The ATM Forum Technical Committee

Voice and Telephony Over ATM to the Desktop Specification

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1 Introduction

1.1 Purpose

This document specifies the ATM Forum's interoperability agreement for supporting voice and telephony over ATM to the desktop. It complies with the Forum's other interoperability agreements.

1.2 Scope

This document specifies the particular features required to provide voice and telephony service in B-ISDN. It describes the functions to be performed by a native ATM terminal and an Interworking Function (IWF) to provide service between B-ISDN attached terminals and/or N-ISDN attached terminals, be they public or private. This document specifies the particular features required to support signalling interworking between B-ISDN and N-ISDN. The requirements for support of UNI 4.0 supplementary services are also described.

This document applies to the transport of a single 64 kbit/s A-law or μ -law (ITU-T Recommendation G.711) voiceband signal. Alternative coding for voice transport (e.g., high quality voice, compressed voice) and silence removal are outside the scope of this specification.

Only the signalling interworking with a N-ISDN and at a Q.SIG (PSS1) interfaces are covered. This document is based on PNNI 1.0, UNI 4.0, ITU-T Recommendation I.580 on N-ISDN interworking and ITU-T Recommendations Q.2931, Q.2951 and Q.2957 on DSS2 signalling. This document covers the interworking scenario where a user on a B-ISDN is accessing N-ISDN services offered by a N-ISDN, through an Interworking Function (IWF).

Three types of attachment are considered in this specification:

- B-TE to Private or Public B-ISDN,
- Private B-ISDN to Public N-ISDN,
- Private B-ISDN to Private N-ISDN.

1.3 Reference configuration



Figure 1-1 Reference configuration

The interworking between a Public B-ISDN and a Public N-ISDN, including the B-ISUP and the N-ISUP protocols, is outside the scope of this specification.

The signalling between a public B-ISDN and a private B-ISDN is specified in UNI 4.0.

1.4 Service requirements

The service requirements for N-ISDN service are described in ITU-T Recommendation I.231 for the circuit mode bearer service and in I.251 and I.257 for the supplementary services supported in this specification.

1.5 References

1.5.1 Normative

ATM Forum Technical Committee, af-pnni-0055.000, Private Network-Network Interface Specification Version 1.0 (PNNI 1.0), March 1996

ATM Forum Technical Committee, af-sig-0061.000, ATM User-Network Interface (UNI) Signalling Specification Version 4.0, June 1996

ATM Forum Technical Committee, af-ilmi-0065.000, Integrated Local Management Interface (ILMI) Specification Version 4.0, July 1996

ISO/IEC 11572:1994, Information technology - Telecommunications and information exchange between systems - Private Integrated Services Network - Circuit mode bearer services - Inter-exchange signalling procedures and protocol

ITU-T G.711-1988, Pulse code modulation (PCM) of voice frequencies

ITU-T I.150-1993, Asynchronous Transfer Mode Functional Characteristics

ITU-T I.356-1993, Integrated Services Digital Network (ISDN); Overall network aspects and functions; B-ISDN ATM layer cell transfer performance

ITU-T I.363.1-1996, B-ISDN ATM Adaptation Layer (AAL) Specification, Types 1 and 2

ITU-T I.363.5-1996, B-ISDN ATM Adaptation Layer (AAL) Specification, Type 5

ITU-T I.430-1993, Integrated Services Digital Network (ISDN); ISDN user-network interfaces; Basic rate usernetwork interface; Layer 1 specifications

ITU-T I.431-1993, Integrated Services Digital Network (ISDN); ISDN user-network interfaces; Primary rate usernetwork interface; Layer 1 specifications

ITU-T Q.921-1993, ISDN User-Network Interface; Data link layer specification

ITU-T Q.931-1993, Digital Subscriber Signalling System No. 1 (DSS1); ISDN User-Network Interface Layer 3 Specification for Basic Call Control

ITU-T Q.951-1992, *Stage 3 Description for Number Identification Supplementary Services using DSS1*, clauses 1, 2, 8

ITU-T Q.951-1993, Stage 3 Description for Number Identification Supplementary Services using DSS1, clauses 3, 4, 5, 6

ITU-T Q.957-1993, Stage 3 Description for Additional Information Transfer Supplementary Services using DSS1), clause 1

ITU-T Q.2931-1995, Broadband Integrated Services Digital Network (B-ISDN); Digital Subscriber Signalling System No. 2 (DSS2); User-Network Interface (UNI) Layer 3 Specification for Basic Call/Connection Control

ITU-T Q.2951-1995, Stage 3 Description for Number Identification Supplementary Services using B-ISDN Digital Subscriber Signalling System No. 2 (DSS2) - Basic Call

ITU-T Q.2957-1995, Stage 3 Description for Additional Information Transfer Supplementary Services using B-ISDN Digital Subscriber Signalling System No. 2 (DSS2) - Basic Call

1.5.2 Informative

ANSI T1.101-1994, Synchronization Interface Standard

ANSI T1.508-1992, Network Performance-Loss Plan for Evolving Digital Networks

ANSI Technical Report No. 27, A Technical report on Echo Cancellers, T1A1.6 Working Group, November 1993

Bellcore TR-NWT-001244, Clocks for the synchronized Network: Common Generic Clock Criteria

ISO/IEC 11571:1994, Information technology - Telecommunications and information exchange between systems - Numbering and subaddressing in Private Integrated Services Networks

ISO/IEC 11579-1:1994, Information technology - Telecommunications and information exchange between systems - Reference configuration for Private Integrated Services Networking (PISN) exchanges

ITU-T G.131-1996, Control of talker echo

ITU-T G.164-1988, Echo suppressors

ITU-T G.165-1993, Echo cancellers

ITU-T G.176-1997, Planning guidelines for the integration of ATM technology into the PSTN

ITU-T G.811-1988, Timing requirements at the outputs of primary reference clocks suitable for plesiochronous operation of international digital links

ITU-T I.231-1988, Circuit-mode bearer service categories

ITU-T I.251.1 (rev. 1)-1992, Integrated Services Digital Network (ISDN) - General structure and service capabilities - Direct-Dialling-In

ITU-T I.251.2 (rev. 1)-1992, Integrated Services Digital Network (ISDN) - General structure and service capabilities - Multiple Subscriber Number

ITU-T I.251.3 (rev. 1)-1992, Integrated Services Digital Network (ISDN) - General structure and service capabilities - Calling Line Identification Presentation

ITU-T I.251.4 (rev. 1)-1992, Integrated Services Digital Network (ISDN) - General structure and service capabilities - Calling Line Identification Restriction

ITU-T I.251.5-1995, Integrated Services Digital Network (ISDN) - Service capabilities - Connected Line Identification Presentation

ITU-T I.251.6-1995, Integrated Services Digital Network (ISDN) - Service capabilities - Connected Line Identification Restriction

ITU-T I.251.8 (rev. 1)-1992, Integrated Services Digital Network (ISDN) - General structure and service capabilities - Sub-addressing supplementary service

ITU-T I.257.1-1995, Integrated Services Digital Network (ISDN) - Service capabilities - User-User Signalling (UUS)

ITU-T I.411-1993, Integrated Services Digital Network (ISDN); ISDN user-network interfaces; ISDN user-network interface - Reference configuration

ITU-T I.413-1993, Integrated Services Digital Network (ISDN); ISDN user-network interfaces; B-ISDN user-network interface

ITU-T I.580-1995, General Arrangements for Interworking Between B-ISDN and 64 kbit/s Based ISDN

ITU-T P.310-1996, Transmission characteristics for telephone band (300-3400 Hz) digital telephones

ITU-T Q.2932-1996, Broadband Integrated Services Digital Network (B-ISDN); Digital Subscriber Signalling System No. 2 (DSS2); Generic Functional Protocol - Core functions

 $\label{eq:ITU-T-X213-1995} | \ \text{ISO/IEC 8348:1996}, \ \textit{Information technology - Open systems interconnection - Network service definition} \\$

1.6	Abbreviations and acronyms
AAL	ATM Adaptation Layer
B-ISDN	Broadband ISDN
B-ISUP	Broadband ISDN User Part
B-TE	Broadband Terminal Equipment
CDV	Cell Delay Variation
CRC	Cyclic Redundancy Check
CS	Convergence Sublayer
CSI	Convergence Sublayer Information
DCC	Data Country Code
DSS1	Digital Subscriber Signalling System #1
DSS2	Digital Subscriber Signalling System #2
GF	Generic Functional Protocol
ICD	International Code Designator
ILMI	Integrated Local Management Interface
ISDN	Integrated Services Digital Network
ISO	International Organization for Standardization
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
IWF	Interworking Function
LLC	Low Layer Capability
MIB	Management Information Base
NSAP	Network Service Access Point
N–BC	Narrowband Bearer Capability
N-ISDN	Narrowband ISDN
N-ISUP	ISDN User Part
N-LLC	Narrowband Low Layer Capability
OAM	Operation, Administration and Maintenance
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
POTS	Plain Old Telephone Service
PNNI	Private Network-Network Interface
PRI	Primary Rate Interface
PRS	Primary Reference Source
PSS1	Private Integrated services Signalling System No. 1

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Q.SIG	PSS1
SAR	Segmentation and Reassembly
SDU	Service Data Unit
SN	Sequence Number
SNP	Sequence Number Protection
TE	Terminal Equipment
TDM	Time Division Multiplexing
UNI	User-Network Interface
VPN	Virtual Private Network
VTOA	Voice and Telephony Over ATM

2 Definition of native ATM terminal equipment

2.1 Functional requirements

The native ATM terminal equipment (B-TE) is able to establish a voice communication through a B-ISDN on a per call basis. The traffic is 64 kbit/s PCM-encoded voice. Each voice call uses one VCC. The B-TE may invoke supplementary services (as described in section 2.5.2) on a per call basis.

2.2 Interfaces

A B-TE has either a public UNI to a public B-ISDN or a private UNI to a private B-ISDN.

2.3 Protocol Reference Model

Figure 2-2 illustrates the protocol reference model at the interface between the B-TE and the B-ISDN network at the S_B reference point.



Figure 2-2 Protocol reference model

2.4 User plane

2.4.1 Physical layer

The physical layer shall be any of the physical layers defined by the ATM Forum.

2.4.2 ATM Layer

The specification of the ATM layer is as defined in section ITU-T Recommendations I.150 and I.361, including OAM cells for F4 and F5 management information flows.

2.4.3 AAL

A B-TE attached to a private B-ISDN has the option of using either the AAL1 specification for voiceband signal transport as described in ITU-T Recommendation I.363.1, or the AAL5 as specified in ITU-T Recommendation I.363.5. However, a B-TE attached directly to a public B-ISDN may have to support the AAL1 specification for voiceband signal transport to conform to the public B-ISDN offerings.

It is considered extremely important that two B-TEs should be able to connect directly to one another and should not require that an interworking function be inserted into the VCC to transcode between different user plane formats (AAL1 for voice and AAL5).

Therefore, it is required that all B-TEs implement a common user plane format - in the sense that they shall accept incoming calls whose signaling indicates the use of that format.

It is not required that all B-TEs establish their outgoing calls using only the common format. A B-TE may choose to use the alternate format instead. If a call is rejected for that reason, a B-TE may choose to retry it using the common format.

Since all B-TEs must implement AAL 5 to support control plane signaling, the common user plane format is mandated to be AAL 5.

Appendix B describes the consequences of interworking between B-TEs using different AALs for voice.

2.4.3.1 Additional requirements for AAL5

Voiceband information shall use the Message Mode service.

The SSCS is null, thus the AAL-SDU is mapped to one CPCS-SDU (no SSCF and no SSCOP).

The CPCS-PDU payload may be up to 40 octets, in increments of 8 octets, with 40 octets as the preferred value. Therefore, the SAR sub-layer is trivial: the AAU bit in the Payload Type of the ATM header is always set to "1" to indicate the end of a SAR-SDU.

The CPCS User-to-User indication (CPCS-UU) (in the CPCS trailer) should be set to 00000000. The use of this octet for other purposes is reserved for further study.

The Length field in the CPCS trailer may be coded as 8 to 40.

The cell rate for AAL5 shall be 8000 \div payload_length cell/s (64 kbit/s x 1 octet/8 bits x 1 cell \div payload_length octets). For a full-fill cell, the cell rate is 200 cell/s. This implies a cell construction delay of 125 μ s x payload_length. For a full-fill cell, the cell construction delay will be 5 ms.

When a B-TE or IWF receives a corrupted AAL5 CPCS-PDU, it shall pass the corrupted data, together with an indication that it is corrupted, from the ATM Adaptation layer to the higher layers according to the procedures of I.363.5. It is the responsibility of the higher layers to process this information. This method will improve the performance of the system as well provide the hooks for alternative encoding algorithms that already have error correction capabilities.

2.4.4 Voice bit stream

The voice stream is encoded as 64 kbit/s PCM μ -Law or A-Law, as described in ITU-T Recommendation G.711. The B-TE may transmit either μ -Law or A-Law signals, according to the guidelines of G.711, and it must be capable of receiving both μ -Law or A-Law signals. Human and acoustical aspects of a speech transmission path are outside the scope of this specification. Nevertheless, the ITU-T P series Recommendations (e.g., P.310) can be taken as guidelines for developing B-TEs.

Section 4.3 contains requirements on a B-TE to provide clocking for the operation of the PCM codec in a specified order of preference. Section 5 contains a recommendation for acoustic echo control.

2.5 Control plane

The physical layer shall be as specified in section 2.4.1.

The ATM layer shall be as specified in section 2.4.2.

The SAAL layer shall be as specified in section 4.1 of UNI SIG 4.0 (af-sig-0061.000).

2.5.1 Basic Call/Connection Control Signalling

The procedures of section 6/Q.2931 "Procedures for the support of 64kbit/s based circuit-mode ISDN services in B-ISDN and access signalling interworking between N-ISDN and B-ISDN" and Appendix II.2.2/Q.2931 shall apply, as modified per UNI 4.0 "Basic Point-to-Point Call".

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For clarity, this section refers to Q.2931 sections numbers because UNI 4.0 is a pointer to Q.2931, with a list of differences. It should be understood that the modifications described in UNI 4.0 apply.

6.2.2/Q.2931 Bearer-service related information: A B-TE shall code the following information elements as described below.

Octet	Information element field	Field value
3	Information transfer capability	- Speech (not to be used for fax or modem), or
		- 3.1 kHz audio
4	transfer mode	Circuit mode
	Information transfer rate	64 kbit/s
4.1	absent	
5	User information layer 1 protocol	- Recommendation G.711 μ-Law, or
		- Recommendation G.711 A-Law
5a-7	absent	

The N-BC information element shall be coded as follows:

The Broadband bearer capability information element shall be coded as follows:

Octet	Information element field	Field value
5	Bearer class BCOB-A	
5a	absent	
6	Susceptibility to clipping Susceptible to clipping	
	User plane connection configuration	Point-to-point

The ATM traffic descriptor information element shall be coded as follows for AAL1 for voice.

Octet	Information element field	Field value if no OAM cells are used (Note 1)	Field value if 1 OAM cells/s is used (Note 2)	Field value with maximal OAM support (Note 3)
5	absent			
6	absent			
7.1 7.2 7.3	Forward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1010 1011 (171 cells/s)	0000 0000 0000 0000 1010 1100 (172 cells/s)	0000 0000 0000 0000 1010 1111 (175 cells/s)
8.1 8.2 8.3	Backward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1010 1011 (171 cells/s)	0000 0000 0000 0000 1010 1100 (172 cells/s)	0000 0000 0000 0000 1010 1111 (175 cells/s)

NOTES

1 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and no cell rate allocation for OAM cells.

2 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and on 1 cell/s allocation for OAM cells.

3 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and the following cell rate allocation for OAM: two percent of the user cell rate and an additional 1 cell/s.

The ATM traffic descriptor information element shall be coded as follows for AAL5.

Octet	Information element field	Field value if no OAM cells are used (Note 1)	Field value if 1 OAM cells/s is used (Note 2)	Field value with maximal OAM support (Note 3)
5	absent			
6	absent			
7.1 7.2 7.3	Forward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1100 1000 (200 cells/s)	0000 0000 0000 0000 1100 1001 (201 cells/s)	0000 0000 0000 0000 1100 1101 (205 cells/s)
8.1 8.2 8.3	Backward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1100 1000 (200 cells/s)	0000 0000 0000 0000 1100 1001 (201 cells/s)	0000 0000 0000 0000 1100 1101 (205 cells/s)

NOTES

1 These values are based on an AAL type 5 payload of 40 octets per cell for user information and no cell rate allocation for OAM cells. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length, encoded in binary format.

2 These values are based on an AAL type 5 payload of 40 octets per cell for user information and on 1 cell/s allocation for OAM cells. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length + 1, encoded in binary format.

3 These values are based on an AAL type 5 payload of 40 octets per cell for user information and the following cell rate allocation for OAM: two percent of the user cell rate and an additional 1 cell/s. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length x 1.02 + 1, encoded in binary format.

The Quality of service parameters information element shall be coded as follow:

Octet	Information element field	Field value
5	QOS-class forward	Unspecified QOS class
6	QOS-class backward	Unspecified QOS class

The AAL parameters information element shall be coded as follows for AAL1 for voice.

Octet	Information element field	Field value	
5	AAL-Type	0000 0000	AAL for voice

The AAL parameters information element shall be coded as follows for AAL5.

Octet	Information element field	Field value
5	AAL-Type	0000 0101 AAL Type 5
6.1 6.2	Forward maximum CPCS-SDU size	Any multiple of 8 between 8 and 40, encoded in binary format
7.1 7.2	Backward maximum CPCS-SDU size	Any multiple of 8 between 8 and 40, encoded in binary format
8 8.1	SSCS-type	absent (preferred), or 0000 0000 null

6.3/Q.2931 Interworking N-ISDN -> B-ISDN: Not applicable to B-TE.

6.4/Q.2931 Interworking B-ISDN -> N-ISDN: Not applicable to B-TE.

6.5/Q.2931 Overlap sending and receiving: Not applicable to B-TE.

6.6/Q.2931 Notification of interworking: Only those procedures relating to a user shall apply.

6.7/Q.2931 Additional features with regards to the provision of N-ISDN services: Only those procedures relating to a user shall apply.

2.5.2 Support of Supplementary Services

The support of the supplementary services defined in this section is optional, as explained in UNI 4.0. However, if they are supported, they shall conform to the requirements of this section. Non-Generic Functional protocol based supplementary services consists of Supplementary Services that do not use the GF protocol as defined by ITU-T Recommendation Q.2932. GF-based supplementary services use the Generic Functional protocol as defined by the ITU-T Recommendation Q.2932.

2.5.2.1 UNI 4.0 supplementary services

The seven supplementary services supported by UNI 4.0 that are applicable to a B-TE are defined in ITU-T Recommendations Q.2951 and Q.2957. These services do not require the use of the generic functional protocol.

- Multiple Subscriber Number (MSN)
- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- Subaddressing (SUB)
- User-to-User Signalling (UUS) Service 1

The requirements of Annex 4/UNI 4.0 apply.

2.5.2.2 Generic Functional protocol based supplementary services For further study.

2.5.3 Notification of adaptive timing recovery

Refer to section 4.3 for a discussion of timing recovery requirements on a B-TE, including a description of the adaptive recovery mechanism used in this specification.

The procedures of this section use the following notification descriptor value in the Notification indicator information element:

Bit 7654321 0011100 Adaptive timing recovery used for transmit (TX) clock

2.5.3.1 Requirements on the B-TE originating a call

If a B-TE originating a call does not have access to a network clock, and has the capability to support adaptive timing recovery, it shall signal its intention to do so to by using notification description "Adaptive timing recovery used for transmit (TX) clock" in a Notification indicator information element in the SETUP message.

If a B-TE originating a call is performing adaptive timing recovery and receives a notification description "Adaptive timing recovery used for transmit (TX) clock" in a Notification indicator information element in any message, it shall stop performing adaptive timing recovery. This case should never happen in a network were all the B-TEs conform to this specification: it is included to allow for future services such as call transfer and conference call.

2.5.3.2 Requirements on the B-TE receiving a call

If a B-TE receiving a call receives a notification description "Adaptive timing recovery used for transmit (TX) clock" in a Notification indicator information element in any message, it shall not perform adaptive timing recovery.

If a B-TE receiving a call satisfy the following requirements, it shall perform adaptive timing recovery and signal its intention to do so to by using notification description "Adaptive timing recovery used for transmit (TX) clock" in a Notification indicator information element in the first message sent in response to the SETUP message:

- it does not have access to a network clock, and

- it has the capability to support adaptive timing recovery, and

- it does not receive a notification description "Adaptive timing recovery used for transmit (TX) clock" in a Notification indicator information element in any message.

2.5.4 AAL negotiation

If a B-TE supporting AAL5 only and not AAL1 for voice receives a SETUP message indicating AAL1 for voice, it shall reject the call with cause value # 93, "AAL parameters cannot be supported".

The B-TE that initiated the call, upon receipt of cause value # 93, AAL parameters cannot be supported" may reattempt the call using AAL5.

2.6 Management plane

ILMI is defined in the ILMI 4.0 specification for the UNI.

3 Interworking Function

3.1 Functional requirements

Refer to section 6.1.1/I.580 for a high-level view of the functional requirements

The interworking function (IWF) converts the voice traffic on the B-ISDN (ATM network) to voice traffic on the N-ISDN (narrowband telephony network). On both sides, the traffic is 64 kbit/s PCM-encoded voice. One ATM VCC is mapped to one N-ISDN channel dynamically on a per call basis. For that purpose, the IWF also maps the B-ISDN signalling information (on VC=5) to N-ISDN signalling information (on the D-channel). To preserve service integrity, the IWF attempts to keep intact as much as possible the information of the signalling channel.

In the opposite direction from the N-ISDN to the B-ISDN, an inverse set of operations to the above is performed.

3.2 Interfaces and protocols

This section specifies mapping requirements to interwork between N-ISDN and B-ISDN at the interworking function (IWF). It also proposes interworking configurations and a protocol reference model. The interworking configurations shown in Figure 3-1 are derived from scenario B case 1 of Annex A/I.580, but contains additional clarification. See section 1.2 for more details. Also, it allows for the use of AAL5 for the transport of voiceband information, as described in section 2.4.3.1. When the N-ISDN is a public network, the signalling protocol between the B-ISDN and the IWF is DSS2, and the signalling between the IWF and the N-ISDN is DSS1. When the N-ISDN is a private network, the signalling protocol between the B-ISDN and the IWF is PNNI signalling or DSS2, and the signalling protocol between the IWF and the private N-ISDN is PSS1 (Q.SIG).

Two B-ISDN private networks can be connected through an N-ISDN network (public or private), or a B-ISDN network can be connected to narrowband devices (N-ISDN or analogue) through a N-ISDN.



Figure 3-1 Interworking configurations

The S and T reference points are described in ITU-T Recommendation I.411. The S_B and T_B reference points are described in ITU-T Recommendation I.413. The Q reference point is described in ISO/IEC 11579-1.

In Figure 3-1, at the T reference point, the user side of the interface is at the left and the network side of the interface is at the right. At the T_B reference point, the user side of the interface is at the left and the network of the interface is at the right if the IWF maps messages one-to-one; if the IWF fully terminates the call control protocol, the user side and the network side can be reversed. See section 3.4.2 for more details.

A private B-ISDN may consist of a multitude of devices, including a private ATM switch or a network of private ATM switches.

A private N-ISDN may consist of a multitude of devices, including a PBX or a network of PBXs.

- Public N-ISDN

In this interworking scenario, the N-ISDN sees the IWF as a B-TE or as a PBX. Depending on which side is the user side, the private B-ISDN sees the IWF either as a B-TE or as a public B-ISDN.

- Private N-ISDN

In this interworking scenario, the private N-ISDN sees the IWF as another N-ISDN node in the private N-ISDN network. Similarly, the private B-ISDN sees the IWF as another B-ISDN node in the private B-ISDN network.

In some scenarios, the IWF may be integrated, as shown in Figure 3-2:

- either with the private B-ISDN, in which case there would be no physical PNNI or UNI between the private B-ISDN and the IWF;

- or with the private N-ISDN, in which case there would be no real Q reference point between the IWF and the private N-ISDN.



Figure 3-2 Additional interworking configurations for integrated IWF

When the IWF is integrated, some of the requirements in the following sections may not apply. For example, if the IWF is integrated with the private B-ISDN, there will be no PNNI at the IWF.

3.3 User plane

3.3.1 Protocol Reference Model

Figure 3-3 illustrates the user information (user plane) interworking applicable to the same reference configuration.



Figure 3-3 User information (U-plane)

In the User plane, the role of the IWF is to convert between TDM encoded voice-band information and ATM cellbased voice-band information.

3.3.2 Physical layer

The physical layer on the B-ISDN side of the IWF shall be any of the physical layers defined by the ATM Forum. On the public N-ISDN side of the IWF, the physical layer as defined in ITU-T Recommendations I.430 or I.431 shall apply. There is no standard physical layer for the private N-ISDN side of the IWF (Q.SIG).

3.3.3 ATM Layer

The specification of the ATM layer is as defined in ITU-T Recommendations I.150 and I.361, including OAM cells for F4 and F5 management information flows. In the case of public N-ISDN interworking, the IWF shall act as the network side of the public UNI.

For private N-ISDN interworking, see section 6.1.2.3/PNNI 1.0 for additional clarification.

3.3.4 AAL

The IWF has the option of using either the AAL1 specification for voiceband signal transport as described in ITU-T Recommendation I.363.1, or the AAL5 as specified in ITU-T Recommendation I.363.5.

The IWF shall support both AAL1 for voice and AAL5, or shall behave like a B-TE in which case the requirements of 2.4.3 apply.

See Appendix B for information on interworking between B-TEs using different AALs for voice.

3.3.4.1 Additional requirements for AAL5

The requirements of 2.4.3.1 apply.

3.3.5 Voice stream

The voiceband information is encoded as 64 kbit/s PCM μ -Law or A-Law, as described in ITU-T Recommendation G.711.

3.4 Control plane

3.4.1 Protocol reference model

Figure 3-4 illustrates the signalling interworking (control plane) applicable to the reference configuration of section 3.2.



Figure 3-4 Signalling (C-plane)

The physical layer shall be as specified in section 3.3.2.

The ATM layer shall be as specified in section 3.3.3.

- Public N-ISDN

In the Control plane, the role of the IWF is to act as a protocol converter between DSS2 and DSS1. The SAAL layer shall be as specified in section 4.1 of UNI SIG 4.0 (af-sig-0061.000).

- Private N-ISDN

In the Control plane, the role of the IWF is to be a protocol converter between PSS1 and DSS2 or PNNI signalling. The SAAL layer shall be as specified in section 4.1/UNI SIG 4.0 (af-sig-0061.000) for DSS2 signalling or in section 6.1.2.2/PNNI.0 (af-pnni-0055.000) for PNNI signalling.

3.4.2 Basic Call/Connection Control Signalling

Interworking between the N-ISDN basic call signalling protocol and the B-ISDN basic call/connection control signalling protocol shall be achieved either by fully terminating each protocol within the IWF or by a one-to-one mapping of messages.

Where each protocol is fully terminated, a message of global significance from one network shall result in the corresponding message being sent to the other network, subject to the other network's protocol being in a suitable state for sending that message. Messages of global significance are SETUP, ALERTING, CONNECT, PROGRESS and the first call clearing message (DISCONNECT, RELEASE or RELEASE COMPLETE). When mapping the first call clearing message from one network onto the first call clearing message to be sent to the other network, the choice of message to be sent will depend on the requirements of the other network's protocol (e.g., a RELEASE COMPLETE message as a first call clearing message from the B-ISDN can result in a DISCONNECT message being sent to the N-ISDN, unless it is the first response to a SETUP message from the N-ISDN). Messages of local significance (all other messages) shall be terminated in the IWF if received and shall be generated in the IWF in

accordance with the requirements of the protocol. Protocol timers shall be run in accordance with the requirements of the two protocols.

Where messages are mapped one-to-one, each message from one network shall be mapped onto the corresponding message of the other network, apart from the exceptions detailed in the remaining subsections of 3.4.2. Protocol timers need not be run except where specified in the remaining subsections of 3.4.2.

In both cases, when mapping a message from one protocol to the other, the IWF shall pass on each received information element that is allowed in the corresponding message of the other protocol and carry out any necessary length adjustment. Other contents shall not be changed, except where specified in the remaining subsections of 3.4.2. Received information elements that cannot be mapped shall be discarded. When sending a message, additional information elements shall be generated in accordance with the requirements of the protocol concerned and the requirements of Annex E/Q.2931 (see 3.4.2.8 for more details).

If the IWF converts between DSS2 and PSS1, a simple one-to-one mapping is impossible and the protocol shall be fully terminated on both sides of the IWF.

3.4.2.1 Public N-ISDN

The procedures of section 6/Q.2931 "Procedures for the support of 64 kbit/s based circuit-mode ISDN services in B-ISDN and access signalling interworking between N-ISDN and B-ISDN" and Annex E/Q.2931 "Mapping functions to support 64 kbit/s based circuit-mode ISDN services in B-ISDN and interworking between N-ISDN and B-ISDN (DSS1/DSS2)" shall apply, as modified per section 2/UNI 4.0 "Basic Point-to-Point Call".

For clarity, this section refers to Q.2931 sections numbers because UNI 4.0 is a pointer to Q.2931, with a list of clarification. It should be understood that the modifications described in UNI 4.0 apply.

The following DSS1 procedures are not supported by UNI 4.0 and should be handled locally by the IWF, if required:

- Annex E/Q.931 Network specific facility selection
- Annex F/Q.931 D-channel backup procedures
- Annex H/Q.931 Message segmentation procedures

The B-TE shall cut-through the voice path when initiating the call.

3.4.2.2 Private N-ISDN

PNNI 1.0 does not explicitly support the procedures for the support of 64 kbit/s based circuit-mode ISDN services in B-ISDN and the mapping functions for B-ISDN/N-ISDN interworking. However, PNNI 1.0 supports the transparent transport of the information elements and messages required to interwork with N-ISDN. The material in this section is thus also applicable to Private N-ISDN interworking.

3.4.2.3 Overlap sending and receiving

If the N-ISDN network uses overlap sending and receiving, the IWF will have to collect the digits and determine when the number is completed, before it can format the addressing information in one the format supported by UNI or PNNI.

3.4.2.4 Instruction indicators

DSS2 and PNNI supports instruction indicators, but not DSS1 and PSS1.

In the N-ISDN to B-ISDN direction, the IWF shall generate the instruction indicator as per the guidelines of Appendix I/Q.2931.

In the B-ISDN to N-ISDN direction, the IWF shall conform to the requirements of sections 5.7, 5.8/Q.2931 for public N-ISDN interworking and the requirements of sections 6.5.7, 6.5.8/PNNI 1.0 for private N-ISDN interworking. In some cases, it means the IWF may need to clear the call on both the B-ISDN and the N-ISDN sides of the IWF.

3.4.2.5 Call clearing

This subsection does not apply if both protocols are fully terminated within the IWF.

Since the call clearing procedures are the same for both public and private N-ISDN interworking, this section covers both cases.

The call clearing sequences of N-ISDN and B-ISDN differ since in B-ISDN does not support the DISCONNECT message. The normal N-ISDN call clearing sequence is DISCONNECT/RELEASE/RELEASE COMPLETE while the normal B-ISDN call clearing sequence is RELEASE/RELEASE COMPLETE.

UNI 4.0 and PNNI 1.0 does not explain how to map the call clearing sequence between N-ISDN and B-ISDN.

In the following requirements, when referring to a message being "mapped", it should be understood that the content of the message should be preserved as much as possible, even when mapping from one message to another (e.g., when mapping a N-ISDN DISCONNECT message to a B-ISDN RELEASE message. By opposition, when referring to a "locally-generated" message, it should be understood that the message is generated by the IWF and does not have significance across the IWF.

Figure 3-5 illustrates the mapping of call clearing messages from N-ISDN to B-ISDN and from B-ISDN to N-ISDN.



Figure 3-5 Mapping of call clearing messages

- N-ISDN -> B- ISDN

A N-ISDN DISCONNECT message shall be mapped to a B-ISDN RELEASE message; the procedures of 5.4.4/ITU-T Q.2931 for public N-ISDN interworking or 6.5.3.3/PNNI 1.0 for private N-ISDN interworking shall apply. The IWF shall respond to the N-ISDN side by sending a locally-generated N-ISDN RELEASE message; the procedures of 5.3.3/ITU-T Q.931 for public N-ISDN interworking or 10.2.3/ISO/IEC 11572 for private N-ISDN interworking shall apply. Normally, the N-ISDN side should respond with a N-ISDN RELEASE COMPLETE message that shall cause the IWF to stop timer T308. In response to the B-ISDN RELEASE message, the B-ISDN side should respond with a B-ISDN RELEASE COMPLETE message that shall cause the IWF to stop timer T308.

A N-ISDN RELEASE message shall be mapped to a B-ISDN RELEASE message; the procedures of 5.4.4/ITU-T Q.2931 for public N-ISDN interworking or 6.5.3.3/PNNI 1.0 for private N-ISDN interworking shall apply. The IWF shall respond to the N-ISDN side by sending a locally-generated N-ISDN RELEASE COMPLETE message. In response to the B-ISDN RELEASE message, the B-ISDN side should respond with a B-ISDN RELEASE COMPLETE message that shall be ignored by the IWF.

If a N-ISDN RELEASE COMPLETE is received as the first call clearing message (e.g., as a response to a SETUP message), it shall be mapped to a B-ISDN RELEASE COMPLETE message.

- B-ISDN -> N- ISDN

A B-ISDN RELEASE message shall be mapped to a N-ISDN DISCONNECT message, and timer T305 shall be started. The IWF shall respond to the B-ISDN side by sending a locally-generated B-ISDN RELEASE COMPLETE message. In response to the N-ISDN DISCONNECT message, the N-ISDN side should respond with a N-ISDN RELEASE message to which the IWF should respond with a locally-generated N-ISDN RELEASE COMPLETE message. On receipt of the N-ISDN RELEASE message, timer T305 shall be canceled. If timer T305 expires, the IWF shall send a N-ISDN RELEASE message to the network with the cause number originally contained in the N-ISDN DISCONNECT message. In addition, the IWF may indicate a second Cause information element with cause No. 102, "recovery on timer expiration".

Note - The default value for timer T305 is defined as 30 s in ITU-T Recommendation Q.931 and in ISO/IEC 11572.

If a B-ISDN RELEASE COMPLETE is received as the first call clearing message (e.g., as a response to a SETUP message), it shall be mapped to a N-ISDN RELEASE COMPLETE message.

3.4.2.5.1 Clearing collision

Clearing collision occurs when the IWF receives a call clearing indication (either a DISCONNECT or a RELEASE message) simultaneously from the B-ISDN and the N-ISDN side specifying the same call (this is done by correlating the call reference values on both sides).

If the IWF receives a N-ISDN DISCONNECT message after sending a N-ISDN DISCONNECT message, it shall stop N-ISDN timer T305, release the call reference, disconnect the information channel (if not already disconnected), send a N-ISDN RELEASE message and start timer T308. The procedures on the B-ISDN side are not affected.

If the IWF receives a N-ISDN RELEASE message after sending a N-ISDN RELEASE message, it shall stop timer T308, release the call reference and disconnect the information channel without sending a N-ISDN RELEASE COMPLETE message. The procedures on the B-ISDN side are not affected.

If the IWF receives a B-ISDN RELEASE message after sending a B-ISDN RELEASE message, it shall stop timer T308, release the call reference and virtual channel without sending a B-ISDN RELEASE COMPLETE message. The procedures on the N-ISDN side are not affected.

Figure 3-6 illustrates the mapping of call clearing messages from N-ISDN to B-ISDN and from B-ISDN to N-ISDN when there is call collision.



Figure 3-6 Mapping of call clearing messages with collision

3.4.2.6 Connection acknowledgment

This subsection does not apply if both protocols are fully terminated within the IWF.

3.4.2.6.1 Public N-ISDN

There is no special requirements for public N-ISDN interworking.

3.4.2.6.2 Private N-ISDN

The PNNI signalling protocol does not support the CONNECT ACKNOWLEDGE message and the associated T313 timer. If possible, the CONNECT ACKNOWLEDGE message shall not be supported on the N-ISDN side by bilateral agreement, as mentioned in the note at the end of section 10.1.6/ISO/IEC 11572.

However, if this is not possible, the IWF shall respond to a N-ISDN CONNECT message by sending a locallygenerated N-ISDN CONNECT ACKNOWLEDGE message. The IWF should ignore a N-ISDN CONNECT ACKNOWLEDGE message.

The IWF shall not support timer T313.

Figure 3-7 illustrates the mapping of call/connection establishment messages from private N-ISDN to private B-ISDN and from private B-ISDN to private N-ISDN when the private N-ISDN makes use of the CONNECT ACKNOWLEDGE message.



Figure 3-7 Mapping of connection establishment messages in private N-ISDN/private B-ISDN

3.4.2.7 Restart procedure

Restart procedures shall operate independently across each interface. On the B-ISDN side, restart procedures shall operate in accordance with section 5.5/Q.2931 or section 6.5.5/PNNI 1.0. On the N-ISDN side, restart procedures shall operate in accordance with section 5.5 of Q.931 or subclause 11.1/ISO/IEC 11572.

As a result of performing a restart of "all interfaces" on the N-ISDN side, the IWF shall initiate a restart of "all virtual channels" on the B-ISDN side.

As a result of performing a restart of "single interface" on the N-ISDN side, the IWF shall initiate a restart of any VCIs on the B-ISDN side that are currently mapped to channels on the N-ISDN interface concerned.

As a result of performing a restart of "indicated channels" on the N-ISDN side, the IWF shall initiate a restart of any VCIs on the B-ISDN side that are currently mapped to the N-ISDN channels concerned.

As a result of performing a restart of "all virtual channels" on the B-ISDN side, the IWF shall initiate a restart of "all interfaces" on the N-ISDN side.

As a result of performing a restart of "all virtual channels in the indicated VPC" on the B-ISDN side, the IWF shall initiate restart of any N-ISDN channels that are currently mapped to VCIs in the indicated VPC.

As a result of performing a restart of "indicated virtual" on the B-ISDN side, the IWF shall initiate restart of any N-ISDN channel that is currently mapped to the indicated VCI.

3.4.2.8 Additional requirements for basic call/connection control

This section provides additional guidance for section 6/Q.2931 and Annex E/Q.2931. To maintain consistency with UNI 4.0, the text of Q.2931 shall apply unless otherwise stated: only the subsections with exceptions are given below. In this implementation agreement, subsections of section 6/Q.2931 and Annex E/Q.2931 are identified by the actual subsection number from that Recommendation with the Recommendation number appended.

The IWF shall map N-ISDN codeset 4, 5, 6 and 7 information elements to B-ISDN. However, it is possible that they will be discarded within the B-ISDN.

Throughout section 6/Q.2931, whenever "AAL for voice" is encountered, it is understood that it may be "AAL1 for voice" or "AAL5 for voice", e.g., in sections 6.2.3/Q.2931, 6.3.3/Q.2931 and 6.7.1/Q.2931.

Figure 6-1/Q.2931 Illustration of the use of N-LLC and B-LLI information elements in Q.2931: The following configuration should be added and is the configuration applicable to this specification.



Annex E/Q.2931 Mapping functions to support 64 kbit/s based circuit-mode ISDN services in B-ISDN and interworking between N-ISDN and B-ISDN (DSS1/DSS2): Since this specification is applicable at both the UNI and at the PNNI, "DSS1" shall be replace by "N-ISDN", and "DSS2" shall be replaced by "B-ISDN" throughout Annex E. The modifications of UNI 4.0 apply.

E.2.1/Q.2931 A B-ISDN user requests the 3.1 kHz audio N-ISDN bearer service: The AAL parameters can be coded as either "AAL for voice" or "AAL5".

E.2.2/Q.2931 A B-ISDN user requests the N-ISDN unrestricted digital information bearer service: Not applicable.

E.2.3/Q.2931 A B-ISDN user requests the N-ISDN telephony teleservice: The AAL parameters can be coded as either "AAL for voice" or "AAL5".

E.2.4/Q.2931 A B-ISDN user requests the N-ISDN videotelephony teleservice based on the unrestricted digital information with tones/announcements bearer capability: Not applicable.

E.3.2/Q.2931 A N-ISDN user requests the unrestricted digital information bearer service: Not applicable.

E.3.4/Q.2931 A N-ISDN user requests the videotelephony teleservice based on the unrestricted digital information bearer capability: Not applicable.

E.4.2.2.1/Q.2931 ATM traffic descriptor for N-BC information transfer capabilities of unrestricted digital information and restricted digital information: Not applicable.

E.4.2.2.2/Q.2931 ATM traffic descriptor for N-BC information transfer capabilities of speech and 3.1 kHz audio:

The following table applies to AAL1 for voice.

Octet	Information element field	Field value if no OAM cells are used (Note 1)	Field value if 1 OAM cells/s is used (Note 2)	Field value with maximal OAM support (Note 3)
7.1 7.2 7.3	Forward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1010 1011 (171 cells/s)	0000 0000 0000 0000 1010 1100 (172 cells/s)	0000 0000 0000 0000 1010 1111 (175 cells/s)
8.1 8.2 8.3	Backward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1010 1011 (171 cells/s)	0000 0000 0000 0000 1010 1100 (172 cells/s)	0000 0000 0000 0000 1010 1111 (175 cells/s)

NOTES

1 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and no cell rate allocation for OAM cells.

2 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and on 1 cell/s allocation for OAM cells.

3 These values are based on an AAL for voice (i.e., AAL type 1 with a payload of 47 octets per cell) for user information and the following cell rate allocation for OAM: two percent of the user cell rate and an additional 1 cell/s.

The following table applies to AAL5.

Octet	Information element field	Field value if no OAM cells are used (Note 1)	Field value if 1 OAM cells/s is used (Note 2)	Field value with maximal OAM support (Note 3)
7.1 7.2 7.3	Forward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1100 1000 (200 cells/s)	0000 0000 0000 0000 1100 1001 (201 cells/s)	0000 0000 0000 0000 1100 1101 (205 cells/s)
8.1 8.2 8.3	Backward peak cell rate (CLP = 0 + 1)	0000 0000 0000 0000 1100 1000 (200 cells/s)	0000 0000 0000 0000 1100 1001 (201 cells/s)	0000 0000 0000 0000 1100 1101 (205 cells/s)

NOTES

1 These values are based on an AAL type 5 payload of 40 octets per cell for user information and no cell rate allocation for OAM cells. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length, encoded in binary format.

2 These values are based on an AAL type 5 payload of 40 octets per cell for user information and on 1 cell/s allocation for OAM cells. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length + 1, encoded in binary format.

3 These values are based on an AAL type 5 payload of 40 octets per cell for user information and the following cell rate allocation for OAM: two percent of the user cell rate and an additional 1 cell/s. For a payload of less than 40 octets, the filed value shall be $8000 \div$ payload_length x 1.02 + 1, encoded in binary format.

E.4.2.4.1/Q.2931 AAL parameters for N-BC information transfer capabilities of unrestricted digital information and restricted digital information: Not applicable.

E.4.2.4.2 AAL parameters for N-BC information transfer capabilities of speech and 3.1 kHz audio

The table in E.4.2.4.2/Q.2931 applies to AAL1 for voice.

The following table applies to AAL5.

Octet	Information element field	Field value
5	AAL-Type	0000 0101 AAL Type 5
6.1 6.2	Forward maximum CPCS-SDU size	Any multiple of 8 between 8 and 40, encoded in binary format
7.1 7.2	Backward maximum CPCS-SDU size	Any multiple of 8 between 8 and 40, encoded in binary format
8 8.1	SSCS-type	absent (preferred), or 0000 0000 null

E.4.2.4.3/Q.2931 AAL parameters for N-BC information transfer capabilities of unrestricted digital information with tones/announcement: Not applicable.

3.4.2.8.1 Called party number, Calling party number and Connected number information elements:

The requirements of section 6 apply.

3.4.2.8.2 Connection identifier information element

This subsection does not apply if both protocols are fully terminated within the IWF. The IWF shall replace the Connection identifier information element by a Channel identifier information element, and vice-versa. At call establishment, the IWF needs to allocate a mapping between a Connection identifier and a Channel identifier. It is outside the scope of this specification to describe how it is done. After a call is established, the IWF needs to maintain a mapping of the Connection identifiers to the Channel identifiers so that the VPI/VCI and channel can be released correctly on call/connection clearing.

3.4.3 Mapping of Supplementary Services Signalling

This section divides supplementary services in two classes:

Non GF-based: Supplementary services which do not require the use of the Generic Functional protocol.

GF-based: Supplementary services that use the Generic Functional protocol.

3.4.3.1 UNI 4.0 Supplementary Services

The IWF shall support the mapping of the supplementary services defined in UNI 4.0 between DSS2 and DSS1. The PNNI 1.0 signalling specification does not explicitly support the procedures for N-ISDN Number identification supplementary services in B-ISDN as described in Annex 4/UNI 4.0. However, these services are implicitly supported by the PSS1 protocol as part of basic call control and by PNNI 1.0 since these services use basic call/connection control procedures. Therefore, the requirements this section are applicable both for public and private N-ISDN interworking.

3.4.3.1.1 Direct Dialing In (DDI)

The requirements of section 6 apply.

3.4.3.1.2 Multiple Subscriber Number (MSN)

The requirements of section 6 apply.

3.4.3.1.3 Calling Line Identification Presentation (CLIP)

The requirements of section 6 apply.

3.4.3.1.4 Calling Line Identification Restriction (CLIR)

The requirements of section 4.11.1/Q.2951 apply.

3.4.3.1.5 Connected Line Identification Presentation (COLP)

The requirements of sections 6 and 5.11.1/Q.2951 apply.

3.4.3.1.6 Connected Line Identification Restriction (COLR)

The requirements of section 6.11.1/Q.2951 apply.

3.4.3.1.7 Subaddressing (SUB)

- N-ISDN -> B-ISDN

The DSS1 Called party subaddress information element is mapped to the DSS2 Called party subaddress information element by the IWF by inserting the second octet and adjusting the length indication without causing other changes to the contents, and by respecting the order of this information element in the DSS2 message.

3.4.3.1.8 User-to-User Signalling (UUS) Service 1

The requirements of section 6.11.1/Q.2957.1 shall apply for public N-ISDN interworking. This supplementary service is not supported by PNNI 1.0 or PSS1 and is thus not applicable to private N-ISDN interworking. The manufacturer specific information transfer capability of the PSS1 Generic Functional protocol offers equivalent functionality.

3.4.3.2 Generic Functional protocol based supplementary services

For further study.

3.4.4 AAL negotiation

If the IWF supports both AAL1 for voice and AAL5, no AAL negotiation procedures are required. Otherwise, it shall behave as a B-TE and the requirements of 2.5.4 are applicable.

3.5 Management plane

ILMI is defined in the ILMI 4.0 and PNNI 1.0 specifications for respectively the UNI and the PNNI. In the case of public N-ISDN interworking, the IWF shall act as the network side of the public UNI. Special MIB elements to manage the IWF are for further study.

MIBs between N-ISDN and the IWF are outside the scope of this specification. In the case of public N-ISDN interworking, the IWF shall act as the user side of the interface (BRI or PRI).

3.7 PNNI Routing Protocol

The PNNI 1.0 specification includes a signalling and routing protocol. Full signalling functionality requires PNNI routing and so PNNI signalling cannot be used without the routing functions.

To support PNNI, the IWF must support the minimum function subset as described in Annex G/PNNI 1.0. This enables the IWF to act as node within a lowest level PNNI peer group.

In a PNNI topology, if an IWF with the minimum function subset has a single connection to the ATM network, it is a stub node from a routing perspective. This means that it terminates and originates calls, but does not have tandem VCs. For reliability, an IWF could be connected by multiple PNNI links to the ATM network. If the network operator does not want the IWF to tandem calls in this topology, the restricted transit bit can be set in the nodal flags advertised in the Nodal IG. This will allow only originating and terminating call traffic.

The IWF could optionally implement the:

- border node subset
- border node with logical group node peer support
- PNNI options subset

Note - In the above configurations, the IWF could be part of an ATM switch that supports the full PNNI protocol.

4 Timing and synchronisation

The voice and telephony services of the N-ISDN are synchronous services. To minimize service defects, public networks are synchronised to primary reference standards. Refer to ANSI T1.101 and ITU-T Recommendation G.811 for accuracy requirements for synchronisation in public networks. A timing relationship is therefore required when a B-TE is connected to the N-ISDN via an IWF. The PCM codec in the B-TE may have to be synchronised to the network clock of the N-ISDN.

4.1 Timing distribution mechanisms

There are three distribution mechanism available:

- Network based (physical layer),
- Adaptive (ATM and/or AAL layer),
- Independent (free running clock),

Timing derived from a network based clock (i.e., the physical layer), when available, is the preferred timing distribution mechanism. However, there are many cases where no network clock distribution mechanism is available: e.g., legacy ATM LANs. Additionally, the 100 Mbit/s Fiber (TAXI) ATM physical interfaces does not have any network based (physical layer) timing distribution mechanism.

The timing distribution mechanism is technically independent of the choice of AAL (i.e., AAL5 or AAL1 for voice). However, ITU-T Recommendation I.363.1 states that network based timing distribution (at the physical layer) shall be used for AAL1 for voice. Within private networks, adaptive or independent timing distribution mechanisms could be used. At a public UNI, the timing distribution mechanism shall be network based (physical layer).

4.1.1 Network based timing distribution (physical layer)

Network based timing distribution mechanisms are shown in Figure 4-1. Network based timing distribution is typically achieved by receiver recovery of the line rate of the physical interface (e.g. DS1, SONET).

In some cases a separate clock distribution network may be used.

Information on the distribution of network timing may be found in Bellcore document TR-NWT-001244. Public networks typically utilize timing traceable to primary references. Some private networks may not be traceable to such references in which case alternative synchronisation mechanisms should be employed.



Figure 4-1 Network Based Timing Distribution Mechanisms

4.1.2 Adaptive timing (ATM and/or AAL layer)

The adaptive clock method is a general method for source clock frequency recovery. No explicit timing information of the source clock is transported by the network: the method is based on the fact that the amount of transmitted data is an indication of the source frequency, and this information can be used to recover the source clock frequency. By averaging the amount of received data over a period of time, CDV (Cell Delay Variation) effects are counteracted. The period of time used for averaging depends on the CDV characteristics.

The implementation of the method is not standardized. One possible method to measure the amount of data is to use the fill level of the AAL user data buffer. The following is the general description of this method and does not preclude other adaptive clock methods.

The B-TE writes the received data into a buffer, and then reads it out using a locally generated clock. Therefore the fill level of the buffer depends on the source frequency and it is used to control the frequency of the local clock. Operations are the following: the fill level of the buffer is continuously measured and the measure is used to drive the phase-locked loop generating the local clock. The method maintains the fill level of the buffer around its medium position. To avoid buffer underflow or overflow, the fill level is maintained between two limits. When the level in the buffer goes to the lower limit, this means the frequency of the local clock is too high compared to the one of the source and so it has to be decreased ; when the level in the buffer goes to upper limit, the frequency of the local clock is too low compared to the one of the source, and so it has to be increased.

4.1.3 Independent timing (free running clock)

Independent timing requires access to a reference clock of sufficient accuracy to minimize the slips at the synchronisation boundary.

If free-running clocking is implemented, then bit slip and cell loss or cell gain will occur. Under these circumstances, the IWF or the B-TE will have to repeat information, insert silence or noise, or discard information (incipient buffer overflow) to recover. This information could be on a cell basis or preferably on an octet basis for better performance. This option will be satisfactory for simple point to point links which only want to transport 64 kbit/s PCM encoded voice. However it is inadequate for quality voice service delivery to the desktop and may cause errors for some fax/modem data rates. For information, for a 64Kbit voice call between 2 PCs, one +50ppm and the other -50ppm the data will slip one octet per 12 seconds. Should the free running option be chosen, the end stations (B-TE or IWF) would normally repeat the last octet on underrun or delete on overrun.

When a free-running clock is used, slips have to be accepted. ISDN quality of service can not reasonable be met. A reasonable slip rate for speech communication might be in the order of one slip every 10 seconds. This corresponds to a frequency accuracy of 10^{-5} for single octet slips or 10^{-4} when slipping more than 8 octets at the time. For facsimile or modem quality of service, the slip requirement has to be decrease by a factor of 10 to 100. Bit error rate requirements can be relaxed to minimize slip rate.

4.2 **IWF** requirements

Figure 4.2 illustrates the basic structure of the interworking function.



Figure 4-2 Basic structure of the IWF

The IWF shall be synchronised to the N-ISDN public or private network clock. This clock is referenced to the Primary Reference Source (PRS). For private N-ISDN that are not referenced to the PRS, a Stratum 4 clock source is required at a minimum.

The IWF shall use its synchronous clock to write PCM data into the packetiser of the AAL transmitter. This leads to a cell rate of 200 cell/s (full cell fill AAL5) or 171 cell/s (AAL1 for voice).

The IWF shall have the ability to receive cell streams from unsynchronised sources, i.e., free running clocks at the B-TEs. This requires provisions to cope with overflows and underflows in the CDV and reassembly buffer. The buffer is read out using the N-ISDN network clock but may be filled by the AAL receiver at a higher or lower rate.

The N-ISDN network clock may be distributed in the ATM network to the B-TEs by transferring synchronisation information through the physical layers.

4.3 B-TE requirements

There are in principle three possible ways to provide clocks for the operation of the PCM codec in the B-TE:

(n) Network based (physical layer),

(a) Adaptive (ATM and/or AAL layer) which is realised, for example, by using a voltage controlled oscillator (VCXO) controlled by the fill level of the CDV buffer,

(i) Independent (free running clock), using a free running oscillator.

The B-TE is required to have a free running oscillator, option (i), if it may be used in a situation where a network clock signal is not accessible or is not active.

Figure 4.3 illustrates the timing relationship in the B-TE. Table 4.1 summarises which clocks have to be applied to the PCM coder and decoder for the various options.

The algorithm to determine the timing recovery mechanism at the terminal, as summarised by Table 4-1, is as described in the following paragraphs. In addition, the B-TE shall conform to the signalling requirements of section 2.5.3.

If a B-TE has access to a network clock (at the physical layer), it shall synchronise both its RX and TX clocks to this network clock.

If no network clock is available, and if the B-TE does not receive an indication from the other end that it will use adaptive timing recovery, the B-TE shall perform adaptive timing recovery on both it's TX and RX clocks if it has the capability to do so.

If no network clock is available, and if the B-TE receives an indication from the other end that it will perform adaptive timing recovery, the B-TE shall use an independent clock on its TX clock to avoid a timing loop. If the B-TE has two distinct clocks for TX and RX, it shall perform adaptive timing recovery on the RX clock if it has the capability to do so. If the B-TE only has one clock (i.e., if RX and TX clocks are the same), it the RX/TX clock shall be run independently (free-running).

If no network clock is available, and if the B-TE does not have the ability to perform adaptive timing recovery, it shall use a free-running clock on both TX and RX. This is the minimum requirement.

Order of preference	R X clock	TX clock
1	n	n
2 - not allowed if other end signals its intention to do adaptive on both RX and TX clocks	а	а
3 - some B-TEs may not support this option if the RX and TX clocks are the same	a	i
4	i	i





Figure 4-3 Timing relationships in the B-TE

A B-TE using a free running clock requires provisions to cope with overflow or underflow in the CDV and reassembly buffer. The buffer is read out using the local free running clock but may be filled by the AAL receiver at a higher or lower rate which is determined by the IWF or the other B-TE.

Note - A B-TE which has the ability to synchronise to the network clock (physical layer) must know if the ATM network it is connected to provides network clock synchronisation over the physical layer. For a 25.6 Mbit/s interface, for example, this may not be a problem since the chips often have a special output providing 8 kHz synchronisation information. This pin has only to be monitored for activity. In case of a 155 Mbit/s Sonet/SDH interface, the transmission clock will be locked to the N-ISDN clock. At the B-TE, the 8 kHz frame clock may be

derived by division from the line clock. In this case, a B-TE can not directly detect if the physical layer is locked to a reference clock or not. One method for the B-TE to determine if network clock synchronisation is achievable would by a "trial and error method" where the B-TE would try to use the clock derived from the physical layer and conclude that the physical layer is not synchronised if slips occur.

5 Delay and echo

Appropriate echo control measures are recommended on all speech connections where the end-to-end delay exceeds that specified in section 3.1/G.131. Additional guidance is provided in G.176.

It is recommended that echo control be provided as close to the source as possible. For connections to the N-ISDN, hybrid echo control should be provided at the IWF on the N-ISDN side, and acoustic echo control should be provided at the voice terminal on the B-ISDN side.

Appendix A provides a brief overview of some of the issues related to echo in N-ISDN.

6 Addressing

The requirements of section 3/UNI Signalling 4.0 and the guidelines of Annex 1/UNI Signalling 4.0 concerning E.164, DCC and ICD addresses apply. Other ATM Forum specifications may apply.

In addition to the numbering plans supported by UNI 4.0 (E.164, or other), there is a need to support telephony numbers in a N-ISDN Private Numbering Plan (PNP). PNP specifies Private N-ISDN numbers (i.e., telephone numbers used within a private numbering plan and not conforming to E.164). Section 6.1 describes the coding requirements for PNP numbers. Otherwise, the requirements for a PNP number are the same as if it was an E.164 number.

In order to be able to be reached directly from the global public N-ISDN and PSTN, a B-TE must have an E.164 number assigned to it in the N-ISDN/PSTN. If the B-TE uses that E.164 number in the B-ISDN in either the E.164 address structure (i.e., IA5 character format) or the private ATM address structure (i.e., ISO NSAP-encoded E.164 format), the IWF can easily map the numbering information between N-ISDN and B-ISDN by doing a simple format translation. If the B-TE uses another format within the private B-ISDN, the IWF will have to maintain a database to map between the non-E.164 ATM addresses used within the private B-ISDN and the corresponding E.164 numbers. This approach requires considerable maintenance at the IWF.

In order to perform a direct format translation between an E.164 address structure (AFI=E.164) in a B-ISDN and an E.164 numbering plan in a N-ISDN, the DSP field has to be set to zero. Otherwise, a mapping database will have to be used.

Similarly, in order to be able to be reached directly from a private N-ISDN, a B-TE must have a PNP or E.164 number assigned to it in the N-ISDN. If an E.164 number is assigned, the previous paragraph applies. If the B-TE has a PNP assigned to it and if it uses that PNP number in the B-ISDN in the private ATM address structure (i.e., ISO NSAP-encoded Local ATM format), the IWF can easily map the numbering information between N-ISDN and B-ISDN by doing a simple format translation. If the B-TE uses another format within the private B-ISDN, the IWF will have to maintain a database to map between the non-PNP ATM addresses used within the private B-ISDN and the corresponding PNP numbers. This approach requires considerable maintenance at the IWF.

In order to perform a direct format translation between a private Local ATM address structure, i.e., ISO NSAPencoded PNP format (AFI=Local ATM format) in a B-ISDN and an PNP numbering plan in a N-ISDN, the HO-DSP, ESI and SEL fields have to be set to zero. Otherwise, a mapping database will have to be used.

Within a private B-ISDN, i.e., without going through the IWF, any addressing format can be used.

For outgoing calls from a B-ISDN through an IWF to a public N-ISDN, a E.164 format must be used.

For outgoing calls from a B-ISDN through an IWF to a private N-ISDN, a E.164 format or a PNP format must be used.

Figure 6-1 summarises the mapping requirements between addressing formats.



Figure 6-1 Mapping of addresses between B-ISDN and N-ISDN

6.1 Addressing within a N-ISDN Private Numbering Plan

In addition to the AFI values for DCC, ICD and encapsulated E.164 formats, ISO/IEC 8348 specifies AFI values for a "Local IDI". The purpose of such an AFI is to accommodate the coexistence of OSI and non-OSI network addressing schemes, particularly in the context of transmission from non-OSI to OSI protocols. In case of the Local IDI format, the IDI is null, i.e., the total length of the IDP is 2 digits (corresponds to one octet according to the preferred binary encoding). Figure 6-2 illustrates the coding.



Figure 6-2 PNP ATM addressing format

The following AFI value shall be used for encoding a private numbering plan (PNP) address as defined in ISO/IEC 11571 in a binary octet format to allow for the coexistence of proprietary or other non-standard network addressing schemes with the standard OSI network addressing scheme:

А	Format	
Hexadecimal	Bits	
0x49	$0\ 1\ 0\ 0\ 1\ 0\ 0\ 1$	Local ATM Format

Use of the Local format for these non-standard addresses ensures that they can not be confused with standard OSI network addresses. In case of a private network using both OSI-NSAP and E.164 with PNP addressing schemes, PNP addresses can be converted into "Local" binary octets OSI-NSAP formats, and vice-versa.

To support PNP addresses within a B-ISDN as "Local" OSI-NSAP addresses, the specification of a preferred encoding of a Local address (and correspondingly, its mapping to PNP addresses) is necessary. The PNP address shall be indicated at the highest level of the private network numbering plan (e.g., level 2); therefore, no "type of number" indication within the OSI-NSAP structure is needed. This is parallel to the encapsulation of E.164 addresses into OSI-NSAP as per UNI 4.0.

The PNP digits should be encoded as binary coded decimal in octets 2 to octets 9 (thus 16 digits) of the NSAP structure, pre-padded by leading "1111" semi-octets. The maximum length of a private numbering plan number is 16 digits.

Appendix A - Echo Cancellation in N-ISDN

(Informative)

A general description of the applications of echo cancellation in N-ISDN/PSTN is provided in ANSI Technical Report No. 27. The material below provides a brief overview of some of the issues related to echo canceller deployments in existing N-ISDN/PSTN networks.

Echo suppressors and echo cancellers are deployed in the existing PSTN and NISDN networks. Deployments are engineered by the network planners to ensure that the characteristics of each link meets network performance standards. Note that this may result in tandem connections each with their own echo cancellation/suppression devices. Echo suppressors are an older technology developed for analogue networks and specified in ITU-T Recommendation G.164. While echo cancellers can have analogue or digital interfaces and perform the echo subtraction using analogue or digital mechanisms, the term is commonly used to refer to those with digital interfaces and digital subtraction algorithms. More formally, these are Type C echo cancellers and they are described in ITU-T Recommendation G.165. Echo cancellation is the currently preferred approach for new deployments.

Echo cancellation techniques are designed to remove echo from voice signals. Non voice services such as voice band data and facsimile can experience difficulties when operating in conjunction with echo cancellers (and even more so with echo suppressors). Inband tone signaling (2100Hz) is defined in G.164 and G.165 to permit modem manufacturers and end users to disable echo suppression and cancellation that may be present on a given connection. Separate tone conventions permit suppressors and cancellers to be turned off independently. Modems that include echo cancellers and facsimile protocols use these tone in various ways to achieve their best performance.

Network based echo arises from impedance mismatches in the network typically associated with the hybrid transformers associated with conversion between 2 wire and 4 wire analogue transmission systems. Echo signals become objectionable with increased levels and increased delays. For links with distance (delay) below a certain threshold (refer to ITU-T Recommendation G.131 and ANSI T1.508 for guidance), additional loss can be inserted to reduce the effects of echoes. Beyond this threshold echo suppression or echo cancellation techniques are recommended.

The echo canceller performance is sensitive to the distance (delay) between the echo canceller device and the source(s) of the echo. Echo cancellation devices typically need to be correctly provisioned for this delay or else have some training or learning mechanisms to determine the best setting.

Acoustic echoes may also occur (e.g. in "speaker phone" or "hands free" telephony equipment) where there is acoustic coupling between the microphone and speaker's transducers. While the acoustic echo phenomenon is similar to network echo, there are a number of differences (e.g., levels, path loss, protection from howling etc.). Control of acoustic echoes is the responsibility of the digital telephone set manufacturer and public networks have not attempted to control for such echoes in the path. The increasing use of PCs equipped with sound cards as digital telephone sets will place additional requirements on the vendors of those equipment to support the necessary acoustic echo suppression.

Appendix B - Interworking between AAL1 and AAL5

(Informative)

Communication between a B-TE using AAL1 for voice and a B-TE using AAL5 for voice requires an interworking function. This specification does not define the details of such an interworking function. One consequence of interworking is the introduction of additional delay.

When mapping from AAL5 to AAL1 for voice there will be an additional cell construction delay of 5.875 ms plus the maximum CDV of the AAL5 connection in the IWF. Assuming a 40 octet fill CPCS-PDU for AAL5, the end-to-end delay would be 5 ms cell construction delay for AAL5 at the source plus CDV buffering delay and 5.875 ms cell reconstruction delay in the IWF plus the CDV delay of the AAL1 connection. Therefore, the total delay would be 10.875 ms plus CDV of AAL5 and AAL1 connections.

When mapping from AAL1 to AAL5 for voice there will be an additional cell construction delay of 5 ms plus the maximum CDV of AAL1 connection in the IWF. Assuming a 40 octet fill CPCS-PDU for AAL5, the end-to-end delay would be 5.875 ms cell construction delay for AAL1 at the source plus CDV buffering delay and 5 ms cell reconstruction delay for AAL5 in the IWF plus the CDV delay of the AAL5 connection. Therefore, the total delay would be 10.875 ms plus CDV of AAL1 and AAL5 connections.

Another consequence of interworking is that signalling information will need to be converted.