AVC-637 TD32(1/15)
TD6(JTC6 on AVMMS)

INTERNATIONAL TELECOMMUNICATION UNION
TELECOMMUNICATION STANDARDIZATION SECTOR
STUDY PERIOD 1993 - 1996

COM 13-R 18-E
March 1994
Original: English

Question: 28/13, 1-3/15

Texte disponible seulement en
Text available only in
Texto disponible solamente en
E

STUDY GROUP 13 - REPORT R 18

SOURCE*: STUDY GROUP 13 (GENEVA MEETING, 7-18 MARCH 1994)

TITLE: REPORT OF WORKING PARTY 2/13 (B-ISDN ASPECTS) - IVS BASELINE DOCUMENT

It is proposed by SG 13 that the IVS Baseline Document has served its purpose in the transfer of information between the relevant groups and that appropriate liaison mechanisms are in place to achieve the desired level of harmonization. SG 13 therefore intends that this be the final version of the IVS Baseline Document. This revised version is based on information available to SG 13 as of March 1994.

INTEGRATED VIDEO SERVICES (IVS) BASELINE DOCUMENT MARCH 1994

LIST OF CONTENTS

* Contact person - Name: TSB
Tel.: +41 22 730 5860
Fax: +41 22 730 5853

1. General

This document was initiated by SG 13 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 Service by SG 13 and other groups.

IVS does not imply any specific service harmonization, but is the concept of aiming at maximum integration of video services through harmonization of the terminal and network capabilities of B-ISDN.

Major areas which require further development are contained in separate annexes to this document.

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. Such service and delivery technology developments are driving the convergence of telecommunications and broadcasting from both a service and network perspective. If the potential advantages of this convergence are to be realized, close and effective-cooperation and co-operation between service providers, network providers, users, terminal developers and standards bodies will be essential.

The Broadband-Integrated Services Digital Network (B-ISDN) will form the foundation of public networks capable of the integrated support of voice, data and video applications, providing the network technology enabler for the convergence of telecommunications and broadcasting,. 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 service (e.g. , picture within picture), ensuring connectivity and a competitive multi-vendor equipment environment.

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

Service integration can occur at may levels within networks and customer equipment. The emergence of B-ISDN standards and network technology provides an opportunity to rationalize 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 maximizing 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 could 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 maximized 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, Recommendation I.211 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;
- minimization of interworking requirements;
- minimization 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 service standardization 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. The requirements of audio signals, both independent and associated with video signals, must be considered.

3. Responsibilities

The following areas of responsibility are SG 13's understanding of the terms of reference for groups interested in Video services on B-ISDN.

SG 1

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 ITU-R SG 11.

SG 8

Responsible for, amongst other tasks:

- Recommendations on coding for still image communication.

SG 9

Responsible for, amongst other tasks:

- Recommendations on the bit rate reduction coding, packaging and transmission of television and sound program 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 SG 15.

SG 11

Responsible for, amongst other tasks:

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

SG 13

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, in collaboration with other groups, ATM Adaptation Layer specifications;
- Identifying network wide impact on B-ISDN service support, including the coordination across ITU bodies necessary to maximize commonality between communicative and distribution video services;

- Establishing the framework for video service support in the B-ISDN;
- General aspects of quality for service and network performance in digital networks including ISDNs;
- Providing coordination across different groups on Integrated Video Services in B-ISDN.

SG 15

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 utilized in the B-ISDN;
- Recommendations for audiovisual aspects;
- Cooperatively assess compatibility between video coding algorithms used for the support of communicative and distribution video services with SG 9

ITU-R SG 11

Responsible for, amongst other tasks:

- Recommendations defining and assessing the subjective and objective performance of digital television coding schemes proposed by SG 9 and supported on the B-ISDN proposed by SG 13;
- Cooperatively asses compatibility of end-to-end performance levels of television and communicative service on the B-ISDN with SG 1.

IEC/ISO

Responsible for, amongst other tasks:

- Through the Moving Picture Experts Group (MPEG), develop generic standards for the representation of compressed digital video and associated audio signals.

4. Range of Services

SG 13 Recommendation I.211, "B-ISDN Service Aspects," identifies two broad service categories; interactive and distribution.

- 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 services type and thus must be considered when developing a framework for B-ISDN video coding studies.

While it is anticipated that B-ISDN will eventually become the ubiquitous telecommunications network, serving all market sectors, the business sector will be first to make use of the new network's features. Indication from computer manufactures suggest there is significant need for broadband, reserved capacity telecommunications to support the emerging generation of multimedia computer equipment. This equipment will integrate the presentation of a wide range of video material with audio, graphics and text in many diverse applications. these will include access to remote information sources, windowed videophone applications and collaborative working methods. A wide range of video material, with different quality requirements, and displayed sizes and shapes, require support. Traditional communicative video service like videoconferencing are likely to have a smaller, but still significant, role and the greater bandwidth may encourage high quality conferencing systems. Despite these early service expectations, the longer term delivery of entertainment television and HDTV to domestic users must also be accommodated in the Integrated Video Services architecture.

5. Evolution to Integrated Video Services in B-ISDN

The development of B-ISDN Recommendations will follow a staged approach and 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 of 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.

Integrated Video Services (IVS) Baseline Document

Annex 1. Work Plan

A1.1 General

The attached tables represent simplistically the anticipated availability of standards related to video network, service and coding activities. the full details of these tables can be found in the relevant reports of the Study Groups.

STANDARDIZATION TIMETABLE 1 - ITU-T SG 13

STUDY GROUP 13 WORK PLAN APRIL 1994

Rec No	New/Rev	Question	Priority	Target	Pages	Area	Domain	Liaisons	Location	Subject
I.37x	N	01/13	M	Nov-94	25	4	10	SG1,11	R127,-	Teleaction
G.965	N	14/13	H	Nov-94	150	4	3	SG11	COM13-6	V5.2 interface for the support of access network
I.355	R	17/13	H	Nov-94	35	4,25	12	SG2,7 15,RSG4,7	R-	ISDN 64 kbit/s connection type availability performance
I.364	R	27/13	H	Nov-94	20	7	10	SG7	R-	Support of broadband connectionless data service on B-ISDN
Rec No	New/Rev	Question	Priority	Target	Pages	Area	Domain	Liaisons	Location	Subject
I.516	N	01/13	M	Jul-95	10	4	7	SG7		Modem interworking
I.311	R	02/13	H	Jul-95	10	7	10	SG11	R-	B-ISDN general network aspects
I.31x	N	02/13	H	Jul-95	20	7	10	SG11	R-	B-ISDN network requirements
I.374	R	03/13	H	Jul-95	10	4	10	JCG/AVMMS	R-	Network capabilities to support multimedia services
I.361	R	05/13	H	Jul-95	20	7	10		R116	B-ISDN ATM layer specification
I.150	R	05/13	H	Jul-95	3	7	10		R116	B-ISDN ATM functional characteristics
I.365x	N	06/13	H	Jul-95	100	7	10	SG7,11	R-	Service specific coordination functions to provide CONS
I.cots	N	06/13	M	Jul-95	20	7	10	SG7,11		SSCF to provide COTS
I.610	R	07/13	H	Jul-95	35	7	13	JCG/TMN	R116,-	OAM in B-ISDN access
I.580	R	09/13	H	Jul-95	30	7	7		R3,-	Interworking between B-ISDN and 64 kbit/s based ISDN
I.58x	N	09/13	M	Jul-95	30	7	7	SG11		Network and terminal compatibility
I.520	R	10/13	M	Jul-95	15	4	7			Network interworking between ISDNs
I.525	R	10/13	M	Jul-95	25	4	7			Interworking between ISDN and networks less than 64 kbit/s
I.372	R	11/13	M	Jul-95	15	4	7			Frame relay network to network
I.555	R	11/13	M	Jul-95	20	4	7		R-	Frame relay interworking
G.960	R	12/13	M	Jul-95	40	4	30		R107	Digital section for ISDN basic rate access
G.961	R	12/13	M	Jul-95	130	4	30		R107	Digital xmission system for basic access metallic lines
G.962	R	12/13	M	Jul-95	60	4	30		R108	Digital xmission system for 1ry acess at 2048 kbit/s
G.963	R	12/13	M	Jul-95	30	4	30		R108	Digital xmission system for 1ry acess at 1544 kbit/s
I.420	R	12/13	M	Jul-95	1	4	3			Basic rate user/network interface
I.421	R	12/13	M	Jul-95	1	4	3			Primary rate user/network interface
I.430	R	12/13	M	Jul-95	120	4	3		R119	Basic rate user/network I/F layer 1 spec
I.431	R	12/13	M	Jul-95	50	4	3		R119	Primary rate user/network I/F layer 1 spec
G96x	N	13/13	M	Jul-95	10	7	3,10	SG15	R97	Access digital section for B-ISDN
I.432	R	13/13	H	Jul-95	50	7	3		R119	B-ISDN user-network interface layer 1 spec
G9xx	N	14/13	H	Jul-95	20	4	3,10	SG15		Access network functional chacteristics
I.32x	N	15/13	H	Jul-95	35	4,7	28	SG7	R-	B-ISDN protocol reference model
I.32y	N	15/13	H	Jul-95	30	4	28			VPN functional model
I.341	N	15/13	H	Jul-95	25	7	27		R98	B-ISDN connection types
I.353	R	16/13	H	Jul-95	15	4,25	12	JCG/QOS/NP	R115	Reference events for defining ISDN performance paramaters
I.35x	N	17/13	H	Jul-95	20	7	12	SG2,7,15,RSG4,7	R-	Availability performance for intl CBR digital paths =/> the primary rate
G.821	R	19/13	H	Jul-95	15	4	12	SG2,4,15,RSG4,9	R8	Error perf of an ISDN intl digital connection at a bit rate < the timing rate
I.352	R	20/13	M	Jul-95	20	4,25	12	SG2,7,11	R114	Network performance objectives for connection processing delays in an ISDN

G.81s	N	21/13	H	Jul-95	20	12,4	8	SG15	R-	Slave clock requirements for SDH operation of intl digital links
I.373	R	22/13	H	Jul-95	20	10	10	JCG/UPT		Network capabilities to support UPT
I.5xz	N	22/13	H	Jul-95	15	10	7	JCG/UPT		UPT mobile interaction
G.ata	N	23/13	H	Jul-95	50	7	27	SG15		Access transport functional architecture
G.atmta	N	23/13	H	Jul-95	50	7	27			ATM transport network architecture
G.tna	N	23/13	H	Jul-95	50	7,11,12	27	SG15		General transport network architecture
G.831	R	24/13	M	Jul-95	10	12	10,13		R6	Management capabilities of transport network based on the SDH
G.804	R	25/13	H	Jul-95	25	7	30		COM13-13	ATM cell mapping into PDH
G.832	R	25/13	H	Jul-95	15	7	30		COM13-5	Transport of SDH elements on PDH networks
I.114	R	26/13	H	Jul-95	10	10	1,10	JCG/UPT	R3	Vocabulary of terms for UPT
Rec·No	New/Rev	Question	Priority	Target	Pages	Area	Domain	Liaisons	Location	Subject
I.334	R	01/13	M	Feb-96	5	4	10	SG2		ISDN numbering/addressing principles
I.460	R	01/13	M	Feb-96	3	4	2			Multiplexing, rate adaption and support I/F
I.500	R	01/13	M	Feb-96	3	4	7			I/W structure
I.5xy	N	01/13	M	Feb-96	15	6	7	SG11		IN interworking
I.37s	N	01/13	M	Feb-96	15	16	10	ICG/SAT	R-	Satellite requirements
I.37w	N	01/13	M	Feb-96	15	15	10	ICG/FPLMTS	R-	FPLMTS network requirements
I.375	N	03/13	H	Feb-96	15	4	10	JCG/AVMMS		Multimedia network requirements
I.37z	N	03/13	H	Feb-96	10	4	23,24	JCG/AVMMS		Multimedia services management
I.5mm	N	03/13	H	Feb-96	20	4	7,23	JCG/AVMMS		Interworking of multimedia services
I.sigr	N	04/13	H	Feb-96	20	7	6	SG11	R-	B-ISDN signalling requirements
I.363x	R	06/13	M	Feb-96	60	7	10	SG11,15	R116,-	AAL types 1,2
I.363y	R	06/13	M	Feb-96	100	7	10	SG11	R116	AAL types 3,4,5
I.core	N	06/13	M	Feb-96	20	7	10	SG7,11	R-	Convergence function for SSCOP above the FR core service
I.371	R	08/13	M	Feb-96	35	7	12,13	SG2,11	R4,-	Traffic control and congestion in B-ISDN
I.581	N	09/13	M	Feb-96	15	7	7		R-	Interworking requirements for B-ISDN
I.570	R	10/13	M	Feb-96	20	4	7	ISO/IEC JTC1		Public/private ISDN interworking
I.6xx	N	11/13	H	Feb-96	20	4	25			Frame relay bearer service OAM
G.964	N	14/13	H	Feb-96	150	4	3	SG11	COM13-14	V5.1 interface (based on 2048 kbit/s) for the support of access network
I.130	R	15/13	H	Feb-96	5	4	19,10	SG11		Method for the characterisation of ISDN services and network capabilities
I.140	R	15/13	H	Feb-96	25	4	19,10	SG11	R-	Attribute technique for characterisation of ISDN services and network capabilities
I.310	R	15/13	H	Feb-96	20	4	10			ISDN network functional principles
I.320	R	15/13	H	Feb-96	20	4	28	SG7		ISDN protocol reference model and its application
I.321	R	15/13	H	Feb-96	10	7	28			B-ISDN reference model and its application
I.324	R	15/13	H	Feb-96	20	4	27			ISDN network architecture
I.325	R	15/13	H	Feb-96	15	4	27			Reference configuration for ISDN connection types
I.327	R	15/13	H	Feb-96	15	7	27			B-ISDN functional architecture
I.32z	N	15/13	H	Feb-96	30	4	28			Reference model for mobility
I.340	R	15/13	H	Feb-96	20	4	27			ISDN connection types
I.350	R	16/13	M	Feb-96	15	4,25	12	JCG/QOS/NP	R114	General aspects of QoS and NP

I.351	R	16/13	M	Feb-96	10	4,25	12	JCG/QOS/NP	R114,-	Relationship among ISDN performance
I.356	R	16/13	H	Feb-96	40	7	12	SG2,15,RSG4,9	R2,R-	B-ISDN ATM layer cell transfer performance
I.35ba	N	17/13	H	Feb-96	20	7	12	SG2,7,15,RSG4,7	R-	B-ISDN availability performance
G.826	R	19/13	H	Feb-96	25	7	12	SG2,4,15,RSG4,9	R2,R-	Error perf parameters and objectives for intl CBR digital paths </= the 1ry rate
I.354	R	20/13	M	Feb-96	20	4,25	12	SG2,7	R115	Network performance objectives for packet mode communication in an ISDN
I.35bcp	N	20/13	H	Feb-96	20	7	12	JCG/QOS/NP,RSG4,7		ISDN connexion processing performance
I.35d	N	20/13	H	Feb-96	20	4,25	12	JCG/QOS/NP	R100	ISDN connexion processing accuracy and dependability performance
G.810	R	21/13	H	Feb-96	5	4	8		R86,R100	Considerations on timing and synchronisation issues
G.811	R	21/13	H	Feb-96	5	4	8		R86,R100	Primary ref clock requirements for plesiochronous operation of intl digital links
G.812	R	21/13	H	Feb-96	6	4	8		R86,R100	Slave clock requirements for plesiochronous operation of intl digital links
G.822	R	21/13	H	Feb-96	20	11	8,12	SG15		Controlled slip rate objectiveson an international digital connexion
G.823	R	21/13	H	Feb-96	20	11	12,8	SG15	R106,128	Control of jitter and wander in digital networks based on the 2048 kbit/s hierarchy
G.824	R	21/13	H	Feb-96	10	11	12,8	SG15	R106	Control of jitter and wander in digital networks based on the 1544 kbit/s hierarchy
G.825	R	21/13	H	Feb-96	10	12	12,8	SG15	R106,128	Control of jitter and wander in digital networks based on the SDH
I.35z	N	22/13	H	Feb-96	10	15	12	ICG/FPLMTS		Framework for mobile performance
I.37y	N	22/13	H	Feb-96	10	10	10	JCG/UPT		UPT network requirements
G.803	R	23/13	H	Feb-96	70	12	27		R106	Architecture of transport networks based on the SDH
G.8int	N	25/13	M	Feb-96	10	12	7	SG15,RSG4,9		Interworking between SDH and other digital hierarchies
G.Rsint	N	25/13	M	Feb-96	10	7	30			Principles / guidelines for the integration of radio and sat systems into sdh
I.112	R	26/13	M	Feb-96	20	4	1			Vocabulary of terms for ISDNs
I.113	R	26/13	M	Feb-96	20	7	1			Vocabulary of terms for broadband aspects of ISDN

STANDARDIZATION TIMETABLE 2 - SG 9

2. SG 9 Digital Secondary Distribution*	1992	1994	1994+
	Outline specification for TV/HDTV coding methods	Draft Recommendation for multi-resolution & multi-service**	Recommendation for HDTV coding methods (1996)
2.1 STM/ATM	STM/ATM	STM/ATM	STM/ATM
2.2 CBR/VBR	CBR and/or VBR	CBR and/or VBR	CBR and/or VBR
2.3 Bit Rate	Equivalent to 1~0.5 bit/pixel	Equivalent to 1~0.5 bit/pixel	Equivalent to 1~0.5 bit/pixel
2.4 Design Features	Hierarchical (multi-resolution) coding: two levels minimum	multi-service**	multi-service**
2.5 Service Types	SDTV/EDTV/HDTV SEcondary Distribution		
2.6 Other Features	Commonality or compatibility with H.262, MPEG 2 & Digital Emission Coding	Commonality or compatability with H.262, MPEG 2 & Digital Emission Coding	Commonality or compatibility with H.262, MPEG & Digital Emission Coding
2.7 Interworking			
2.8 Dependencies	ITU-R SG11 SG 15, MPEG SG 13	ITU-R SG11 SG 15, MPEG SG 13	ITU-R SG11 SG 15, MPEG SG 13

* SG 9 is also working in the field of digital contribution in accordance with Questions 28/9 and 29/9.
** Provision will be made to carry more than one TV program in a channel.

STANDARDIZATION TIMETABLE 3 - ITU-T SG 15

3. ITU-T SG 15 ATM Video Coding Experts Group	1994	1994+
	Recommendation H.262 (video coding) completed	Rec. H.22x, H.32x Rec. AV.222 (Conversational system) AV.223 (Distribution system) AV.245 (Communication procedure) AV.25X (Audio coding) AV.321 (Broadband video telephone) AV.331 (Multipoint system) AV.42X (Call control
3.1 STM/ATM	ATM	ATM
3.2 CBR/VBR	CBR and/or VBR	
3.3 Bit Rate	Range up to several tens of Mbit/s	
3.4 Design Features	- Universal coding (in terms of services, quality resolution, application and bit rate) - Extension capability to HDTV quality	
3.5 Service Types	- Conversational - Distribution - Retrieval	
3.6 Other Features	Compatibility with H.261, MPEG 2 & SG 9	
3.7 Interworking	Terminal interworking	
3.8 Dependencies	SG 13 - AAL Spec SG 13 - QOS and network performance SG 1 Stage 1 Service Descriptor MPEG - Generic Coding SG 9- Secondary Distribution Coding	

STANDARDIZATION TIMETABLE 4 - ITU-R SG 11

4. TV BROADCASTING		1993	1994	1994+
TG 11/3	New Recommendation Digital Terrestrial Broadcasting		6MHz (except Japan)	7MHz and 8 MHz
WP 11B				
	User requirement for Secondary distribution	for conventional TV	for High-definition TV	
	Generic coding functional requirement			
	User requirement for bit-reduction coding system		Draft Recommendation for conventional, enhanced and high-definition TV	
	Bit - reduction coding standards to be used in Studios			Recommendations
WP 11D				
	Program Delivery Control (PDC)		Recommendations	
	Conditional Access Broadcasting Systems		Recommendations	
	Reference Model for data broadcasting		Recommendations	
	Broadcasting of time and data information in coded form		Recommendations	

STANDARDIZATION TIMETABLE 5 - ITU-R SG10

RADIO BROADCASTING	1993	1994	1994+
TG 10/2			
Digital Audio Bit-Rate reduction techniques for emission contribution, distribution and Commentary/NCA applications Draft Recommendation			
WP 10C			
Higher bit rate in studio production		Revision of Rec. 646	
Digital Interface for BC Studios			Revision of Rec.647
User Data Channel			Revision of Draft New Rec.
Measurement of Digital Audio Equipment	New Rec.		New Rec.
Digital Quality Parameters	New Rec.		New Rec.
Subjective Assesment for Digital Parameters with revision of Rec.562			

STANDARDIZATION TIMETABLE 6 - MPEG

5. MPEG Generic audiovisual coding	1994	1994+
	MPEG2 (CD Nov. 1993)	(MPEG2 1995) MPEG4 IS 1997
5.1 STM/ATM	(Independent of transmission or storage medium)	(Independent of transmission or storage medium)
5.2 CBR/VBR	CBR and / or VBR	CBR and / or VBR
5.3 Bit rate	Up to several tens of Mbit/s	of order 10 kbit/s
5.4 Design features	Generic coding	Very low rate
5.5 Servise types	Digital VTR Digital disk Cable/satellitr/terrestrial TV Telecommunications	Restricted bandwith communication channels Restricted storage apps. Portable and hand-held displays
5.6 Other features	Low delay mode Scalability Compatibility Cell loss resiliance	
5.7 Interworking	Compatibility mode for interworking with H.261, MPEG1 Scalability mode offers interworking between terminals of different capabilities	PSTN videophone compatibility desirable
5.8 Dependencies	SG 9 SG 15	SG 15

STANDARDIZATION TIMETABLE 7 - ITU-T SG 1

Rec. Number	Title	Target Date	Priority
F.813	Virtual Path Service for Reserved and Permanent Communications	Sept 1994	H
F.MDS	Multimedia Distribution Services	Feb. 1996	H
F.310	Broadband videotex Services	Feb.1996	M
F.722	Broadband video-telephony Services	May 1995	M
F.732	Broadband video-conference Services	May 1995	M
F.821	Broadband TV distribution Services	Feb.1996	M
F.822	Broadband HDTV distribution Services	Feb.1996	M
F.MDV	Multimedia Delivery Services	Feb.1996	H
F.UCTBS	Broadband Unstructured circuit transport bearer service	May 1995	H
F.TG704	Broad band Transparent transport of G.704 frame bearer service	May1995	H
F.FG704	Broadband Transport of individual time slot frames of a G.704 frame bearer service	May 1995	H
F.N64	Broadband n*64 kbit/s bearer service	May 1995	H
F.811	Broadband Connection Oriented Bearer Service	May 1995 Revised	H

Integrated Video Services (IVS) Baseline Document

Annex 2. Network Aspects

Note: With the ending of the current study period of ITU-T SG 13 new or enhanced versions of I-series Recommendations relating to Broadband ISDN have become available. While Annex 2 of the IVS Baseline attempts to reflect major impacts on video service support, the original text of Recommendations should be used as the basis of detailed assessments.

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 characterized in many ways, including:

The direction of information flow: video services may be bidirectional, e.g., videotelephony and videoconference, or essentially unidirectional, e.g., video distribution services for business and entertainment.

The symmetry of information flow: messaging, retrieval and distribution services are characterized by asymmetrical information flows.

The origin of the source material: how video signals enter the network (e.g., direct from camera, from storage media, via satellite or other delivery mechanisms) can also provide a means of characterizing service information flows.

<u>Telecommunications Services</u>

The term service is used and understood in many different ways however, a recognized definition is given in CCITT Recommendation I.210. Two families of telecommunication services have been identified:

- Bearer services
- Teleservices

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

A teleservice provides the full capacity for communication by means of terminal and network functions, and possibly also functions provided by dedicated centers. Teleservices are end-to-end services, communication requires standardized functions within the terminals involved. All protocols of the seven layers of the OSI model must conform to the relevant ITU standards for communication to take place.

These two types of basic services may support 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.
- Multicast, broadcast 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 service 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,

A2.2.1 Application of Virtual Channel and Virtual Path Connections

Revised Recommendation I.311 (1993) identifies the following point-to-point applications of Virtual Channel Connections:

- User-user applications - in which the Virtual Channel Connection extends between the TB or SB reference points.
- User-network application - in which the Virtual Channel Connection extends between a TB or SB reference point and a network node. The user-network application of a VCC can be used to provide customer equipment access to a network element.
- Network-network application - in which the Virtual Channel Connection extends between two network nodes. The network-network application of the Virtual Channel Connection includes network traffic management and routing.

Revised Recommendation I.311 (1993) identifies the following applications of Virtual Path Connections:

- User-user application - in which the Virtual Path Connection extends between TB os SB reference points. The ATM network elements transport all cells associated with a VPC along the same route.
- User-network application - in which the Virtual Path Connection extends between a TB or SB reference point and a network node.
- Network-network application - in which the Virtual Path Connection extends between two network nodes. The network-network application includes a network traffic management and routing. At the network nodes where the Virtual Path Connection is terminated, the Virtual Channels within the Virtual Path are switched or cross-connected to Virtual Channels within other Virtual Paths.

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. Signaling in necessary for the flexible implementation of multiparty and multiconnection calls for customers with multisite, multimedia communication needs over the full range of service bandwidths from low bit rate videotelephones to HDTV.

A2.3.1 B-ISDN Signaling Principles

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

- Capabilities to control ATM virtual channel path connections
  - Establish, maintain and release ATM Virtual Channel Connections (VCCs)s and Virtual Path Connections (VPCs). Establishment can be on-demand, semi-permanent or permanent, and should comply with requested connection characteristics.
  - 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. The simultaneous establishment of multiple connections should not be significantly slower than the establishment of a single connection.
  - Add and remove connection form an existing call.
  - Ability to correlate (when requested) connections composing a multiconnection call. This correlation is handled by the origination and destination B-ISDN switches, which may be public or private.
  - Reconfigure a multiparty call including an existing call or splitting the original multiparty call into more calls.

- Processing related function
  - 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 for service interworking.

A.23.2 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:

- Selection Switching

Distribution services of all types (e.g., 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 programs.

- 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 100ms.

- Supplementary service aspects for further study.

A2.4. Call and Connection Control

The mature 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 e.g., 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 Rate

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

The following areas relating to service bit rate still under investigation.

A2.6.1 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.

Recommendations I.211 also notes that the transfer over B-ISDN of signals at service bit rates above 135.631 Mbit/s (e.g., TV signals near 140 Mbit/s) specified in ITU-R Recommendation 721 requires further study.

A2.6.2 Maximum Service Bit Rate Supported by the 622.08 Mbit/s Interface

Agreement as been reached that the B-ISDN (I.413, I.432) at 622.08 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 Mbit/s is common and this results in an ATM cell transfer capacity of 599.040 Mbit/s. The maximum service bit rate which can be supported on this interface may be equal to less than 542.526 Mbit/s (ref. Recommendation I.211 Revised, 1993). The actual maximum service bit rate is for further study.

A2.6.3 Bit rate Assurances

Parameters for constant and variable it 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.

A2.6.4 The Specification of Service Bit Rate Parameters

- constant and variable bit rates are expressed by a number of parameters related to the traffic characteristics described in Recommendation I.371.
- the time period over which the rate is specified is influenced by service timing and buffering constraints and the capabilities oc 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., 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.
- multiple parameters may be require if a unique time period cannot be agreed as meeting the requirements of all services.
- the parameters selected must be of a form and nature which allows the network to exercise the option of statically multiplexing VBR services, where appropriate, in a manner which does not violate the agreed QOS.
- for reasons including network operation, interworking and service development - a number of specific bit rates will be standardized. This comment applies to both constant and variable bit rate service support.

A2.6.6 Allocation and control of network resources

- does not present new problems for CBR services
- there may be advantages form the statistical multiplexing of uncorrelated VBR sources. The applicability of VBR coding to specific service types is for further study. (Ref I.211 Revised - 1993)

A2.6.7 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 rely on AAL protocols which are B-ISDN service specific, nor on higher layer protocols which are application specific.
- The design of an optimal set of ATM layer traffic controls should trade-off minimizing network and end-system complexity for maximizing network utilization.
- ATM layer traffic controls should maintain the ATM layer Quality of Service even under congestion conditions.

A2.6.8 Traffic Control and Congestion Control - The Impact on Video Services

Draft Recommendation I.371 entitled Traffic Control and Congestion Control describes the parameters and procedures necessary to protect the network and user in order to achieve network performance objectives.

Traffic Contract

A traffic contact at the TB reference point is specified between the user and network for each ATM connection.

In establishing a connection on an ATM network, a suer must specify the characteristics of the offered traffic and the required QOS for the connection. Additional information describing the maximum cell delay variation is also required. It has not yet been decided whether this cell delay variation tolerance is negotiated on a subscription on per connection basis.

Traffic Specification

The Source Traffic Descriptor is a set of traffic parameters used during connection setup to capture the intrinsic characteristics of the connection requested by the source. So far, only the peak cell rate of an ATM connection has been specified. However, a user may request two levels of priority for an ATM connection as indicted by the Cell Loss Priority (CLP) bit. In this case, traffic parameters describing both the high priority flow (CLP=0) and aggregate flow (CLP=0+1) are included in the source traffic descriptor.

Quality of Service

The ATM layer QOS is defined by a set of parameters such as delay, delay variation and cell loss ratio. The network provides and ATM layer QOS for both the high priority (CLP=0) and aggregate flow (CLP=0+1) components of an ATM connection. The network must meet the requested QOS as long as the user complies with the traffic contract.

Impact of Cell Delay Variation

When cells from two or more ATM connections are multiplexed, cell of one connection may be delayed while cells of another connection are being inserted at the output of the multiplexer. Similarly, some cells may be delayed while physical layer overhead or OAM requires that this cell delay variation be specified.

Usage Parameter Control/Network Parameter Control

The UPC/NPC mechanism should not discard or tag cells in an ATM connection if the source conforms to the Source Traffic Descriptor negotiated at connection establishment. A method to determine whether a traffic flow is conforming to the negotiated peak cell rate at a given interface is defined in Recommendations I.356 and I.371.

When an ATM connection uses the CLP capability as requested by the user, network resources are allocated to both high and low priority traffic flows. By allocating adequate resources and by controlling both the high priority (CLP=0) flow and the aggregate (CLP=0+1) flow, a network operator may provide the requested QOS classes.

A2.6.8.2 SG 15 ATM Experts Group Perspective

The Experts Group is concerned about UPC/NPC algorithm standardization. The NPC technique must be mirrored in the terminal to ensure that no violation of the network agreement (which could lead to discarded cells) occurred. It is therefore considered essential that, for anything other than peak rate monitoring, the UPC/NPC algorithm must be standardized; it cannot be left to individual operators to choose.

The Experts Group is interested in receiving information on the details of UPC/NPC algorithm development as they emerge. These algorithms may have a significant impact on video service provision an efficient utilization of network resources from a service point of view. Iw would be impractical to consider that terminals could adapt to different parameter control algorithms depending on which network, or combination of networks is used.
The Experts Group will continue to study UPC mechanisms form the video services viewpoint. Current indications are that Leaky Budget techniques have some advantages in terms of efficient implementation.

A2.6.9 Cell Loss Resilience and Coding Aspects

There are three factors which contribute to cell loss:

i) header errors due to transmission errors, and consequent cell discard to minimize misdelivery;
ii) overflows or underflows of the smoothing buffer in the adaptation layer, due to large variations in queuing delay;
iii) buffer overflows in the ATM layer due to instantaneous cell traffic overload.

Two possible approaches dealing with cell loss have been recognized: One deals with a two-channel transmission scheme in which all the information essential for the reconstitution of the picture is transmitted with a high QOS in one channel, and the remaining information is transmitted with a lower QOS in the second channel which, due to network loading, may be subject to cell loss. The second approach is a cell loss compensation scheme using a Reed-Solomon random error corrector with octet interleaving. this is the method used in Rec. I.363 for AAL Type 1 for video applications.
As noted in Recommendation I.211 (Revised 1993), the layered signal approach offers improved cell loss protection. Decomposing the coded video information into separate layers allows information from different layers to be placed in "separate" cell streams using the CLP indicator. This technique therefore provides some measure of statistical error protection. The available information on this subject is given in Rec. I.356.

A2.7. Quality of Service Aspects

Customer control of video and image service 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 characterizing 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 factor involved:

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

Quality of service is defined in 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 Recommendation I.350 the relationships described in these terms, "the user oriented QOS values provide a valuable framework for network design bu they are not directly usable in specifying performance requirements for particular connections. Similarly, the NP parameters primarily determine the QOS, but they do not necessarily describe the quality in a way that is meaningful to users". Both types of parameters are needed and their values qualitatively related if a network is to be effective in serving its user.

A2.7.1 Quality of Service Indication and Negotiation

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

- QOS related to Virtual Channel Connections (VCCs)

  The user of a VCC is provided with one of a number of QOS classes supported by the network. The QOS class of a given connection will not change for the duration of the connection, any renegotiation of the QOS class may require establishment of a new connection.

- QOS related to Virtual Path Connections (VPCs)

  A user with a VPC is provided with one of a number of QOS classes supported by the network. The QOS class will not change for the duration of the VPC.

Several Recommendations make reference to QOS negotiation issues.

- Recommendation I.211 (Revised 1993), indicates QOS is negotiated at all 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. For several reasons, including network operation, interworking and service development, a limited number of specific QOS will be standardized.

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.

The Cell Loss Ratio for high priority cells will be defined, and assured by the network if cell traffic does not exceed the negotiated values. Handling of low priority cells is for further study.

- 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 fo r the duration of the call. Changes to these parameters may be negotiated within call period and the details of this negotiation are for further study. In both cases, a set of discrete bit rates will be chosen.

A2.7.2 General Aspects of ISDN Performance

A2.7.2.1 Performance Parameters

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.356 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 (Recommendation 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.356 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 of 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 SG 13/6 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 e -9 to e -10. This particularly the case for high bit rate video services.

- 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 synchronization.

- 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 filed. The need for and methods of measuring the severely errored cells 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.

A2.7.2.2 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.

Misinserted 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.

When dummy information is provided to AAL users, a primitive is provided, giving the status of this information (invalid). For services requiring a high quality of service, a method for the correction of cell losses is provided in Rec. I.363.

- Errored and severely Errored Cells

Bit errors occurring in the ATM cell information field are Transferred to the user as they occur. It is possible however that some AAL types may improve bit error and cells loss performance provided by the ATM layer. These AAL issues will continues to be studied in SG 13 who will also continue studies on the relationship of transmission network performance objectives to ATM network performance.

- 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 of dummy information and result in bit errors in user information.

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 bonded, network delays.

Cell jitter (the variable delay in cell arrivals) must be buffered within 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 service components of a multimedia service, particularly the video and audio, is both bounded and acceptable.

The need for end-to-end timing recovery has been recognized by the ATM Experts Group of SG 15. Precise requirements are under consideration. The availability of a network reference clock will be essential to ensure timing recovery necessary for high quality video applications.

Additional information with respect to AAL1 can be found in Annex 3.

A2.9 Network Parameters Impacting on Video Coding Definition

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

The areas are requiring clarification are listed in the following

A2.9.1 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 recognized that there is 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, through, 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 SG 15 Experts Group believes that 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 ratio is sufficiently small, no cell loss protection may be necessary. Table 1 provides some network performance requirements obtained from some example service quality figures. The table concentrates on bit error and cell loss error correction techniques. Layered coding concealment techniques are however under consideration and lead to different figures.

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

A2.9.2 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 to expensive and delay maybe excessive for conversational services if multiple consecutive lost cells must be detected and corrected. Cell loss burst behavior may be modelled by the Gilbert model (a two-state Markov model requiring four transition probabilities, with one state representing no cell loss and 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 form that of low priority cells and, if so, how?
- How can we estimate the average interval time, T, in which no cell occurs? If T>>1 (bit rate x CLR), the requirement for CLR might be relieved.

Annex 2 Table 1: SERVICE AND NETWORK REQUIREMENTS

Service	Bit Rate	QOS Req. (***)	Required BER/CLR without error handling in AAL	AAL Type	Required BER/CLR after single bit rate correction on cell basis on AAL (*)	Required BER/CLR after single bit EC on cell basis & addit. cell loss correction in AAL (**)
Communication						
Videophone	64 kbps/2 Mbps CBR (H.261)	30 min error free	BER< 1.e-6 CLR<1.e-7 (BCH (511.4 93) FEC in user layer	Type 1	In user layer	BER<... CLR<8.e-5
Videophone	2 Mbps VBR	30 min error free	BER<3e-10 CLR<1e-7	Type 2	BER<1.2e-6 CLR<4e-8	BER<2.3e-5 (CLR=1.e-6) CLR<8e-5
Videoconference	5 Mbps VBR	30 min error free	BER<1e-10 CLR<4e-8	Type 2	BER<8e-7 CLR<4e-8	BER<1.8e-5 (CLR=1e-6) CLR<5e-5
Videodistribution						
TV Distribution	20-50 Mbps VBR	2 hrs error free	BER<3e-12 CLR<1e-9	Type 2	BER<1.2e-7 CLR<1e-9	BER<6e-6 (CLR=1.e-6) CLR<8e-6
MPEG 1 core	1.5 Mbps VBR	20 min error free	BER<4e-10 CLR<1e-7	Type 2	BER<1.4e-6 CLR<1e-7	BER<2.5e-5 (CLR=1.e-6) CLR<9.5e-5
MPEG 2 core	10Mbps VBR	30 min error free	BER<6e-11 CLR<2e-8	Type 2	BER<5.4e-7 CLR<2e-8	BER<1.5e-5 (CLR=1.e-6) CLR<4.e-5

Table 1 (continued)

(*) Payload scrambling polynomial 1+x**43 produces double, correlated bit errors

(**) Based on parity cell built from 31 consecutive data cells. The cell losses are assumed to be isolated. With this simple correction scheme, single cell losses are corrected if combined with cell loss detection by cell numbering. Also non-corrected but errors in a cell are handled by replacing this faulty cell by a dummy correlated errors due to this cell by the cell parity mechanism. The BER calculations are done in the assumption that all double ATM link errors (2 times 2 correlated errors due to payload scrambling) can be detected.

(***) QOS requirements, as visualized by viewers; not directly related to channel errors.

Notes:

- These values are calculated under the assumption that cell losses are isolated. If cell losses lend to occur successively, another cell loss ration and another cell loss correction technique may be required.
- It was assumed that one cell loss always causes picture degradation. The visual perception of the picture, however, may be acceptable even if cell loss concealment technique is not used. Therefore there is a possibility that these requirements will be relaxed.

A2.9.3 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. The CLP bit may be over written (tagged) by the network, thus it cannot be relied on to provide an end-to-end indication.

A related issue is the use of the ATM header codepoints to support video/multimedia services. In this respect, the following understanding has been reached:

- The CLP bit remains separate from the 3 bit Payload Type field. However the significance of the CLP bit value when some Payload Type codepoints occur (i.e., resource management codepoints for example) remains open at this time.

- In the Payload Type Field a capability exists the support of user-user applications.

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?

A2.9.4 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; minimize the rate and keep it constant. For the B-ISDN (with the possible advantage 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 will be an important parameter that is monitored (Ref. to Rec. I.371).

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 33-40ms) 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 resources for low and high priority cell loss classes. Will the high priority cell loss class continues to exists in the future?

A2.9.5 Multimedia Connections

Multiplexing of multiple media has been carried out within the terminal device for circuit switched networks. The B-ISDN offers the flexibility to use the virtual channel (i.e., cell) based multiplexing instead. An important factor in the choice between terminal-based or cell-based multiplexing is whether there will be penalty caused by the use of an ensemble of virtual channels instead of one composite one, although the overall rate characteristic, for example, would be the same. The choice of multiplexing options must therefore take into consideration a number of variables, including:

- transmission cost
- control cost
- flexibility
- efficiency
- overall service performance

Cell sequence integrity within a Virtual Path Connection will be maintained to enable OAM performance monitoring. The availability of sequence integrity at this level will influence options available for the support video/multimedia services.

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 unlikely to be a problem unless the cumulative differential delay exceeds some tens of milliseconds from 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 form the standpoint of network resource management?
- At what stage of B-ISDN development will it be possible for the signalling and control needed to minimize and control cross-media delay be available?

A2.9.6 Bit Error Ratios

Cell payloads will be subject to 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 ratio are required by those working on video coding schemes for the B-ISDN.

For interworking between video codecs on 64 kbit/s ISDN networks, the B-ISDN bit ratio must be no greater than that for the 64kbit/s ISDN. It should also be noted that the Recommendation H.261 coding scheme for 64kbit/s ISDN provides bit error correction, so this would not be a necessary function of the AAL in this case.

SG 13 should work in close collaboration with the video coding experts to define any capability within the AAL concerning bit error detection or correction.

A2.9.7 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 fo 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.

A2.10 Access Network Issues

The ATM Experts Group of SG 15 recognizes that access to B-ISDN may, at least for a considerable interim period, be via other networks such as LANs and MANs. Video services must also be supported over these access networks. The implications of the differing network characteristics, in terms of resource allocation, timing requirements, protocol conversations, UPC control, etc, require study.

A2.11 Network Capabilities to Support Charging for B-ISDN Services

Recommendation I.311 (Revised) notes that services to be taken into account in B-ISDNs based on ATM technique include connection oriented as well as connectionless services, in different communication configurations as e.g., point-to-point, multipoint, (including multicast, broadcast) and other communication configurations.

Network capabilities to support the charging of these B-ISDN services are for further study.

A2.12 Multipoint Networking

A2.12.1 General

There exists an obvious need to consider the multipoint capabilities which may be required in both early and longer term B-ISDN. Applications requiring multipoint networking include broadcast and narrowcast video/multimedia and multicast data. The stage introduction of signalling capabilities agreed for B-ISDN indicates that initially multipoint connections would need to be established on the basis of user requests to be network provider, rather than established on demand using network signalling.

Relevant issues requiring further study include:

- the impact of multipoint access, including multiaccess and multi-CPE access. This would also include the development of procedures to co-ordinate the access of multiple terminals to a single interface.
  The need for multipoint access is for further study.

- multipoint network connectivity to allow the interconnection, at the network level, of more than one ATM source-destination pair, which are assumed to be on two or more physical interfaces. This will require ATM multicasting and also possibly ATM channel merging. Multicasting refers to the capability of an ATM switch or network to copy an input ATM cell stream and deliver it to multiple output ATM cell streams. ATM channel merging refers to the capability of a single ATM switch or network to combine multiple ATM cell streams and deliver to a single ATM output stream. Higher layer information is used to determine from which source a particular cell originated.

The need for limitations to the number of multicast or merged streams supported at UNI/NNIs is for further study.

A2.12.2 Connection Topologies

Multipoint connection topologies requiring consideration include:

- point-to-multipoint i.e., a single source broadcasting identical information to more than one endpoint. This basic topology can be expanded and used as a building block in describing other multipoint topologies.

- multipoint-to-point - Revised Recommendation I.150 notes that for multipoint-to-point virtual channel connection, cell sequence integrity is preserved for cells from each VCC endpoint of VCC.

- multipoint-to-multipoint

Further study is required of the arrangements suitable for ATM link implementation, identifier management, bandwidth issues, connectively descriptions, establishment and billing.

A2.12.3 Multipoint Functional Requirements

For further study

A2.12.4 Multipoint Performance Requirements

For further study

Integrated Video Services (IVS) Baseline Document

Annex 3. ATM Adaptation Layer

A3.1 ATM Adaptation Layer Type 1

AAL type 1 for circuit transport is stable and included in the 1993 version of Rec. I.363. Review and assessment of protocol for this layer service is useful and necessary to progress in development of AAL Type 1 and/or Type 2 for video signal transport, which is scheduled to be complete in 1995 by joint and cooperative work between SG 13 and coding groups.

This Annex 3 presents figures and explanations of AAL Type 1 as a supplement to I.363. The purpose of this Annex 3 is to provide tutorial information on AAL Type 1, with general background and overview information on some data transfer. It does not describe detailed procedures. Note that figures included in this Annex 3 are not to the same as those included in Recommendation I.363.

Five layer service provided by AAL type 1 to an AAL user;

1. Circuit transport (e.g., G.702 signals
   as 1.544, 2.048, 6.312, 8.448 Mbits.
   N-ISDN signals such as 64, 384, 1536, 1920 kbit/s)
3. Video signal transport
4. Voice-band signal transport
5. High-quality audio signal transport

AAL User

Specific layer service is realized by a DEFINED SET of CS functions and protocols. (All CS functions and protocols are not always necessary for a specific layer service.)

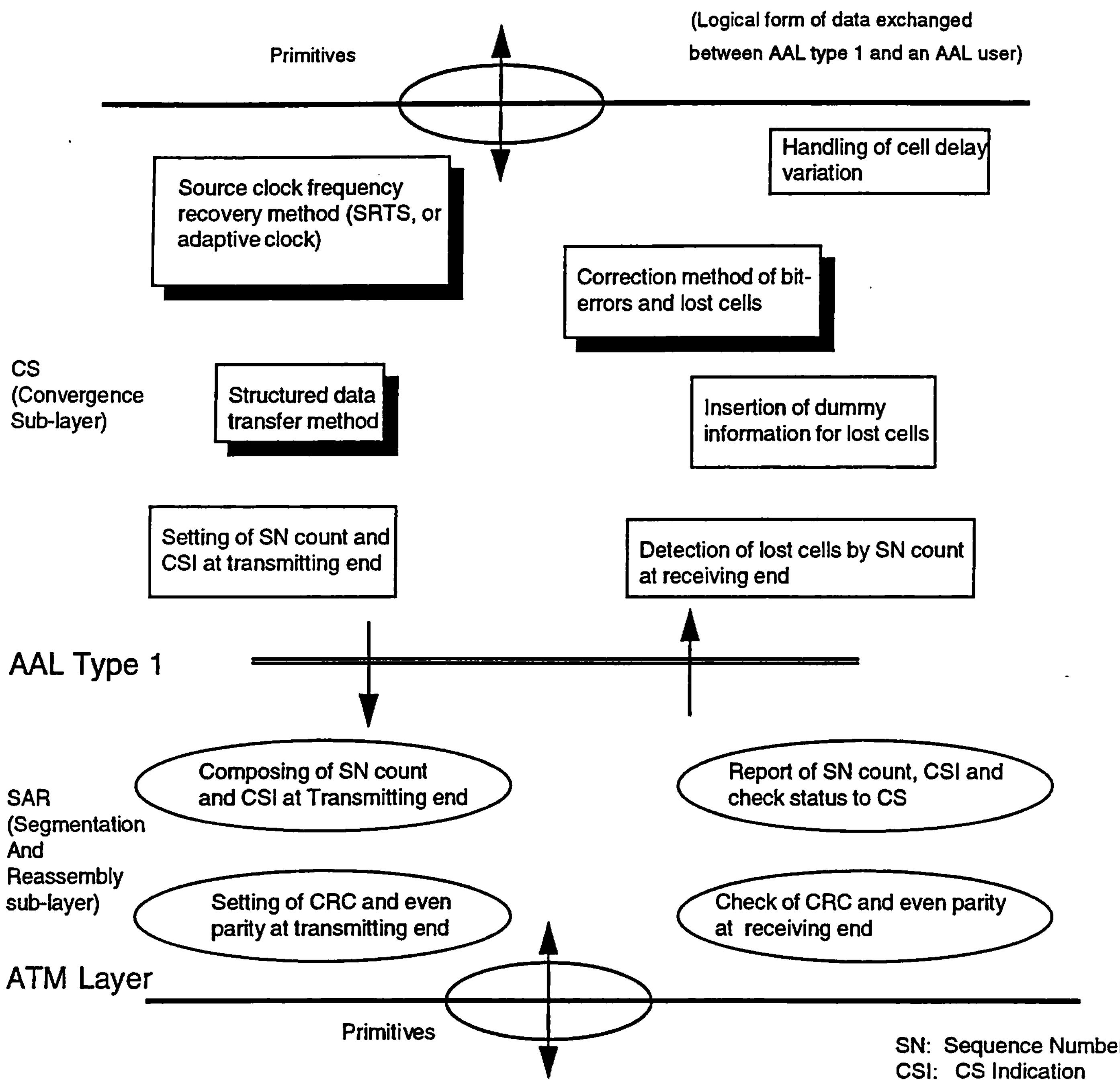


Fig. 1 Overall Structure of AAL Type 1

	SN field (4 bits)		SNP field (4 bits)		AAL user information field (47 octets)	
Cell header (ATM layer)	CSI	SN count	CRC	Even parity	P field (8 bits) (if necessary)	

CSI (CS Indication): 1 bit

- Conveyance of SRTS information, when SRTS is used, for SN count 1, 3, 5, and 7,
- Indicating existence of P field, when structured data transfer method is used, for SN count 0, 2, 4 and 6,
- Indicating the first cell of the octet-interleaved matrix, when error correction method of Reed-Solomon code combined with octet interleaver is used,
- CSI value is provided by CS to SAR at transmitting end, and reported by SAR to CS at receiving end.

SN count (Sequence Number count): 3 bits

- Counter numbered modulo 8,
- Counter value is provided by CS to SAR at transmitting end, and reported by SAR to CS at receiving end.

CRC (3 bits) end even parity (1 bit)

- Polynominal to be used is x3+x+1,
- Two modes of operation; Correction mode capable of single-bit error correction, and Detection mode capable of multiple-bit error detection.

P field (Pointer field): 8 bits

- This field is placed, when structured data transfer method is used,
- Indicating the first octet of the structured data within AAL user information field,
- The pointer should be used once every sequence count cycle (i.e. 8 cells).

Fig. 2 Format and Coding of AAL Type 1

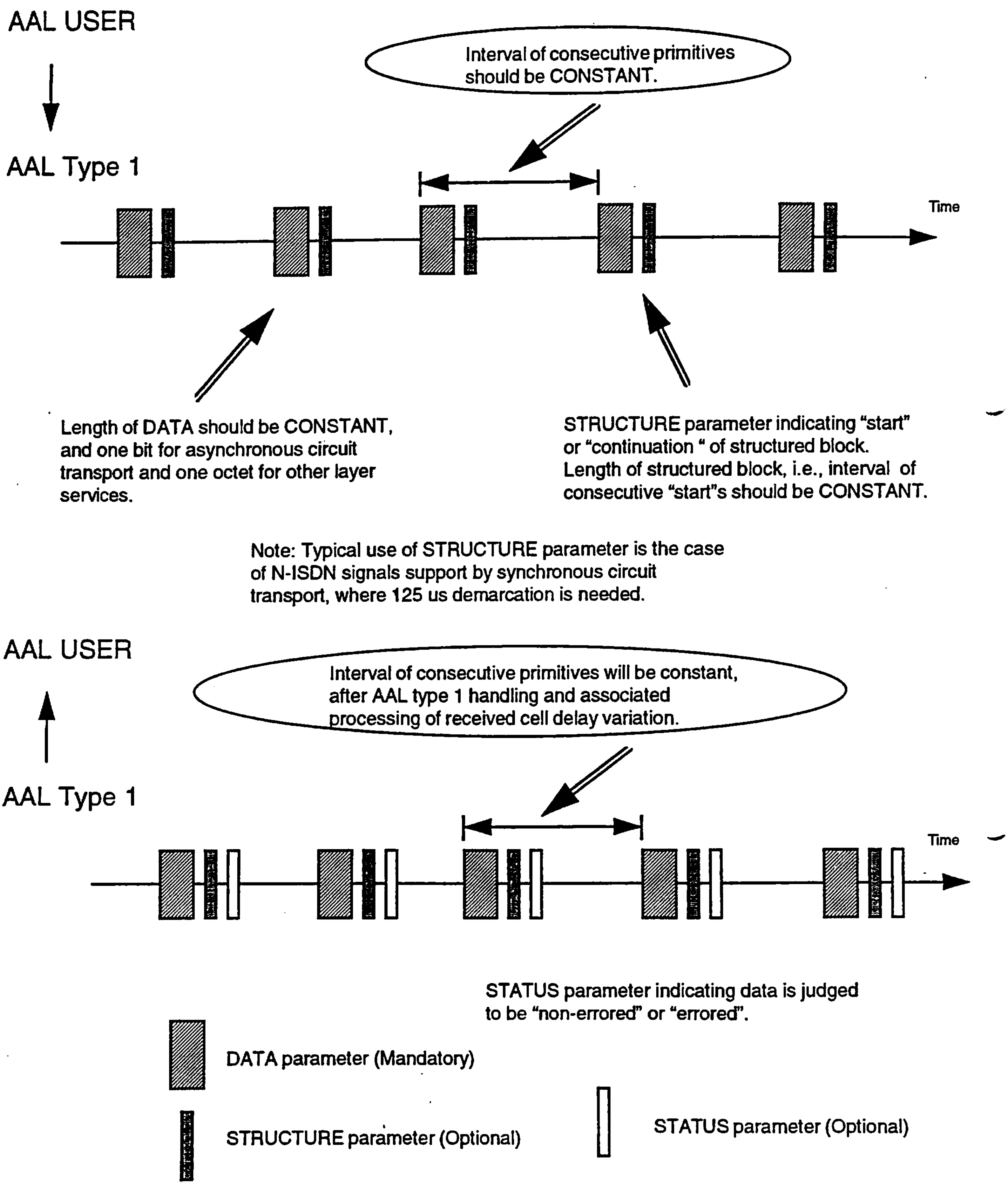
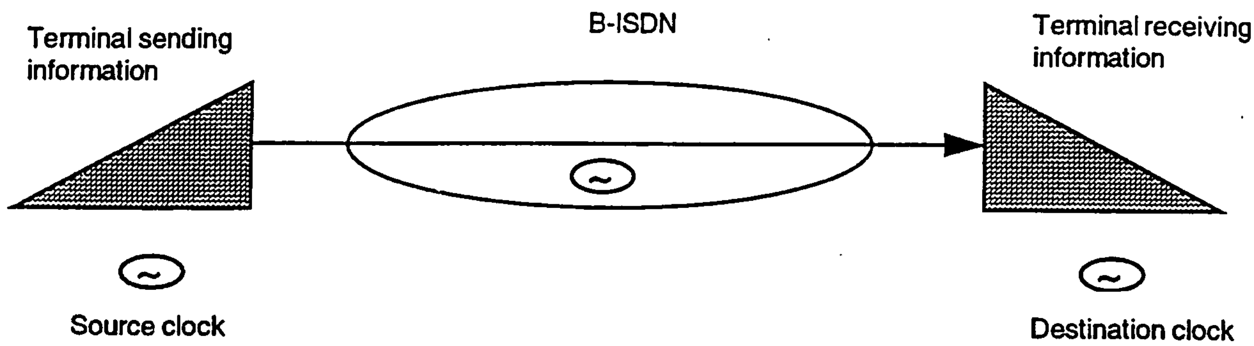


Fig. 3 Primitives between AAL Type 1 and an AAL User



Source clock frequency recovery is required when;

- Source clock is not locked to the network clock, and
- Destination clock should be locked to source clock.

Examples of need for source clock frequency recovery are;

- G.702 signals transport by asynchronous circuit transport, when source and destination clock are not locked to the network clock,
- Camera clock delivery from source to destination terminal, when sufficient jitter performance is required.

Note 1: Source clock frequency recovery is not always required for a given layer service. It will also depend on detailed layer service requirement such as jitter performance.

Three methods are recommended;

- Use of adaptive clock supported by AAL type 1 (See Fig. 5),
- Use of SRTS supported by AAL type 1 (See Fig. 6),
- Use of synchronization pattern within AAL user information flow (AAL type 1 is not involved for source clock frequency recovery).

Note 2: SRTS will provide for better jitter performance but require complicated protocol compared to adaptive clock method.

Note 3: For service aspects of timing and synchronization issues including source clock frequency recovery, see Recommendation I.211 section 2.5.

Fig. 4 Source Clock Frequency Recovery

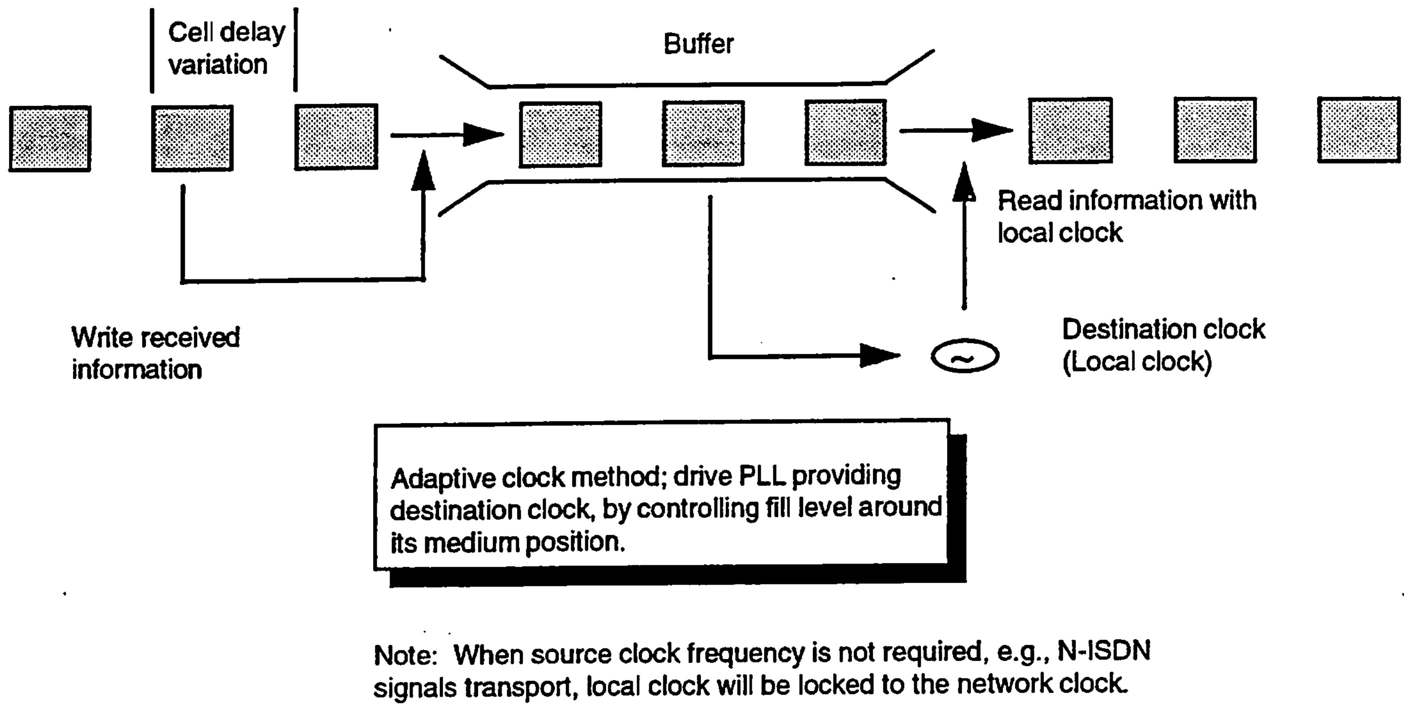


Fig. 5 Handling of Cell Delay Variation and Adaptive Clock

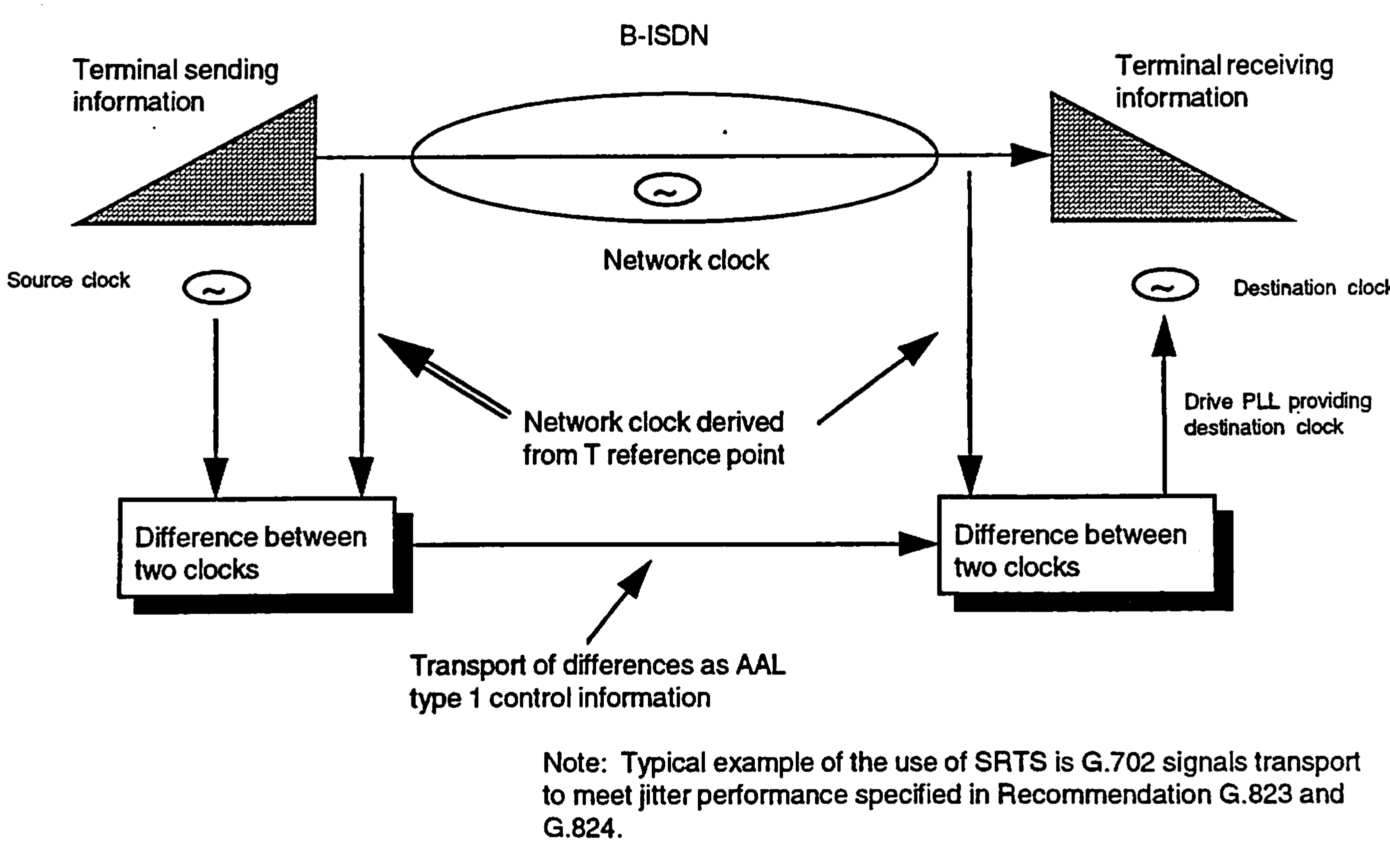


Fig. 6 SRTS (Synchronous Residual Time Stamp)

A.3.2 Status Report on AAL1&2 (AAL Types 1 and 2) for video signal transport

----

1. Introduction

Studies have been carried out between SG13, SG15 ATM Video Coding Experts Group and SG9 (formerly CMTT) on AAL1&2 for video signal transport through Liaison documents and the IVS (Integrated Video Services) Baseline Document. This document is an updated version on the issues being discussed, and summarizes the status of study from network viewpoints.

2. AAL1

2.1 Error protection

In the March 1993 version of I.363, SG13 specifies the method of correcting lost cells and bit errors in a cell payload, namely the long interleaver method, for distributive services according to requirement by SG9. At its July 1993 meeting, SG13 also developed an error correction method for delay sensitive services, namely the short interleaver method, based on Liaison documents from SG15 and SG9. These two methods are summarized below:

(1) Long interleaver method

- 128-cell interleaving matrix with (128, 124) RS code, and vertical reading at the transmitter.
- Error correction: 4 cell loss within 128 cells, 2 cell loss and 1 errored octet in each row of 128 octets, 2 errored octets in each row if there is no cell loss.
- Delay including both ends at AAL1-SAP: 2x124 cells, i.e., 2.65 ms (34 Mbit/s) and 2.01 ms (45 Mbit/s).

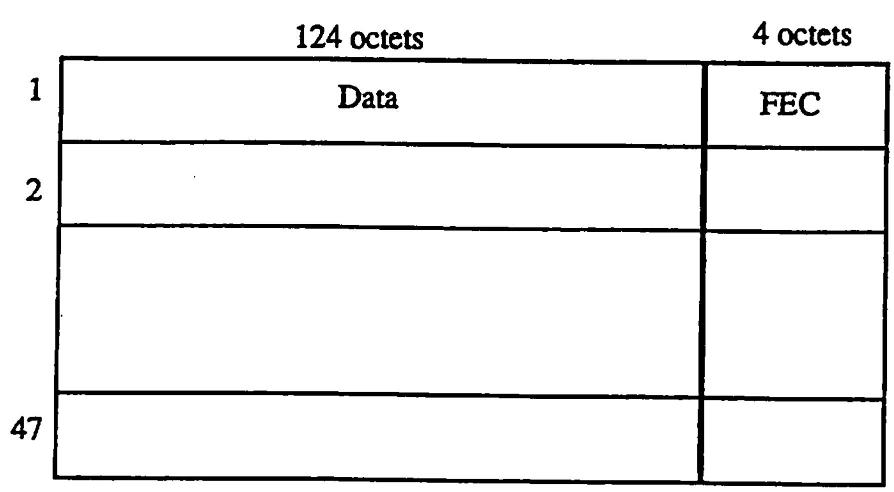


Fig. 1 Long interleaver method with vertical reading

(2) Short interleaver method

- 16-cell interleaving matrix with (94, 88) RS code, and diagonal reading at the transmitter.
- Error correction: 1 cell loss within 16 cells, or 3 errored octets in each row of 94 octets.
- Delay including both ends at AAL1-SAP: 14.7 ms (384 kbit/s), 3.67 ms (1536 kbit/s), 2.93 ms (1920 kbit/s).

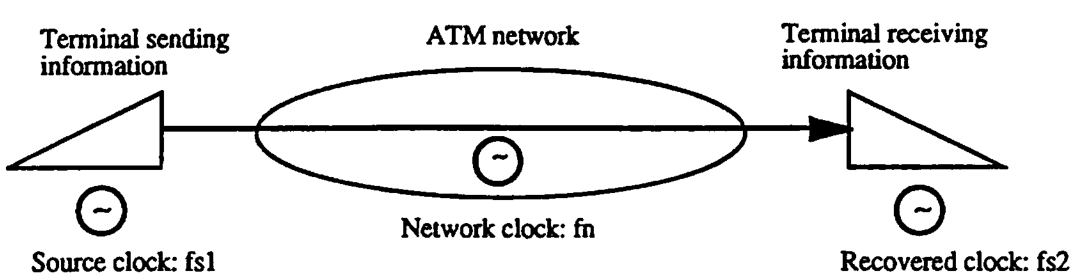
	88 octets	6 octets
1	Data	FEC
2		
8		

Fig. 2 Short interleaver method with diagonal reading

2.2 Source clock frequency recovery for asynchronous circuit transport

The term asynchronous, when applied to a CBR service, means that the bit clock of the CBR source at the ingress AAL1-SAP is not locked to a network clock. Recommendation I.363 specifies two methods whereby this source clock frequency can be recovered at the egress AAL1-SAP. The two methods are SRTS and adaptive clock.

The SRTS method, however, itself requires that a common network clock be available at the sending and receiving AAL1 entities. SG 13 is studying the issue of extending the SRTS method to a plesiochronous environment, i.e., an exact common clock is not available at both ends, and where CPE does not know the exact clock situation between two ends. Such a case is typical in international networks and a network comprising public and private networks.



Note: Source clock frequency recovery makes fs2 equal to fs1, where fs1/fs2 is not frequency locked to fn.

Fig. 3 Source clock frequency recovery of AAL1

3. Support of MPEG-2 signal transport over ATM network

It was reported that SG9 has a plan to study transport of MPEG-2 signals over an ATM network for digital TV distribution services, according to requirements from ITU-R SG 10 and

11. The AAL1&2 group has the following consensus:

- when MPEG-2 Transport Stream (TS) is supported by CBR services of an ATM network for digital TV distribution services, it is adequate to use AAL1 with/without a long or short interleaver method depending on error protection and delay requirements. The exact method of source clock frequency recovery of the AAL1 to be used also needs to be assessed.

- when MPEG-2 is supported by VBR services, the group wishes to have as much communality between H.262/H.22X and MPEG-2 as possible. TS may not be appropriate due to its redundancy with ATM functionality. A collaborative study with SG9 and SG15 (ATM Video Group) will be taken.

4. AAL2

4.1 Functions of H.22X

Fig. 4 depicts possible protocol stack of AAL2 and higher layers. AAL2 referred to hereinafter is AAL to support class B services, as defined in I.362. It is assumed that H.22X will support the following functions:

- Audiovisual multiplexing, i.e., multiplexing video, voice and data in terms of H.32X system aspects.
- Synchronization between higher layers, e.g., between video and voice.

In addition, support of CPE-to-CPE (in-channel) signalling (i.e., H.24X) may be provided by H.22X.

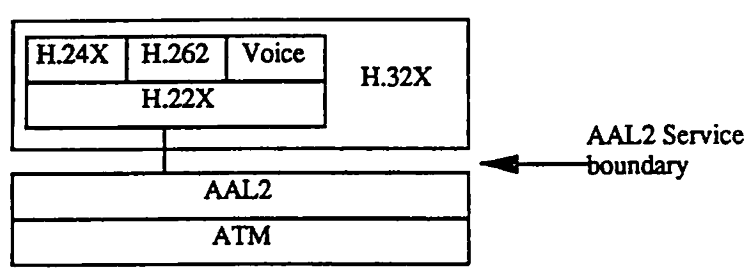


Fig. 4 Protocol stack of AAL2 and higher layers

4.2 Objectives of AAL2

The objectives of AAL2 are:

- to support interactive video/audiovisual applications, e.g., H.32X, in VBR services,
- to support distributive video applications in VBR services (ffs), and
- to maintain future expandability to support other possible applications.

4.3 Functions of AAL2

The following paragraph gives candidate functions and considerations regarding AAL2.

(1) Framing of H.22X data: cell-by-cell or packet with trailer

Since data exchanged between H.22X and AAL2 is variable length, AAL2 supports framing of such data. There are two alternatives for framing methods:

- cell-by-cell approach like AAL1, 3/4, and
- variable packet with trailer like AAL5.

When the packet method is adopted, protocol overhead will be reduced compared to the cell-by-cell method. It is anticipated, however, that in the packet method, a delay variation is introduced at the receiver due to variable waiting time, i.e., the receiver needs the whole packet for its trailer operation, and then passes the valid packet to H.22X.

In the following items (2) and (3), the two methods are further compared in terms of error protection functions and ATM user-to-user indication.

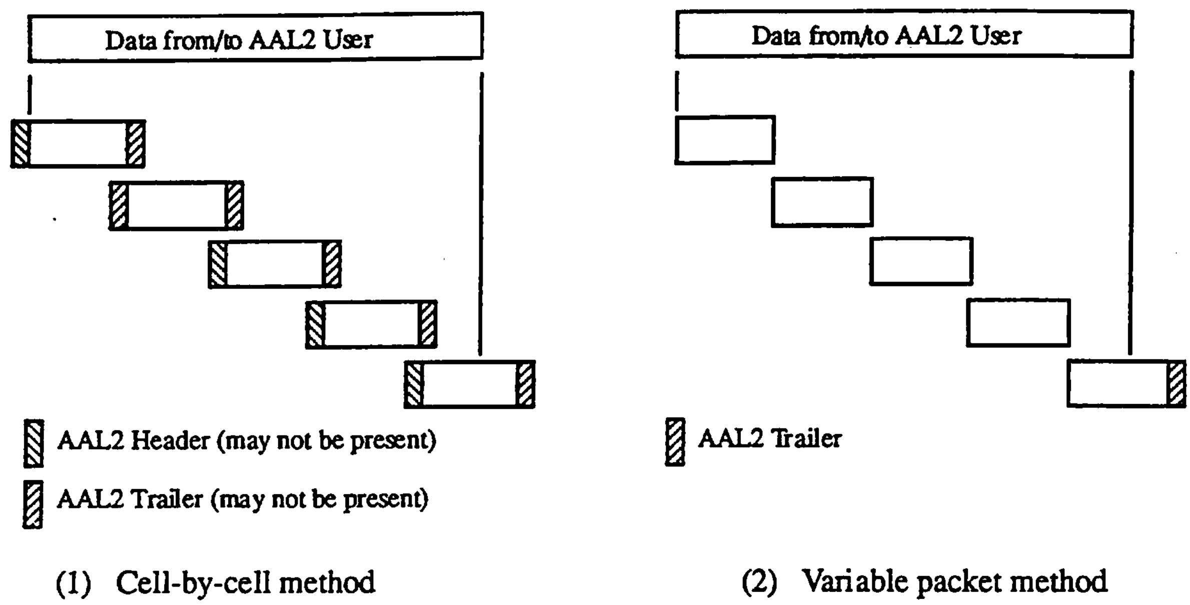


Fig. 5 Two methods of framing AAL2 user data

(2) <u>Error protection: detection only or detection/correction</u>

The need for an error protection function in AAL2 and the required performance should be addressed. The error protection capability of AAL1 can be used for assessing this issue in AAL2, which is described in section 2.1 of this document. Will AAL2 have to support:

- detection of cell losses?
- detection of bit errors of the cell payload?
- correction of cell losses?
- correction of bit errors of the cell payload?

SG13 needs answers from SG15 on this issue. Performance requirements of error protection capabilities also need to be clarified. Such performance requirements may depend on a specific application with a specific bit-rate, for example:

- 30 minutes error free for video conference with 1.5 - 10 Mbit/s?
- Two or three hours error free for video program transmission with 10 - 20 Mbit/s?
- Others?

One possible method for supporting correction of both cell losses and bit errors is the use of FEC and octet interleaving as specified in AAL1. Those methods may require a cell-by-cell AAL2 header such as sequence numbering, e.g., to detect positions of lost cells/octets in an interleaving matrix. It should be noted that correction of lost cells will be achieved more efficiently in AAL, as compared to higher layers.

(3) Handling of ATM user-to-user indication

An AAL2 user is able to use two information streams on a cell basis by using the ATM user-to-user indication in the cell header. Moreover, each information stream may be utilized with a cell-by-cell based cell loss priority. Use of the ATM user-to-user indication is not possible when an AAL5-like method is adopted.

(4) Handling of Cell Loss Priority

This function will allow an AAL2 user to use cell-by-cell based CLP. AAL2 should support mapping of this information between the AAL2 user and the ATM layer.

(5) Timing/synchronization issues

A study should be directed to clarify timing/synchronization issues in VBR services with real-time constraints. Issues to be addressed include: 1) what timing relationship should be supported between a transmitter and receiver at AAL2-SAP? and 2) can the effect of cell delay variation be removed at the receiver side of AAL2-SAP?

(6) Multiplexing of information

Although H.22X will provide for multimedia multiplexing of audio-visual services, support of multiplexing in AAL2 needs to be examined as possible advancement for future standardization. One possible use of AAL2 multiplexing is to support multipoint-to-multipoint connections as depicted below.

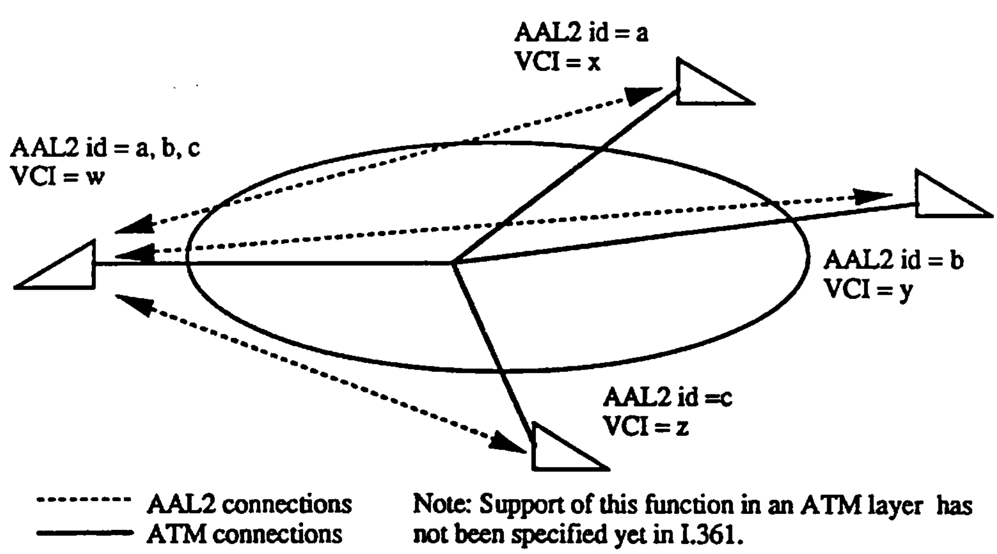


Fig. 6 An example of multipoint-to-multipoint configuration over AAL2/ATM

Integrated Video Services (IVS) Baseline Document
Annex 4. Video Service Interworking

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.

Integration of video services is recognized as a key objective for ATM Video Coding. It is an agreed target for the video coding systems under study by the SG 15 identified several methods by which interworking between video services can be achieved. The following three methods have been identified as relevant to video service interworking:

Negotiation, or switchable encoder, approach

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

Simulcast

Transmitting terminals contain multiple encoders, operating at a variety of resolutions and quality levels so that broad interconnectivity can be achieved by transmitting multiple parallel encoded signals. 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 utilize 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 commensurate with their input signal quality. Three additional terms relevant to layered coding also requires definition:

- Flexible layering. Identified within the SG 15 Experts Group, this concept allows for a choice to be made as to how many layers should be used for particular services or in particular applications. Those layers not required can be "switched off" with possible advantages in complexity and transmitted rate while allowing for broad interworking. Note that full layering and single coders are special cases.
- Embedded bit stream. This is an implementation of layered coding, in which the receiving device need only accept the data stream it can usefully process. The layers could, for example, be transmitted in separate VCs in an ATM network.
- Syntactic extension. This is another layered coding implementation, in which the decoder accepts the entire generated bit stream and separates from it that information it can usefully process by means of the common syntax.

A range of issues needs to be considered in comparing these different approaches, including complexity, coding rate penalties and performance. They may provide different levels of compatibility, impose different constraints on coding algorithm design and are better matched to different applications. For example, negotiation would seem inappropriate for multipoint and distribution services, whereas simulcast seems in appropriate for storage applications (e.g., store and forward video mail).

Layered coding seems suited to the widest application range, with the additional benefit of providing inherent cell loss tolerance. "Flexible layering" appears to provide broad interworking capability with few restrictions, and is currently one of the options under study by the SG 15 Experts Group. Layered coding has been identified by SG 9 as suitable for extension to future video systems such as super HDTV.

Studies continue to identify applicability, advantages and disadvantages of the various techniques.

Constraints imposed by compatibility may be unacceptable in certain specific situations. The main objective for contribution television applications, for example, is to achieve the best picture quality. Due to possible post-processing, it is not desirable to lose any information. Therefore, it may not be appropriate to assign a lower priority to some cells of the bit stream as is possible with e.g., layered coding. Instead (or, perhaps, in addition) Forward Error Correction methods may be required to maintain the very high end-to-end QOS objective.

It is recognized 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 represents an important area that will influence video coding and its being studied actively in the SG 15 Experts Group. SG 9 has, however, recognized that standardization of a hierarchy of picture formats could impose constraints on the production process.

Integrated Video Services (IVS) Baseline Document
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, despite the inherently varying information content of a motion video sequence (being dependant 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, quantizer stepsize, etc.) ensures that the buffer neither empties nor overflows. Such codecs operate in 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 resource 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 an d 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 in 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. Measurements indicate that under certain conditions the statical multiplexing gain of some VBR video services applications is potentially small.

Studies are also required to quantify the reduction in delay resulting from VBR coding and relate this to limits obtained from human factors investigations.

A5.2 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., 34-35 Mbit/s and 140Mbit/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 this plesiochronous-oriented adaptation.

Many of the more efficient picture coding schemes in use today employ:

- conditional replenishment - where only those parts of the picture that have changed are coded and transmitted; and
- statistical coding - where the shortest codewords are allocated to the most commonly occurring values.

Both techniques result in the data in the video encoder being generated at a variable rate. When the codec is part of a fixed bit rate transmission network, a buffer (which must not be allowed to overflow) is used in the encoder to match the variable data rate generated to the fixed rate of the network. If the buffer is in danger of overflowing, the amount of data being generated is deliberately decreased by, for example, selecting a coarser quantizer until the buffer level is again within the specified range. Such a coding scheme can be considered as a fixed bit rate/variable quality scheme.

When a CBR video codec is to be connected to an ATM network, a problem appears because the internal adaptation does not perform extra functions required by ATM networks. Adaptation is necessary in addition to the built-in adaptation of the codec. Such a scheme does not make it easy to combine functions of both adaptations .

According to the service classification defined in 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 existing plesiochronous-adapted codecs to ATM networks, two methods have been identified: circuit emulation and direct connection, both using a dedicated AAL.

- Circuit emulation - The codec is used as if it 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.

Consideration must also be given to capabilities required to support existing and emerging coding schemes.

A5.3 Compatibility Aspects

It is important to consider the various applications of coded video signals and to maximize 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 communication applications. For example, there may be requirements 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.

A5.4 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 minimizing th 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.

Integrated Video Service (IVS) Baseline Document
Annex 6. Multimedia Service Support

Note: Aspects of multimedia service support are described in a number of I-series Recommendations. Of particular relevance are the 1992 and 1993 revisions to Recommendations I.211, I.311 and I.374. Details concerning the status of these recommendations may be found in Annex 8.

A6.1 Multimedia Service Categories

Multimedia services may be categorized on the basis of how they appear to the customer or the support capabilities required within a network.

Possible examples include:

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

Multimedia Database e.g., browsing through catalogues, educational tutorial;

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

Guidance on important aspects which need to be taken into account in supporting and developing services for B-ISDN can be found Recommendation I.211 - B-ISDN Service Aspects.

The methodology for all multimedia services is the responsibility of SG 13 (Multimedia and B-ISDN). This group is responsible for the investigation of the network capabilities required to ensure the support of multimedia services. Additional Information relating to Network Capabilities for Multimedia Service Support may be found in Recommendation I.374.

Revised Recommendation I.140 indicates that multimedia services are characterized by service dependent attributes (service attributes) describing the means of communication offered by interactive or distribution services and by service components describing the characteristics of the information t o be transferred. The revised form of this Recommendation provides details of the service attributes for Broadband ISDN monomedia and multimedia bearer and teleservices.

Also of interest in the development of multimedia and hypermedia applications is the standardization activity within ISO/IEC JTC1/SC29/WG12 - responsible for the coded representation of audio, picture, multimedia and hypermedia information. This group is developing standards for the coded representation of final form multimedia and hypermedia information objects that will be interchanged as a whole within or across services and applications, by any means of interchanged (storage media, local area, wide area telecommunication or broadcast networks). The Multimedia Hypermedia Expert's Group (MHEG) works with: SG 8/Q.9 (Protocols for Audiovisual Telematic Services), JTC1/SC18/WG8 (Document Processing and Communication for Hypermedia, Multimedia, Publishing and Office Systems - Document Description and Processing Languages), SC29/WG11 - Moving Picture Expert's Group and JTC1/SC18/WG3 - Open Document Architecture.

A6.2 Network Issues and Capabilities

A6.2.1 Multimedia Service Attributes

Multimedia services have multivalued service attributes which distinguish them from traditional telecommunications services such as voice or data. Service attribute examples include: information transfer rate, traffic, type, structure, symmetry. The definitions of service attributes, connection attributes, and their possible values to be used for the description of multimedia services and the appropriate connections to support them are in Rec. I.140 (Revised).

A multimedia service may involve multiple parties, multiple connections, the addition/deletion of resources and users within a single call. From a network perspective, user multiplexed media enter the network as a single information stream and are carried over monomedia bearer services.

Framework Recommendation I.374 describes the concepts of service components and service control elements for multimedia services. A service component is a part of a service which describes a monomedia communication related to a single information type. A user involved in a call may request the allocation or deallocation of individual service components from his call. Each service component of a telecommunications service is associated with transfer capability information. In this way it is possible to describe multimedia services separately from the specification of the information types of the service components they include.

The transfer capability parameters of an ATM connection may include:

- telecommunication service class
- network resources (e.g., peak bit rate, mean bit rate, burstiness)
- quality of service (e.g., cell loss ratio, cell transfer delay, cell delay variation)
- symmetrical/asymmetrical connection

The relationship between service components and transfer capability is important. Under normal conditions it would be expected that a one-to-one relationship applies i.e., a service component would use a single transfer capability for a given call. However it is possible that the characteristics at the user-network interface may differ from those used within the network for the same service component.

Framework Recommendation I.374 also defines service control elements as procedures executed at the calling and called sides to provide a multimedia service. Service control elements are used for: call control, connection control and media control. The separation of call, connection and media control is expected to applicable to any network - however the extent of application may be constrained b y the available network infrastructure e.g., switching and signaling capable of supporting multiple connections. The need for, and extent of, separate definition for networks without a Broadband service support capability is for further study.

Network capabilities required for the support of multimedia services have been initially identified under the general areas of connection management, service management, multimedia interaction, multimedia multiplexing and multimedia resource management. SG 13 has identified and prioritized the following open issues for Recommendation I.374 which require resolution:

	ISSUE	Priority
1.	The restrictions necessary to multimedia capabilities to allow them to be offered over 64kbit/s ISDN	LP
2.	The number of distinct information types - this affects whether multimedia can be defined as involving two cases of the same information type.	MP
3.	Mechanisms for the allocation and deallocation of service components by users.	MP
4.	Transfer of control (ownership) between users during a multimedia call ana mechanisms to achieve this:	
	- in relation to charging	LP
	- in relation to multiparty conferencing	MP
5.	Mandatory versus optional service components.	LP
6.	Additional service attributes and sub-attributes required for multimedia services.	HP
7..	Issues involved in call set-up which are service related (not media or connection related)	MP
8.	Services where the right to release the call is asymmetric - 'invitation to release' and 're-connect' may be needed as service control elements to achieve this.	LP
9.	Relationship between the concepts in I.374 and those of the evolving Intelligent Networks.	HP
10.	The splitting of a multipoint call into more than one call.	MP
11.	The support of connections in which the call originator is not one of the endpoints of the connection, e.g., when information is obtained form a distributed data base on behalf of the user.	MP
12.	Network capabilities for synchronization between service components/media.	HP
13.	Network capabilities required to support the inerchange of synchronized multimedia objects.	HP
14.	Impact of usage parameter control on multimedia services.	HP
15.	Charging capabilities for multimedia services.	LP
16.	Multipoint networking, including the use of media specific services.	MP
17.	The support of network/service interworking for multimedia services.	MP
18.	Determination of the performance limits acceptable to network operators for the signalling an d resource management facilities provided within a network to support dynamic reallocation of capacity. Studies in this areas should include consideration of the user's perception of acceptable delay between a request for change in a particular medium's presentation and the realization of that change.	MP
19.	The need for decoupling the transfer capability of the access network and transport network.	LP
20.	Need for additional services component attributes and refinements to I.210 and I.140 and mapping techniques between attributes in I.374 and I.140/I.210.	HP

	ISSUE	Priority
21.	Can the transport system support both connection-oriented and connectionless connections within the same multimedia session.	LP
22.	Relationship between functions described in I.374 an similar functions in existing reference models.	MP

Note:
LP stands for low priority, results expected at the end of the study period 1993-1996.
MP stands for medium priority, results expected for 1995.
HP stands for high priority, results expected for 1994.

A6.2.2 Signalling for Multiparty Multimedia Services Capability

Call and connection control multimedia services is a new issue for public network standards. Work in this area is at a very early stage of development. SG 11 are currently investigating the functionality required of call and connection control.

Revised Recommendation I.311 (1992) identifies the following signalling capabilities as being the basic capabilities required to support simple multiparty and multiconnection call:

- Capabilities to control ATM virtual channel and virtual path connections:
- Capability to support simple multiparty and multiconnection call:
- Processing related functions:

Further information concerning the signalling for multimedia within Recommendation I.311 can be found in Annex 2, Section 2.3.1.

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 multiplexing multimedia connection (e.g., using Recommendation 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.
- Signaling to support multiconnection 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 call establishment must convey at least the following information:
  - » source traffic characteristics;
  - » required QOS class.

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

- Methods for characterizing source traffic are for further study. Traffic characteristics may include: average rate, peak rate, burstiness and peak duration. Again it may prove difficult to characterize multimedia services where media is multiplexing on the basis of anything other than a virtual channel or a virtual path.
- Traffic characteristics are negotiated with network at call establishment and may be renegotiated, by the user request, during the call. The network may limit the frequency of these renegotiation. 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 and VPs.
- Agreed parameters for usage control are for further study, however additional material describing developments in this area may be found in Recommendation I.371.
- 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.

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 capability, the QOS appropriate to his VP.

A6.2.4.4 Resource Renegotiation

The ATM Experts Group of SG 15 is considering the conversion of 64 kbit/s multiplexing signals to B-ISDN to Virtual Channel multiplexing signals either in Terminal Adaptors (TA) or B-ISDN/64 kbit/s ISDN Interworking Units (IWU). Current user multiplex structures (e.g., Recommendation H.221) can reconfigure their internal rate allocation in the order of 20msec.

Resource allocation in user-multiplexed structures such as those described in Recommendation H.221 corresponds to a redistribution of a fixed resource allocation.

Resource allocation for Virtual Channel multiplexed structures corresponds to a change in the allocation of resources within a communication network. The performance achievable and required in terms of network resource allocation is for further study.

A6.3 Multimedia Multiplexing

A6.3.1 The SG 15 ATM Experts Group

The Experts Group is considering the support of B-ISDN of audiovisual and other multimedia services. Virtual Channel based multiplexing has been identified as a long term target, but early service implementation may have to use other means of multiplexing, since:

- interworking with audiovisual equipment on other networks (64 kbit/s ISDN) will require a user mutltiplex mode of operation.
- an understanding that the network will not be able to support Virtual Channel based multimedia multiplexing at the early stages of standardization.

Table 6.1 summaries multimedia multiplexing options as viewed by the Experts Group (October 1992)

The Experts Group has also developed a reference terminal configuration (Figure 6.1) which shows where the alternate multiplexing options are performed. The Experts Group is concerned about the measurement of Traffic Descriptors by the user at the AAL-SAP and by the network at the Reference Point, and of the effect on Cell Delay Variation from multiplexing and the NT2.

In supporting multiple media and different streams representing the medium (e.g., different layers of a layered video signal representation), the Expert's Group recognizes the value of matching the channel Quality of Service to the characteristics of the signal being carried.

It is the Experts Groups understanding that all Virtual Channels in a given Virtual Path will have the same Quality of Service, though two different Cell Loss Ratios will be available according to the selected value of the Cell Loss Priority bit. It therefore seems that there is no advantage in supporting the different bit streams in different Virtual Channels of the one Virtual Path. Furthermore, the efficient delivery of layered video signals in configurations that provide for interworking between terminals of different capabilities, the different signal streams may need to be routed over different parts of the network.

These considerations imply optimum service support will require the establishment of multiple Virtual Paths, each carrying a subset of the total number of multiplexing signal streams.

Annex 6, Table 1

	Cell (VC) Multiplex	SAR Multiplex	CS Multiplex	User Multiplex - MPEG (packet multiplex) approach	User Multiplex - H.221 at higher rate (bit multiplex) approach
Transmission overhead (for multiplex indication)	0	≥ 4/384	≥4/(packet size) 1	≥ 4bytes per packet	16/(p*640)
Multiplexing delay				Depends on packet interval	No delay due to multiplexing
H.320 compatibility	Switch/simulcast			Switch/simulcast (Different multiplexing components)	Switch/simulcast (Common multiplexing components)
MPEG1 Compatibility	Not at system level			Switch/simulcast (Common multiplexing components)	Switch/simulcast (Different multiplexing components)
Multimedia identification	HLC or user-user signaling	Identifier in each cell (IT?)	Identifier in each AAL-SDU	Packet header (Stream ID)	User-user signaling (BAS)
Bit rate identification	Cell signaling	User-user signaling		Packet header (rate bound)	User-user signaling (BAS?)
Cross-media synchronization	Guaranteed on one VP, requirement	Guaranteed (single VC)			
Separation of audio & video for continuous presence multipoint operation	Easy, but copy function requied by network or MCU, otherwise mesh connection is needed	Difficult, but possible with MCU		Difficult, but possible with MCU (easier than H.221 approach)	Difficult, but possible with MCU
Transmission of low bit rate	Trade-off between delay and transmission efficiency			Low rates accommodated in multiplex (400 bit/s minimum)	Low rates accommodated in multiplex (minimum?)
Influence of one cell loss	Restricted to one medium			One or multiple media may be affected (Depends on AAL)	Multiple media be affected
Ease of implementation	Uses existing network functionality	Additional terminal functions		Requires additional terminal functionality, but easy.	Requires additional terminal functionality. H.221 has been implemented in LSI. Extension to higher rate requires study.
Quality of Service (QOS)	Option of matching QOS to requirements of each medium (not available in B-ISDN Release 1.)	QOS must be that of the most demanding (sensitive) medium			
Management	Cost of multiple VCs is unknown 2	Single VC Used			
Flexibility/Interworking	Can interwork terminals with different media capabilities. Flexible and arbitrary addition, control & routing of media	All communicating terminals must use the same fixed multiplex structure			

1. These figures assume 4 bits as a media identifier - one per cell SAR and one per packet for CS multiplexing. This is the minimum overhead, assuming streaming mode of transmission. (CS multiplex packets contain an illegal number of cells.) Both thee schemes could use packet-based transport, which would involve an additional overhead of 192/(Packet size+192) - (UW). Average wasted bits - 192 per packet. UW (Unique Word) is the fraction of overhead due to the GOB start code (or similar) used to indicate the start of the GOB/slice (UW - word length/packet size), assuming one GOB (slice) per packet.

2. However, SG 11 currently has a requirement for multiple VCs (between the same slice) to be established with a single exchange (1).

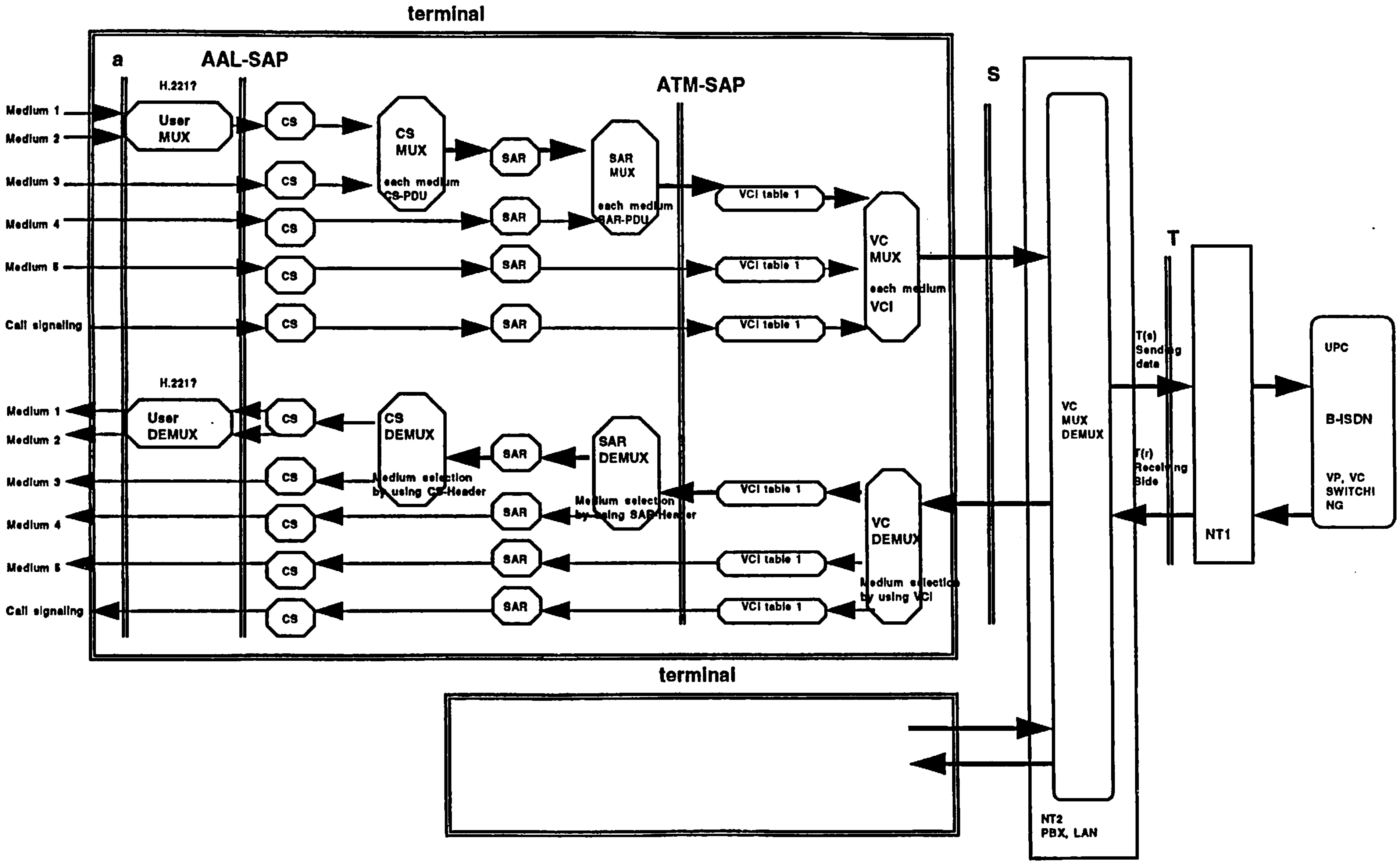


Fig. 6.1 Reference Terminal Configuration Developed by ATM Experts Group

Integrated Video Services (IVS) Baseline Document
Annex 7. Service Requirements

A7.1 Requirements Associated with Recommendation H.261

A7.1.1 Status Notation

(A)	Agreed
(P)	Preferable
(M)	Mandatory
(T)	Target
(FFS)	Implementation method is for further study

A7.1.2 Bit Rate

Up to several 10s Mbit/s	(A)

A7.1.3 Codec Source Format

QCIF/CIF	(A)
"601" Class	(FFS)
EDTV	(?)
HDTV	(?)

A7.1.4 Compatibility

Encoder Decoder

Recommendation H.320 ---------------------------->Recommendation H.32X (terminal) (A,M)
Recommendation H.32X ---------------------------->Recommendation H.320 (terminal) (A,M)

Recommendation H.261---------------------------->Recommendation H.262 (P,FFS)
Recommendation H.262 ---------------------------->Recommendation H.261 (P,FFS)

MPEG1 ---> Recommendation H.262 (P,FFS)
MPEG2 ---> Recommendation H.262 (P,FFS)

"SG 9/1"* ---> Recommendation H.262 (P,FFS)

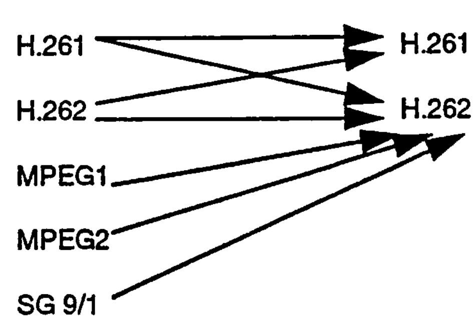


Figure 7.1: Coder/Decoder Compatibility Relationships

* Secondary distribution, which include classes above "601"

A.7.1.5 Applications

"PAL/NTSC" at 3-5 Mbit/s and delay=? (T,FFS)

"CCIR Rec 601" at 8=10 Mbit/s and delay = ? (T,FFS)

CTV	Cable TV Distribution on optical networks, copper, etc
ENG	Electric News Gathering (including SNG, Satellite News Gathering)
IPC	Interpersonal Communications (videoconferencing, videophone, etc)
ISM	Interactive Storage Media (optical disks, etc)
NDB	Networked Database Services (via ATM, etc)
RVS	Remote Video Surveillance
SSM	Serial Storage Media (Digital VTR, etc)
STV	Satellite TV Broadcasting
TTV	Terrestrial TV Broadcasting

A7.1.6 Delay

Less than about 150 ms at bit rate >2Mbit/s (FFS)

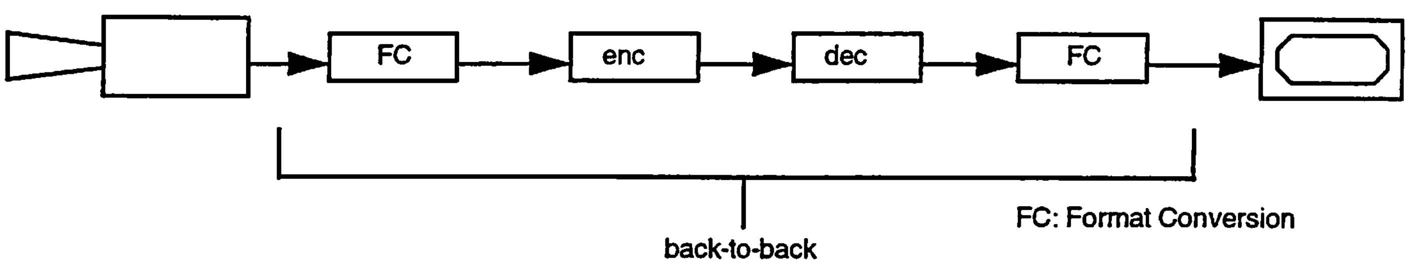


Fig. 7.2: Delay Sources in a Coding Chain

A.7.1.7 Codec Complexity

Complex/high performance
vs
simple/low performance
ex. pure intra-codec

A.7.1.8 ATM

VBR and CBR	(A,M)
Cell Loss resilience	(M,FFS)
Bit error resilience	(M,FFS)
High/low priority cell utilization	(P,FFS)
High/low priority cell independent rate control	(P,FFS)
Usage Parameter Control	(M,FFS)

A.7.1.9 Multipoint

Continuous presence possible (P,FFS)

- Time-sliced decoding
- Editing without decoding-recoding

Mix of Recommendation H.320 and Recommendation H.32X	(M,S)

A7.1.10 Recommendation H.32X Terminal

- with:

Recommendation H.320	(A,FFS)
Network database	(P,FFS)
Distributive service	(P,FFS)
Multipoint	(A,FFS)
Stored bit stream	(P,FFS)

- Multimedia multiplexing	(M,FFS)
- Audio quality > ?	(FFS)
- Relative audio/video delay < ?	(FFS)
- Video clock recovery	(FFS)
- Encryption/scrambling	(FFS)

A7.2 Traffic Characteristics

Traffic Characteristics of voice telephony calls are well known and documented. Similar studies should be undertaken to determine the traffic patterns of video users. (Question source - ITU-R SG 11)

A7.3 HDTV/HRI Based Services

ITU-R SG 11 have identified the need for further studies are needed to determine the signalling requirements for various types of HDTV/HRI based services.

Integrated Video Services (IVS) Baseline Document

Annex 8. Related Recommendations

ITU-T SG 13

Recommendation I.113 - Vocabulary of terms for Broadband aspects of ISDN (Revised Geneva 1992)

Recommendation I.140 -Attribute Technique for the Characterization of Telecommunication Services Supported by an ISDN and Network Capabilities of an ISDN. (Revised Geneva 1993)

Recommendation I.150 -B-ISDN ATM Functional Characteristics (Revised Geneva 1993)

Recommendation I.210: Principles of Telecommunication Services Supported by an ISDN and the Means to Describe them. (revised Geneva 1993)

Recommendation I.211- B-ISDN Service Aspects (Revised Geneva 1993)

Recommendation I.311 - B-ISDN General Network Aspects (Revised Geneva 1993)

Recommendation I.321 - B-ISDN Protocol Reference

Recommendation I.327 - B-ISDN Functional Architecture

Recommendation I.356- B-ISDN ATM Layer Cell Transfer Performance (Revised Geneva 1993)

Recommendation I.361 - B-ISDN ATM Layer Specification (Revised Geneva 1993)

Recommendation I.362 - B-ISDN Adaptation Layer (AAL) Functional Description (Revised Geneva 1992)
Recommendation I.363 - B-ISDN Specification (Revised Geneva 1993)
Recommendation I.371: Traffic Control and Congestion Control (Geneva 1993)

Recommendation I.374 - Network Capabilities fo r the Support of Multimedia Services (Geneva 1993)

Recommendation I.413 - B-ISDN User-Network Interface (Revised Geneva 1993)

Recommendation I.432 -B-ISDN User-Network Interface Physical Layers Specification

Recommendation I.610 - B-ISDN OPerations and Maintenance Principles and Functions (Revised Geneva 1993)

IEC/ISO

IS 11172-1 "Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to about 1.5 Mbit/s. Part 1: Systems"

IS 11172-2 "Coding of Moving Pictures and Associated Audio of Digital Storage Media at up to about 1.5 Mbit/s. Part 2: Video"

IS 11172-3 "Coding of Moving Pictures and Associated Audio of Digital Storage Media at up to about 1.5 Mbit/s. Part 3 Audio"

Integrated Video Services (IVS) Baseline Document
Annex 9. Vocabulary and Abbreviations

A9.1 Objective and Rationale

The objective of this annex is to provide a basis for the consistent and unambiguous use of terms and abbreviations between the various groups participating in IVS co-ordination studies and contributions to the IVS Baseline Document.

A9.2 Abbreviations Used in the IVS Baseline Document

A: Agreed

AAL: ATM Adaptation Layer

ATM: Asynchronous Transfer Mode

BER: Bit Error Rate

B-ISDN: Broadband Integrated Services Digital Network

B-UNI: Broadband User-Network Interface

CBR: Constant Bit Rate

CDV: Cell Delay Variation

CEQ: Customer Equipment

CIF: Common Intermediate Format

CLP: Cell Loss Priority

CLR: Cell Loss Ratio

Codec: Coder/Decoder

CRF: Connection Related Function

CTV: Cable TV Distribution

DSM: Digital Storage Media

EDTV: Enhanced Definition Television

ENG: Electronic News Gathering

FC: Format Conversion

FEQ: Forward Error Correction

FFS: For Further Study

GFC: Generic Flow Control

HDTV: High Definition Television

HRI: High Resolution Imagery

IEC: International Electrotechnical Commission

IPC: Interpersonal Communications (videoconferencing, videophone, etc.)

ISDN: Integrated Services Digital Network

ISM: Interactive Storage Media (optical disks, etc.)

ISO: International Standards Organization

ITU: International Telecommunications Union

IVS: Integrated Video SErvice

IWU: Interworking Unit

M: Mandatory

OAM: Operations and Maintenance

MPEG: Moving Picture Experts Group

NBD: Network Database Services

NNI: Network Node Interface

NP: Network Performance

NPC: Network Parameter Control

UPC: Usage Parameter Control

NTSC: National Television System Committee

P: Preferable

PAL: Phase Alternation Line

PCM: Pulse Code Modulation

PDU: Protocol Data Unit

QCIF: Quarter Common Intermediate Format

QOS: Quality of Service

RVS: Remote Video Surveillance

SAP: Service Access Point

SDU: Service Data Unit

SN: Sequence Number

SNG: Satellite News Gathering

SSM: Serial Storage Media (digital ,VTR, etc.)

STV: Satellite TV Broadcasting

T: Target

TA: Terminal Adaptor

TTV: Terrestrial TV Broadcasting

TV: Television

UNI: User-Network Interface

VBR: Variable Bit Rate

VC: Virtual Channel

VCC: Virtual Channel Connection

VP: Virtual Path

VPC: Virtual Path Connection

VTR: Video Tape Recorder

A9.3 Vocabulary

A9.3.1 Scope and Intent

This section of the IVS Baseline Document contains vocabulary terms and expressions used within the Baseline text. The section has been complied with the intent of providing guidance to the wide and diverse community of users of the Baseline Document.

Contributors are urged to ensure that defined terms are used wherever possible and that terms with multiple definitions (for example - media) are always clearly identified by context.

Where possible, the source of the definition is provided. For example, Recommendation I.113 provides the primary source and reference for the vocabulary of terms for Broadband aspects of ISDN. Annex B of Rec I.113 (Geneva 1993) provides a list of abbreviations used in B-ISDN Recommendations.

A9.3.2 Vocabulary of Terms Used by Contributors to the IVS Baseline

add/remove: when connection elements can be established and released while other connection elements of the same connections still exist, the configuration of this connection is described as add/remove. (Rec I.140)

Asynchronous Transfer Mode (ATM): a transfer mode in which the information is organized into cell, it is asynchronous in the sense that the recurrence of cells containing information from an individual user is not necessarily periodic. (Rec. I.113)

broadband: a service or system requiring transmission channels capable of supporting greater than the preliminary rate. (Rec. I.113)

broadcast: a value of the service attribute "communications configuration" which denotes unidirectional distribution
to all users. (Rec I.113)

broadcast communication: unidirectional communication from a single access-point to an unlimited number of unspecified destination access-points. (Rec I.113)

broadcasting: within this document this term is most commonly used in reference to coding to support a television broadcasting service.

cell: unit os information of fixed length, and consisting of a header and an information field. It is identified by a label at the asynchronous transfer mode layer of the B-ISDN protocol reference model. (Rec I.113)

cell delay variation: Reference Recommendation I.356

cell error ratio: Reference Recommendation I.356

cell loss ratio: Reference Recommendation I.356

cell misinsertion rate: Reference Recommendation I.356

cell tagging: Reference Recommendation I.371

cell transfer delay: Reference Recommendation I.356

communication configuration: the spatial arrangement for transferring information between two or more access points. It completes the structure associated with a telecommunication service as it associates the relationship between the access points involved and the flow of information between the access points. Possible values include: point-to-point, multipoint and broadcast, (Rec I.140)

connection: provide the capability of transferring information between end-points. It represents the association between end-point together with the incremental information regarding the information transfer integrity. (Rec I.311)

connection (ATM): An ATM Layer connection consists of the concentration of ATM layer links in order to provide an end-to-end transfer capability to access points. (Rec I.140)

connection configuration: describes the spatial arrangement for transferring information on a given connection. It consists of two sub-attributes, topology and dynamics. (Ref I.140)

constant bit rate:

1. a type of telecommunications service characterized by a service bit rate specified by a constant value (Rec I.113)
2. a constant value bit rate arising from specific coding algorithm.

continuous presence: a form of multipoint conferencing, in which the video signal from more than one remote participating site can be viewed simultaneously.

distributive: see distribution

multipoint-to-point service: e.g., televoting

network interworking: Ref. I.500-series recommendations. Refers to the functions and requirements for interworking of networks with different low layers capabilities in order to support the interworking of services across the network boundary.

network node interface: the interface at the network node which is used to interconnect with another network node. (Rec I.113)

peak cell rate: in the Source Traffic Descriptor specifies an upper bound on the traffic than can be submitted on an ATM connection. (Recommendation I.371)

point-to-multipoint: Ref Rec. I.140

point-to-point communication: communication between only two access points. (Rec I.140)

point-to-point connection: a connection in which only two end points are provided. (Rec I.140)

periodic frame: a transmission segment which is repeated at intervals of equal duration and may be delineated by incorporating fixed periodic patterns into the bit stream. (Rec I.113)

physical frame: a segment of a serial logical bit stream at an interface, partitioned into successive segments. (Rec I.113)

picture formats: the parameters that define picture resolutions (horizontal and vertical pixels, luminance and chrominance sampling patterns and relationships) and frame rates.

quality of service: the collective effect of service performances which determine the degree of satisfaction of a user of a specific service. (Rec I.350)

resource management: Ref Rec I.371

service bit rate: the bit rate which is available to a user for the transfer of user information. (Rec I.113)

service/s interworking: Ref Draft Rec I.501. Specific ISDN service descriptions, as viewed from a user perspective are provided in the I.200-series. Network interworking recommendations are provided in the I.500-series. Recommendation I.580 describes general arrangements for interworking between B-ISDN and 64kbit/s-ISDN.

simulcast: the simultaneous transmission of the same video signal encoded in multiple formats and, in particular, at different quality levels for reception by decoders of different capabilities.

switch (virtual channel): a virtual channel switch is a network element that connects virtual channel links; it terminates virtual path connections and translates virtual channel identifier values and is directed by control plane functions.

switch (virtual path): a virtual path switch is a network element that connects virtual path links; it translates Virtual Path Identifiers (not Virtual Channel Identifiers) values and is directed by control plane functions.

switched (connection): Ref I.140 and I.311

teleservice: Ref Rec I.210

traffic parameters: a specification of a particular traffic aspect. It may be qualitative or quantitative. (Ref I.371)

transfer mode: aspects covering transmission, multiplexing and switching in a telecommunications network. (Rec I.113)

usage parameter control: the taking of appropriate action if usage monitoring establishes that the negotiated values of the information transfer parameters of a virtual channel or virtual path are exceeded. (Rec I.113)

variable bit rate service; a type of telecommunication service characterized by a service bit rate specified by statistically expressed parameters which allow the bit rate to vary within defined limits. (Rec I.113)

variable bit rate coding: a type of service coding (e.g., video) characterized by a varying output bit rate.

video-on-demand: the ability to request, on a per-terminal basis and with individual control, the delivery of particular video material. An example would be selection and playback of a movie over the network.

virtual channel: a concept used to describe unidirectional transport of ATM cells associated by a common unique identifier value. (Rec I.113)

virtual channel connection: a concatenation of virtual channel links that extends between two points where the adaptation layer is accessed. (Rec I.113)

virtual channel switch: a network element which connects virtual channel links; it terminates virtual path connections and it translates virtual channel identifier value. (Rec I.113)

virtual path: a concept used to describe unidirectional transport of ATM cells belonging to virtual channels that are associated by a common identifier value. (Rec I.113)

virtual path connection: a concatenation of virtual path links that extends between the point where the virtual channel identifier values are assigned and the point where those values are translated or removed. (Rec I.113)

virtual path switch: is a network element which connects virtual path links, it translates virtual path identifier values and is directed by control plane functions.

Integrated Video Services (IVS) Baseline Document

Annex 10

Summary List of Open Issues

The following is a summary list open issues across both network and coding aspects. It is intended to provide guidance on relevant issues requiring contributions to the various groups. Most of the current issues impact on the network aspects, requiring attention by SG 13 but many of these issues also rely on requirements inputs from the coding groups. No priority order is intended by the numbering.

1. Timing of availability of multiconnection calls for support of VC-based multimedia/audiovisual services.
2. Circuit emulation, particularly for support of 64 kbit/s ISDN services and interworking:
   - interworking with mixed ATM/STM networks
   - circuit emulation capabilities, including supplementary services
   - H.320 terminal support (H.320 via B-ISDN and N.320 to H.32x)
3. Clarification of resource management and QoS issues.
4. Advantage of VBR is unclear.
5. Alignment of H.32x and B-ISDN standardization timetables and other non-ITU activities.
6. Preferred multiplexing option (VC or user multiplex). Signalling requirements for support of multimedia (separation of call and connection). Multiconnection support should be given consideration as a matter of urgency and signalling requirements followed up with SG 11 . The need for multipoint communication functionality must be taken into account.
7. Cell loss figures (high and low priority), and what they mean (there is a need for a network performance model).
8. UPC _ mechanisms, time constant (for VBR and CBR), standardization of parameters across networks and close relationship between UPC/traffic shaping and video coding.
9. Differential delay between audiovisual/multimedia components (audio, video, etc.)
10. Tariffing issues (basis for charging, cost of low and high priority cells, charging for different QoS).
11. Transmission of multiple layers of video signal (QoS on each VP/VC, end-to-end signalling, synchronization).
12. AAL Type 2 functionality.
13. Difference of Cell Delay Variation measurement measurement terminal by network and terminal, due to presence of NT2.
14. Flexibility of assignment of rate of high and low priority channels.
15. Impacts on coding imposed by non-network constraints (TV studio operations).
16. Timing of MPEG digital compression system is seen by ITU-R SG11 as premature for some expected digital TV implementations.
17. Broadcast television seen by ITU-R SG11 as driving force for digital video due to size of market.
18. Hierarchical coding methods are seen as important by ITU-R SG11.
19. High rates used for digital television, so less potential for statistical multiplexing and so CBR is assumed by ITU-R SG11.
20. Digital terrestrial broadcasting is the current highest priority in ITU-R SG11. Harmonization with other delivery means will be carried out later.
21. Acquisition time delay on channel change (including display oscillator, access control, decoder acquisition, and all other elements in the system) must be constrained (<1 sec.?)
22. Parameters to be negotiated between user and network at call establishment.
23. QOS that can be requested by a user from ATM network.
24. Network information (e.g., congestion) available during a call.
25. CDV characteristics on A.
26. Method and accuracy of end-to-end timing recovery over ATM.
27. Range of input/output picture formats, integration into hierarchical structure.
28. Application of, and migration to , progressive picture formats.
29. Graceful degradation is important, once error correction capabilities are exceeded.