay and jitter are caused by network congestion. We will look into how we can, through a better utilization of network resources and control over congestion, use UBR class of service for video transport.

## 2. MPEG-2 and AAL-5

MPEG-2 format was adopted by the ISO and later ITU-T as a standard for video compression, other formats used for video

conference like H.261, CIF and SIF proved to be unsatisfactory for moving picture (movies) compression. MPEG-2 breaks down a single frame/field of picture into blocks, macro blocks, and slices, while classifies pictures as intra frame (I), Predictive (P) and Bi-directional (B) when encoded, employing both spatial and temporal techniques to compress the video streams using algorithms like Discrete Cosine Transform (DCT), prediction and variable length coding (VLC).

The output of the process of video stream compression and digitization, prior to being presented to the network platform (ATM in our case), results in a variable length packetized elementary stream (PES) which constitutes transport Streams (TS), transport streams are fixed length 188 bytes long.

ATM Adaptation Layer 5 (AAL5) is one of two adaptation layers, the other one being AAL1, that are used to interface between video service layer and the ATM layer.

AAL5 is the best suited layer to be used for UBR class of service for reasons that will be mentioned later.

Video transport over ATM AAL-5 has been standardized by ITU-T standard ITU-T J.82, it has also been treated by ATM Forum Implementation Agreement (IA).

Video Transport Streams are mapped on the AAL5 payload and through segmentation and reassembly (SAR) are mapped to ATM 53byte cells..

ATM FORUM IA specifies the following rules for TS=n=2 mapping on AAL5:

1. AAL5 with null service specific convergence shall be used.
2. AAL5 PDU shall contain at least TS packets of the single program transport stream.
3. An AAL5 PDU shall contain only one MPEG-2 transport packet, if that MPEG-2 transport packet is the last TS of the single program stream.

The mapping of TS packets in AAL-5 PDU is shown in Fig 1.

## AAL5 disadvantages

Common part sub layer does not support forward error correction (FEC), only error detection and discard the errored SDU.

Encapsulation process affects jitter which affects the clock recovery.

AAL-5 is divided in two major parts:

1. Convergence Sublayer (CS)
  - a. service specific convergence sub layer (SSCS) that can be null
  - b. Common part convergence sub layer (CPCS)

2. Segmentation and Reassembly (SAR)

AAL-5 cells contain fields used for congestion notification . ATM congestion is marked in the AAL-5 header.

Fig. 2 depicts AAL5 sublayer structure

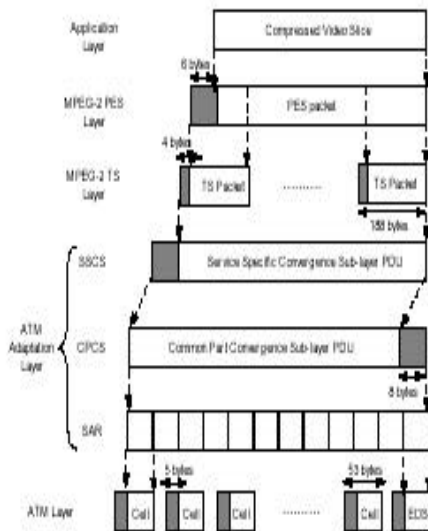


Figure 1 - Mapping of MPEG Video Slices on AAL5

This mapping process is called PCR-unaware process, which ignores whether the TS contains a program clock reference (PCR) time stamp , unlike the PCR –aware process which has to take into consideration the mapping process ..etc.

PCR field in the transport stream (TS) is one of the crucial fields when dealing with real time service transfer. It indicates the program clock reference transported across the network in order to synchronize the source with the destination. Any deviation in this PCR will cause the Audio or Video application not to function properly.

## AAL-5 Advantages

The most commonly used AAL to carry best efforts traffic through the ATM, PDU of which is 48 bytes, unlike the AAL-1 where PDU is 47 bytes.

Usually null CS (see Fig.2) eliminates HW support and complexity is moved to service level .

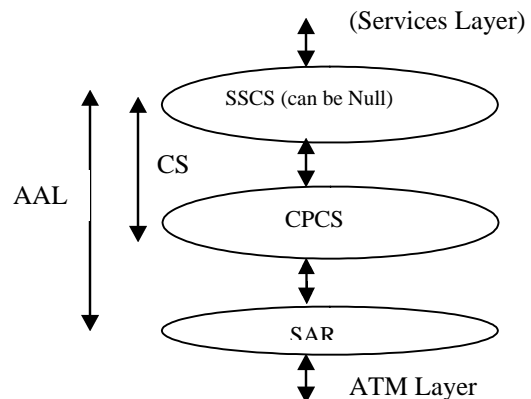


Fig.2 AAL5 Sublayer Structure

## 3. Congestion Control

UBR class of service does not offer any service guarantees , the network makes no guarantees at all on the cell loss, delay, or delay variation (jitter). To compensate for this We either implement pre-standard congestion control mechanism or support adequate buffering to minimize the probability of cell loss.

AAL-5 was intended for connection oriented data transfer, but has evolved to address MPEG-2 compressed video also.

AAL-5 is extremely efficient protocol with very little over head (48bytes payload) . on the other hand, it has no function that can cope with timing relationship (it is, therefore necessary to include PCR within the TS, if We were to take care of clock synchronization) , mis-sequenced ATM cells, nor can it carry more than one connection per PVC.

### 3.1 Service Classification and Prioritization

In order to confront the congestion issue, We have to prioritize and classify our service. Based on this, classes like CBR and VBR as well as UBR were introduced, however, since We have already decided to use the UBR class of service, We need to limit ourselves to this class, but at the same time We want to look into how We can use ways and means that would make this class capable of carrying real time services like video.

As it was mentioned earlier, MPEG-2 defines three types of encoded picture frames, Intra (I), Predictive (P) and Bi-directional (B). These three types are of different importance, based on the nature of each one, the I type is encoded independently of any other picture, it is therefore, extremely important to give this type the highest possible priority, since the loss of this frame will seriously affect the quality of picture and will also affect the following picture frames which will be encoded based on it. P frame comes next in importance because it relies, beside the intra coding, on the previously encoded frame (the I frame), hence it should receive the next priority. The least important frame is the B frame, which is encoded according to the past and future frames (I, P and B) .

It is very essential that our system treats these three types differently when considering any discard mechanism in case congestion forces us to apply any.

Similarly Data Partitioning is a mechanism to split the video streams according to importance and passes the ones with higher priority on channels with high performance and the less important ones on channels with less performance. This would mean that for spatially compressed video, I picture frames would be transported on the high performance channels. For temporally compressed elements 1<sup>st</sup> Discrete Cosine Transform (DCT) coefficient + header,...etc. would be transported on the high

performance channel, while coefficients from 2 to 64 would be transported on a lower priority channel with less performance.

### 3.2 Available algorithms to deal with congestion

A number of algorithms were devised to deal with the congestion consequences, in our case deterioration of picture quality.

Mainly systems create buffers to avoid cell loss, however buffers have major negative impacts on real time services like video because of delay.

Buffers normally result in queues, queues are an acceptable alternative to discarding packets. Some algorithms are used to drain the queue from accumulated traffic.

For real time services minimum time is required for the packet to stay in the queue, Algorithms normally take care of prioritizing the traffic based on the nature of the service.

Here are some examples of those algorithms used.

#### 3.2.1 Random Early Detection (RED)

RED is applied to the output buffer and it determines when packets should be discarded .

It randomly and intentionally drops packets once the traffic reaches one or more configured thresholds in situations where congestion forces any new packet to be discarded

WRED (weighted RED) uses RED algorithm, but instead of discarding any traffic without looking at the content, it discards it after classifying the result

Other algorithms include partial packet discard (PPD), early packet discard (EPD) and late packet discard (LPD).

LPD defines the threshold between PPD threshold and the limit of a queue permitting end of message (EOM) cells to be transmitted so subsequent frames are not corrupted

#### 3.2.2 Resource Reservation Protocol (RSVP)

RSVP is a protocol through which an application can reserve resources from source to destination. An exchange of messages initialized by the source with a PATH message and a response from the receiver with a RESV message installs a reverse routing state in each router along the path and provides receiver with information about the characteristics of the sender traffic and end-to-end path

#### 3.2.3 FIFO

First in First out (FIFO) is a single queue system (for the difference from other algorithms which maintain several queues at a time). Packets arriving at this queue are treated

as they arrive i.e. the first in will be the first out. No preference is given to any packet.

The simple technique works with no priority system, it is just an alternative for a NO-QUEUE system which might cause loss of cells in case a slightest congestion shows up.

This system is good in systems where the output bandwidth exceeds normal traffic patterns, it also protects from short duration bursts as well as being good for systems with sporadic and small congestion.

#### 3.2.4 Selective discard.

This system classifies the traffic into two categories so that in case of congestion it decides which packet to drop, decision is made based on tagging each packet with a drop preference (DP) tag. The system monitors the bandwidth on a per flow basis, frames determined to be DP by the input policing function will have this status inserted in the packet header, which will be detected by other DP aware devices in the network. This status will be propagated throughout the network.

## 4. STC Network

Saudi Telecom Company (STC) owns a vast state of the art ATM network with a versatile vendors Equipments covering almost all of the Kingdom, this network is used, so far, for Data Transfer Only, although STC is contemplating to build a Next Generation Network (NGN) on top of it. This would mean inclusion of intelligent and real time services .

With regard to the congestion control techniques discussed in this article, these equipments provide all of the mechanisms mentioned in Para 3. to control the congestion.

Table 1. shows the average occupancy of the STC ATM backbone , having the maximum not exceeding 12% of the total available bandwidth. This would enable immediate application of video services because of the low usage and under utilization of bandwidth and absence of congestion.

Table 1. STC ATM backbone bandwidth utilization.

Link	Switch #1	Link Capacity	Switch #2	NOV1-7, 02 utilization
Core1-Core2	Core 1	STM-16	Core 2	4.75%
Core1-Core3	Core1	STM-16	Core3	5%
Core1-Core4	Core1	STM-16	Core4	12%
Core2-Core4	Core2	STM-16	Core4	3%
Core2-Core3	Core2	STM-16	Core3	4.75%
Core3-Core4	Core3	STM-16	Core4	6%

## 5. Conclusion

It is clear that UBR class of service can be used not only for data transport but also for real time services like Video and Audio.

Saudi Telecom Co (STC) is in a position, provided that systems are configured to meet the requirements, to use the existing ATM platforms to generate video services.

The ATM technology installed in Saudi Telecom Co (STC) Network provides all of the congestion control algorithms mentioned in this article. This will enable it to easily configure and activate them in case a decision is made to provide video services over the existing ATM backbone.

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Eltayeb omer eltayeb, holds an Msc degree from the university of Manchester institute of science and technology (UMIST), U.K In electrical engineering , joined STC Oct 1997. his major field of research interest is transport of video over digital networks .