CCITT SGXV
Working Party XV/1
Experts Group for ATM Video Coding

Document AVC-68
July 27, 1991

SOURCE : STUDY GROUP XVIII
TITLE : LIAISON STATEMENTS AND IVS BASELINE DOCUMENT
Purpose: Report

The following documents from Study Group XVIII meeting in June 1991 are contained for consideration of the Experts Group meeting in Santa Clara (14-23 August):

END

Questions: 2,13,22/XVIII

SOURCE: WP XVIII/8, Geneva meeting, 11-28 June 1991

TITLE: LIAISON STATEMENT TO SG XV ATM VIDEO CODING EXPERTS GROUP, CMTT/3 for action and CMTT/4 for information, CCIR IWP 11/9 AND ISO/IEC MPEG ON IVS STUDIES

SG XVIII reviewed liaison documents received from SG XV ATM video coding experts group and CMTT at the June 1991 meeting, and recognized that the initiative of SG XVIII in coordinating IVS studies through the baseline document, proposed by SG XVIII at the Matsuyama meeting, has been welcome.

The proposal of SG XV to establish a framework of Recs for IVS to provide a division of work among the relevant groups has merit and will be further addressed at our next meeting.

The status of cell loss priority discussions is attached as Annex 1. Furthermore, agreement on UPC issues is summarized in Annex 2. These Annexes will provide answers to SG XV.

The IVS baseline document has been updated to incorporate new text as proposed by SG XV and CMTT, and to reflect the agreements reached so far in SG XVIII.

The updated revision of IVS baseline document appears in Annex 3.

ANNEX 1

ANNEX 1

(to the LS)

~~APPENDIX 2 to ANNEX B~~

Status of discussions on CLP issues

The following are questions and answers which have been identified and obtained from services viewpoints by SWP 8-3.

Q.1) There are two ways of using CLP capabilities of B-ISDN, i.e. cell by cell basis and per connection basis. In the latter case, will a CLP be defined per VPC basis as well as per VCC basis ?

A.1) No constraint from services viewpoints. This should be studied mainly by ATM and Resource Management aspects.

Q.2) In the case of CLP bit capability, will a single level of CLR (Cell Loss Ratio), i.e. high priority level, be defined, or will two levels of CLR be defined respectively ?

A.2) CLR for CLP=0 (high priority) should be defined, and assured by the network if a cell traffic does not exceed the negotiated values. For CLP=1 traffic, the following two options exist;

1) defined and assured by the network;
2) not defined and no assurance provided.

In the case of option 2), a given CLR may be maintained by the network engineering.

Q.3) Will the CLP bit, set by a user as high priority, be changed by the network, e.g. for violation tagging in case that a cell traffic exceeds the negotiated values ?

A.3) Since CLR of CLP=0 is assured by the network, there may be no impacts on services and for users, whether the network will override the bit or not.

Q.4) Will the specific value of CLR be explicitly declared by a user, or will the CLR indication be implicitly associated with specific service requests, e.g.

- a standardized service will by definition include the specification of all relevant QOS values, or
- a standardized QOS class will by definition include the specification of all relevant QOS values ?

A.4) Under study.

Q.5) Will the network provide for several CLR, or will the network accommodate CLR requests from users within a very limited number of CLR ? And how many CLR will be required ?

A.5) Under study.

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ANNEX 2

(to the LS)

Status of agreements on UPC issues

ANNEX 2

~~Liaison Statement to SG XV~~

SG XVIII-8 agreed on the definition of one traffic parameter : the Peak Cell Rate. The exact definition is as follows :

Location

At the ATM layer SAP for a VCC.

Basic event

Request to send an ATM_SDU (48-octet information field).

Definition

The peak cell rate R_P is the inverse of the minimum inter-arrival time T_0 of the basic event above.

Usage Parameter Control will be based on Peak Cell Rate as defined above. At this moment, it is not expected that a specific UPC mechanism be standardized. A maximum Cell Delay Variation will be allocated to the Customer Equipment (CEQ) between the ATM SAP and the interface at the T Reference Point.

UPC enforcing actions may be dropping cells or optionally tagging high priority cells by changing the Cell Loss Priority bit from 0 to 1. This is agreed because cells subject to tagging are in excess to the negotiated traffic contract. Cells complying with this contract are neither subject to discard nor to tagging. There will be negotiation for both priority flows. This is because SG XVIII-8 came to the agreement that both high priority and low priority flows will be subject to UPC and enforcement. It appears therefore possible, but left to network operators' decision, to provide for a given QOS on each flow.

~~Liaison Statement to Study Group I~~

At this moment, SG XVIII-8 agreed on the definition of one traffic parameter : the Peak Cell Rate. The exact definition is as follows :

Location

At the ATM layer SAP for a VCC.

Basic event

Request to send an ATM_SDU (48-octet information field).

Definition

The peak cell rate R_P is the inverse of the minimum inter-arrival time T_0 of the basic event above.

There is no traffic parameter agreed at the moment for Variable Bit Rate sources beyond Peak Cell Rate. Additional parameters are likely to be defined in the future, according to the following requirements :

Any traffic parameter involved in a traffic descriptor should :

- Be understandable for the user or his terminal ; conformance should be possible ;
- Participate in resource allocation schemes meeting network performance requirements ;
- Be enforceable by the UPC and NPC.

Peak rate is a mandatory parameter to be declared. Additional standardized parameters beyond peak rate should provide for a significant improvement of network utilization.

It is premature at this moment to say that "bursty" might be a relevant parameter.

ANNEX 3

(to the LS)

will reproduce the baseline document on NS

CCITT

Temporary Document 30 (PLEN)

Appendix 4 to Annex B of

WPXVIII/8 Report

R70

STUDY GROUP XVIII

Geneva 11 - 28 June 1991

ANNEX 3

SOURCE: SWP 8-3 (Editor of IVS Baseline)

TITLE: IVS BASELINE DOCUMENT

The IVS Baseline Document has been modified to incorporate new text as proposed by CMTT/3 and CCITT SGXV and to also reflect the enchancements to the document discussed n the drafting group.

The text has been updated to reflect the results of SWPs of WPXVIII/8 at this June 1991 Geneva meeting. Further updating may be required to reflect changes arising from the activities of other relevant working parties of SGXVIII.

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AVC-68

INTEGRATED VIDEO SERVICES (IVS) BASELINE DOCUMENT

June 1991

LIST OF CONTENTS

1. General

This document is initiated by CCITT SGXVIII to gather information related to Integrated Video Services (IVS) support on B-ISDN. The document contains aspects related to the work of several other groups to provide a consolidated overview of Integrated Video Service issues and the areas that need to be addressed and specified to become both technically and commercially viable. The prime purpose is to provide a common basis for the ongoing study of Integrated Video Services by SGXVIII and other groups.

It is the intention of SGXVIII to maintain and update this baseline document until such time as the relevant information is transferred to Recommendations for which different groups are responsible.

To keep this document updated and complete, all groups involved are invited to study it carefully and provide appropriate input.

Major areas which require further development are contained in separate annexes to this document. The scope of the document should be expanded as necessary to meet the requirements of the different groups involved.

2. Objectives

Video and image services represent an increasingly important form of communications. With the establishment of powerful and extensive broadband network facilities, customer interest in video and image services is expected to lead to growing demands for greater service variety and higher quality.

The B-ISDN will form the foundation of public networks capable of the integrated support of voice, data and video applications. In addition to a consistent broadband transmission and switching fabric, the B-ISDN will provide common interfaces for the support of all customer services and supplementary services (e.g. picture within picture), ensuring connectivity and a competitive multi-vendor equipment environment.

Integrated network support and delivery of the various service classes can provide advantages in terms of efficient handling of service types within the network and a consistent application environment within the customer's premises.g. common display, control, etc.

Service integration can occur at many levels within networks and customer equipment. The emergence of B-ISDN standards and network technology provides an opportunity to rationalise video service support by developing a framework for the integration of interactive and distribution video service delivery. Video service integration will provide a means of maximising the rate and extent of video service development and application within both the residential and business market sectors. The objective is therefore to develop a communications environment which can provide effective and flexible video service support, across all service types, together with positive incentives for new service development and deployment.

From a customer viewpoint, the integrated support of video services would offer lowered costs and enhanced flexibility.

To achieve this flexibility and provide integrated service support requires alignment and consistency between related service standards.

Video service integration benefits will be maximised under conditions offering commonality of User-Network Interface, signalling and control formats, coding techniques and display devices across a range of service types.

Draft Recommendation I.211 "B-ISDN Service Aspects" provides a classification of services to be supported by the B-ISDN, and basic considerations of the network capabilities required by the B-ISDN. For video service applications, it identifies the key objective of maximum integration through common coding and integration of control and signalling systems, and also provides an overview of the related coding and service interworking issues.

The development of common coding schemes will help to fulfil the following objectives:

- economic provision of multiservice terminals and customer equipment;
- ease of adaptation of terminal equipment for different services;
- minimisation of interworking requirements;
- minimisation of transcoding requirements within the network.

To achieve these objectives it will be necessary that there be close cooperation and liaison between all the B-ISDN video services standardisation groups. It is the prime purpose of this baseline document to provide the vehicle for achieving this liaison and the required level of cooperation and commonality of direction.

3. Responsibilities

CCITT SGXVIII

Responsible for, amongst other tasks:

- Recommendations on all network aspects of the B-ISDN, including the network architecture, transport techniques, User-Network Interface, access and inter-exchange signalling and ATM Adaptation Layer specifications;
- identifying network wide impact on B-ISDN service support, including the coordination across ITU bodies necessary to maximise commonality between communicative and distribution video services;
- establishing the framework for video service support in the B-ISDN;
- general aspects of quality of service and network performance in digital networks including ISDNs;
- providing coordination across different groups on Integrated Video Services in B-ISDN.

The following areas of responsibility are SGXVIII's understanding of the terms of reference for the other groups interested in video services on B-ISDN.

CCITT SGXV

Responsible for, amongst other tasks:

- Recommendations on video coding algorithms necessary to support a range of different quality communicative video services on the B-ISDN;
- Recommendations for transmission systems and equipment utilised in the B-ISDN;
- Recommendations for audiovisual system aspects;
- cooperatively assess compatibility between video coding algorithms used for the support of communicative and distribution video services with CMTT.

CCITT SGXI

Responsible for, amongst other tasks:

- Recommendations on Stage 2 and Stage 3 service descriptions for the Stage 1 service descriptions as provided by SGI.

CCITT SGVIII

Responsible for, amongst other tasks:

- Recommendations on coding for still image communication.

CCITT SGI

Responsible for, amongst other tasks:

- Recommendations on the service attributes including end-to-end service quality of all communicative services supported on the B-ISDN;
- Stage 1 service descriptions;
- cooperatively assess compatibility of end-to-end performance levels of television and communicative services on the B-ISDN with CCIR SG11.

CMTT

Responsible for, amongst other tasks:

- Recommendations on the bit rate reduction coding, packaging and transmission of television and sound programme signals in all portions of the telecommunications network; this includes contribution, primary distribution and secondary distribution signals;
- cooperatively assess compatibility between video coding algorithms used for the support of communicative and distributive video services with CCITT SGXV.

CCIR SG11

Responsible for, amongst other tasks:

- Recommendations defining and assessing the subjective and objective performance of digital television coding schemes proposed by the CMTT and supported on the B-ISDN proposed by CCITT SGXVIII;
- cooperatively assess compatibility of end-to-end performance levels of television and communicative service on the B-ISDN with CCITT SGI.

IEC/ISO

Responsible for, amongst other tasks:

- Through the Moving Picture Experts Group (MPEG), develop standards for storage and retrieval of moving images and sound for Digital Storage Media (DSM);
- development of standards for display devices.

4. Range of Services

CCITT Recommendation I.211, "B-ISDN Service Aspects", identifies two broad service categories;

interactive and distribution.

Full integration of the coding schemes to be adopted for all video services, including the following, should be pursued:

- distribution services including entertainment and information;
- conversational services including videotelephony and videoconferencing;
- messaging services including moving picture mail;
- retrieval services including film libraries and high resolution images.

Video service applications in B-ISDN fall across this entire range of broad service types and thus must be considered when developing a framework for B-ISDN video coding studies.

5. Evolution to Integrated Video Services in B-ISDN

It is clear that the development of B-ISDN Recommendations will follow a staged approach. Similarly it can be expected that the achievement of the objectives for IVS as described in Section 2 will also necessitate a staged approach. Assuming that there is agreement on the long term objectives of Section 2, a number of issues on the evolution to IVS emerge. It is the intent of Annex 1, 'Work Plan', to provide the means of identifying the stages in development of Recommendations relevant to achievement of the objectives of IVS.

Annex 1. Work Plan

The timeframes of this work plan indicate when Recommendations are expected. The contents should reflect the development status of B-ISDN network capability and the stages in development of Recommendations relevant to video coding for the B-ISDN.

1990-92

- Communicative video services support on B-ISDN (point-to-point);
- Initial services - videotelephony and videoconferencing (using limited point-to-multipoint connections),
  - lower rate retrieval;
  - point-to-point switched;
- Initial emphasis on CBR mode;
- Assessment of VBR vs CBR advantages/disadvantages;
- AAL Type 1 SAR and CS to be specified;
- AAL Type 2 SAR specified;
- Degree of compatibility with existing coding standards required;
- SGXVIII Recommendation on network aspects of Integrated Video Services on B-ISDN;
- Studies on layered coding for service interworking in IVS;
- Traffic control and resource management.

1992-94

- Lower rate (?) distribution services;
- Support of VBR mode emerging, with use of CLP;
- AAL Type 2 CS specified;
- Digital television distribution.
- Enhanced multimedia video services;
- Enhanced signalling and control capabilities;
- Recommendations relating to service interworking in IVS and B-ISDN
- Architecture aspects.
- Video coding for ATM
  - Communicative
  - Distribution

1994+

- Wide range of interactive and distribution services;
- Digital HDTV services - broadcast and non-broadcast applications

Integrated Video Services (IVS) Baseline Document

Annex 2. Network Aspects

A2.1. Information Flows

The nature of service information flows within a communications network influences the design and dimensioning of switches and transmission links and interfaces. Approaches to network resource management are also influenced by the characteristics of the service information flow. This issue is particularly relevant to the B-ISDN given the diverse range of video service types and qualities to be supported.

Video service information can be characterised in many ways, including:

- The direction of information flow: video services may be bidirectional, eg. videotelephony and videoconference, or essentially unidirectional, eg. video distribution services for business and entertainment.
- The symmetry of information flow: messaging, retrieval and distribution services are characterised by asymmetrical information flows.
- The origin of the source material: how video signals enter the network (eg. direct from camera, from storage media, via satellite or other delivery mechanisms) can also provide a means of characterising service information flows.

Telecommunications Services

The term service is used and understood in many different ways however. A recognised defination is given in CCITT Recommendation I.210. Two families of telecommunciation services have been identified :

- Bearer services
- Teleservices

Bearer services provide the network capabilities to transfer information between points of access to the network. The communication between the users will only function if the two parties, by prearrangement, have chosen compatible terminals and communication protocols. In terms of standardisation, bearer services are "interface services", i.e. if the interfaces to the network are functionally identical, the invoked bearer service can be used. Thus compatibility is ensured if the protocols used for layer 1, 2 and 3 comply with CCITT standards.

In contrast, a teleservice provides the full capacity for communication by means of terminal and network functions, and possibly also functions provided by dedicated centres. The teleservices are end-to-end services. In addition to standardised bearer functions, communication requires standardised functions within the terminals involved. In the case of teleservices, all protocols of the 7 layers of the OSI model have to conform to the relevant CCITT standards for communication to take place.

These two types of basic services may be supported by a range of supplementary services e.g. call diversion, call waiting, closed user group etc. Supplementary services are mainly offered by the network and they can only be used in conjunction with the basic services.

A2.2. Switch Functionality

The switching infrastructure of a Broadband ISDN may be required to support a variety of switched services. For example,

- Point-to-point switching e.g videotelephony
- Point-to-multipoint: bi-directional e.g. videoconferencing, multimedia conferencing
- Point-to-multipoint unidirectional e.g. broadcast distribution services, switched distribution services.
- Multipoint-to-point e.g. televoting services, transfer of charging information to service providers.
- General and selective broadcast switching e.g. switched and unswitched distribution services.

An ATM based B-ISDN will have the ability to support one-to-many call distribution through multicast switching. The basic multicast capability could be used with appropriate connection management to support a wide range of multipoint services e.g. conference calls, message broadcasting, video-on-demand, etc. This may result in significant simplification in multimedia terminal design and could also support the flexible deployment of multimedia and multipoint bridges.

A2.3. Signalling Requirements

The proposed service diversity of Broadband ISDN may require some associated enhancement of signalling protocols to accommodate the expanded service range. Signalling is necessary for the flexible implementation of multiparty and multiconnection calls for customers with multisite, multimedia communication needs.

B-ISDN Signalling Principles

B-ISDN Recommendation I.311 identifies the following signalling capabilities as being needed:

- Capabilities to control ATM virtual channel and virtual path connections
    - Establish, maintain and release ATM VCCs and VPCs.
    - Support point-to-point, point-to-multipoint and broadcast communication configurations.
    - Negotiate traffic characteristics of a connection at connection establishment.
    - Renegotiate source traffic characteristics of an established connection.
- Capability to support simple multiparty and multiconnection call
    - Symmetric and asymmetric simple calls
    - Simultaneous establishment and removal of multiple connections within a call
    - Add and remove connection from an existing call
    - Ability to correlate (when requested) connections composing a multiconnection call.
    - Reconfigure a multiparty call including an existing call or splitting the original multiparty call into more calls.
- Processing related functions
    - Capability to reconfigure an established connection e.g to pass through an intermediate processing facility such as a conference bridge.
    - Support for interworking between different coding schemes.
    - Support interworking with non-B-ISDN services e.g. 64 kbit/s ISDN.

Signalling for point-to-multipoint video services

- requires further study

The full signalling requirements for distribution services are for further study, however likely additional requirements include :

o Selection Switching

Distribution services of all types (eg. data, audio, video, image and multimedia) require a means of selecting items from the range available at the point of service distribution.

These actions correspond to a customer/viewer changing programmes.

- Fast Call Establishment

Switched access to distribution services (such as television) will require significantly shorter call establishment delays than existing networks. While set up delays of 2-3 seconds may be acceptable in a voice network, the tolerable delay for a user changing a TV channel is unlikely to exceed 100 ms.

- *Supplementary service aspects for further study.*

A2.4. Call and Connection Control

The B-ISDN will offer independent call and connection control facilities. This concept has two aspects :

- Separate specification of call and connection control within the network;
- Call control information flows may take a different route to the connection control information flows.

The multimedia and multipoint nature of many B-ISDN services will require flexible means of connection control. It should, given that network resources are available, be possible to add or remove parties from a multiparty call and add or remove services from a multimedia call.

Call establishment and termination, which may require multiple connections, and other network related operations during a call, must be common across multiple interworking video services.

A2.5. Storage Requirements

Many video and image services may require the network to store some aspect of the service. For example, video and messaging services will require network resident storage facilities, as will many forms of database. The efficiency and economy of such services is strongly influenced by the ease of coding and decoding images for storage and the characteristics of the storage medium itself eg. compact disk, videotape, magnetic disk.

Storage related issues :

- Efficient image compression/decompression algorithms to achieve cost efficient use of available storage capacity.
- Image coding times which reflect the nature of the intended service. e.g. for deferred delivery services, such as video mail, acceptable coding solutions may take the form of slow, but efficient coding and fast decoding.
- Others for further study.

A2.6. Service Bit Rates

The B-ISDN will be based on ATM techniques which are well suited to supporting source traffic which is time varying. The establishment of virtual connections which involve the transfer of information only when required will mean that the resources of the network can be closely matched to the needs of the source traffic.

The following areas relating to service bit rates are still under investigation :

Maximum Service Bit Rate Supported by the 155.52 Mbit/s Interface

The transfer capability of the 155.52 Mbit/s interface provides a payload capacity of 149.76 Mbit/s. Allowing for ATM cell overheads, the maximum service bit rate which can be supported is equal to or less than 135.631 Mbit/s. The actual maximum service bit rate is for further study. The actual value depends on the capacity required by signalling, operations and maintenance and ATM adaptation overheads.

The granularity of the actual service bit rates offered by networks is for further study.

Maximum Service Bit Rate Supported by the 622.08 Mbit/s Interface

Agreement has been reached that the B-UNI (I.413, I.432)at 622 Mbit/s should be based on a single ATM stream rather than a multiplexed structure of four 155 Mbit/s streams. The overhead structure of the UNI and the NNI at 622 Mbi/s is common and this results in an ATM cell transfer capacity of 599.040 Mbit/s.

Bit Rate Assurances

Parameters for constant and variable bit rates agreed at call set up time are assured for the duration of the call. No assurance is given concerning additional traffic above the level initially negotiated.

The specification of service bit rate parameters

- specification of the bit rate of CBR services requires only a single parameter
- specification of the service bit rate of VBR services is expected to require multiple parameters, e.g peak and

average rates, burst length etc.

• the time period over which the rate is specified is influenced by service timing and buffering constraints and the capabilities of the network interface.

• options for the specification of service bit rate include cells per unit time, bits per unit time or nx64 kbit/s.

• for CBR services there are two options to be addressed in specifying service bit rates;

-the service bit rate is the actual bit rate. i.e. a user must generate traffic at the exact bit rate.

-the service bit rate means a ceiling to be supported by the B-ISDN. A user can generate traffic at any bit rate less than the service bit rate. The use of a CBR service in this manner is for further study.

o multiple parameters may be required if a unique time period cannot be agreed as meeting the requirements of all services.

o the parameters selected must be of a form and nature which allows the network to exercise the option of statistically multiplexing VBR services, where appropriate, in a manner which does not violate the agreed QOS.

Control and monitoring of source traffic behaviour

• required for CBR and VBR services to ensure agreed parameters are not exceeded.

• it may be possible for the network to accept non-negotiated traffic, however it will not be possible to give the same quality of service assurance in such circumstances.

Allocation and control of network resources

• does not present new problems for CBR services

• large savings may be possible from the statistical multiplexing of uncorrelated VBR sources.

Traffic Control and Resource Management

The objectives of ATM layer traffic control have been identified as the following :

- ATM layer traffic controls should support a set of ATM layer Quality of Service classes sufficient for all foreseeable B-ISDN services

- ATM layer traffic controls should not reley on AAL protocols which are B-ISDN service specific, nor on higher layer protocols which are application specific.

- The design of an opticmal set of ATM layer traffic controls should tradeoff minimizing network and end-system complexity for maximising network utilisation.

- ATM layer traffic controls should maintain the ATM layer Quality of Service even under congestion conditions.

Usage Parameter Control and Network Parameter Control

Usage Parameter Control (UPC) and Network Parameter Control (NPC) are similar functions

- UPC is performed withing the Connection Related Function (CRF) on VPs only.

- NPC functions are performed on VC or VP links at the access point where they are terminated in the network.

- the need to standardise the UPC/NPC algorithm is for further study.

- UPC should accommodate any implementation of the customer equipment (CEQ).

- UPC should not assume any specific Generic Flow Control (GFC) mechanism. However any GFC should meet cell delay variation requirements and may introduce traffic shaping procedures for that purpose.

- to ensure, and protect, network performance - both CLP=0 and CLP=1 traffic flows must be allocated resources. Both must be controlled at the UPC/NPC. The impact on cell sequence integrity on a VCC requires further investigation.

- at the cell level, the UPC/NPC acts in response to user violation of the traffic parameters agreed at call establishment. For non-complying cells, the network need not respect the end-to-end performance contract.

- at the cell level actions of the UPC/NPC include the following :

* cell passing

* cell rescheduling (a combination of traffic shaping and usage parameter control)[1]

* cell tagging[1]

* cell discard

When the cell tagging option is excercised, non-compliant CLP=0 cells may be overwritten to CLP=1. Non-complying cells are merged with the remaining CLP=1 flow before it is subject to control. Non-complying CLP=1 cells are discarded. If no resources have been allocated to CLP=1, non-complying CLP=0 cells are discarded. Appropriate network dimensioning makes it possible to provide a given QOS for low priority cells.

Traffic parameters and descriptors

For a VCC, traffic descriptors are specified at the ATM SAP[2] or at some reference event in the ATM layer (such an event taking place before any ATM multiplexing and before any cell delay variation occurs).

A VPC is specified at some reference event, if possible similarly defined at the ATM layer. This issue is for further study.

Peak Rate

The following definition applies to both CBR and VBR connections and addresses the intrinsic peak cell rate generated by a given source before being altered by any ATM or Physical Layer mechanism. The basic event used as a reference is the Request to send an ATM_SDU (48 octet information field, note however this field may not be completely filled with user information.

The peak cell rate R_p is the inverse of the minimum inter-arrival time T_0 of the basic event described earlier. The unit to assess this inter-arrival time must be much smaller than the cell time and is for further study.

Resource allocation - preventative actions.

Three options are under consideration :

- *resource allocation*, i.e. allocating different resource levels for CLP=0 and CLP=1 traffic

- *usage/network parameter control*

- *traffic shaping*, i.e. rescheduling cells according to the declared peak cell rate value on individual ATM connections.

Resource allocation - reactive actions to relieve existing congestion conditions

Two options are under consideration :

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1 Optional

2 As agreed by CCITT SGXVIII/8-7, Geneva, 11-28 June 1991

- Normal conditions. i.e. those under which the peak cell rate in the traffic descriptor specifies an upper bound on the traffic that can be submitted on an ATM connection. Enforcement of this bound by a UPC allows the network operator to allocate sufficient resources to ensure that the performance objectives (e.g. for CLR) can be achieved. For low priority traffic, some adaptive rate control facilities at the ATM layer or above may be used. Such cell based reative techniques are for further study.

- *Failure conditions* - for further study.

A2.7. Quality of Service Aspects

Customer control of video and image service quality is an issue of both technical and economic importance. The flexibility to select the required service quality based on tariff, application, or other considerations requires the availability of suitable mechanisms for characterising different qualities.

It is generally accepted that quality of service is largely a users view of a service as opposed to the network providers view. Definition is difficult because of the nature of the key factors involved :

- different users;
- different services;
- subjective dependence on the users view of the service.

Quality of service is defined in CCITT Recommendation I.350 as "the collective effect of service performances which determine the degree of satisfaction of a user of a specific service". Network Performance is defined as " the ability of the network or network portion to provide the functions related to communications between users. Also, NP is a statement of the performance of a connection element or a concatenation of connection elements employed to provide a service". The relationship between QOS and NP is of vital importance. In CCITT Recommendation I.350 the relationship is described in these terms, "the user oriented QOS values provide a valuable framework for network design but they are not directly usable in specifying performance requirments for particular connections. Similarly, the NP parameters primarily determine the QOS, but they do not necessaryily describe the quality in a way that is meaningful to users".Both types of parameter are needed and their values qualitatively related if a network is to be effective in serving its users.

A2.7.2 Quality of Service (QOS) Provided by the ATM Layer

Draft Recommendation I.150 describes B-ISDN ATM functional characteristics including Quality of Service. Issues covered include :

- QOS related to Virtual Channel Connections (VCCs)

While the detailed specification of QOS classes is for further study, it has been agreed that the user will be provided with one class per VCC. The agreed QOS cannot be changed during a call. Renegotiation may require the establishment of a new connection.

- QOS related to Virtual Path Connections (VPCs)

Again the detail specification of specific QOS classes is for further study and only one class may be assigned to each VPC. The QOS is agreed at call/connection establishment and does not change for the duration of the VPC. Since an individual VPC may carry VC links of various QOS classes, the VPC QOS must meet the needs of the most demanding VC. This is of particular relevance in the consideration of multiplexing schemes for multimedia service support.

- QOS related to Cell Loss Priority (CLP)

The CLP indicator in the cell header may be set by either the user or the service provider. In the case of video services, the CLP bit is set by the layered coding provider to indicate lower priority cells which may be discarded by the network without violating the negotiated QOS.

The following questions and answers, relating to the status of discussions on CLP issues, have been identified and obtained from a services viewpoint within CCITT SGXVIII.

Q1). There are two ways of using CLP capabilities of B-ISDN, i.e. cell by cell basis and per connection basis. In the latter case, will a CLP be defined per VPC basis as well as per VCC basis?

A1) No constraint from services viewpoints. This should be studied mainly by ATM and Resource Management aspects.

Q2) In the case of CLP bit capability, will a single level of CLR (Cell Loss Ratio) i.e. high priority level, be defined, or will two levels of CLR be defined respectively?

A2) CLR for CLP=0 (high priority) should be defined, and assured by the network if a cell traffic does not exceed the negotiated values. For CLP=1 traffic, the following two options exist;

1) defined and assured by the network

2) not defined and no assurance given

In the case of option 2), a given CLR may be maintained by the network engineering.

Q3) Will the CLP bit, set by a user as high priority, be changed by the network e.g. for violation tagging in case that a cell traffic exceeds the negotiated values?

A3) Since CLR of CLP=0 is assured by the network, there may be no impacts on services for users, whether the network will override the bit or not.

Q4) Will the specific value of CLR be explicitly declared by a user, or will the CLR indication be implicitly associated with specific service requests, e.g.

- a standardised service will by definition include the specification of all relvant QOS values, or

- a standardised QOS class will by definition include the specification of all relevant QOS values?

A4) Under study.

Q5) Will the network provide for several CLR, or will the network accommodate CLR requests from users within a very limited number of CLR? And how many CLR will be required?

A5) Under study.

A2.7.3 Quality of Service Indication and Negotiation

Several Recomendations make reference to QOS negotiation issues.

Recommendation I.211, indicates QOS is negotiated at call setup or possibly during a call. It is for further study to determine whether specific QOS parameters will be explicitly indicated (.e.g. by a specific cell loss ratio) or implicitly associated with specific service requests. Services making use of the Cell Loss Priority indication on a cell-by-cell basis will need to indicate the intended use of this indicator at call establishment. This indication is needed to allow appropriate network resource allocation and usage parameter control.

Recommendation I.211 also comments on CBR and VBR service bit rates. For both CBR and VBR services the service bit rate parameters are negotiated at call establishment and supported for the duration of the call. Changes to these parameters may be negotiated within the call period and the details of this negotiation are for further study. In both cases, a set of discrete bit rates will be chosen.

Recommendation I.150 (QOS of the ATM layer) specifies that the QOS of virtual channel connections cannot be changed for the duration of the connection. Renegotiation of the QOS class may require establishment of a new connection. The QOS of a virtual channel connection will not change for the duration of the VPC and must meet the most demanding QOS of theVC links carried.

A2.7.4 General Aspects of ISDN Performance

Recommendation I.350 defines Quality of Service and Network Performance principles and illustrates how the QOS and NP concepts are applied in digital networks. Draft new Recommendation I.35B defines performance parameters and performance objectives for the ATM layer of a Broadband ISDN.

ATM cell transfer performance parameters are specified on the basis that the sequence of cells on a virtual channel is preserved (reference I.121). In principle, a point-to-multipoint connection might cause out of sequence cells.

ATM performance parameters subject to definition and specification within I.35B are :

• Cell Loss Ratio

Simulation results indicate that cell loss events may occur in clusters rather than independently. One or more parameters describing the distribution or relative frequency of consecutive cell loss events in ATM networks should therefore be considered.

The response to lost cells for CBR and VBR is under study within CCITT WPXVIII/6, however two service independent methods are available :

- replacement of lost cells by a fixed bit pattern;

- correction for lost cells through the use of forward error correcting codes.

The effect of discarding cells will be service dependent. For example video services may require discarded cell ratios of 10^{-9} to 10^{-10}. This is particularly the case for high bit rate video services.

o Cell Misinsertion Ratio - the number of misinserted cells within a specified time interval. Cell misinsertion may exert a major influence on QOS since it is more difficult to deal with misinserted than lost cells. Inserted cells result in an increased information flow for the VC concerned and the cell misinsertion ratio selected must ensure that no load problems arise. For some services. misinserted cells may result in loss of terminal synchronisation.

• Cell Error Ratio

• Cell Transfer Delay - the end-to-end cell transfer delay consists of :

- inter-ATM node transmission delay;

- queueing, switching and routing processes in ATM nodes. As an objective this delay component should be of the order of 20ms. In practice, the delay of one ATM switching element is likely to be less than 1ms, although it may vary with the traffic load on the switch.

• Mean Cell Transfer Delay

• Cell Delay Variation

• Severely errored cell ratio - severely errored cells arise when a successfully delivered cell has N or more bit errors in its information field. The need for and methods of measuring the severely errored cell ratio are for further study.

• Cell Transfer Capacity - the definition of this parameter is for further study. Some of the issues to be considered are :

- the relationship between this parameter and the user's a priori request for capacity

- the effects of ATM flow control mechanisms, including the requirements on the user to apply and respond to these mechanisms.

- the limits on the cell loss ratio when the connection is operating at its cell transfer capacity.

- the unit of time over which the parameter is measured.

Relationship Between ATM Layer NP and the QOS of the AAL for CBR Services

• Lost and Misinserted Cells

The Sequence number (SN) in the adaptation layer; header can be used to detect lost and misinserted cells.

Detection mechanisms are for further study.

Inserted cells may be discarded without disrupting the user information flow.

Lost cells may be substituted by dummy cells in order to adjust the number of bits (bit count integrity) however this results in bit errors in the user information. The content of the dummy cells require further study.

- Errored and Severely Errored Cells

Bit errors occurring in the ATM cell information field are transferred to the user as they occur.

- Cell Transfer Delay

To compensate for the variation of cell delay, arriving cells are buffered at the receiving side Adaptation layer. Buffering and cell assembly increase the transfer delay of user information. Lost cell detection mechanisms may also increase the overall transfer delay.

Excessive cell transfer delay may cause substitution by dummy information and result in bit errors in user information.

Variable Bit-Rate Services - Adaptation Layer Performance Specifications

For further study.

A2.8. Timing Issues

The support of real-time services over an ATM network requires mechanisms to achieve timing recovery and compensate for variable, although bounded, network delays.

Cell jitter (the variable delay in cell arrivals) must be buffered within the codec. The size of the required buffers is determined by the cell jitter and the service bit rate.

For multimedia services there is a need to ensure differential delay between the various services, particularly the video and audio, is acceptable.

A2.9 Open Issues

The following text is an input from SGXV Experts Group for ATM Video Coding, on the subject of Networks Aspects

Network Parameters Impacting on Video Coding Definition

A number of parameters and operational procedures concerning the B-ISDN network will have significant impact on the definition of appropriate coding schemes for the support of video services.

The areas requiring definition are listed below:

Cell loss ratio

This is an important determinant of the quality of service achievable for a video application. It determines the means, and even necessity, for providing cell loss protection for different services. It is recognised that there is a degree of flexibility in this figure, since the network operators have some flexibility to dimension the network to provide certain cell loss ratios if they are considered essential for some video services, while the codec design can also be changed to accommodate different figures. Progress needs to be made, though, perhaps by considering the impact of a range of cell loss ratios on both network and codec. The cell loss ratios for both priority levels need to be defined. The SGXV Experts Group believesthat guaranteed overall cell loss ratios, for both priority levels, will be essential to satisfy video quality of service requirements. Guaranteed performance, at least within certain time intervals, will also be required.

If the cell loss ratio is sufficiently small, no cell loss protection may be necessary. For example, a high quality videoconference connection operating at 10 Mbit/s would suffer only one cell loss every 10 hours with a CLR of 10-9. This may be acceptable even if the cell loss caused visible degradation.

Studies are required to determine the quality of service parameters available to the user, and to relate these to cell loss ratio.

Cell loss burst behavior

It is understood that cell losses may occur in bursts. This impacts on the means of cell loss protection; the use of forward error correction may be too expensive and delay may be excessive for conversational services if multiple consecutive lost cells must be detected and corrected. Cell loss burst behavior may be modeled by the Gilbert model (a two-state Markov model requiring four transition probabilities, with one state representing no cell loss and the other constant cell loss).

Open questions remaining are:

- How will the cell loss burst behavior depend upon the service rate?

- Will the burst behavior of high priority cells differ from that of low priority cells and, if so, how?

- How can we estimate the average interval time, T, in which no cell loss occurs? If $T >> 1/(\text{bitrate} \times \text{CLR})$, the requirement for CLR might be relieved.

Use of CLP bit

The CLP bit is seen as a useful mechanism to provide protection against cell loss by controlling that information which might be lost. It is crucial that, after a cell is labeled "high priority" by a terminal device, this is not changed by the network.

Open questions:

- Will there be separate negotiations for the two priority levels?

- Will the usage monitoring structure encourage use of both high and low priority cells?

- What options are available in selecting the quality of service?

Usage parameters

The rate statistics required of a video encoder have a significant impact on its performance (in terms of picture quality and delay). For circuit switched networks, the target was straightforward; minimise the rate and keep it constant. For the B-ISDN (with the possible advantages of variable rate over constant rate operation), entirely different rate control strategies may be appropriate, and these could have a significant impact on codec performance. At this stage, the only clear decision is that peak rate will be an important parameter that is monitored.

In our group the term "window" means the policing time for the average bit rate. The following methods are considered for policing in the network:

Jumping window:

There is no time interval between two successive windows.

Moving window (sliding window):

The window is sliding at a time step smaller than the window size.

Stepping window:

There is a time interval between two successive windows, which always start at a valid cell.

Leaky bucket:

Cells are put into a buffer and taken from the buffer at an average bit rate. If the buffer overflows, cells are discarded.

If a codec does not know when the network measuring window starts, it should control the bit rate by sliding window (the most severe method). Is there any way in which the starting time of the network measuring window can be known?

Open questions:
- What parameters will be used for policing and admission control?

- What policing mechanism will be used?

- What averaging intervals can be used to measure mean, peak, etc.? Longer intervals (significantly greater than a video frame period which is typically 33-40 ms) are preferred for video services.

- When the network capacity is very large, the bit rate requirements of a single user will be relatively small. In this situation it seems there will be very little difference in the required network resources for low and high priority cell loss classes. Will the high priority cell loss class continue to exist in the future?

Multimedia connections

Multiplexing of multiple media has been carried out within the terminal device for circuit switched networks. The B-ISDN already offers the flexibility to use cell-based or virtual channel based multiplexing instead. An important factor in the choice between terminal-based or cell-based multiplexing is whether there will be a penalty caussed by the use of an ensembleof virtual channels instead of one composite one, although the overall rate characteristic, for example, would be the same. Most importantly, would the two options have the same transmission costs?

Some multimedia connections (most obviously associated audio, stereo in particular, and video channels) require synchronism. A concern arises, therefore, if the differential delay between virtual channels became noticeable in some service applications. This is unlikely to be a problem unless the cumulative differential delay exceeds some tens of millisecondsfrom end to end.

Open questions:

How will multimedia services be handled in the B-ISDN?

What signalling methods are being proposed?

What kind of multimedia multiplexing method is preferred from the standpoint of network resource management?

Bit error rates

Cell payloads will be subject to a small probability of transmission error on the B-ISDN. The statistics of such errors will determine the need for, and type of, error correction mechanism and the overhead necessary to achieve this. It could also influence approaches to, and efficiency of, video coding and choice of codeword assignment scheme. Estimates of the likely bit error rates are required by those working on video coding schemes for the B-ISDN.

For interworking between video codecs on N-ISDN and B-ISDN networks, the B-ISDN bit error rate must be no greater than that for the N-ISDN. It should also be noted that the H.261 coding scheme for N-ISDN provides bit error correction, so this would not be a necessary function of the AAL in this case.

SGXVIII should work in close collaboration with the video coding experts to define anycapability within the AAL concerning bit error detection or correction.

Cell delay and jitter

The fixed component of end-to-end network delay contributes to the total service end-to-end delay and therefore is a determining factor in the overall quality of service. Estimates of the limits of B-ISDN delay are required to quantify such performance and determine its impact on video encoders and decoders.

The variation in delay, or jitter determines the size of receiver buffers necessary for its removal, and therefore again influences total end-to-end delay. The expected statistics of cell delay jitter need to be known to determine the impact on the video coding system and overall quality of service.

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Annex 3. ATM Adaptation Layer

A3.1 AAL Type 1

A3.1.1 Service provided by AAL type 1

The services provided by AAL type 1 to the higher layer are:

- transfer of service data units with a constant source bit rate and the deliv ... bit rate;
- transfer of timing information between source and destination;
- indication of lost or errored information which is not recovered

A3.1.2 Interaction with the management

The following indications may be passed from AAL type 1 in ... agement plane:

- errors in the transmission of user information;
- lost or misinserted cells (further study is req ... essary to distinguish between lost and misinserted cells);
- cells with errored AAL Protocol Inform ... study is required if this indication is necessary for services supported by
- loss of timing/synchronization.

A3.1.3 Functions in AAL type 1

The following functions may be ... order to enhance the service provided by the ATM layer:

a) segmentation a ... rmation;

b) handling of

c) handlin ... ells;

d) sour ... ry at the receiver;

e) r ... r bit errors;

f) ... it errors;

g) moni ... information field for bit errors and possible corrective action (the use of this function ... service is for further study);

h) other service specific functions are for further study.

The allocation of these functions to the CS or SAR is for further study.

NOTE : As a result of agreements at the June 1991 meeting of SGXVIII the text relevant to the SAR of AAL1 will be amended to reflect these agreements in a subsequent revision of this Baseline document.

Note - For circuit emulation a need has been identified to monitor the end-to-end QOS. This may be achieved by calculating a CRC for the CS-PDU payload, carried in one or more cells, and transmitting the result in the CS-PDU or by the use of an OAM cell. Further study is required.

A3.1.4 Segmentation and reassembly sublayer

Functions of the SAR sublayer.

The definitions of these functions are for further study.

The SAR functions are performed on a ATM-SDU basis.

SAR-PDU structure and coding.

SN:Sequence Number (4 bits); to detect lost or misinserted cells.

A specific value of the sequence number may indicate a special purpose, e.g. the existence of Convergence Sublayer functions.

The exact counting scheme is for further study.

SNP: Sequence Number Protection (4 bits).

The SNP field may provide error detection and correction capabilities. The polynomial to be used is for further study.

FIGURE 1/I.363

SAR-PDU format for AAL type 1

A3.1.5 Convergence Sublayer

Functions of the CS

The CS may include the following functions

a) For high quality audio and video forward error correction may be performed to protect against bit errors. This may be combined with bit interleaving to give more secure protection against errors.

b) For some services, this sublayer provides the clock recovery capability for the receiver e.g. by monitoring the buffer filling. This requires no specific field in the CS-PDU.

c) For services requiring explicit time indication, this may be provided by means of a time stamp pattern inserted in the CS-PSU. Other mechanisms may be used to provide this function.

d) Further sequence number processing may be performed at this sublayer. The handling of lost and misinserted cells is also performed in this sublayer.

A3.2 AAL Type 2

A3.2.1 Service provided by AAL type 2

The services provided by AAL type 2 to the higher layer may include:

- transfer of service data units with a variable source bit rate;
- transfer of timing information between source and destination;
- indication of lost or errored information which is not recovered by AAL type 2.

A3.2.2 Interaction with the management

The following indications may be passed from the AAL type 2 in the user plane to the management plane:

- errors in the transmission of user information;
- loss of timing/synchronization;
- lost or misinserted cells (further study is required on whether it is necessary to distinguish between lost and misinserted cells);
- cells with errored AAL-PCI (further study is required to determine if this indication is necessary for all services supported by this AAL type).

A3.2.3 Functions in AAL type 2

The following functions may be performed in the AAL type 2 in order to enhance the service provided by the ATM layer:

a) segmentation and reassembly of user information;

b) handling of cell delay variation;

c) handling of lost and misinserted cells;

d) source clock frequency recovery at the receiver;

e) monitoring of AAL-PCI for bit errors;

f) handling of AAL-PCI bit errors;

g) monitoring of user information field for bit errors and possible corrective action (the use of this function for voice service is for further study);

The allocation of these functions to the CS or SAR is for further study. Other service specific functions are for further study.

A.3.2.4 Segmentation and Reassembly sublayer

Functions of the SAR

For further study.

The SAR functions are performed on a ATM-SDU basis. As the SAR accepts variable length CS-PDUs from the convergence sublayer the SAR-PDUs may need to be partially filled.

SAR-PDU structure and coding.

The SAR-PDU structure and coding requires urgent further study.

A3.2.5 Convergence Sublayer

Functions of the CS

The functions to be performed are for further study.

The convergence Sublayer may perform the following functions:

a) clock recovery for variable bit rate audio and video services by means of the insertion of a time stamp or real time synchronization word in the CS-PDU. Other mechanisms may be used to provide this function;

b) sequence number processing may be performed to detect the loss or misinsertion of ATM-SDUs. The handling of lost and misinserted ATM-SDUs is also performed in this sublayer;

c) for audio and video services forward error correction may be performed.

A3.2.6 SGXVIII intends to continue its studies on AAL Type 2 in both the SAR and Convergence Sublayers to further elaborate the above text. As the AAL Type 2 will be used for a range of different services, SGXVIII is aiming to first establish the functional requirements for AAL type 2, identifying those which may be mandatory or optional for particular service applications. Inputs to SGXVIII from the specialist video coding groups of CCITT SGXV, CMTT and CCIR SG11 on video and image service AAL functional requirements are requested.

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Annex 4. Video Service Interworking

Layered coding has been identified in Draft Rec I.211 as a promising means of facilitating interworking between video services, as well as providing protection in the event of cell loss.

A review of alternative approaches to interworking would be appropriate; in particular, the following methods should be considered and contrasted:

- Simulcast Approach. The simultaneous transmission of the same video signal in different formats.
- Negotiation, or switchable coders. The sender and receiver negotiate on an appropriate video signal format to use, within the capabilities of both devices.
- Layering, or embedded bitstream. A hierarchical coded signal representation permits different grades of the same signal to be recovered.

The B-ISDN will be capable of delivering a range of service applications (e.g. communicative real-time video, video retrieval or store-and-forward video, distributive services), using signal formats covering a wide range from videophone resolution to HDTV and at a range of qualities for any given signal format. The suitability of the alternative video service interworking approaches should be considered in the context of this range of applicability to satisfy the agreed key objective of maximum video service integration.

Open Issues:

- Terminology

It is noted that there are differences in terminology used by the different groups and that this will require further study.

Additional Text

Text proposals have been received from SGXV and CMTT/3, both of which have been included in the Baseline. It is SG XVIII's intention that there be a later merging of similar input material.

The following text is an input from CMTT/3 on the subject of video service interworking:

Video service interworking is desirable and has advantages for a wide range of applications, e.g. conversational, distributive services and multimedia applications.

Attention must be paid to the fact that ensuring interworking within a heirarchy of resolution standards may impose some constraints on the production techniques. These constraints include limitations like resolution, aspect ratio and other aspects of production.

Interworking may be achieved by several compatibility techniques,

e.g. :

- simultcasting, in which two or more encoded signals are transmitted in parallel, and which can be decoded separately;

- embedded bit stream (layered coding);
- syntactic extension, in which a new decoder can interpret a new syntax as well as a sub-set of that syntax generated by an existing encoder;
- switchable encoder.

These techniques will result in different levels of compatibility and will constrain the design of coding algorithms.

Layered coding has been identified in draft Recommendation I.211 as a promising means of facilitating interworking between video services, as well as providing protection in the event of cell loss. For comparison, however, non-layered coding methods should be considered.

Some advantages of a layered coding system are;

- ease of extension to future video systems (e.g. from HDTV to super HDTV with e.g. 4000 x 4000 pixels);
- ease of compatibility among various video services.

Some of the disadvantages of layered coding are:

- possible reduced coding efficiency when compared to non-layered systems if motion compensation is applied.
- possible increase of the complexity of encoder and decoder.

The main objective for contribution applications is to achieve the best picture quality. As compatibility approaches may cause some constraints on the performance of the algorithm it is not clear whether these approaches are suitable for contribution applications. Due to possible post-processing applicaiton it is not desirable to lose any information. Therefore, it is not appropriate to assign a lower priority to some cells of the bit stream as is possible with e.g. layered coding.

At the present time further studies are needed to define preferred solutions.

The following text is an input from SGXV Experts Group for ATM Video Coding, on the subject of video service interworking:

Integration of video services is recognised as a key objective for ATM Video Coding. It is an agreed target for the video coding system under study by the SGXV Experts Group. Several options exist for interworking between services.

Negotiation Approach

At the commencement of a connection, terminals negotiate a set of parameters with which both can cope. A set of standards of increasing quality would be defined and a basic capability assumed for all terminals.

Simulcast Approach

Transmitting terminals contain multiple encoders, operating at a variety of resolutions and quality levels so that broad interconnectivity can be achieved. Receiving terminals could be simple devices able to receive one of the bit streams, or could contain multiple decoders allowing a selection.

Layered Signal Approach

A hierarchical representation of the video signal is defined. Coders transmit a baseband signal which provides a basic quality service. Incremental signals, which can be used along with the baseband to recover a high quality signal, are also transmitted. Receiving terminals utilise the baseband and an appropriate number of incremental signals to recover the video signal to the quality which they are capable of displaying. Transmitting terminals provide the number of signals which is commensurate with their input signal qulaity. Note that `embedded bitstream'and `syntactic extension' techniques are also versions of layered coding (see TG CMTT/3 liaison statement to SGXVIII dated 17 April 1991 for the terminology).

A range of issues needs to be considered in comparing these different approaches, including complexity, coding rate penalties and performance. Negotiation would seem inappropriate for multipoint and distribution services, whereas simulcast seems inappropriate for storage applications (e.g store and forward video mail). Layered coding seems suited to the widest application range. `Flexible layering' in which any number of layers can be used in any particular application, appears to provide broad interworking capability with few restrictions, and is currently one of the options being studied.

It is recognised that to provide easy interworking or conversions between services, and to use common display components on a terminal device intended to access multiple video services, the definition of a family of picture formats would be beneficial. Picture formats represent an important area that will influence video coding and it is being studied actively in the SGXV Experts Group.

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Annex 5. Coding Aspects

A5.1 Constant Bit Rate (CBR) and Variable Bit Rate (VBR) coding

Restrictions of traditional circuit switched networks have meant that all commercial digital video codecs operate at a constant bit rate, this despite the inherently varying information content of a motion video sequence (being dependent on changing image complexity, degree of motion, frequency of scene changes, etc.). The internally varying

rate in these codecs is smoothed by buffering, and dynamic control of codec parameters (sensitivity, quantiser stepsize, etc.) ensures that the buffer neither empties nor overflows. Such codecs operate in a fixed rate, but variable quality, mode.

ATM Networks will support Variable Bit Rate (VBR) coded video, allowing the transmitted bit rate to reflect the information content of the changing video signal, limited by the maximum channel capacity and parameters agreed with the network management system.

A VBR codec can therefore (usually) maintain a fixed quality, variable bit rate mode of operation. The possible advantages of this are:

- Because data is not transmitted when the information content is low, and because high rates are only used when necessary, VBR codecs are expected to deliver a given overall quality at a lower average rate than a CBR codec;
- The reduction in buffer size and easing of constraints on rate control means that there could be savings in codec complexity and cost;
- Reduced buffering may mean that end-to-end delays will be reduced; this is an important consideration for communicative services such as videotelephony and videoconferencing.

There may be substantial savings in average bit rate through the use of VBR coding and statistical multiplexing of multiple sources on the one network. Studies are required to confirm this advantage under realistic network conditions and to determine its sensitivity to the type of application (videoconferencing, television distribution, etc.) and method of coding. Delayed contribution D.962 from FRG to the Nov/Dec 1990 meeting of CCITT SGXVIII reported the results of some measurements which indicate that under certain conditions the statistical multiplexing gain of some VBR video service applications is only small.

Studies are also required to quantify the reduction in delay resulting from VBR coding, and to look at this saving in relation to limits obtained from human factors investigations.

A5.2 Compatibility Aspects.

It is important to consider the various applications of coded video signals and to maximise commonality where possible to achieve a truly integrated video services structure. A particularly important area for compatibility is in the coded representation of video for communications and storage.

Stored video has some constraints that are not applicable for communications applications. For example, there may be the requirement for fast forward, and reverse play. The constraints may differ depending on whether tape or disk based storage is used.

Efforts to provide commonality between stored and transmitted video formats have already been initiated by the ISO/IEC MPEG group. If, however, the coding techniques cannot be made identical, care should be taken to ensure that compatibility can be facilitated readily.

Compatibility between an IVS signal format and existing or emerging standard digital video formats for circuit switched networks should be the objective during the interim period before full B-ISDN support.

A5.3 Cell-based (ATM) Transport aspects relating to Video Coding

Transmission of video information in cells requires consideration of several factors:

- Error protection. A layered coding approach (see Annex 4) appears attractive as a means of minimising the effect of cell loss, particularly if it occurs in bursts. This requires separation of the video information into high and low priority components and appropriate setting of the cell loss priority (CLP) bit in the ATM cell header.

- Error propagation. Mechanisms to avoid propagation of errors in the event of a cell loss need to be investigated.

A5.4 Open Issues

FFS

A5.5 Additional Text

The following text is an input from CMTT/3 on Coding Aspects

It is suggested that this text be inserted in A5.1, after the end of the existing first paragraph.

Current codecs and ATM networks

Existing CBR codecs have been designed to be used in current plesiochronous networks. Their output bit-rate matches the rates of the plesiochronous hierarchy e.g 34-45 Mbit/s and 140 Mbit/s. As a consequence, their design includes an adaptation to plesiochronous networks in the form of an error detection and correction unit. As existing CBR video codecs have been defined, it is not envisaged to remove the internal adaptation so they are plesiochronous-oriented.

When a CBR video codec is to be connected to an ATM network, one problem appears because the internal adaptation does not perform extra functions required by ATM networks. Therefore, an additional adaptation is necessary. It is called the AAL (ATM Adaptation Layer), and it has to be added beside the built-in adaptation of the codec. Such a scheme is not an optimum one, because it is not easy to combine functions of both adaptations.

Adapting CBR codecs to ATM Networks

According to the service classification defined in CCITT Rec.I.362, constant bit rate (CBR) video services pertain to Class A. Codecs are currently available performing these services. For the time being, most of existing codecs, if not all, have been designed to be connected to plesiochronous networks. For the connection of these codecs to ATM networks, two ways have been identified : circuit emulation and direct connection, both using a dedicated AAL.

- Circuit emulation - The codec is used as it if were connected to a plesiochronous network. The signal is inserted into the relevant PDH frame structure which is then carried transparently over ATM networks through a specific AAL. As a result, particular requirements of the CBR signal components (video, audio, data) are not taken into account.

- Direct connection - The definition of the AAL takes advantage of error correction which is already performed in the codec itself. In this case the different components of the signal (video, audio, data) may be carried in the ATM network in separate VCs.

Annex 6. Multimedia Service Support

A6.1 Multimedia Service Categories

Multimedia services may be categorizied on the basis of how they appear to the customer e.g.

Multimedia Call and Conference, e.g. multiparty, multimedium calls;

Multimedia Mail, e.g. extending the text based electronic mail model to include other media;

Multimedia Database ,e.g. browsing through catalogs, educational tutorials;

Shared Resource Multimedia Applications e.g. sharing network resources among users as a means of containing costs for expensive or infrequently used facilities.

The methodology for all multimedia services is the responsibility of CCITT SGXVIII/5-2 (Multimedia and B-ISDN). This group is responsible for the investigation of the network capabilities required to ensure the support of multimedia services. Work has commenced on a framework for a recommedation covering these issues.

A6.2. Network Issues

A6.2.1 Multimedia Service Attributes

Multimedia services have attributes which distinguish them from traditional telecommunications services such as voice or data. A multimedia service may involve multiple parties, multiple connections, the addition/deletion of resources and users within a single communications session.

The following service attributes are multi-valued in a multimedia call

- information transfer rate;
- traffic type
- timing end-toend
- information transfer capability;
- structure;
- symmetry;
- type of user information;
- higher layer protocols (layer 4 through 7 protocol functions);
- quality of service.
- information access protocol
- others for further study

A6.2.2 Multimedia Call Modelling

For further study.

A6.2.3 Signalling for Multiparty Multimedia Services

Call and connection control for multimedia services is a new consideration for public network standards. Work in this area is at a very early stage of development. CCITT SGXI are currently investigating the functionality required of call and connection control.

B-ISDN Recommendation I.311 identifies the following signalling capabilities as being the basic capabilities required to support simple multiparty and multiconnection calls :

- Capabilities to control ATM virtual channel and virtual path connections;
- Establish, maintain and release ATM VCCs and VPCs;
- Support point-to-point, point-to-multipoint and broadcast communication configurations;

- Negotiate traffic characteristics of a connection at connection establishment;
- Renegotiate source traffic characteristics of an established connection.

• Capability to support simple multiparty and multiconnection call;

- Symmetric and asymmetric simple calls;
- Simultaneous establishment and removal of multiple connections within a call;
- Add and remove connection from an existing call;
- Ability to correlate (when requested) connections composing a multiconnection call;
- Reconfigure a multiparty call including an existing call or splitting the original multiparty call into more calls.

• Processing related functions;

- Capability to reconfigure an established connection e.g to pass through an intermediate processing facility such as a conference bridge;
- Support for interworking between different coding schemes;
- Support interworking with non-B-ISDN services e.g. 64 kbit/s ISDN.

Support of multimedia services on B-ISDN will permit the use of virtual channels for separate service components of the multimedia connections.

Issues that must be studied in this area include:

• Interworking with a terminal multiplexed multimedia connection (e.g. using H.221);

• differential delays between virtual channels (particularly important for audio and associated video).

• Network usage parameter control and charging based on ensembles of virtual channels within one or multiple virtual paths.

• Signalling to support multi-connection calls within a single call, or use of multiple calls (each supporting one connection).

A6.2.4 Multimedia Traffic Control and Resource Management

A6.2.4.1 Connection Admission Control

Recommendation I.311 indicates:

• in the case of multimedia and multiparty services, connection admission control procedures are performed for each VC or VP connection;

• Signalling messages sent by a user at call establishment must convey at least the following information :

- source traffic characteristics;
- required QOS class.

These parameters may be difficult to determine in those cases where media is multiplexed on anything other than a virtual channel or virtual path basis.

• Methods for charactersing source traffic are for further study. Traffic characteristics may include : average rate, peak rate, burstiness and peak duration. Again it may prove difficult to characterise multimedia services where media is multiplexed on the basis of anything other than a virtual channel or a virtual path.

• Traffic characteristics are negotiated with the network at call establishment and may be renegotiated, by user request, during the call. The network may limit the frequency of these renegotiations Further study is required to determine the impact of such potential restrictions on multimedia calls.

A6.2.4.2 Usage Parameter Control

Recommendation I.311 indicates :

- Usage parameter control is performed on VCs and VPs at the access point where they are terminated within the network. This implies multiple usage parameter control for a multimedia service where individual services are carried on separate VCs or VPs.
- Agreed parameters for usage control are for further study.
- Actions proposed when traffic violates the call establishment agreement include :
    - discarding cells which exceed the pre-negotiated traffic levels;
    - tagging of violating cells;
    - releasing the connection.

A6.2.4.3 Resource Management

Recommendation I.150 specifies

- For VPs are required to carry VCs with a range of QOS values, the VPC QOS corresponds to the most demanding VC link carried.

The impact of this arrangement on options for multimedia service support is for further study.

Recommendation I.xxx specifies:

- Where the network has no knowledge of the QOS of the VCs within a VP connection, it is the users responsibility to determine, in accordance with the network capabilities, the QOS appropriate to his VP.

A6.3. CPN Issues

A6.4 CPE Issues

A6.5 Open Issues

- multimedia service interworking (between services, terminals and networks)
- interworking terminals and the determination of basic levels of compatibility e.g. H.221 requires all terminals to include a PCM coder-decoder, hence all compliant terminals are compatible with 3.1 kHz audio and speech terminals.
- media multiplexing options and their impact on required network capabilities e.g. sychronisation to provided bounded cross media delays.
- service negotiation issues.
- multimedia service interaction e.g. voice activated video.
- multipoint networking (e.g. multicast, broadcast and conference connections for multimedia)
- charging issues