



ATIS-1000017.2008(R2013)

Interworking between the ISDN User-Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS)

AMERICAN NATIONAL STANDARD FOR TELECOMMUNICATIONS



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## ATIS-1000017.2008(R2013), *Interworking between the ISDN User-Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS)*

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American National Standard for Telecommunications

# **Interworking between the ISDN User-Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS)**

**Alliance for Telecommunications Industry Solutions**

Approved September 2008

**American National Standards Institute, Inc.**

## **Abstract**

This Standard describes ANSI parameter, field, and field value extensions to the Q.1980.1 Narrowband Signalling Syntax (NSS) to provide a normalized set of telephony parameters. NSS enables mapping from multiple telephony protocols in use today into a common parameter set.

## FOREWORD

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The information contained in this Foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI's requirements for an ANS. As such, this Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the Standard.

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TABLE OF CONTENTS

---

1 Scope, Purpose, and Application ..... 1

    1.1 Scope..... 1

    1.2 Purpose ..... 1

    1.3 Application ..... 1

2 Normative References..... 1

3 Definitions, Acronyms, & Abbreviations ..... 2

    3.1 Definitions..... 2

    3.2 Acronyms & Abbreviations ..... 2

4 Overview of DSS1 to SIP/NSS Interworking ..... 3

5 DSS1 Messages to SIP Request/Responses ..... 3

    5.1 Call Setup ..... 3

        5.1.1 SETUP Message Information Elements ..... 4

        5.1.2 ALERTING Message Information Elements ..... 6

        5.1.3 PROGRESS Message Information Elements..... 6

        5.1.4 CONNECT Message Information Elements ..... 7

    5.2 Mid-Call Messages ..... 7

        5.2.1 FACILITY Message Information Elements ..... 7

        5.2.2 NOTIFY Message Information Elements..... 8

    5.3 Call Clearing ..... 9

        5.3.1 First Call Clearing Message Information Elements ..... 9

        5.3.2 DSS1 Cause Values ..... 10

6 SIP Request/Response to DSS1 Message ..... 12

    6.1 Pre-Call..... 12

    6.2 Call Setup ..... 12

        6.2.1 INVITE Message Headers ..... 12

        6.2.2 SIP Progress Response Headers..... 14

        6.2.3 ACK Message Headers ..... 15

    6.3 Mid-Call Requests ..... 15

        6.3.1 INFO Message Headers ..... 15

        6.3.2 NOTIFY Message Headers ..... 16

    6.4 Call Clearing ..... 17

        6.4.1 BYE Message Headers ..... 17

        6.4.2 CANCEL Message Headers ..... 18

        6.4.3 SIP Response Codes ..... 18

Annex A ..... 23

    A.1 Scope, ..... 23

    A.2 DSS1 Messages to SIP Request/Responses ..... 23

        A.2.1 Call Setup ..... 23

            A.2.1.1 SETUP Message Information Elements ..... 23

            A.2.1.2 CALL PROCEEDING Message Information Elements ..... 27

            A.2.1.3 ALERTING Message Information Elements ..... 27

            A.2.1.4 PROGRESS Message Information Elements ..... 27

            A.2.1.5 CONNECT Message Information Elements ..... 28

            A.2.1.6 CONNECT ACKnowledgement Message Information Elements ..... 28

        A.2.2 Mid-Call Messages ..... 29

            A.2.2.2 NOTIFY Message Information Elements ..... 31

    A.2.3 Call Clearing ..... 31

        A.2.3.1 DISCONNECT Message Information Elements ..... 32

        A.2.3.2 RELEASE Message Information Elements ..... 32

        A.2.3.3 RELEASE COMPLETE Message Information Elements ..... 33

        A.2.3.4 DSS1 Cause Values ..... 33

A3 SIP Request/Response to DSS1 Message ..... 36

    A3.1 Pre-Call ..... 36

    A3.2 Call Setup ..... 36

        A3.2.1 INVITE Message Headers ..... 36

        A3.2.2 SIP Progress Response Headers ..... 40

        A3.2.3 ACK Message Headers ..... 40

    A3.3 Mid-Call Requests ..... 41

        A3.3.1 INFO Message Headers ..... 41

        A3.3.2 NOTIFY Message Headers ..... 42

    A3.4 Call Clearing ..... 43

        A3.4.1 BYE Message Headers ..... 43

        A3.4.2 CANCEL Message Headers ..... 43

        A3.4.3 SIP Response Codes ..... 44

TABLE OF FIGURES

---

FIGURE 1 - SCOPE OF THIS STANDARD ..... 1

TABLE OF TABLES

TABLE 1 – DSS1 TO SIP CALL SETUP MESSAGES.....	3
TABLE 2 – DSS1 SETUP IE TO SIP INVITE REQUEST HEADERS.....	4
TABLE 2.A - DSS1 BEARER CAPABILITY IE TO SDP OFFER/ANSWER HEADERS.....	6
TABLE 3 – DSS1 ALERTING IE TO SIP 180 RINGING RESPONSE HEADERS .....	6
TABLE 4 – DSS1 PROGRESS IE TO SIP 183 SESSION PROGRESS RESPONSE HEADERS.....	6
TABLE 5 – DSS1 CONNECT IE TO SIP 200OK RESPONSE HEADERS .....	7
TABLE 6 – DSS1 TO SIP MID-CALL MESSAGES.....	7
TABLE 7 – DSS1 FACILITY IE TO SIP INFO HEADERS .....	7
TABLE 8 – DSS1 NOTIFY IE TO SIP NOTIFY HEADERS .....	8
TABLE 9 – DSS1 TO SIP CALL CLEARING MESSAGES.....	9
TABLE 10 – DSS1 FIRST CALL CLEARING IE TO SIP BYE/CANCEL REQUEST HEADERS.....	9
TABLE 11 – DSS1 CAUSE VALUE TO SIP RESPONSES .....	10
TABLE 12 – SIP REGISTER/OPTIONS/UPDATE/PRACK TO DSS1 MESSAGES .....	12
TABLE 13 – SIP INVITE TO DSS1 CALL SETUP MESSAGES .....	12
TABLE 14 – SIP INVITE HEADERS TO DSS1 SETUP MESSAGE IE .....	13
TABLE 15 – SIP INVITE SDP HEADERS TO DSS1 SETUP MESSAGE IE .....	14
TABLE 16 – SIP PROGRESS RESPONSE TO DSS1 PROGRESS MESSAGES .....	15
TABLE 17 – SIP ACK HEADER TO DSS1 CONNECT/ACKNOWLEDGEMENT MESSAGE IES .....	15
TABLE 18 – SIP MID-CALL REQUEST TO DSS1 MID-CALL MESSAGES.....	15
TABLE 19 – SIP INFO HEADERS TO DSS1 FACILITY IE .....	16
TABLE 20 – SIP NOTIFY HEADERS TO DSS1 NOTIFY IE .....	17
TABLE 21 – SIP TO DSS1 CALL CLEARING MESSAGES.....	17
TABLE 22 – SIP BYE TO DSS1 DISCONNECT/RELEASE/RELEASE COMPLETE IES .....	17
TABLE 23 – SIP CANCEL TO DSS1 DISCONNECT/RELEASE/RELEASE COMPLETE IES.....	18
TABLE 24 – SIP TO DSS1 CALL REDIRECTION RESPONSES.....	18
TABLE 25 – SIP TO DSS1 CLIENT ERROR RESPONSES .....	19
TABLE 26 – SIP TO DSS1 SERVER ERROR RESPONSES.....	21
TABLE 27 – SIP TO DSS1 GLOBAL ERROR RESPONSES.....	22
TABLE A. 1 – DSS1 TO SIP CALL SETUP MESSAGES [CORRESPONDS TO TABLE 1].....	23
TABLE A. 2 – DSS1 SETUP IE TO SIP INVITE REQUEST HEADERS [CORRESPONDS TO TABLE 2].....	24
TABLE A.2.A - DSS1 Bearer Capability IE to SDP Offer/Answer Headers [Corresponds to Table 2] .....	26
TABLE A. 3 – DSS1 CALL PROCEEDING IE TO SIP 100 TRYING RESPONSE HEADERS.....	27
TABLE A. 4 – DSS1 ALERTING IE TO SIP 180 RINGING RESPONSE HEADERS [CORRESPONDS TO TABLE 3].....	27
TABLE A. 5 – DSS1 PROGRESS IE TO SIP 183 SESSION PROGRESS RESPONSE HEADERS [CORRESPONDS TO TABLE 4].....	28
TABLE A. 6 – DSS1 CONNECT IE TO SIP 200OK RESPONSE HEADERS [CORRESPONDS TO TABLE 5] .....	28
TABLE A. 7 – DSS1 CONNECT ACK IE TO SIP ACK REQUEST HEADER .....	29
TABLE A. 8 – DSS1 TO SIP MID-CALL MESSAGES [CORRESPONDS TO TABLE 6].....	29
TABLE A. 9 – DSS1 FACILITY IE TO SIP INFO HEADERS [CORRESPONDS TO TABLE 7] .....	30
TABLE A. 10 – DSS1 NOTIFY IE TO SIP NOTIFY HEADERS [CORRESPONDS TO TABLE 8] .....	31
TABLE A. 11 – DSS1 TO SIP CALL CLEARING MESSAGES [CORRESPONDS TO TABLE 9].....	31
TABLE A. 12 – DSS1 DISCONNECT IE TO SIP BYE/CANCEL REQUEST HEADERS [CORRESPONDS TO TABLE 10].....	32
TABLE A. 13 – DSS1 RELEASE IE TO SIP BYE/CANCEL REQUEST HEADERS .....	33
TABLE A. 14 – DSS1 RELEASE COMPLETE IE TO SIP BYE/CANCEL REQUEST .....	33
TABLE A. 15 – DSS1 CAUSE VALUE TO SIP RESPONSES [CORRESPONDS TO TABLE 11].....	34
TABLE A. 16 – SIP REGISTER/OPTIONS/UPDATE/PRACK TO DSS1 MESSAGES [CORRESPONDS TO TABLE 12] .....	36
TABLE A. 17 – SIP INVITE TO DSS1 CALL SETUP MESSAGES [CORRESPONDS TO TABLE 13] .....	36
TABLE A. 18 – SIP INVITE HEADERS TO DSS1 SETUP MESSAGE IE [CORRESPONDS TO TABLE 14] .....	36
TABLE A. 19 – SIP INVITE SDP HEADERS TO DSS1 SETUP MESSAGE IE [CORRESPONDS TO TABLE 15].....	40
TABLE A. 20 – SIP PROGRESS RESPONSE TO DSS1 PROGRESS MESSAGES [CORRESPONDS TO TABLE 16].....	40

TABLE A. 21– SIP ACK HEADER TO DSS1 CONNECT/ACKNOWLEDGEMENT MESSAGE IEs [CORRESPONDS TO TABLE 17] ..... 40

TABLE A. 22 – SIP MID-CALL REQUEST TO DSS1 MID-CALL MESSAGES [CORRESPONDS TO TABLE 18]..... 41

TABLE A. 23 – SIP INFO HEADERS TO DSS1 FACILITY IE [CORRESPONDS TO TABLE 19] ..... 41

TABLE A. 24 – SIP NOTIFY HEADERS TO DSS1 NOTIFY IE [CORRESPONDS TO TABLE 20] ..... 42

TABLE A. 25 – SIP TO DSS1 CALL CLEARING MESSAGES [CORRESPONDS TO TABLE 21]..... 43

TABLE A. 26 – SIP BYE TO DSS1 DISCONNECT/RELEASE/RELEASE COMPLETE IEs [CORRESPONDS TO TABLE 22] ..... 43

TABLE A. 27 – SIP CANCEL TO DSS1 DISCONNECT/RELEASE/RELEASE COMPLETE IEs [CORRESPONDS TO TABLE 23]43

TABLE A. 28– SIP TO DSS1 CALL REDIRECTION RESPONSES [CORRESPONDS TO TABLE 24]..... 44

TABLE A. 29 – SIP TO DSS1 CLIENT ERROR RESPONSES [CORRESPONDS TO TABLE 25] ..... 44

TABLE A. 30 – SIP TO DSS1 SERVER ERROR RESPONSES [CORRESPONDS TO TABLE 26]..... 47

TABLE A. 31 – SIP TO DSS1 GLOBAL ERROR RESPONSES [CORRESPONDS TO TABLE 27]..... 47

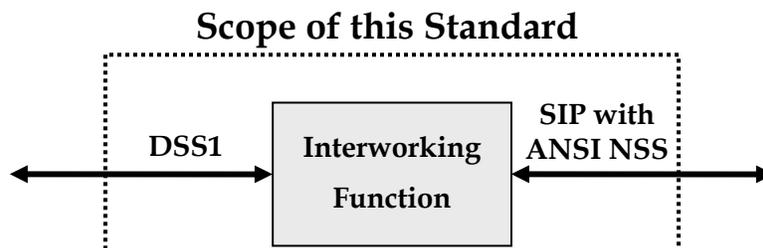
American National Standard for Telecommunications –

# Interworking between the ISDN User-Network Interface Protocol and the Session Initiation Protocol (SIP) with ANSI Extensions to the Narrowband Signaling Syntax (NSS)

## 1 SCOPE, PURPOSE, AND APPLICATION

### 1.1 Scope

This standard defines the interworking relationship between the D-channel layer-3 functions and protocol employed across an ISDN User-Network Interface and an interface using the Session Initiation Protocol (SIP) augmented by the Narrowband Signaling Syntax (NSS) with ANSI Extensions.



**Figure 1 - Scope of this Standard**

### 1.2 Purpose

This standard defines how the ISDN User-Network Interface and SIP/NSS protocols should be used in combination with call control functions to support the ANSI narrowband services.

### 1.3 Application

The interworking between the above two signaling protocols typically may occur at a gateway between an enterprise and a SIP-based service provider network, or may be performed within an IP-PBX system that uses SIP to connect to that service provider.

## 2 NORMATIVE REFERENCES

The following standards contain provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

ATIS-1000113.2005(R2010), July 2005, *Signalling System No.7 (SS7) Integrated Services Digital Network (ISDN) User Part*.<sup>1</sup>

ATIS-1000607.2000(R2009), August 2000, *Integrated Services Digital Network (ISDN) – Layer 3 Signalling Specification for Circuit Switched Bearer Service for Digital Subscriber Signalling System Number 1 (DSS1)*.<sup>1</sup>

ATIS-1000008.2006(R2011), January 2006, *ANSI Extensions to the Narrowband Signaling Syntax (NSS)*.<sup>1</sup>

ISO 3166-1 (1997), *Codes for the representation of names of countries and their subdivisions - Part 1: Country codes*.<sup>2</sup>

ITU-T Q.1980.1, *The Narrowband Signalling Syntax – Syntax Definition*.<sup>3</sup>

RFC 3261, *SIP: Session Initiation Protocol*, June 2002.<sup>4</sup>

RFC 3966, *The tel URI for Telephone Numbers*, December 2004.<sup>4</sup>

### 3 DEFINITIONS, ACRONYMS, & ABBREVIATIONS

---

#### 3.1 Definitions

This Recommendation uses the terms defined in Q.1980.1 and defines no additional terms.

#### 3.2 Acronyms & Abbreviations

ANSI	American National Standards Institute
ATIS	Alliance for Telecommunications Industry Solutions
CUG	Closed User Group
DSS1	Digital Subscriber Signaling System One
GW	Gateway
IETF	Internet Engineering Task Force
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IP	Internet Protocol
ITU-T	International Telecommunications Union - Telecom Sector
IWF	Interworking Function
NSS	Narrowband Signaling Syntax
OOS	Out Of Service
PBX	Private Branch Exchange
PI	Progress Indication
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request For Comment

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<sup>3</sup> This document is available from the International Telecommunications Union. < <http://www.itu.int/ITU-T/> >

<sup>4</sup> This document is available from the Internet Engineering Task Force (IETF). < <http://www.ietf.org> >

SDP	Session Description Protocol
SIP	Session Initiation Protocol
SS7	Signalling System 7

#### 4 OVERVIEW OF DSS1 TO SIP/NSS INTERWORKING

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The document shows bi-directional conversions, both from the DSS1 to SIP with NSS as well as from SIP with NSS to DSS1. The bi-directional mappings between DSS1 and standard SIP are specified elsewhere, and this document is not intended to supersede or modify these mappings in any way. The intent of this document is specifically to provide the mappings between DSS1 and NSS parameters for DSS1 parameters which cannot be currently mapped to standard SIP. If a DSS1 parameter can be mapped to standard SIP messages, then no mapping to NSS parameters will be provided.

Section 5 below shows a conversion of DSS1 messages to SIP Requests/Responses and the mapping of DSS1 specific information elements to SIP headers and session descriptions. The focus is on mapping to NSS parameters. Mappings to standard SIP messages are only provided where this is necessary for understanding.

Section 6 below shows a conversion of SIP Requests/Responses to DSS1 messages and the mapping of SIP headers and session descriptions to DSS1 specific information elements. The focus is on mapping from NSS parameters. Mappings from standard SIP messages are only provided where this is necessary for understanding.

Annex A is informative and shows protocol mappings to support existing implementations.

The flows show that as specific information is added to the DSS1 messages or the SIP headers and session descriptions, the basic nature of a flow may vary with additional DSS1 messages or different SIP responses required to support the supplemental information. An example of this is the progress information element may be present in one of various DSS1 messages and may result, based on its value, in different SIP responses being sent.

The basis for this document is SIP version 2.0, T1.607, T1.609, and T1.679.

NOTE – Overlap sending used in ETSI is not used by ANSI. Extensions would be required if this document is shared with ITU-T SG11 work for similar interworking purposes.

#### 5 DSS1 MESSAGES TO SIP REQUEST/RESPONSES

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This section shows the mapping from DSS1 messages to SIP requests and responses. Note that information elements for which there is no mapping are omitted.

##### 5.1 Call Setup

Table 1 shows the mapping of call setup messages going from DSS1 to SIP.

**Table 1 – DSS1 to SIP Call Setup Messages**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
SETUP	INVITE Request (if overlap sending used by the PSTN/PBX to the gateway, the INVITE is held until all the digits are collected in subsequent INFO messages)
CALL PROCEEDING	No mapping.
ALERTING	180 RINGING Response / 183 SESSION PROGRESS response

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
PROGRESS	183 SESSION PROGRESS Response
CONNECT	200 OK Response
CONNECT ACKNOWLEDGE	No mapping.

**5.1.1 SETUP Message Information Elements**

Table 2 shows the mapping of DSS1 SETUP message information elements to SIP headers and NSS bodies.

**Table 2 – DSS1 SETUP IE to SIP INVITE Request Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Called Party Number	To: RFC 3966 E.164 number mapped into a SIP URI as per RFC 3961/19.1.6
Called Party Subaddress	To: RFC 4715 isdn-subaddress mapped into a sip URI as per RFC 3261/19.1.6
Calling Party Number	From: RFC 3966 E.164 number mapped into a SIP URI as per RFC 3261/19.1.6  NOTE: If there is no calling party number, the From Header should be set to 'Anonymous@anonymous.invalid' at the domain of the gateway.  NOTE: Privacy is indicated with the use of the Privacy Header  If the calling party number is marked as private, the From Header should be set to: 'Anonymous@anonymous.invalid' at the domain of the gateway.  NOTE: The From Header is a required header in SIP session establishment and the P-Asserted Identity can be used to provide a more complete definition of the Calling Party information.
Calling Party Subaddress	From: RFC 4715 isdn-subaddress mapped into a sip URI as per RFC 3261/19.1.6
There is no charge number IE in ISDN	SIP:  P-DCS-Billing-Info: Billing-Correlation-ID/FEID; charge= "addr-spec"  Where the last is one type of Billing-Info-param is Charge-param asserting an Acct-Charge-URI.

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Dialed Number Information Service (DNIS)	NSS MIME body parameters: <sup>1</sup> GEA, <b>diad</b> ,dd,a,a,a,a,# (type,noa,npicni,pi,si,#)
Network Specific Facilities	NSS Trunk Group information (RFC 4904): <sup>1</sup> ton, nip, nid, nsp ind maps to the trunk-context Network-specific facility specification maps to trgp (and is specific to the trunk-context)
Original Called Number (OCN)	From: header Referred-By: header Note: Dependency on INVITE, UPDATE and REFER message usage.
Original Dialed Number (ODN)	To: header History-Info: header
Redirecting Number	History-Info: header
User Entered Code <sup>2</sup>	NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.
User to User	NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: "a" is protocol discriminator and 1*(2Hex) carries the user-user information.

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa [Here we're mapping from ISDN into NSS]

IAM,

NOTE: Other headers not identified above that are required / optional for the INVITE Request are not dictated by the contents of the PRI SETUP message. These are populated independent of the contents of the PRI SETUP message.

<sup>2</sup> This IE is an extension to standard ISDN

**Table 2.a - DSS1 Bearer Capability IE to SDP Offer/Answer Headers**

DSS1 INFORMATION ELEMENT SENT TO IWF	RESULTING SDP OFFER/ANSWER
Bearer Capability: Voice is converted to speech or 3.1 kHz / data is converted 56 rate adapted or 64 unrestricted based upon subscription at PRI or bandwidth information provided in SDP	Media lines (m): m=audio for speech, 3.1 kHz, m= audio/clearmode for 64 kbps data, 56 rate adapted data or packet data Bandwidth (b): Used with (m) to determine Bearer Capability if provided
High Layer Compatibility	No action
Low Layer Compatibility	No action

**5.1.2 ALERTING Message Information Elements**

Table 3 shows the mapping of DSS1 Alerting message information elements to SIP headers and NSS bodies.

**Table 3 – DSS1 ALERTING IE to SIP 180 Ringing Response Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #8	This is not included in the 180 Ringing response. A 183 Session Progress response must be sent with session value of media instead of the 180 Ringing response <sup>1</sup>

NOTE: Other headers not identified above that are required / optional for the 180 Ringing response are not dictated by the contents of the PRI ALERTing message. These are populated independent of the contents of the PRI ALERTing message.

<sup>1</sup> Session header is not a standard SIP header.

**5.1.3 PROGRESS Message Information Elements**

Table 4 shows the mapping of DSS1 Progress message information elements to SIP headers and NSS bodies.

**Table 4 – DSS1 PROGRESS IE to SIP 183 Session Progress Response Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #1, #8	183 with SDP containing Answer
Progress Indicator - #10	Send 183 without SDP

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Cause	----- (ignore as only causes that provide supplementary information are allowed)

NOTE: Other headers not identified above that are required / optional for the 183 Session Progress response are not dictated by the contents of the PRI PROGRESS message. These are populated independent of the contents of the PRI PROGRESS message.

#### 5.1.4 CONNECT Message Information Elements

Table 5 shows the mapping of DSS1 Connect message information elements to SIP headers and NSS bodies.

**Table 5 – DSS1 CONNECT IE to SIP 200OK Response Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #2	Warning 399 : miscellaneous warning

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

#### 5.2 Mid-Call Messages

Table 6 shows the mapping of mid-call messages from DSS1 to SIP.

**Table 6 – DSS1 to SIP Mid-Call Messages**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
FACILITY	INFO
NOTIFY	NOTIFY

#### 5.2.1 FACILITY Message Information Elements

Table 7 shows the mapping of DSS1 Facility message information elements to SIP headers and NSS bodies.

**Table 7 – DSS1 FACILITY IE to SIP INFO Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Operation Value from Code Set	NSS MIME body parameters: <sup>1</sup> GNO,dd  Note: "a" values inserted as appropriate and "dd" is the 2-digit notification indicator selected from the list provided in NSS standard.

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Network Specific Facilities (parameterized network management data)	NSS Trunk Group information (RFC 4904): <sup>1</sup> ton, nip, nid, nsp ind maps to the trunk-context Network-specific facility specification maps to trgp (and is specific to the trunk-context)
User Entered Code	NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.
User to User Information	NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: "a" is protocol discriminator and 1*(2Hex) carries the user-user information.

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa [Here we're mapping from ISDN into NSS]

FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNect message. These are populated independent of the contents of the PRI CONNect message.

### 5.2.2 NOTIFY Message Information Elements

Table 8 shows the mapping of DSS1 Notify message information elements to SIP headers and NSS bodies.

**Table 8 – DSS1 NOTIFY IE to SIP NOTIFY Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Operation Value from Code Set	NSS MIME body parameters: <sup>1</sup> GNO,dd Note: "dd" is the 2-digit notification indicator selected from the list in the NSS standard
Redirect Status	History-Info: header
Redirection Information	History-Info: header (response code 489 Bad Event)

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
User –User Indicators	NSS MIME body parameters: <sup>1</sup> UUI,aaaa,a,a,a,[a] Note: “a” is value selected from the list in the NSS standard

<sup>1</sup> The MIME body starts with parameters (“a” values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa

, [This should be either RES (resume) or SUS (suspend) depending on value of notification indicator IE]

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

### 5.3 Call Clearing

Table 9 shows the mapping of call clearing messages from DSS1 to SIP.

**Table 9 – DSS1 to SIP Call Clearing Messages**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
First Call Clearing message (i.e., DISCONNECT, RELEASE, or RELEASE COMPLETE as appropriate)	BYE Request / CANCEL Request / 4xx, 5xx, 6xx Response, depending of SIP session status
RESTART	BYE Request / CANCEL Request if an active call on the B-channel (s) restarted
RESTART ACKNOWLEDGE	200 OK Response if appropriate based upon what initiated the initial RESTART

NOTE: Responses are not “mapped”, they are generated locally. This applies to the full section 5.3. Cause #16 is the generic call clearing message.

#### 5.3.1 First Call Clearing Message Information Elements

Table 10 shows the mapping of DSS1 First Call Clearing message information elements to SIP headers and NSS bodies.

**Table 10 – DSS1 First Call Clearing IE to SIP BYE/CANCEL Request Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Cause - #16	----- (normal clearing indication, see below for non normal clearing )

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Operation Value from Code Set	NSS MIME body: VER,1.00 PRN,aaaaa,aaa,aaaaa,aaaa REL, GNO,dd Note: dd = 13 – other party disconnected, when applicable
User-User Information	UUS

NOTE: Other headers not identified above that are required / optional for the BYE / CANCEL Request response are not dictated by the contents of the PRI DISConnect message. These are populated independent of the contents of the PRI DISConnect message.

### 5.3.2 DSS1 Cause Values

The following Cause Code mapping is aligned with ATIS-1000679.2004(R2010). When the Cause Value Information Element indicates anything other than 'normal clearing' the following response codes should be used:

**Table 11 – DSS1 Cause Value to SIP Responses**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
Cause Value Information Element	SIP Response / SIP Request
<b>ITU_T Causes</b>	
#1 – unallocated number	404 Not Found Response
#2 – no route to specified transit network	404 Not Found Response
#3 – no route to destination	404 Not Found Response
#6 - channel unacceptable	400 Bad Request Response
#7 – call awarded and being given in an established channel	----- (not sent when clearing call)
# 16 – normal clearing	BYE / CANCEL Request (see above)
#17 – user busy	486 Busy Here Response
#18 – no user responding	480 Temporarily Unavailable Response
#19 - user alerting, no answer	480 Temporarily Unavailable Response
#20 – subscriber absent	480 Temporarily Unavailable Response
#21 – call rejected	403 Forbidden
#22 – number changed	410 Gone
#26 – non selected user clearing	404 Not Found Response

<b>DSS1 MESSAGE NAME SENT TO IWF</b>	<b>RESULTING SIP REQUEST/RESPONSE</b>
#27 – destination out of order	502 Bad Gateway
#28 – Address Incomplete	484 Address Incomplete
#29 –Facility Rejected	501 Not Implemented
#30 – response to status enquiry	----- (not sent when clearing call)
#31 - normal, unspecified	480 Temporarily Unavailable
#34 – circuit/ channel congestion	503 Service Unavailable
#38 – network out of order	503 Service Unavailable
#41 - temporary failure	503 Service Unavailable
#42 – switching equipment congestion	503 Service Unavailable
#43 – access information discarded	----- (not sent when clearing call)
#44 – requested channel not available	480 Temporarily Unavailable Response
#47 – resource unavailable	503 Service Unavailable
#55 – Incoming Class Barred with CUG	403 Forbidden
#57 – bearer capability not authorized	403 Forbidden
#58 – bearer capability not presently available	500 Server Internal Error
#63 – service or option not available, unspecified	480 Temporarily Unavailable Response
#65 – bearer capability not implemented	500 Server Internal Error
#79 – service or option not implemented, unspecified	501 Not Implemented
#81 – invalid call reference value	400 Bad Request Response
#87 – user not member of CUG	500 Server Internal Error
#88 – incompatible destination	500 Server Internal Error
#96 – mandatory information element missing	400 Bad Request Response
#97 – message type nonexistent, or not implemented	400 Bad Request Response
#99 – information element not exist or not implemented	400 Bad Request Response
#100 – invalid information contents	400 Bad Request Response
#101 – message not compatible with call state	400 Bad Request Response
#102 – Recover on timer expiry	480 Temporarily Unavailable Response

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
#111 – Protocol error, unspecified	500 Server Internal Error
#127 – Interworking, unspecified	500 Server Internal Error
<b>National ISDN Specific Causes</b>	
#4 – vacant code	400 Bad Request Response
#8 – prefix 0 dialed in error	400 Bad Request Response
#9 – prefix 1 dialed in error	400 Bad Request Response
#10 – prefix 1 not dialed	400 Bad Request Response
#11- excessive digits received, call proceeding	----- (not sent for call clearing)
#101 - protocol error, threshold exceeded	400 Bad Request Response

6 SIP REQUEST/RESPONSE TO DSS1 MESSAGE

This section shows the mapping from SIP requests and responses to DSS1 messages, i.e. the reverse of mappings found in Section 5. Note that headers for which there is no mapping are omitted.

6.1 Pre-Call

Table 12 shows the pre-call-setup messages used in SIP. Note that there is no equivalent messages used in DSS1.

**Table 12 – SIP Register/Options/UPDATE/PRACK to DSS1 Messages**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
REGISTER (sent from gateway to VoIP Network)	None
OPTIONS	None

6.2 Call Setup

Table 13 shows the call setup message mapping from SIP to DSS1.

**Table 13 – SIP INVITE to DSS1 Call Setup Messages**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
INVITE Request	SETUP

6.2.1 INVITE Message Headers

Table 14 shows the mapping of INVITE headers to SETUP information elements.

**Table 14 – SIP INVITE Headers to DSS1 Setup Message IE**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Request-URI: User-part	Called Party Number IE
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
From: e.164 format for address	<p>Calling Party Number:</p> <p>NOTE: If the From Header should be set to 'unavailable' at the domain of the gateway, then the calling party number may be omitted.</p> <p>Privacy is indicated with the use of the Privacy Header</p> <p>If the From Header should be set to 'Anonymous@anonymous.invalid' at the domain of the gateway, the calling party number is marked as private.</p> <p>The Calling Party information may be derived from the P-Asserted Identity or the CGN parameter from an NSS MIME body.</p>
From: RFC 4715 isdn-subaddress mapped into a sip URI as per RFC 3261/19.1.6	Calling Party Subaddress
Privacy	<p>Calling Party Number (octet 3a)</p> <p>NOTE: Takes precedence over From Header if present</p>
P-Asserted-Identity	<p>Calling Party Number</p> <p>NOTE: Takes precedence over From Header if present</p>
To: e.164 format for address	<p>Called Party Number:</p> <p>If To header is different from the Request-URI:</p> <ul style="list-style-type: none"> <li>- To header maps to Redirecting Party Number</li> <li>- Request-URI maps to Called Party Number</li> </ul> <p>If the To header has operator dialing (0+,0-,00-) it must be parsed into a Operator System Access</p>
To: RFC 4715 hasdn-subaddress mapped into a sip URI as per RFC 3261/19.1.6	Called Party Subaddress

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
History-Info: header	[Redirecting Number IE] Note: Could also be found in Referred-By header
History-Info: header	Original Dialed Number (ODN)
History-Info: header	Redirecting Number/Party (RP)
History-Info: header	[Redirecting Number]
NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: "a" is protocol discriminator and 1*(2Hex) carries the user-user information.	User to User
NSS Trunk Group information (RFC 4904): <sup>1</sup> ton, nip, nid, nsp ind maps to the trunk-context Network-specific facility specification maps to trgp (and is specific to the trunk-context)	Network Specific Facilities

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,aaaaa,aaa,aaaa,aaaa [Here the NSS could be mapped from any protocol]

IAM,

NOTE: Other information elements not identified above that are required / optional for the SETUP Message are not dictated by the contents of the INVITE Request. These are populated independent of the contents of the INVITE Request.

**Table 15 – SIP INVITE SDP Headers to DSS1 Setup Message IE**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Media lines (m): m=audio for speech, 3.1 kHz, m= data for 64 kbps data, 56 rate adapted data or packet data Bandwidth (b): Used with (m) to determine Bearer Capability if provided	Bearer Capability: Voice is converted to speech or 3.1 kHz / data is converted 56 rate adapted or 64 unrestricted based upon subscription at PRI or bandwidth information provided in SDP
Could use UTI NSS parameter	High Layer Compatibility
Could use USI NSS parameter	Low Layer Compatibility

### 6.2.2 SIP Progress Response Headers

Table 16 shows the mapping from SIP provisional response codes to DSS1 progress messages.

**Table 16 – SIP Progress Response to DSS1 Progress Messages**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
100 TRYING Response	No mapping.
180 RINGING Response	ALERTING
181 CALL FORWARDED Response	PROGRESS – PI#10
182 CALL QUEUED Response	PROGRESS – PI#10
183 SESSION PROGRESS Response	PROGRESS – PI#8
200 OK Response (to INVITE)	CONNECT

### 6.2.3 ACK Message Headers

Table 17 shows the mapping of SIP ACK headers to DSS1 Connect ACK information elements.

**Table 17 – SIP ACK Header to DSS1 Connect/Acknowledgement Message IEs**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

### 6.3 Mid-Call Requests

Table 18 shows the mapping of mid-call messages from SIP to DSS1.

**Table 18 – SIP Mid-Call Request to DSS1 Mid-Call Messages**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
INFO Request	FACILITY / conversion to inband information
NOTIFY Request	NOTIFY (as appropriate)
UPDATE	None
PRACK	None
REFER	Dependent on action indicated in REFER. Can result in SETUP message

#### 6.3.1 INFO Message Headers

Table 19 shows the mapping of SIP INFO headers and NSS bodies to DSS1 Facility information elements.

**Table 19 – SIP INFO Headers to DSS1 FACILITY IE**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> GNO,dd Note: “a” values inserted as appropriate and “dd” is the 2-digit notification indicator selected from the list provided in NSS standard.	Operation Value from Code Set
NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.	User Entered Code
NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: “a” is protocol discriminator and 1*(2Hex) carries the user-user information.	User to User
NSS Trunk Group information (RFC 4904): <sup>1</sup> ton, nip, nid, nsp ind maps to the trunk-context Network-specific facility specification maps to trgp (and is specific to the trunk-context)	Network Specific Facilities (parameterized network management data)

<sup>1</sup> The MIME body starts with parameters (“a” values inserted according to standard):

VER,aaaa  
 PRN,t1113,aaa,aaaa,aaaa  
 FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

**6.3.2 NOTIFY Message Headers**

Table 20 shows the mapping of SIP Notify headers and NSS bodies to DSS1 Notify message information elements.

**Table 20 – SIP NOTIFY Headers to DSS1 NOTIFY IE**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> GNO,dd Note: “dd” is the 2-digit notification indicator selected from the list in the NSS standard	Operation Value from Code Set
History-Info: header	Redirect Status
History-Info: header	Redirection Information
NSS MIME body parameters: <sup>1</sup> UUI,aaaa,a,a,a,[a] Note: “a” is value selected from the list in the NSS standard	User –User Indicators

<sup>1</sup> The MIME body starts with parameters (“a” values inserted according to standard):

VER,aaaa

PRN,t1113,aaa,aaaa,aaaa

FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

**6.4 Call Clearing**

Table 21 shows the mapping of call clearing messages from SIP to DSS1.

**Table 21 – SIP to DSS1 Call Clearing Messages**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
BYE Request	DISCONNECT
CANCEL Request	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE)
200 OK response (to BYE / CANCEL)	None

**6.4.1 BYE Message Headers**

Table 22 shows the mapping of SIP BYE headers to corresponding DSS1 message information elements.

**Table 22 – SIP BYE to DSS1 Disconnect/Release/Release Complete IEs**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

NOTE: The Cause Value of 'Normal Clearing' is sent when a BYE Request leads to the generation of the PRI DISConnect / RELEase/ RELEase COMplete Message. A failure cause in these messages will be generated by a 4xx, 5xx, 6xx response.

**6.4.2 CANCEL Message Headers**

Table 23 shows the mapping of CANCEL headers to corresponding DSS1 message information elements.

**Table 23 – SIP CANCEL to DSS1 Disconnect/Release/Release Complete IEs**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

NOTE: The Cause Value of 'Normal Clearing' is sent when a CANCEL Request leads to the generation of the PRI DISConnect / RELEase/ RELEase COMplete Message. A failure cause in these messages will be generated by a 4xx, 5xx, 6xx response.

**6.4.3 SIP Response Codes**

Table 24 shows the mapping of SIP final response codes to DSS1 Progress messages.

**Table 24 – SIP to DSS1 Call Redirection Responses**

SIP REQUEXT/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
300 Multiple Choices Response	PROGRESS – PI#10
301 Moved Permanently Response	PROGRESS – PI#10
302 Moved Temporarily Response	PROGRESS - PI#10
303 See Other Response	PROGRESS – PI#10
305 Use Proxy Response	PROGRESS – PI#10
380 Alternate Service Response	PROGRESS – PI#10

The actions indicated below in Table 25 for these Client Error Responses should only be performed if no retry of the Request is occurring.

**Table 25 – SIP to DSS1 Client Error Responses**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
400 Bad Request Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
401 Unauthorized Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 57
402 Payment Required	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21
403 Forbidden Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 57
404 Not Found Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 3
405 Method Not Allowed Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
406 Not Acceptable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
407 Proxy Authorization Required Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21
408 Request Timeout Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 102
409 Conflict Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 41

SIP REQUEXT/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
410 Gone Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
411 Length Required Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
413 Request Entity Too Large Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
414 Request URI Too Long Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
415 Unsupported Media Type Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 79
420 Bad Extension Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
480 Temporarily Unavailable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 18
481 Call Leg / Transaction Does Not Exist Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
482 Loop Detected Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
483 Too Many Hops Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
484 Address Incomplete Response	If enbloc sending - DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 28

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
485 Ambiguous Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
486 Busy Here	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 17
487 Request Canceled	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
488 Not Acceptable Here	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127

The actions indicated below in Table 26 for these Server Error Responses should only be performed if no retry of the Request is occurring.

**Table 26 – SIP to DSS1 Server Error Responses**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
500 Server Internal Error Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 41
501 Not Implemented Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 79
502 Bad Gateway Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 38
503 Service Unavailable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 63
504 Gateway Timeout Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 102

SIP REQUEXT/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
505 Version Not Supported Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127

The actions indicated below in Table 27 for these Global Failure Responses should only be performed if no retry of the Request is occurring.

**Table 27 – SIP to DSS1 Global Error Responses**

SIP REQUEXT/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
600 Busy Everywhere Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 17
603 Decline Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21
604 Does Not Exist Anywhere Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
606 Not Acceptable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 58

**Annex**  
(informative)

**A.1 Scope,**

This Informative Annex provides protocol mappings in support of existing deployments.

EDITORIAL NOTE: The tables below may mirror those found in Clauses 5 and 6 and are so noted. These mirrored tables have added information that is not normative as those found above.

**A.2 DSS1 Messages to SIP Request/Responses**

This section shows the mapping from DSS1 messages to SIP requests and responses. Note that information elements for which there is no mapping are omitted.

**A.2.1 Call Setup**

Table A.1 shows the mapping of call setup messages going from DSS1 to SIP.

**Table A. 1 – DSS1 to SIP Call Setup Messages [corresponds to Table 1]**

<b>DSS1 MESSAGE NAME SENT TO IWF</b>	<b>RESULTING SIP REQUEST/RESPONSE</b>
SETUP	INVITE Request (if overlap sending used by the PSTN/PBX to the gateway, the INVITE is held until all the digits are collected in subsequent INFO messages)
CALL PROCEEDING	100 TRYING Response (No Response if 100 TRYING already sent)
ALERTING	180 RINGING Response / 183 SESSION PROGRESS response
PROGRESS	183 SESSION PROGRESS Response
CONNECT	200 OK Response
CONNECT ACKNOWLEDGE	ACK Request (No Request if ACK already sent)

**A2.1.1 SETUP Message Information Elements**

Table A.2 shows the mapping of DSS1 SETUP message information elements to SIP headers and NSS bodies.

**Table A. 2 – DSS1 SETUP IE to SIP INVITE Request Headers [corresponds to Table 2]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Called Party Number	To: e.164 format for address
Called Party Subaddress	To: Tag field
Calling Party Number	<p>From: e.164 format for address</p> <p>NOTE: If there is no calling party number, the From Header should be set to 'unavailable' at the domain of the gateway.</p> <p>NOTE: Privacy is indicated with the use of the Privacy Header</p> <p>If the calling party number is marked as private, the From Header should be set to: 'Anonymous@anonymous.invalid' at the domain of the gateway.</p> <p>NOTE: The From Header is a required header in SIP session establishment and the P-Asserted Identity can be used to provide a more complete definition of the Calling Party information</p> <p>NSS MIME body parameters: <sup>1</sup> CGN,dd,a,a,a,a,#</p>
Calling Party Subaddress	From: tag field
There is no charge number IE in ISDN	<p>SIP:</p> <p>P-DCS-Billing-Info: Billing-Correlation-ID/FEID; charge= "addr-spec"</p> <p>Where the last is one type of Billing-Info-param is Charge-param asserting an Acct-Charge-URI:</p> <p>and</p> <p>NSS MIME body parameters: <sup>1</sup> CHN,dd,a,# (noa,np,i,#)</p>
Dialed Number Information Service (DNIS)	<p>NSS MIME body parameters: <sup>1</sup> GEA,<b>diad</b>,dd,a,a,a,a,# (type,noa,np,i,cni,pi,si,#)</p>

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Network Specific Facilities	NSS MIME body parameters: <sup>1</sup> NSF,a,a,1*(2Hex),1*(2Hex) (ton, nip, nid, nsp ind) Note: "a" and 1*(2Hex) are selected or inserted values Note: If sent between a legacy PBX and application server, the NSF may be binary encapsulated in an application/ISUP MIME body part.
Original Called Number (OCN)	SIP (in response to REFER): From: header Referred-By: header and NSS MIME body parameters: <sup>1</sup> OCN,dd,a,a,# (noa, npi, pi, #)
Original Dialed Number (ODN)	SIP: To: header and NSS MIME body parameters: <sup>1</sup> GEA, <b>diad</b> ,dd,a,a,a,a,# (type,noa,npi,cni,pi,si,#)
Originating Line Information (OLI) <sup>2</sup>	NSS MIME body parameters: <sup>1</sup> OLI,dd Note: "dd" is line information selected from NSS standard.
Progress Indicator - #1, #3	Subject: 'call is from non ISDN interface'
Redirecting Number	(For INVITEs in response to REFER) NSS MIME body parameters: <sup>1</sup> OCN,dd,a,a,# (noa,npi,pi,#) (For INVITEs in response to REFER) NSS MIME body parameters: <sup>1</sup> RNI,d,a,dd,a (ri,orr,rc,rr) FDC,mi,orr**,a,1*(2Hex) Note: In FDC, "a" indicates selected instruction and 1*(2Hex) contains proprietary orr code

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
User Entered Code <sup>2</sup>	NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.
User to User	NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: "a" is protocol discriminator and 1*(2Hex) carries the user-user information.

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa [Here we're mapping from ISDN into NSS]

IAM,

NOTE: Other headers not identified above that are required / optional for the INVITE Request are not dictated by the contents of the PRI SETUP message. These are populated independent of the contents of the PRI SETUP message.

<sup>2</sup> This IE is an extension to standard ISDN

**Table A.2. a – DSS1 Bearer Capability IE to SDP Offer/Answer Headers [corresponds to Table 2a]**

DSS1 INFORMATION ELEMENT SENT TO IWF	RESULTING SDP OFFER/ANSWER
Bearer Capability: Voice is converted to speech or 3.1 kHz / data is converted 56 rate adapted or 64 unrestricted based upon subscription at PRI or bandwidth information provided in SDP	Media lines (m): m=audio for speech, 3.1 kHz, m= data for 64 kbps data, 56 rate adapted data or packet data  Bandwidth (b): Used with (m) to determine Bearer Capability if provided
High Layer Compatibility	
Low Layer Compatibility	

**A2.1.2 CALL PROCEEDING Message Information Elements**

Table A.3 shows the mapping of DSS1 Call Proceeding message information elements to SIP headers and NSS bodies.

**Table A. 3 – DSS1 CALL PROCEEDING IE to SIP 100 Trying Response Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Channel Identification	-----

NOTE: Other headers not identified above that are required / optional for the 100 Trying response are not dictated by the contents of the PRI CALL PROCEEDing message. These are populated independent of the contents of the PRI CALL PROCEEDing message.

NOTE: If 100 Trying response has already been sent, a second 100 Trying should not be sent unless additional information can be provided (such as Display)

**A2.1.3 ALERTING Message Information Elements**

Table A.4 shows the mapping of DSS1 Alerting message information elements to SIP headers and NSS bodies.

**Table A. 4 – DSS1 ALERTING IE to SIP 180 Ringing Response Headers [corresponds to Table 3]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #8	This is not included in the 180 Ringing response. A 183 Session Progress response must be sent with session value of media instead of the 180 Ringing response <sup>1</sup>

NOTE: Other headers not identified above that are required / optional for the 180 Ringing response are not dictated by the contents of the PRI ALERTing message. These are populated independent of the contents of the PRI ALERTing message.

<sup>1</sup> Session header is not a standard SIP header.

**A2.1.4 PROGRESS Message Information Elements**

Table A.5 shows the mapping of DSS1 Progress message information elements to SIP headers and NSS bodies.

**Table A. 5 – DSS1 PROGRESS IE to SIP 183 Session Progress Response Headers [corresponds to Table 4]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #1, #8	Session: value of media (1)
Progress Indicator - #10	Send 100 Trying response instead of 183 Session Progress response
Cause	----- (ignore as only causes that provide supplementary information are allowed)

NOTE: Other headers not identified above that are required / optional for the 183 Session Progress response are not dictated by the contents of the PRI PROGRESS message. These are populated independent of the contents of the PRI PROGRESS message.

(1) Session header is not a standard SIP header.

**A2.1.5 CONNECT Message Information Elements**

Table A.6 shows the mapping of DSS1 Connect message information elements to SIP headers and NSS bodies.

**Table A. 6 – DSS1 CONNECT IE to SIP 200OK Response Headers [corresponds to Table 5]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Progress Indicator - #2	Subject: 'call is to non ISDN interface'  (if allowed, if not Warning 399 : miscellaneous warning )

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

**A2.1.6 CONNECT ACKnowledgement Message Information Elements**

Table A.7 shows the mapping of DSS1 Connect Acknowledgement message information elements to SIP headers and NSS bodies.

**Table A. 7 – DSS1 CONNECT ACK IE to SIP ACK Request Header**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

NOTE: Other headers not identified above that are required / optional for the ACK Request are not dictated by the contents of the PRI CONNect ACKnowledge message. These are populated independent of the contents of the PRI CONNect ACKnowledge message.

*A2.2 Mid-Call Messages*

Table A.8 shows the mapping of mid-call messages from DSS1 to SIP.

**Table A. 8 – DSS1 to SIP Mid-Call Messages [corresponds to Table 6]**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
FACILITY	INFO
NOTIFY	NOTIFY
STATUS	No Action (related to status of call on PRI)
STAUS ENQUIRY	No Action (related to status of call on PRI)
SERVICE (B-channel for B-channel Availability Signaling)	No Action if no call on B-channel (modify B-channel/ status). If B-channel goes to OOS, send BYE Request / CANCEL Request if an active call on the B-channel being taken OOS Note: when a SERVICE is initiated by the Gateway based upon a gateway failure, the same actions as receiving a SERVICE apply
SERVICE ACKNOWLEDGE (B-channel for B-Channel Availability Signaling)	No Action (acknowledging SERVICE sent to SIP)
SERVICE (D-channel)	Should not receive, as the gateway does not support NFAS. If received with OOS and can be processed, send BYE Request / CANCEL Request to all calls on PRI
SERVICE ACKNOWLEDGEMENT (D-channel)	Should not receive as the gateway does not support NFAS and gateway should never send SERVICE

A2.2.1 FACILITY Message Information Elements

Table A.9 shows the mapping of DSS1 Facility message information elements to SIP headers and NSS bodies.

**Table A. 9 – DSS1 FACILITY IE to SIP INFO Headers [corresponds to Table 7]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Operation Value from Code Set	NSS MIME body parameters: <sup>1</sup> GNO,dd Note: “a” values inserted as appropriate and “dd” is the 2-digit notification indicator selected from the list provided in NSS standard.
Network Specific Facilities (parameterized network management data)	NSS MIME body parameters: <sup>1</sup> NSF,a,a,1*(2Hex),1*(2Hex) (ton, nip, nid, nsp ind) Note: “a” and 1*(2Hex) are selected or inserted values Note: If sent between a legacy PBX and application server, the NSF may be binary encapsulated in an application/ISUP MIME body part.
User Entered Code	NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.
User to User Information	NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: “a” is protocol discriminator and 1*(2Hex) carries the user-user information.

<sup>1</sup> The MIME body starts with parameters (“a” values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa [Here we’re mapping from ISDN into NSS]

FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNect message. These are populated independent of the contents of the PRI CONNect message.

**A2.2.2 NOTIFY Message Information Elements**

Table A.10 shows the mapping of DSS1 Notify message information elements to SIP headers and NSS bodies.

**Table A. 10 – DSS1 NOTIFY IE to SIP NOTIFY Headers [corresponds to Table 8]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Operation Value from Code Set	NSS MIME body parameters: <sup>1</sup> GNO,dd Note: “dd” is the 2-digit notification indicator selected from the list in the NSS standard
Redirect Status	NSS MIME body parameters: <sup>1</sup> RDS,a Note: “a” is the value selected from the list in the NSS standard
User –User Indicators	NSS MIME body parameters: <sup>1</sup> UUI,aaaa,a,a,a,[a] Note: “a” is value selected from the list in the NSS standard
Notification indicator	Note: This determines which NSS message to send

<sup>1</sup> The MIME body starts with parameters (“a” values inserted according to standard):

VER,aaaa

PRN,isdn\*,aaa,aaaa,aaaa

RES (resume) or SUS (suspend). [depending on value of Notification indicator IE]

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

**A2.3 Call Clearing**

Table A.11 shows the mapping of call clearing messages from DSS1 to SIP.

**Table A. 11 – DSS1 to SIP Call Clearing Messages [corresponds to Table 9]**

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
DISCONNECT	BYE Request / CANCEL Request / 4xx, 5xx, 6xx Response
RELEASE	200 OK Response (or BYE Request / CANCEL Request / 4xx, 5xx, 6xx Response, if RELEASE is first disestablishment message)

DSS1 MESSAGE NAME SENT TO IWF	RESULTING SIP REQUEST/RESPONSE
RELEASE COMPLETE	No Action (or CANCEL Request / 4xx, 5xx, 6xx Response, if RELEASE COMPLETE is first disestablishment message)
RESTART	BYE Request / CANCEL Request if an active call on the B-channel (s) restarted
RESTART ACKNOWLEDGE	200 OK Response if appropriate based upon what initiated the initial RESTART

**A2.3.1 DISCONNECT Message Information Elements**

Table A.12 shows the mapping of DSS1 Disconnect message information elements to SIP headers and NSS bodies.

**Table A. 12 – DSS1 DISCONNECT IE to SIP BYE/CANCEL Request Headers [corresponds to Table 10]**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Cause - #16	----- (normal clearing indication, see below for non normal clearing )
Operation Value from Code Set	NSS MIME body: VER,1.00 PRN,aaaaa,aaa,aaaaa,aaaa REL, GNO,dd Note: dd = 13 – other party disconnected, when applicable
User-User Information	UUS

NOTE: Other headers not identified above that are required / optional for the BYE / CANCEL Request response are not dictated by the contents of the PRI DISConnect message. These are populated independent of the contents of the PRI DISConnect message.

**A2.3.2 RELEASE Message Information Elements**

Table A.13 shows the mapping of DSS1 Release message information elements to SIP headers and NSS bodies.

**Table A. 13 – DSS1 RELEASE IE to SIP BYE/CANCEL Request Headers**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Cause - #16	----- (normal clearing indication, see below for non normal clearing )

NOTE: This is only applicable if the RELEase message is the first clearing message processed. If a DISConnect message has already been sent and the RELEase message is in response to the DISConnect message, it is assumed that the BYE / CANCEL Request has already been sent.

NOTE: Other headers not identified above that are required / optional for the BYE / CANCEL Request response are not dictated by the contents of the PRI RELEase message. These are populated independent of the contents of the PRI RELEase message.

**A2.3.3 RELEASE COMPLETE Message Information Elements**

Table A.14 shows the mapping of DSS1 Release Complete message information elements to SIP headers and NSS bodies.

**Table A. 14 – DSS1 RELEASE COMPLETE IE to SIP BYE/CANCEL Request**

DSS1 INFORMATION ELEMENTS SENT TO IWF	RESULTING SIP HEADERS OR NSS
Call Reference	Call-ID: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.
Cause - #16	----- (normal clearing indication, see below for non normal clearing )

NOTE: This is only applicable if the RELEase COMplete message is the first clearing message processed. If a RELEase message has already been sent (e.g., in response to a DISConnect message) and the RELEase COMplete message is in response to the RELEase message, it is assumed that the BYE / CANCEL Request has already been sent.

NOTE: Other headers not identified above that are required / optional for the BYE / CANCEL Request response are not dictated by the contents of the PRI RELEase COMplete message. These are populated independent of the contents of the PRI RELEase COMplete message.

**A2.3.4 DSS1 Cause Values**

When the Cause Value Information Element indicates anything other than 'normal clearing' the following response codes should be used:

**Table A. 15 – DSS1 Cause Value to SIP Responses [corresponds to Table 11]**

<b>DSS1 MESSAGE NAME SENT TO IWF</b>	<b>RESULTING SIP REQUEST/RESPONSE</b>
<b>Cause Value Information Element</b>	<b>SIP Response / SIP Request</b>
<b>ITU_T Causes</b>	
#1 – unallocated number	404 Not Found Response
#2 – no route to specified transit network	404 Not Found Response
#3 – no route to destination	404 Not Found Response
#6 - channel unacceptable	400 Bad Request Response
#7 – call awarded and being given in an established channel	----- (not sent when clearing call)
# 16 – normal clearing	BYE / CANCEL Request (see above)
#17 – user busy	486 Busy Here Response
#18 – no user responding	480 Temporarily Unavailable Response
#19 - user alerting, no answer	480 Temporarily Unavailable Response
#20 – subscriber absent	480 Temporarily Unavailable Response
#21 – call rejected	403 Forbidden
#22 – number changed	410 Gone
#26 – non selected user clearing	404 Not Found Response
#27 – destination out of order	404 Not Found Response
#28 – Address Incomplete	484 Address Incomplete
#29 –Facility Rejected	501 Not Implemented
#30 – response to status enquiry	----- (not sent when clearing call)
#31 - normal, unspecified	BYE / CANCEL Request (same as #16) or 404 Not Found Response
#34 – circuit/ channel congestion	503 Service Unavailable
#38 – network out of order	503 Service Unavailable
#41 - temporary failure	503 Service Unavailable
#42 – switching equipment congestion	503 Service Unavailable
#43 – access information discarded	----- (not sent when clearing call)
#44 – requested channel not available	480 Temporarily Unavailable Response

<b>DSS1 MESSAGE NAME SENT TO IWF</b>	<b>RESULTING SIP REQUEST/RESPONSE</b>
#47 – resource unavailable	503 Service Unavailable
#55 – Incoming Class Barred with CUG	403 Forbidden
#57 – bearer capability not authorized	403 Forbidden
#58 – bearer capability not presently available	501 Not Implemented
#63 – service or option not available, unspecified	480 Temporarily Unavailable Response
#65 – bearer capability not implemented	501 Not Implemented
#79 – service or option not implemented, unspecified	501 Not Implemented
#81 – invalid call reference value	400 Bad Request Response
#87 – user not member of CUG	503 Service Unavailable
#88 – incompatible destination	400 Bad Request Response
#96 – mandatory information element missing	400 Bad Request Response
#97 – message type nonexistent, or not implemented	400 Bad Request Response
#99 – information element not exist or not implemented	400 Bad Request Response
#100 – invalid information contents	400 Bad Request Response
#101 – message not compatible with call state	400 Bad Request Response
#102 – Recover on timer expiry	408 Request Timeout Response
#111 – Protocol error, unspecified	400 Bad Request Response
#127 – Interworking, unspecified	500 Server Internal Error
<b>National ISDN Specific Causes</b>	
#4 – vacant code	400 Bad Request Response
#8 – prefix 0 dialed in error	400 Bad Request Response
#9 – prefix 1 dialed in error	400 Bad Request Response
#10 – prefix 1 not dialed	400 Bad Request Response
#11- excessive digits received, call proceeding	----- (not sent for call clearing)
#101 - protocol error, threshold exceeded	400 Bad Request Response

A3 SIP REQUEST/RESPONSE TO DSS1 MESSAGE

This section shows the mapping from SIP requests and responses to DSS1 messages, i.e. the reverse of mappings found in Section 5. Note that headers for which there is no mapping are omitted.

A3.1 Pre-Call

Table A.16 shows the pre-call-setup messages used in SIP. Note that there is no equivalent messages used in DSS1.

**Table A. 16 – SIP Register/Options/UPDATE/PRACK to DSS1 Messages [corresponds to Table 12]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
REGISTER (sent from gateway to VoIP Network)	None
OPTIONS	None

A3.2 Call Setup

Table A.17 shows the call setup message mapping from SIP to DSS1.

**Table A. 17 – SIP INVITE to DSS1 Call Setup Messages [corresponds to Table 13]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
INVITE Request	SETUP
ACK Request	CONNECT ACKNOWLEDGE (optional for user side of PRI)

A3.2.1 INVITE Message Headers

Table A.18 shows the mapping of INVITE headers to SETUP information elements.

**Table A. 18 – SIP INVITE Headers to DSS1 Setup Message IE [corresponds to Table 14]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Request-URI: User-part	Called Party Number IE
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
<p>From: e.164 format for address                      NSS MIME body parameters: <sup>1</sup>                      CGN,dd,a,a,a,a,#</p>	<p>Calling Party Number:                      NOTE: If the From Header should be set to 'unavailable' at the domain of the gateway, then the calling party number may be omitted.                      Privacy is indicated with the use of the Privacy Header                      If the From Header should be set to 'Anonymous@anonymous.invalid' at the domain of the gateway, the calling party number is marked as private.                      The Calling Party information may be derived from the P-Asserted Identity or the CGN parameter from an NSS MIME body.</p>
<p>From: tag field</p>	<p>Calling Party Subaddress</p>
<p>Privacy</p>	<p>Calling Party Number (octet 3a)                      NOTE: Takes precedence over From Header if present</p>
<p>P-Asserted-Identity</p>	<p>Calling Party Number                      NOTE: Takes precedence over From Header if present</p>
<p>To: e.164 format for address</p>	<p>Called Party Number:                      If To header is different from the Request-URI:                      - To header maps to Redirecting Party Number                      - Request-URI maps to Called Party Number                      If the To header has operator dialing (0+,0-,00-) it must be parsed into a Operator System Access                      If the To header has equal access dialing (101xxxx) it must be parsed into a Transit Network Selection</p>
<p>To: Tag field</p>	<p>Called Party Subaddress</p>
<p>Subject: 'call is from non ISDN interface'</p>	<p>Progress Indicator - #1, #3</p>
<p>P-DCS-Billing-Info: Billing-Correlation-ID/                      FEID; charge= "addr-spec"                      Where the last is one type of Billing-Info-param is Charge-param asserting an Acct-Charge-URI:</p>	<p>[Charge Number is not a standard ISDN IE, and some implementations may map it to either the Called or Calling Party Number]</p>

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> CHN,dd,a,# (noa,npi,#)	There's no charge number IE in ISDN
NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.	User Entered Code
NSS MIME body parameters: <sup>1</sup> OCN,dd,a,a,# (noa, npi, pi, #)	[Redirecting Number IE] Note: Could also be found in Referred-By header
NSS MIME body parameters: <sup>1</sup> GEA, <b>diad</b> ,dd,a,a,a,a,# (type,noa,npi,cni,pi,si,#)	Original Dialed Number (ODN)
NSS MIME body parameters: <sup>1</sup> OLI,dd Note: "dd" is line information selected from NSS standard.	Originating Line Information (OLI)
(For INVITEs in response to REFER) NSS MIME body parameters: <sup>1</sup> OCN,dd,a,a,# (noa,npi,pi,#)	Redirecting Number/Party (RP)
(For INVITEs in response to REFER) NSS MIME body parameters: <sup>1</sup> RNI,d,a,dd,a (ri,orr,rc,rr) FDC,rni,orr**,a,1*(2Hex) Note: In FDC, "a" indicates selected instruction and 1*(2Hex) contains proprietary orr code	[Redirecting Number]
NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: "a" is protocol discriminator and 1*(2Hex) carries the user-user information.	User to User

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
<p>NSS MIME body parameters:  NSF,a,a,1*(2Hex),1*(2Hex) (ton, nip, nid, nsp ind)  Note: "a" and 1*(2Hex) are selected or inserted values  Note: If sent between a legacy PBX and application server, the NSF may be binary encapsulated in an application/ISUP MIME body part.</p>	<p>Network Specific Facilities</p>

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,aaaaa,aaa,aaaa,aaaa [Here the NSS could be mapped from any protocol]

IAM,

NOTE: Other information elements not identified above that are required / optional for the SETUP Message are not dictated by the contents of the INVITE Request. These are populated independent of the contents of the INVITE Request.

**Table A. 19 – SIP INVITE SDP Headers to DSS1 Setup Message IE [corresponds to Table 15]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Media lines (m): m=audio for speech, 3.1 kHz, m= data for 64 kbps data, 56 rate adapted data or packet data Bandwidth (b): Used with (m) to determine Bearer Capability if provided	Bearer Capability: Voice is converted to speech or 3.1 kHz / data is converted 56 rate adapted or 64 unrestricted based upon subscription at PRI or bandwidth information provided in SDP
Could use UTI NSS parameter	High Layer Compatibility
Could use USI NSS parameter	Low Layer Compatibility

**A3.2.2 SIP Progress Response Headers**

Table A.20 shows the mapping from SIP provisional response codes to DSS1 progress messages.

**Table A. 20 – SIP Progress Response to DSS1 Progress Messages [corresponds to Table 16]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
100 TRYING Response	CALL PROCEEDING (if not in 'Outgoing Call Proceeding' state, e.g., a CALL PROCEEDING message has not already been sent)
180 RINGING Response	ALERTING
181 CALL FORWARDED Response	PROGRESS – PI#10
182 CALL QUEUED Response	PROGRESS – PI#10
183 SESSION PROGRESS Response	PROGRESS – PI#8
200 OK Response (to INVITE)	CONNECT

**A3.2.3 ACK Message Headers**

Table A.21 shows the mapping of SIP ACK headers to DSS1 Connex ACK information elements.

**Table A. 21– SIP ACK Header to DSS1 Connect/Acknowledgement Message IEs [corresponds to Table 17]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

A3.3 Mid-Call Requests

Table A.22 shows the mapping of mid-call messages from SIP to DSS1.

**Table A. 22 – SIP Mid-Call Request to DSS1 Mid-Call Messages [corresponds to Table 18]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
INFO Request	FACILITY / conversion to inband information
NOTIFY Request	NOTIFY (as appropriate)
UPDATE	None
PRACK	None
REFER	Dependent on action indicated in REFER. Can result in SETUP message

**A3.3.1 INFO Message Headers**

Table A.23 shows the mapping of SIP INFO headers and NSS bodies to DSS1 Facility information elements.

**Table A. 23 – SIP INFO Headers to DSS1 FACILITY IE [corresponds to Table 19]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> GNO,dd Note: “a” values inserted as appropriate and “dd” is the 2-digit notification indicator selected from the list provided in NSS standard.	Operation Value from Code Set
NSS MIME body parameters: <sup>1</sup> GED,d,d,# (tod,es,#) FDC,ged,tod**,a,1*(2Hex) Note: a= instruction selected and 1*(2Hex) contains entered code type of digits Note that zero or more occurrences of this GED-FDC set is possible.	User Entered Code
NSS MIME body parameters: <sup>1</sup> UUS,a,1*(2Hex) Note: “a” is protocol discriminator and 1*(2Hex) carries the user-user information.	User to User

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> NSF,a,a,1*(2Hex),1*(2Hex) (ton, nip, nid, nsp ind) Note: "a" and 1*(2Hex) are selected or inserted values Note: If sent between a legacy PBX and application server, the NSF may be binary encapsulated in an application/ISUP MIME body part.	Network Specific Facilities (parameterized network management data)

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa

PRN,t1113,aaa,aaaa,aaaa

FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNect message. These are populated independent of the contents of the PRI CONNect message.

### A3.3.2 NOTIFY Message Headers

Table A.24 shows the mapping of SIP Notify headers and NSS bodies to DSS1 Notify message information elements.

**Table A. 24 – SIP NOTIFY Headers to DSS1 NOTIFY IE [corresponds to Table 20]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
NSS MIME body parameters: <sup>1</sup> GNO,dd Note: "dd" is the 2-digit notification indicator selected from the list in the NSS standard	Operation Value from Code Set
NSS MIME body parameters: <sup>1</sup> RDS,a Note: "a" is the value selected from the list in the NSS standard	Redirect Status
NSS MIME body parameters: <sup>1</sup> RNI,d,a,dd,a Note: "a" and "d" values selected from NSS standard as appropriate.	Redirection Information
NSS MIME body parameters: <sup>1</sup> UUI,aaaa,a,a,a,[a] Note: "a" is value selected from the list in the NSS standard	User –User Indicators

<sup>1</sup> The MIME body starts with parameters ("a" values inserted according to standard):

VER,aaaa  
 PRN,t1113,aaa,aaaa,aaaa  
 FAC,

NOTE: Other headers not identified above that are required / optional for the 200 OK response are not dictated by the contents of the PRI CONNECT message. These are populated independent of the contents of the PRI CONNECT message.

**A3.4 Call Clearing**

Table A.25 shows the mapping of call clearing messages from SIP to DSS1.

**Table A. 25 – SIP to DSS1 Call Clearing Messages [corresponds to Table 21]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
BYE Request	DISCONNECT / RELEASE / RELEASE COMPLETE
CANCEL Request	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE)
200 OK response (to BYE / CANCEL)	None

**A3.4.1 BYE Message Headers**

Table A.26 shows the mapping of SIP BYE headers to corresponding DSS1 message information elements.

**Table A. 26 – SIP BYE to DSS1 Disconnect/Release/Release Complete IEs [corresponds to Table 22]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

NOTE: The Cause Value of 'Normal Clearing' is sent when a BYE Request leads to the generation of the PRI DISConnect / RELEase/ RELEase COMplete Message. A failure cause in these messages will be generated by a 4xx, 5xx, 6xx response.

**A3.4.2 CANCEL Message Headers**

Table A.27 shows the mapping of CANCEL headers to corresponding DSS1 message information elements.

**Table A. 27 – SIP CANCEL to DSS1 Disconnect/Release/Release Complete IEs [corresponds to Table 23]**

SIP HEADERS OR NSS SENT TO IWF	RESULTING DSS1 INFORMATION ELEMENTS
Call-ID	Call Reference: The gateway will maintain the Call-ID to Call Reference association for the duration of the call.

NOTE: The Cause Value of 'Normal Clearing' is sent when a CANCEL Request leads to the generation of the PRI DISConnect / RELEase/ RELEase COMplete Message. A failure cause in these messages will be generated by a 4xx, 5xx, 6xx response.

**A3.4.3 SIP Response Codes**

Table A.28 shows the mapping of SIP final response codes to DSS1 Progress messages.

**Table A. 28– SIP to DSS1 Call Redirection Responses [corresponds to Table 24]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
300 Multiple Choices Response	PROGRESS – PI#10
301 Moved Permanently Response	PROGRESS – PI#10
302 Moved Temporarily Response	PROGRESS - PI#10
303 See Other Response	PROGRESS – PI#10
305 Use Proxy Response	PROGRESS – PI#10
380 Alternate Service Response	PROGRESS – PI#10

The actions indicated below in Table A.29 for these Client Error Responses should only be performed if no retry of the Request is occurring.

**Table A. 29 – SIP to DSS1 Client Error Responses [corresponds to Table 25]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
400 Bad Request Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
401 Unauthorized Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 57
402 Payment Required	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21
403 Forbidden Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 57
404 Not Found Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 3

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
405 Method Not Allowed Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
406 Not Acceptable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
407 Proxy Authorization Required Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21
408 Request Timeout Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 102
409 Conflict Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 41
410 Gone Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
411 Length Required Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
413 Request Entity Too Large Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
414 Request URI Too Long Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
415 Unsupported Media Type Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 79
420 Bad Extension Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
480 Temporarily Unavailable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 18
481 Call Leg / Transaction Does Not Exist Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
482 Loop Detected Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
483 Too Many Hops Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
484 Address Incomplete Response	If enbloc sending - DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 28
485 Ambiguous Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
486 Busy Here	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 17
487 Request Canceled	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127
488 Not Acceptable Here	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127

The actions indicated below in Table A.30 for these Server Error Responses should only be performed if no retry of the Request is occurring.

**Table A. 30 – SIP to DSS1 Server Error Responses [corresponds to Table 26]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
500 Server Internal Error Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 41
501 Not Implemented Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 79
502 Bad Gateway Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 38
503 Service Unavailable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 63
504 Gateway Timeout Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 102
505 Version Not Supported Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 127

The actions indicated below in Table A.31 for these Global Failure Responses should only be performed if no retry of the Request is occurring.

**Table A. 31 – SIP to DSS1 Global Error Responses [corresponds to Table 27]**

SIP REQUEST/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
600 Busy Everywhere Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 17
603 Decline Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 21

SIP REQUEXT/RESPONSE SENT TO IWF	RESULTING DSS1 MESSAGE
604 Does Not Exist Anywhere Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 1
606 Not Acceptable Response	DISCONNECT (if CALL PROCEEDING, ALERTING, and / or PROGRESS has been sent, if not RELEASE COMPLETE) – Cause 58