TTC標準
Standard

JJ-90.30

Common interconnection interface
Between IMS operator's networks
(English Edition)

Version 12.0

May 16, 2024

Introduction

This document provides the TTC original standard formulated by TTC signaling working group. The working group translated JJ-90.30 Japanese version 11.0. (May 18, 2023) into English, and issued JJ-90.30 English version on May 18, 2023.

In case of dispute, the original to be referred is the Japanese edition of the text.

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<Reference Information>

1. Relationship to International Recommendations

This Standard conforms to TTC specification TS-3GA-29.165.

2. Change History

Edition and Version	Date of issue	Contents of changes
English Edition, Version 1.0	May 21, 2015	Published
English Edition, Version 2.1	September 11, 2015	Addition of ENUM-SIP interworking procedures, SIP-ISUP interworking procedures for number portability information and mandating support of AMR-NB on the II-NNI between mobile operator's networks.
English Edition, Version 3.0	May 26, 2016	Mandating the Communication Diversion (CDIV) service, removing the option item regarding the SIP extension "join" and the addition of the SIP routing procedures and the setting condition of Request-URI over the II-NNI when ENUM server(s) are out of order.
English Edition, Version 4.0	May 25, 2017	Limiting the IP version used for control plane and user plane, clarification of the unavailable user identity, mandating of authorization of early media, reflecting 3GPP release-13 specifications into JJ-90.30, mandating of setting of SDP message body to an initial INVITE, mandating not using of PSI, mandating cup parameter, specifying the provision of communication progress information, and adding transcoding call flow example between fixed and mobile IMS network.
English Edition, Version 4.1	June 5, 2017	Editorial correction of Annex Table a.4.2.8
English Edition, Version 5.0	August 29,2018	Added conditions of lower layer of SIP/SDP IP packet prioritization between IMS operator networks, support condition of SIP method, selection of the terminating IBCF, setting conditions of international calling party number, signalling conditions of 00XY/0AB0/1XY and inter-operator accounting, SDP setting conditions, mandatory speech coded between fixed and mobile IMS networks, change codecs and media direction attribute of an established media, method for providing early media, monitoring for extinction of RTP/RTCP packet, and fault detection and restoration detection for SIP layer. Clarified scope of this Standard, SIP transport, setting condition of hostport part in Request-URI, DTMF transport, applicability condition for 100rel/precondition, and bandwidth control. Added new Appendix of procedure for converting multiple early dialogs to a single dialog and procedure for termination of the precondition mechanism and updated related references. Modified structure of whole this Standard.

English Edition, Version 6.0	May 23, 2019	Mandating the usage of UDP, Clarification of the description of fragment UDP, Clarification of terminology, Clarification of the valid number of digits, Clarifications on the setting of calling party number into an SIP request outside an existing dialog, Communication diversion from an actual number to a logical number, and Application of SIP session timer for all sessions and clarification of session timer duration. Deletion of details for operator choice for MIME type, Clarification of option item related to setting of the originating area information, Clarification of option item related to setting of the contractor number, Added the method of controlling maximum number of simultaneous SIP sessions, and Clarification of the condition for use of OPTIONS method. Correction on the History-Info header field example, and updated related references.
English Edition, Version 7.0	May 21, 2020	Clarifications on the setting condition of the Allow header field and the signalling for international calls, improvement of the descriptions for 0AB0 service calls, addition of the exceptions of the P-Asserted-Identity header field transparency, modification of the descriptions for the P-Access-Network-Info header field, addition of a new inter-operator identifier, clarifications on the interworking conditions of the History-Info header field, addition of supplementary information on the setting condition of the P-Asserted-Identity header field when no calling party number is available in the IAM message, editorial corrections, and references updates.

English Edition, Version 8.0	November 12, 2020	Specifying the port number for SIP, clarification of the II-NNI signalling requirement for 3xx response, handling of the header field related to Calling party number and Called party number, P-Asserted-Identity header field handling when valid calling party number is not available, clarification of the handling of local-number-digits setting in tel URI, clarification of presentation codition for network specific number calls, handling of verstat parameter for service calls and network specific number calls, clarification of the description for the reliability of provisional response, clarification of setting condition of P-A-N-I header, correction for the syntax of "operator-specific-GI", condition for the SIP Timer setting when early media is applied, modifications on the P-Charging-Vector header field setting into the SIP responses, editorial enhancements for the additional information in the P-Charging-Vector header, editorial enhancements on the header field in the hi-targeted-to-uri, modification of the condition for the SDP setting in Re-INVITE request, clarification of the b=line handling, clarification of the option item for MIME type, clarification of the condition for the SDP setting when early media is applied, removal of the option item for communication progress information, clarification of the signalling requirements for supplementary services, clarification of the II-NNI signalling requirement for HOLD service, clarification of the maximum length for the contents of the SIP message, removal of the option item related to CDIV service, modification of the CW service applicability, modification of the CAT service applicability, clarification of the subadressing, clarification of the signalling requirements for OPTIONS request used for restoration detection, readability enhancement for option item tables, specifying the port number for ENUM, Specifying the port number for DNS, modification the codition related to fault detection and restoration detection, and editorial Correction for the value of the SIP_DISPLAYNAME. Reference update to 3GPP release-16 specifications, Editorial Corrections, and reference updates.
English Edition, Version 9.0	May 20, 2021	Correction of the description related to 30x SIP response, setting of Callig party's category for international calls without valid Calling party number, clarification on setting condition of "isub-encoding" tel URI parameter, Modification of 200 OK message coding examples following PRACK message, modification of message coding examples related to History-Info header field, modification of OPTIONS message coding examples, and Reference version updates.
English Edition, Version 10.0	May 19, 2022	Clarification of DSCP value for RTCP, reference update to 3GPP release-16 specifications, Editorial Corrections, and reference updates.

English Edition, Version 11.0	May 18, 2023	Clarification of the maximum length of SIP URI and host port in Request-URI, clarification on signaling condition of 00XY, clarification of setting condition of hi-targeted-to-uri in History-Info header, reference update to 3GPP release-16 specifications, Editorial Corrections, and reference updates.
English Edition, Version 12.0	May 16, 2024	Clarification of setting the verstat parameter, clarification of procedures for providing Unallocated (unassigned) number talkie when the ENUM answer is received, clarification of setting of the IOI value and additional information in P-Charging-Vector, clarification of conditions for restoration detection using OPTIONS method, clarification of procedures when the OPTIONS method is set in the Allow header, and reference updates.

3. References

3.1. Normative references

The following documents are the latest version of normative references at the time of publication. All the references may be revised with technical change(s). This Standard provides clarifications on the signalling requirements in conformance with the following normative references. Therefore, careful analysis of the technical impact on this Standard is required when updating a version of an existing normative reference.

[TS 23.003]   "Numbering, addressing and identification", TTC specification TS-3GA-23.003(Rel16)v16.10.0, The Telecommunication Technology Committee, January 2023.

[TS 24.229]   "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3", TTC specification TS-3GA-24.229(Rel16)v16.10.0, The Telecommunication Technology Committee, October 2021.

[TS 29.165]   "Inter-IMS Network to Network Interface (NNI)", TTC specification TS-3GA-29.165(Rel16)v16.6.0, The Telecommunication Technology Committee, October 2021.

[JJ-90.10]   "Inter-Carrier Interface based on ISUP", TTC standard JJ-90.10 version 7.1, The Telecommunication Technology Committee, September 2006.

[JJ-90.27]   "Technical Specification on SIP Network-to-Network Interface (NNI) for Communication Diversion (CDIV)", TTC standard JJ-90.27 version 9.2, The Telecommunication Technology Committee, May 2024.

[JJ-90.28]   "Network-to-Network Interface (NNI) specification for emergency calls", TTC standard JJ-90.28 version 4.3, The Telecommunication Technology Committee, May 2024.

[JJ-90.31]   "Common interconnection interface for carrier ENUM", TTC standard JJ-90.31 version 5, The Telecommunication Technology Committee, November 2020.

[JJ-90.32]   "Common interconnection interface for SIP domain name resolution based on DNS", TTC standard JJ-90.32 version 4, The Telecommunication Technology Committee, May 2021.

[JT-Q763]   "ISUP formats and codes", TTC standard JT-Q763 version 21.1, The Telecommunication Technology Committee, September 2006.

[JT-Y1221]   "Traffic control and congestion control in IP based networks", TTC standard JT-Y1221 version 2, The Telecommunication Technology Committee, February 2013.

[TS-1008]   "Technical Specification on ISDN Calling and Called Party Subaddress Information Transferring", TTC Technical Specification TS-1008 version 2, The Telecommunication Technology Committee, October 2014.

[TS-1018]   "Technical Specification on SIP interface for CUG/PNP", TTC Technical Specification TS-1018 version 7.3, The Telecommunication Technology Committee, May 2024.

3.2. Normative references (TTC standards based on IETF RFCs)

The following documents are TTC standards based on the standards developed by IETF, referenced in this Standard.

[RFC 2474]   "Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers", TTC standard JF-IETF-RFC2474, The Telecommunication Technology Committee, May 2009.

[RFC 2597]   "Assured Forwarding PHB Group", TTC standard JF-IETF-RFC2597, The Telecommunication Technology Committee, May 2010.

[RFC 3261]   "Session Initiation Protocol", TTC standard JF-IETF-RFC3261, The Telecommunication Technology Committee, June 2005.

[RFC 3246]   "An Expedited Forwarding PHB (Per-Hop Behaviour)", TTC standard JF-IETF-RFC3246, The Telecommunication Technology Committee, May 2010.

[RFC 3323]   "A Privacy Mechanism for the Session Initiation Protocol (SIP)", TTC standard JF-IETF-RFC3323, The Telecommunication Technology Committee, June 2005.

[RFC 3550]   "RTP: A Transport Protocol for Real-Time Applications", TTC standard JF-IETF-RFC3550, The Telecommunication Technology Committee, August 2018.

[RFC 3551]   "RTP Profile for Audio and Video Conferences with Minimal Control", TTC standard JF-IETF-STD65, The Telecommunication Technology Committee, June 2005.

[RFC 3966]   "The tel URI for Telephone Numbers", TTC standard JF-IETF-RFC3966, The Telecommunication Technology Committee, June 2005.

[RFC 4458]   "Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)", TTC standard JF-IETF-RFC7131, The Telecommunication Technology Committee, August 2006.

[RFC 4566]   "SDP: Session Description Protocol", TTC standard JF-IETF-RFC4566, The Telecommunication Technology Committee, March 2007.

[RFC 4694]   "Number Portability Parameters for the "tel" URI", TTC standard JF-IETF-RFC4694, The Telecommunication Technology Committee, November 2007.

[RFC 4715]   "The Integrated Services Digital Network (ISDN) Subaddress Encoding Type for tel URI", TTC standard JF-IETF-RFC4715, The Telecommunication Technology Committee, March 2007.

[RFC 4733]   "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals", TTC standard JF-IETF-RFC4733, The Telecommunication Technology Committee, May 2009.

[RFC 5009]   "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media" TTC standard JF-IETF-RFC5009, The Telecommunication Technology Committee, May 2017.

[RFC 7044]   "An Extension to the Session Initiation Protocol (SIP) for Request History Information", TTC standard JF-IETF-RFC7044, The Telecommunication Technology Committee, November 2014.

[RFC 7131]   "Session Initiation Protocol (SIP) History-Info Header Call Flow Examples", TTC standard JF-IETF-RFC7131, The Telecommunication Technology Committee, November 2014.

[RFC 7315]   "Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3GPP", TTC standard JF-IETF-RFC7315, The Telecommunication Technology Committee, August 2018.

[RFC 7913]   "P-Access-Network-Info ABNF Update", TTC standard JF-IETF-RFC7913, The Telecommunication Technology Committee, August 2018.

[RFC 8119]   "Session Initiation Protocol (SIP) Cause URI parameter for Service Number translation", TTC standard JF-IETF-RFC8119, The Telecommunication Technology Committee, August 2017.

[RFC 8496]   "P-Charge-Info: A Private Header Field (P-Header) Extension to the Session Initiation Protocol (SIP)", TTC standard JF-IETF-RFC8119, The Telecommunication Technology Committee, May 2019.

3.3. Non-normative references

The following documents are non-normative references, referenced in this Standard.

[TS 24.628]   "Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification", TTC specification TS-3GA-29.628(Rel16)v16.2.0, The Telecommunication Technology Committee, October 2020.

[TS 26.114]   "IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction", TTC specification TS-3GA-26.114(Rel16)v16.11.0, The Telecommunication Technology Committee, January 2022.

[TS 29.162]   "Interworking between the IM CN subsystem and IP networks", TTC specification TS-3GA-29.162(Rel16)v16.0.0, The Telecommunication Technology Committee, October 2020.

[TS 29.163]   "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks", TTC specification TS-3GA-29.163(Rel16)v16.4.0, The Telecommunication Technology Committee, October 2020.

[IR.34]   "Guidelines for IPX Provider networks (Previously Inter-Service Provider IP Backbone Guidelines)", GSMA IR.34 Version 13.0, October 2016.

[IR.92]   "IMS Profile for Voice and SMS", GSMA IR.92 Version 9.0, April 2015.

[JT-Q3402]   "NGN UNI Signalling Profile (Protocol Set 1)", TTC standard JT-Q3402 version 3, The Telecommunication Technology Committee, May 2015.

[TR-1014]   "General overview of NGN architecture", TTC Technical Report TR-1014 version 1, The Telecommunication Technology Committee, June 2006.

4.   Industrial Property

The status of the "IPR Licensing Statement" submission is provided on the TTC web site.

5.   Responsible Working Group

Signalling Working Group

1. Overview

1.1. Scope

[TS 29.165] specifies the Inter-IMS Network to Network Interface (II-NNI) between IMS networks. Based on [TS 29.165], this Standard specifies the Network to Network Interface between the IMS networks commonly used within Japan.

Figure 1.1-1 illustrates the II-NNI-related architecture diagram given in Subclause 5.1 of [TS 29.165].
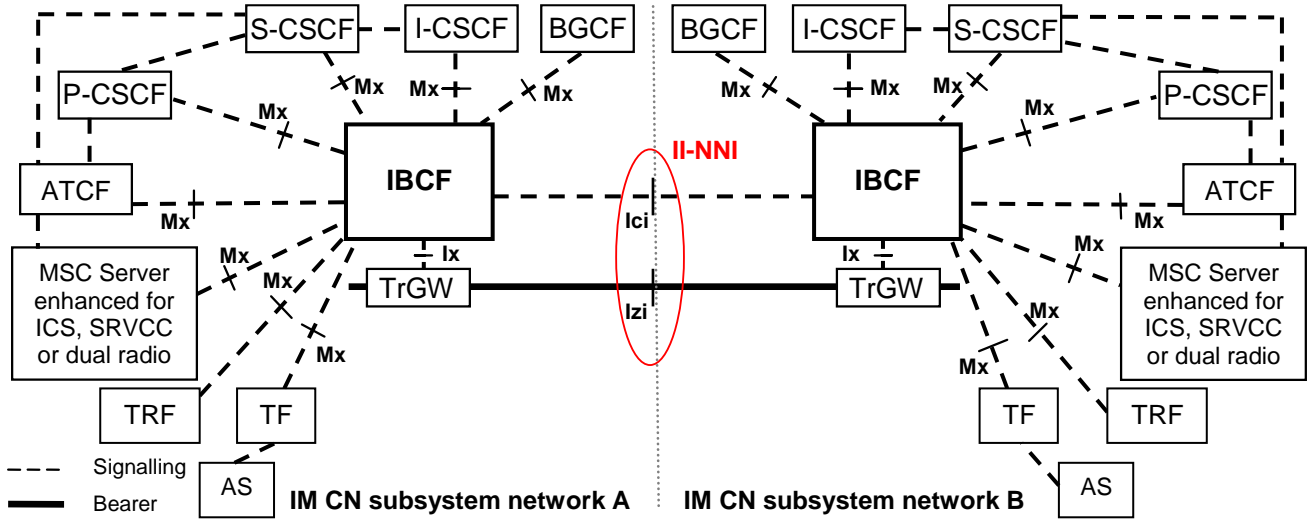
Figure 1.1-1 / JJ-90.30: II-NNI between IMS networks

This Standard specifies the non-roaming II-NNI between the IMS home network on originating side (IMS home network A) and terminating side (IMS home network B). Figure 1.1-2 illustrates the II-NNI traversal scenario given in Clause 4 of [TS 29.165].

NOTE 1: The II-NNI specifications for the roaming and the emergency calls are outside the scope of this Standard, while these specificasions are inside the scope of [TS 29.165]. [JJ-90.28] specifies the II-NNI specifications for emergency calls in Japan.

NOTE 2: This Standard does distinguish mobile IMS networks from fixed IMS networks unlike [TS 29.165], if necessary.

NOTE 3: In this Standard, "II-NNI" represents the non-roaming II-NNI.
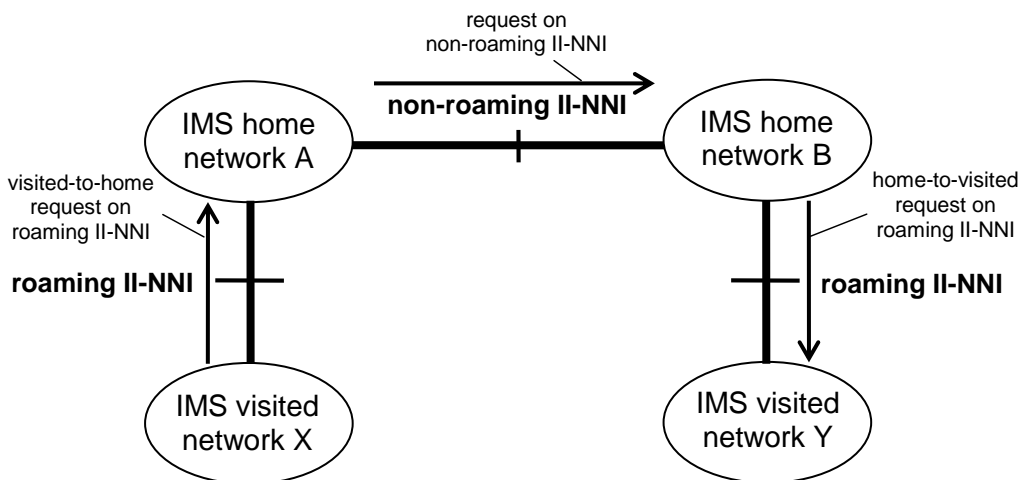
Figure 1.1-2 / JJ-90.30: II-NNI traversal scenario

1.2. Objective

This Standard is aimed to improve the interconnectivity between the IMS networks within Japan by providing a common interpretation of the interconnection conditions using the II-NNI.

The references to the option item tables relating to the selectable items as the interconnection conditions are enclosed in "{" and "}".

1.3. Contents

This Standard specifies national supplementary specifications against [TS 29.165] within the scope described in Subclause 1.1 of this Standard.

1.3.1. Mandatory signalling requirements

The mandatory signalling requirements on the II-NNI are described as follows. These requirements shall be followed by an operator who implements an interconnection interface compliant to this Standard.

   1) Supported II-NNI traversal scenarios (Clause 3)

   2) II-NNI specifications

      a) Layer 3 (Subclause 4.1)

      b) Layer 4 (Subclause 4.2)

      c) SIP/SDP (Subclause 4.3)

   3) Clarifications of [TS 29.165] by means of option item selections (Annex a)

   NOTE: Basically, the mandatory signalling requirements are described in the main body of this Standard; however, to improve the readability, the mandatory signalling requirements could be described in a normative annex.

1.3.2. Optional signalling requirements

The optional signalling requirements on the II-NNI are described as follows. The signalling requirements are applied based on inter-operator agreements.

   1) SIP/SDP signalling requirements on the II-NNI

      a) Subaddress (Annex b)

      b) Bandwidth control (Annex c)

      c) Restoration detection using OPTIONS method (Annex d)

1.3.3. Informative items

The following information is described as a reference to achieve smooth interconnections. Note that this information is not a normative specification.

   1) List of option items (Appendix i)

   2) SIP-ISUP interworking (Appendix ii)

   3) Considerations on network operation and maintenance (Appendix iii)

   4) Void

   5) Procedures for termination of the precondition mechanism (Appendix v)

   6) Procedures for converting multiple early dialogs to a single dialog (Appendix vi)

   7) Examples of callflow and message coding (Appendix vii)

8) SIP message generation considering the maximum length of SIP message (Appendix viii)

2. Terminologies and abbreviations

2.1. Terminologies

The terms and definitions in this Standard comply with [TS 29.165], [JJ-90.31] and [JJ-90.32].

2.2. Abbreviations

The abbreviations in this Standard comply with [TS 29.165]. Abbreviations used in this Standard are shown below.

18x	A SIP status-code in the range 180 through 189
3PCC	3rd Party Call Control
3PTY	Three-Party
3xx	A SIP status-code in the range 300 through 399
4xx	A SIP status-code in the range 400 through 499
5xx	A SIP status-code in the range 500 through 599
6xx	A SIP status-code in the range 600 through 699
ACR	Anonymous Communication Rejection
AMR(-NB)	Adaptive Multi-Rate (- NarrowBand)
AMR-WB	Adaptive Multi-Rate - WideBand
AOC	Advice of Charge
CA	Charge Area
CAT	Customized Alerting Tone
CCBS	Completion of Communications to Busy Subscriber
CCNL	Completion of Communications on Not Logged-in
CCNR	Communication Completion on No Reply
CDIV	Communication Diversion
CONF	Conference
CRS	Customized Ringing Signal
CUG	Closed User Group
CW	Communication Waiting
DNS	Domain Name System
DSCP	Differentiated Services Code Point
DTMF	Dual Tone Multi Frequency
ECT	Explicit Communication Transfer
ENUM	E.164 Number Mapping
FA	Flexible Alerting
GRUU	Globally Routable User agent URIs
HOLD	Communication HOLD
ICB	Incoming Communication Barring
ICID	IM CN subsystem Charging Identifier
II-NNI	Inter-IMS Network to Network Interface
IMS	IP Multimedia Subsystem
IOI	Inter Operator Identifier
IP	Internet Protocol
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISDN	Integrated Services Digital Network
ISUP	Integrated Services Digital Network (ISDN) User Part
MBMS	Multimedia Broadcast Multicast Service
MCID	Malicious Communication Identification
MCPTT	Mission Critical Push-To-Talk
MGCF	Media Gateway Control Function
MGW	Media Gateway
MIME	Multipurpose Internet Mail Extension
MSRP	Message Session Relay Protocol
MTSI	Multimedia Telephony Service for IMS

MTU	Maximum Transmission Unit
MWI	Message Waiting Indication
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
PHB	Per Hop Behavior
PNM	Personal Network Management
POI	Point of Interface
PSAP	Public Safety Answering Point
PSI	Public Service Identity
PSTN	Public Switched Telephone Network
RACF	Resource and Admission Control Functions
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SCTP	Stream Control Transmission Protocol
SIP	Session Initiation Protocol
SDP	Session Description Protocol
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TCP	Transmission Control Protocol
TRF	Transit and Roaming Function
UDP	User Datagram Protocol
UNI	User to Network Interface
URI	Uniform Resource Identifiers

3.   Supported II-NNI traversal scenarios

The operator who implements an interconnection interface compliant to this Standard shall support the non-roaming II-NNI between the two home IMS networks.

4.   II-NNI specifications

4.1. Layer 3

This Subclause specifies the conditions for the layer 3 interface.

4.1.1. IP version

The operator who implements an interconnection interface compliant to this Standard shall support IPv4 over the II-NNI. The applicability of IPv6 shall be determined in the inter-operator agreements. {Appendix Table i.4-1 Item 1}

> NOTE: The IPv4 fallback procedure when the applicable IP version is not match between originating and terminating UE during the session establishment in the circumstance where both IPv4 and IPv6 are used over the II-NNI is described in Subclause 4.3.5.1.3.2 of this Standard.

The IP version applied for SIP message and media shall be same in the session.

4.1.2. Differentiated Services Code Point (DSCP)

The DSCP value shall be set as described in Table 4.1.2-1 for IP packets of SIP message and RTP packets of speech media (m=audio) respectively as with [IR.34].

Annex Table 4.1.2-1 / JJ-90.30: Setting value of DSCP

Protocol	IP Transport	
	PHB	DSCP
RTP (m=audio) (NOTE 4)	EF (NOTE 1)	101110 (NOTE 1)
SIP	AF31 (NOTE 2)	011010 (NOTE 2)
NOTE 1: The element conforms to [RFC 3246].		
NOTE 2: The element conforms to [RFC 2597].		
NOTE 4: The element includes RTCP.		

NOTE 3: It should be noted that the DSCP values defined in this Subclause need to be set despite the priority of call (e.g., call from priority users, emergency call).

4.2. Layer 4

The operator who implements an interconnection interface compliant to this Standard shall support UDP over the II-NNI as a transport protocol for SIP. The port number to accept for SIP signalling using UDP shall be "5060".

In order to guarantee exchanging a SIP message of which size is larger than 1,300 bytes, IP fragmented UDP packet shall be supported between the IMS networks.

NOTE: As mentioned in [RFC 3261], it is assumed that the Ethernet MTU of the interface between the IMS networks is 1,500 bytes in this Standard.

4.3. SIP/SDP

4.3.1. SIP requests/responses

In this Standard, ACK/BYE/CANCEL/INVITE/PRACK/UPDATE/OPTIONS methods shall be supported over the II-NNI, in accordance with Subclause 6.1.1.2 of [TS 29.165].

The applicability of MESSAGE/REFER/NOTIFY/SUBSCRIBE methods over the II-NNI and whether sending an OPTIONS request outside an existing dialog from the its own network over the II-NNI are determined by the inter-operator agreements. {Appendix Table i.4-3 Item 1, 2, 3, 4}{Appendix Table i.4-20 Item 2} The applicable SIP methods shall not be changed during the established dialog.

In this Standard, OPTIONS method is applicable only for fault detection and restoration detection of the opposing IBCF. The signalling requirements for the OPTIONS request and the response to the request are in accordance with Annex d of this Standard. Domestic IMS operator's network shall be able to send a 200 (OK) response to the OPTIONS request regardless whether the IMS operator applies restoration detection using OPTIONS method or not, in order to notify the restoration of the IBCF which has been detected a fault.

The IMS network which implements an interconnection interface compliant to this Standard shall set an Allow header field into an initial INVITE request and 18x/200 (OK) response corresponding to the initial INVITE request. Only the mandatory SIP methods at the II-NNI in this Standard and the applicable SIP methods within a INVITE dialog which determined in the inter-operator agreements (refer to Appendix Table i.4-3 Item 1, 2) shall be included in the Allow header field.

NOTE: If the OPTIONS method is set in the Allow header, the IMS network ignores it.

The SIP request and the response inside the existing dialog may include an Allow header field, then the Allow header field shall include the SIP methods identical to the methods set in the Allow header field in the initial INVITE request or 200 (OK) response corresponding to the initial INVITE request. An example of an Allow header filed including only the mandatory SIP methods at the II-NNI is shown below.

Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE

4.3.1.1. Conditions for sending back 503 (Service Unavailable) response

For fault detection and detouring, an IBCF in a terminating IMS network may send a 503 (Service Unavailable) response to the initial INVITE request received from an originating IMS network, only when the IBCF is temporarily unable to process the SIP request due to temporary overloading or fault. When the IBCF in the terminating IMS network receives a 503 (Service Unavailable) response from a functional entity in its own network, the IBCF in the terminating IMS network shall change the SIP response code from 503 (Service Unavailable) to 500 (Server Internal Error) prior to forwarding the response to the originating IMS network.

4.3.1.2. Conditions for sending back 3xx response

The IMS network which implements an interconnection interface compliant to this Standard shall not send a 3xx response to the initial INVITE request received from the originating IMS network.

4.3.2. Request-URI

Called party number is set to the Request-URI of the SIP request outside an existing dialog. The Request-URI shall be set according to Subclauses in this Subclause, whether or not a called party number is a target number to be queried to carrier ENUM. The PSI specified in [TS 23.003] shall not be populated in the Request-URI.

When a donor carrier of the called terminating number block provides carrier ENUM interface specified in [JJ-90.31], an originating IMS network shall obtain a SIP domain name corresponding to a called party number by using the carrier ENUM interface. The details of carrier ENUM interface are determined in the inter-operator agreements. {Appendix Table i.4-4 Item 1}

When a called party number is a target number to be queried to ENUM, an originating IMS network sends an ENUM query to obtain a NAPTR resource record(s) corresponding to the target number via the carrier ENUM interface for routing SIP request outside an existing dialog.

1) If the originating IMS network successfully obtained a NAPTR resource record(s), the Request-URI of the SIP request outside an existing dialog shall be updated with the received NAPTR resource record(s). If the originating IMS network obtains multiple NAPTR resource records, the NAPTR resource record used for updating the Request-URI is selected based on the operator policy.

2) If the originating IMS network failed to obtain a NAPTR resource record(s), the originating IMS network may forward the SIP request to the IMS network of the donor carrier of the called terminating number block, only if the inter-operator agreement exists. {Appendix Table i.4-4 Item 2}

When a called party number is not a target number to be queried to carrier ENUM, an originating IMS network uses the SIP domain name of a donor carrier corresponding to the called party number which is provisioned in the originating IMS network, in order to route the SIP request outside an existing dialog.

4.3.2.1. URI scheme

The Request-URI of a SIP request outside an existing dialog shall be SIP URI.

4.3.2.2. telephone-subscriber part

The formats of the called party number are listed in Table 4.3.2.2-1.

Table 4.3.2.2-1 / JJ-90.30: Format of called party number

Format	Application
+[Country code] [National number]	International network calls
+81ABCDEFGHJ	Regional fixed-line phone calls, IP phone calls (Category A)
+81CD177	
+81A0CDEFGHJK	Mobile/PHS/wireless pager calls, IP phone calls (Category B)
+81AB0-	Service calls
1XY	Network specific number calls (e.g., 104, 115 171, 188, 189)
00XY-	Network specific number calls (connection that uses the carrier identification codes)

A global number format is used for the Request-URI of a SIP request outside an existing dialog, except that network specific number calls such as 1XY is set to the called party number. For network specific number calls such as 1XY, the called party number shall be set into the Request-URI of a SIP request outside an existing dialog as a local number format with "phone-context" parameter set to +81. Network specific numbers not listed in Table 4.3.2.2-1 may be used only when the operators agree to use, as described in Subclause 8.1 of [TS 29.165].
{Appendix Table i.4-5 Item 1}

When an originating IMS network requests the routing based on the called party number in a SIP request outside an existing dialog, visual-separators shall not be included in the description of the global number and local number. Also, for the Request-URI, the minimum number of DIGIT (number from 0 to 9) in the global-number-digits and the HEXDIG (number from 0 to 9, alphabet from A to F, "*", and "#") in the local-number-digits, in the telephone-subscriber part of

SIP URI is 3 and the maximum is 26. A SIP request outside an existing dialog containing the phonedigits that are less than the minimum phonedigits or more than the maximum phonedigits in the global-number-digits or local-number-digits, shall not be forwarded over the II-NNI.

4.3.2.2.1. "npdi" tel URI parameter

IMS networks shall support number portability parameters (i.e., "rn" and "npdi" tel URI parameters) defined in [RFC 4694]. When an originating IMS network successfully obtains NAPTR resource record(s) via carrier ENUM interface specified in [JJ-90.31] or retains the DNS cache, then the originating IMS network shall set "npdi" tel URI parameter to the SIP request outside an existing dialog, in order to indicate to a terminating IMS network that the originating IMS network has already successfully obtained the NAPTR resource record(s) via carrier ENUM interface. When receiving a SIP request outside an existing dialog containing "npdi" tel URI parameter, a terminating IMS network shall consider that ENUM access has been done in the preceding IMS network, and then should not access carrier ENUM.

> NOTE 1: In [JJ-90.31], 60 seconds is recommended for TTL value of NAPTR resource record.

If the originating IMS network failed to obtain NAPTR resource record(s) via the carrier ENUM interface and if the originating IMS network connects to the IMS network of the donor carrier of the called terminating number block based on the inter-operator agreement, the originating IMS network shall not set "npdi" tel URI parameter in the forwarded SIP request outside an existing dialog.

> NOTE 2: When the terminating IMS network receives a SIP request outside an existing dialog not including "npdi" tel URI parameter and the called party number is a target number to be queried to ENUM, the terminating IMS network needs to continue routing process if the inter-operator agreement exists.

When the called party number is not a target number to be queried to ENUM, the originating IMS network shall not set "npdi" tel URI parameter in the forwarded SIP request outside an existing dialog.

> NOTE 3: When the terminating IMS network receives a SIP request outside an existing dialog not including "npdi" tel URI parameter and the called party number is not a target number to be queried to ENUM, the terminating IMS network needs to continue routing process.

4.3.2.2.2. "rn" tel URI parameter

When a NAPTR resource record being used for updating the Request-URI includes "rn" tel URI parameter set to a network routing number, the retrieved "rn" tel URI parameter containing the network routing number shall be set into the Request-URI of a SIP request outside an existing dialog. The maximum number of DIGIT (number from 0 to 9) in the hex-phonedigits for the global-hex-digits of "rn" tel URI parameter is 26.

> NOTE: In the period that there are both interconnection on ISUP interface defined in [JJ-90.10] and interconnection on SIP interface defined in this Standard, a NAPTR resource record containing "rn" tel URI parameter can be returned from carrier ENUM interface defined in [JJ-90.31].

4.3.2.2.3. Other tel URI parameters

The terminating IMS network should continue routing process even if telephone-subscriber part contains the other unrecognized tel URI parameter(s).

4.3.2.3. hostport part

When an originating IMS network obtains NAPTR resource record(s) via carrier ENUM interface, a SIP domain name in the SIP URI included in the NAPTR resource record(s) shall be set to the hostport part of the Request-URI in accordance with Subclause 4.3.2 of this Standard. When a called party number is not the target number to be queried to ENUM or an originating IMS network fails to obtain NAPTR resource record(s) and forwards the SIP request to an IMS network of a donor carrier, then a SIP domain name of the IMS network of the donor carrier shall be set to the hostport part of the Request-URI.

Note that the SIP domain name(s) and/or IP address(es) of hostport part to accept shall be determined in the inter-operator agreements. {Appendix Table i.4-5 Item 1}

4.3.2.4. SIP URI parameters

4.3.2.4.1. "user" SIP URI parameter

When a global E.164 number or a local number such as 1XY is set in the telephone-subscriber part, a "user" SIP URI

parameter with the value of "phone" shall be set in the SIP URI in accordance with Subclause 8.1 of [TS 29.165].

4.3.2.4.2. "cause" SIP URI parameter

When translating a logical number (e.g., the number for toll free services beginning with "0120" or "0800", the number for unified service beginning with "0570") into an actual number, an IMS network shall populate "cause" SIP URI parameter with the value of 380. A terminating IMS network should continue the call processing even if the Request-URI does not contain the "cause" SIP URI parameter.

The IMS network shall not send the "cause" SIP URI parameter to user equipment via UNI, and shall remove the "cause" SIP URI parameter received from international networks (overseas).

4.3.2.4.3. Other SIP URI parameters

When receiving unrecognized SIP URI parameter(s) in the Request-URI of a SIP request outside an existing dialog, a terminating IMS network ignores the SIP URI parameter(s).

4.3.2.5. Examples of the Request-URI

Table 4.3.2.5-1 shows examples of the Request-URI of a SIP request outside an existing dialog in accordance with the formats of called party number listed in Table 4.3.2.2-1.

Table 4.3.2.5-1 / JJ-90.30: Examples of the Request-URI

Format	Request-URI example
+[Country code] [National number]	sip:+12121111111@example.ne.jp;user=phone
+81ABCDEFGHJ	sip:+81311111111;npdi@example.ne.jp;user=phone
+81CD177	sip:+81422177;npdi@example.ne.jp;user=phone
+81A0CDEFGHJK	sip:+819011111111;npdi@example.ne.jp;user=phone
	sip:+815011111111@example.ne.jp;user=phone (NOTE 1)
+81AB0-	sip:+8112011111111;npdi@example.ne.jp;user=phone
	sip:+8157011111111@example.ne.jp;user=phone (NOTE 1)
1XY	sip:104;phone-context=+81@example.ne.jp;user=phone (NOTE 2)
00XY-	sip:003603222222222;phone-context=+81@example.ne.jp;user=phone (NOTE 2)
NOTE 1: An example that the SIP URI not containing "npdi" tel URI parameter is shown here since the called party number is not subject to number portability.	
NOTE 2: The "npdi" tel URI parameter is not set since the called party number is not a target number to be queried to ENUM.	

4.3.3. Selection of the terminating IBCF

Based on inter-operator agreement, an originating IMS network shall apply either of the following two methods for selection of a terminating IBCF based on a SIP domain name of the terminating IMS network in the Request-URI of a SIP request outside an existing dialog. {Appendix Table i.4-6 Item 1}

1) The method using the inter-operator DNS interface

 - A method that an originating IMS network uses the inter-operator DNS interface for SIP domain name resolution specified in [JJ-90.32] and selects a terminating IBCF based on the received DNS answer.

2) The method not using the inter-operator DNS interface

 - A method that an originating IMS network selects a terminating IBCF based on the IP address (list) notified by the terminating IMS network operator beforehand.

If a terminating IMS network provides the inter-operator DNS interface for SIP domain name resolution, then an originating IMS network shall apply 1) as a method for selection of a terminating IBCF. If a terminating IMS network does not provide the inter-operator DNS interface for SIP domain name resolution and applies 2) for selection of a terminating IBCF, then details of a method for selection (e.g., round robin) is determined in the inter-operator agreement.

Note that whether continue processing a call or reject a call when an originating IMS network can not obtain any normal DNS answer from DNS servers in a terminating IMS network is selected by originating IMS network's operator, then the condition is determined in the inter-operator agreements.

4.3.4. SIP headers and parameters

4.3.4.1. Calling party number presentation and restriction

Calling party number presentation and restriction is realized by means of delivering network asserted user identity information and presentation/restriction information.

Table 4.3.4.1-1 shows the terminologies used in this Subclause.

Table 4.3.4.1-1 / JJ-90.30: Terminologies related to calling party number

Terminology	Definition
Network asserted user identity information	This is information with respect to an identity of a user that is asserted by the network through authentication or other means (or verified by the network when an identity is provided by the user) in a trusted network. An example of network asserted user identity information is an E.164 number reachable to the user. Note that subaddress information provided by the originating user equipment can be included.
anonymous URI	This is the URI used to make URI information anonymous. The anonymous URI (sip:anonymous@anonymous.invalid) is defined in [RFC 3323] and [TS 23.003].
unavailable URI	This is the URI used to represent the URI information is not available. The unavailable URI (sip:unavailable@unknown.invalid) is defined in [TS 23.003].
Presentation/restriction information	This is information specifying whether a user is allowing or prohibiting the presentation of its network asserted user identity information to another user receiving a SIP message.

4.3.4.1.1. Trust relationship

A trust relationship with respect to the P-Asserted-Identity header field shall exist between the two interconnected networks which implements an interconnection interface compliant to this Standard, in order to deliver a network asserted user identity information in accordance with Subclause 6.1.1.3.1 of [TS 29.165].

4.3.4.1.2. Conditions for setting of calling party number

Calling party number is delivered between the IMS networks in accordance with following conditions.

1) The calling party number is delivered in a SIP request outside an existing dialog (INVITE, MESSAGE, SUBSCRIBE or REFER request).

2) The originating IMS network populates each parameter value of the P-Asserted-Identity header field in a SIP request outside an existing dialog using information elements defined in Table 4.3.4.1.2-1.

   a) The originating IMS network shall populate the P-Asserted-Identity header field containing "TEL_URI" into a SIP request outside an existing dialog unless no valid calling party number was available from an international network. The originating IMS network can populate a single P-Asserted-Identity header field containing "TEL_URI" and shall not populate two or more P-Asserted-Identity header field containing "TEL_URI". If "TEL_URI" is populated, then the originating IMS network may populate "TEL_DISPLAYNAME".

   b) The originating IMS network may populate the P-Asserted-Identity header field containing "SIP_URI" into a SIP request outside an existing dialog. If "SIP_URI" is populated, then the originating IMS network may populate "SIP_DISPLAYNAME". When populating either the anonymous URI or the unavailable URI into "SIP_URI", then the originating IMS network shall populate "id" for "PRIVACY". When the valid calling party number is not available (e.g., an incoming call originated by an international network), the unavailable URI shall be set into the "SIP_URI".

   c) Setting conditions shall conform to Table 4.3.4.1.2-2.

3) When populating "id" for "PRIVACY", the originating IMS network shall populate a From header field containing either the anonymous URI or the unavailable URI.

4.3.4.1.2A. Conditions for presentation of calling party number

Calling party number delivered between IMS networks is handled at the terminating network according to the following conditions:

1) The terminating IMS network uses the received parameter in the P-Asserted-Identity header field for presentation of calling party number. The terminating IMS network shall not use the contents received in the From header field delivered between IMS networks as the calling party number indicated to the terminating user.

   a) When receiving a SIP request outside an existing dialog not containing the P-Asserted-Identity header field, the terminating IMS network handles the request as if the request includes the P-Asserted-Identity header field containing either the anonymous URI or the unavailable URI. In this case, the terminating IMS network shall handles the request as if "PRIVACY" is set to "id".

   b) When receiving a SIP request outside an existing dialog containing the P-Asserted-Identity header field, the terminating IMS network obtains the calling party number (which is notified to the terminating user/user equipment) according to the following procedures:

      i) When "PRIVACY" is set to "id", the call is treated as calling party number presentation is restricted. Cause of no ID shall be obtained from "SIP_DISPLAYNAME" according to Table 4.3.4.1.2-2 and Table 4.3.4.1.2-5.

      ii) When "PRIVACY" is set to "none" or not included in the request, the call is treated as calling party number presentation is allowed. If "TEL_DISPLAYNAME" is available, the calling party number (which is notified to the terminating user/user equipment) shall be obtained from "TEL_DISPLAYNAME" according to Table 4.3.4.1.2-2 and Table 4.3.4.1.2-4. If "TEL_DISPLAYNAME" is not available, the number notified to the terminating user/user equipment shall be obtained from "TEL_URI" according to Table 4.3.4.1.2-2 and Table 4.3.4.1.2-3.

Table 4.3.4.1.2-1 / JJ-90.30: Elements associated with handling of calling party number

	Information elements	Definition
1	SIP_URI	This indicates network asserted user identity reachable from the IMS network. (NOTE) The addr-spec part of the SIP URI in the P-Asserted-Identity header field of a SIP request outside an existing dialog is symbolized by the "SIP_URI".
2	SIP_DISPLAYNAME	This indicates network asserted user identity associated with the SIP_URI, including information other than the number to be displayed to the terminating user. The display-name part of the SIP_URI in the P-Asserted-Identity header field of a SIP request outside an existing dialog, which is character strings in UTF-8 encoding, is symbolized by the "SIP_DISPLAYNAME". When it is enclosed in quotation marks, the "SIP_DISPLAYNAME" is character strings left after removing the double quotation marks (DQUOTE). Omission of the SIP_DISPLAYNAME indicates that a display format different from the SIP_URI is not particularly desired. In this case, IMS network should take into account that the character strings of the SIP_URI are used for display, unless special restrictions are specified.
3	TEL_URI	This indicates network asserted user identity consisting of an E.164 number reachable from a Public Switched Telephone Network (PSTN). The global-number-digits or local-number-digits in the telephone-subscriber part of the tel URI in the P-Asserted-Identity header field of a SIP request outside an existing dialog is symbolized by the "TEL_URI". Omission of the TEL_URI indicates that the originating user has no E.164 number for receiving incoming calls.
4	TEL_DISPLAYNAME	This indicates network asserted user identity consisting of a dial number by which the originating user can be reached based on a numbering plan. The display-name part of the tel URI in the P-Asserted-Identity header field of a SIP request outside an existing dialog is symbolized by the "TEL_DISPLAYNAME". When it is enclosed in quotation marks, the "TEL_DISPLAYNAME" is the character strings left after removing the double quotation marks (DQUOTE). Omission of the TEL_DISPLAYNAME indicates that a dial number different from the number indicated by the TEL_URI is not particularly desired, or that the network does not have accurate information pertaining to the dialling numbering plan that can be used by the terminating user. In this case, the IMS network should take into account that the character strings of the TEL_URI are equivalent to the TEL_DISPLAYNAME, unless special restrictions are specified.
5	PRIVACY	This indicates presentation/restriction information identifying whether the presentation of network asserted user identity to the terminating user is allowed. The value (priv-value) of the Privacy header field of a SIP request outside an existing dialog is symbolized by the "PRIVACY".
NOTE: An unavailable URI or anonymous URI may be set as a SIP_URI.		

Table 4.3.4.1.2-2 / JJ-90.30: Conditions for presentation of calling party number

Component	Mapping condition	Notes
Calling party number (Contractor number)	TEL_URI	Used as a number identifying the originating user. Visual-separators are not used. Specific setting contents are listed in Table 4.3.4.1.2-3.
Generic number (Notified number)	TEL_DISPLAYNAME	Used when a number other than the calling party number is notified to the terminating side. Visual-separators are not used. Specific setting contents are listed in Table 4.3.4.1.2-4.
Presentation/restriction	PRIVACY	"none" and "id" are treated as displayable and not displayable, respectively. Values other than "none" and "id" shall not be set. (NOTE 1) (NOTE 2) Assumed to be displayable when the Privacy header field is absent. When both calling party number (Contractor number) and generic number (notified number) are set, this component is treated as the presentation/restriction for the generic number (notified number), and the calling number (Contractor number) is uniformly handled as the number being restricted to display.
Cause of no ID	SIP_DISPLAYNAME	Character strings in UTF-8 encoding. When the presentation/restriction information is restriction, the character strings in Table 4.3.4.1.2-5 can be populated to indicate the cause. If the "SIP_DISPLAYNAME" is not set, or if character strings not listed in Table 4.3.4.1.2-5 are set, then the call is handled as if "Unavailable" were specified.
NOTE 1: "id" is set to the Privacy header field in case of an unavailable URI or an anonymous URI. NOTE 2: Any interworking procedures between the IMS network which implements the SIP interface compliant to this Standard and the network not compliant to this Standard is outside the scope of this Standard.			

Table 4.3.4.1.2-3 / JJ-90.30: TEL_URI format

TEL_URI	Number digit	Usage
+[Country-code] [National number]	Max 15 digits	Originating call on international network (overseas)
+81ABCDEFGHJ	10 or 11 digits	Originating call on local fixed telephone network Originating call on IP phone (category A)
+81A0CDEFGHJK	12 digits	Originating call on mobile/PHS network Originating call on IP phone (category B)
Arbitrary digits	Max 16 digits	Operator-originating call, etc. (NOTE)
NOTE: Only when IAM→ Initial INVITE request interworking procedure as specified in Appendix ii.2.4.3.3 of this Standard is invoked, "TEL_URI" containing network specific numbers can be sent over the II-NNI.			

Table 4.3.4.1.2-4 / JJ-90.30: TEL_DISPLAYNAME format

TEL_DISPLAYNAME	Number digit	Usage
0ABCDEFGHJ	9 or 10 digits	Originating call on local fixed telephone network Originating call on IP phone (category A)
0A0CDEFGHJK	11 digits	Originating call on mobile/PHS network Originating call on IP phone (category B)
0AB0-	Max 16 digits	Logical number
Arbitrary digits	Max 16 digits	Operator-originating call, etc.
NOTE: For international calls, the TEL_DISPLAYNAME is not populated as an originating IMS network will remove the TEL_DISPLAYNAME if received from international networks (overseas).			

Table 4.3.4.1.2-5 / JJ-90.30: Character strings indicating cause of no ID in SIP_DISPLAYNAME

SIP_DISPLAYNAME	Meaning
Unavailable	No caller ID: service unavailable
Anonymous	No caller ID: rejected by user
Interaction with other service	No caller ID: service conflict
Coin line/payphone	No caller ID: call from public telephone

4.3.4.1.3. Calling party's category ("cpc" tel URI parameter)

The calling party's category represents a subscriber's category that an originating user retains or a network-asserted attribute to a call, such as the "calling subscriber with priority" indication or "test call" indication. In order to deliver the calling party's category information between IMS networks, a trust relationship with respect to the "cpc" tel URI parameter shall exist between the two interconnected networks in accordance with Subclause 6.1.1.3.1 of [TS 29.165].

4.3.4.1.3.1. Trust relationship

Trust relationship with respect to the "cpc" tel URI parameter shall exist between the two interconnected networks which implements an interconnection interface compliant to this Standard, in accordance with Subclause 6.1.1.3.1 of [TS 29.165] in order to deliver a calling party's category between the IMS networks. The IMS network shall not send the "cpc" tel URI parameter to user equipment via UNI.

4.3.4.1.3.2. Setting conditions of "cpc" tel URI parameter

The calling party's category information is exchanged between the IMS networks by populating the "cpc" tel URI parameter into the URI of the P-Asserted-Identity header field. If the URI of the P-Asserted-Identity header field is tel URI scheme, then a "cpc" tel URI parameter can be populated into the par part. Likewise, if the URI of the P-Asserted-Identity header field is SIP URI scheme, then a "cpc" parameter can be populated after (;) appended after global-number-digits as a tel URI parameter. The calling party's category is populated in the userinfo part (appended after ";" and right before "@") of the SIP URI in the P-Asserted-Identity header field using the syntax of "cpc" described in Table 4.3.4.1.3.2-1, if the SIP URI in the P-Asserted-Identity header field is not including global-number-digits. If more than one URI is populated into the P-Asserted-Identity header field, then the same "cpc" tel URI parameter shall be populated into all the URIs. If the calling party's category information is not specified, the call is handled as if "ordinary" is specified.

The syntax of a "cpc" tel URI parameter that conforms to [RFC 3966] is described in Table 4.3.4.1.3.2-1. The originating IMS network shall not set the cpc-value not listed in Table 4.3.4.1.3.2-1 for the "cpc" tel URI parameter.

NOTE: Any interworking procedures between the IMS network which implements the SIP interface compliant to this Standard and the network not compliant to this Standard is outside the scope of this Standard.

Table 4.3.4.1.3.2-1 / JJ-90.30: The syntax of "cpc" tel URI parameter

```
par =/ cpc
cpc = cpc-tag "=" cpc-value
cpc-tag = "cpc"
cpc-value = "ordinary" / "priority" / "test" / "payphone"
```

4.3.4.1.3.3. Examples of "cpc" tel URI parameter

Examples of a calling party's category are shown below:

1) Example when "calling subscriber with priority" is populated into the P-Asserted-Identity header field including a SIP URI:

P-Asserted-Identity: <sip:+81312345678;cpc=priority@example.ne.jp;user=phone>

2) Example when "calling subscriber with priority" is populated into the P-Asserted-Identity header field including a tel URI.

P-Asserted-Identity: <tel:+81312345678;cpc=priority>

3) Example when "calling subscriber with priority" is populated into the P-Asserted-Identity header field including both SIP URI and tel URI.

   P-Asserted-Identity: <sip:+81312345678;cpc=priority@example.ne.jp;user=phone>,

   <tel:+81312345678;cpc=priority>

4) Example when "calling subscriber with priority" is populated into the P-Asserted-Identity header field including a SIP URI which has a userinfo part other than global-number-digits.

   P-Asserted-Identity: <sip:unavailable;cpc=priority@unknown.invalid>

4.3.4.1.4. International calling party number ("verstat" tel URI parameter)

An international calling party number is a calling party number provided by international carriers. Given that there is a possiblity that the international calling party number is not sufficiently verified, it is necessary to distinguish the international calling party number from the calling party number provided by the national operators.

4.3.4.1.4.1. Trust relationship

Trust relationship with respect to the "verstat" tel URI parameter shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard in order to deliver the information for identifying the international calling party number. The IMS network shall not send the "verstat" tel URI parameter to user equipment via UNI.

4.3.4.1.4.2. Setting conditions of "verstat" tel URI parameter

Upon receiving an initial INVITE request from an international operator's network, and then forwarding the initial INVITE request to another national operator's IMS network compliant to this Standard, the IMS network shall include the valid calling party number and the "verstat" tel URI parameter set to "No-TN-Validation" defined in [TS 24.229] in the tel URI of the P-Asserted-Identity header field if the valid calling party number is available, in order to indicate that the calling party number is an international network calling number. Note that the terminating IMS network may ignore the "verstat" tel URI parameter set in the SIP URI of the P-Asserted-Identity header field.

The syntax of the "verstat" tel URI parameter is shown in Table 4.3.4.1.4.2-1 according to Subclause 7.2A.20 of [TS 24.229].

Table 4.3.4.1.4.2-1 / JJ-90.30: The syntax of "verstat" tel URI parameter

par =/ verstat verstat = verstat-tag "=" verstat-value verstat-tag = "verstat" verstat-value = "No-TN-Validation"

4.3.4.1.4.3. Examples of "verstat" tel URI parameter

Examples of "verstat" tel URI parameter are shown below:

1) Example when "verstat" tel URI parameter is populated into the P-Asserted-Identity header field including a tel URI.

   P-Asserted-Identity: <tel:+13123456789;verstat=No-TN-Validation>

2) Example when "verstat" tel URI parameter is populated into the P-Asserted-Identity header field including both SIP URI and tel URI.

   P-Asserted-Identity: <sip:+13123456789@example.ne.jp;user=phone>,

   <tel:+13123456789;verstat=No-TN-Validation>

   or

   P-Asserted-Identity: <sip:+13123456789;verstat=No-TN-Validation@example.ne.jp;user=phone>,

<tel:+13123456789;verstat=No-TN-Validation>

4.3.4.1.5. Supplementary services

The setting conditions of a P-Asserted-Identity header field or a Privacy header field if various supplementary services are invoked are specified in the following Subclauses of this section.

4.3.4.1.5.1. Service calls / Network specific number calls

Upon receiving a SIP request outside an existing dialog destined for any 00XY format number, +81AB0 format number or 188/189 number of 1XY format from an originating IMS network, a service provider's IMS network populates the P-Asserted-Identity and the Privacy header field into an outgoing SIP request outside an existing dialog toward a terminating IMS network under the following conditions.

1) The tel URI in the P-Asserted-Identity header field

   - The received global-number-digits shall be included in the tel URI.

   - The received display-name in the name-addr part containing the tel URI shall be included, if received.

   - The received "cpc" tel URI parameter set to the value defined in Subclause 4.3.4.1.3.2 of this Standard shall be included, if received. However, if a service provider's IMS receives the "cpc" tel URI parameter set to "payphone" in a SIP request outside an existing dialog destined for any 00XY format number, it shall discard the received value and set it to "ordinary". Note that the "cpc" tel URI parameter set to "ordinary" can be omitted.

   - Depending on an operator policy, the received "isub" tel URI parameters may be included, if received.

   - The received "verstat" tel URI parameter set to "No-TN-Validation" shall be included, if received.

2) The SIP URI in the P-Asserted-Identity header field

   - The received display-name in the name-addr part containing the SIP URI shall be included, if received both the Privacy header field set to "id" and the display-name.

   - In the cases other than the above, the SIP URI in the P-Asserted-Identity header field may be set to the value other than the received value.

3) The Privacy header field

   - The Privacy header field set to "id" shall be populated, if received.

4.3.4.1.5.2. Call diversion

The setting conditions of a P-Asserted-Identity header field or a Privacy header field if the diverting IMS network invokes call diversion are specified in Subclause 3.1.5 of [JJ-90.27].

4.3.4.2. Reliability of provisional response (100rel)

As specified in [TS 29.165], 100rel shall be supported over the II-NNI. However all user/IMS network equipment do not support and use 100rel; therefore, whether to apply 100rel to all session is decided as a use condition in the inter-operator agreements. {Appendix Table i.4-7 Item 2}

In this Standard, applying "100rel" means the initial INVITE request includes the Supported header field containing "100rel" SIP option-tag, the 18x response to the initial INVITE request includes the Require header field containing "100rel" SIP option-tag, and the PRACK transaction is processed for the 18x response.

When interconnecting to the network where "100rel is not applied to all sessions", the call shall not be released, even if the SIP option tag "100rel" is not included in the Supported header field of an received initial INVITE request or in the Require header field of a received 18x response.

4.3.4.3. Negotiation of resource reservation (precondition)

As specified in [TS 29.165], the applicability of the SIP extension functionality "precondition" over the II-NNI for exchanging the resource reservation status using SDP offer/answer is decided in the inter-operator agreements.

{Appendix Table i.4-7 Item 3}

However, the precondition is not applied over the II-NNI between an IMS network using precondition and an IMS network not using precondition, and the IMS network using precondition shall not populate the parameters related to the precondition (i.e., SIP option tag "precondition" in the Supported header field, SDP attributes ("a=curr:", "a=des:" and "a=conf:")) in a SIP request outside an existing dialog. Also, in order to avoid media clipping, the IMS network using precondition shall send the SIP request outside an existing dialog not containing "a=inactive" after resource reservation for the call was completed.

NOTE: Appendix v in this Standard describes the procedure for termination of the precondition mechanism that an originating IMS network applying the precondition mechanism connects to a terminating IMS network not applying the precondition mechanism after resource reservation in the originating IMS network.

4.3.4.4. Originating area information (P-Access-Network-Info header field)

This Subclause describes the procedure for delivering a CA code allocated per charge area (5 digits number) as originating area information included in a SIP request outside an existing dialog over the II-NNI.

4.3.4.4.1. Trust relationship

Trust relationship with respect to the P-Access-Network-Info header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in accordance with Subclause 6.1.1.3.1 of [TS 29.165] and this Standard in order to deliver originating area information between the IMS networks.

The IMS network shall not send the P-Access-Network-Info header field to user equipment via UNI, and shall remove the P-Access-Network-Info header field received from international networks (overseas).

4.3.4.4.2. Setting conditions of the P-Access-Network-Info header field

An originating IMS network shall populate a single P-Access-Network-Info header field in a SIP request outside an existing dialog destined for any 00XY format number, +81AB0 format number, 188/189 number of 1XY format or the number which is agreed to set originating area information in the SIP request between the originating and the terminating operators.

If the called party number is a number other than above numbers, based on the operator policy, the IMS network may populate a single P-Access-Network-Info header field in a SIP request outside an existing dialog.

The setting conditions of header field parameters in the P-Access-Network-Info header field are specified in following Subclauses.

4.3.4.4.2.1. Access category ("access-type" and "access-class" header field parameters)

An originating IMS network shall set either "access-type" or "access-class" header field parameter into the P-Access-Network-Info header field. For the value of "access-type" and "access-class" header field parameter, the values defined in Subclause 3.4 of [RFC 7315] or Subclause 7.2A.4 of [TS 24.229] shall be used.

NOTE 1: [TS 24.229] defined that "access-type" indicates a specific type of access network used by a served UE, and "access-class" indicates a type of access network used by a served UE where the value of particular "access-type" is unknown.

NOTE 2: The value in both header parameters does not always indicate a type of access network actually used by a served UE in this Standard.

4.3.4.4.2.2. Operator specific geolocation information ("operator-specific-GI" header field parameter)

An originating IMS network shall include "operator-specific-GI" header field parameter set to the CA code in the P-Access-Network-Info header field. The syntax for of "operator-specific-GI" header field parameter is shown in Table 4.3.4.4.2.2-1 according to [RFC 7913].

Table 4.3.4.4.2.2-1 / JJ-90.30: The syntax of "operator-specific-GI"

operator-specific-GI = "operator-specific-GI" EQUAL charge-area-code
charge-area-code = 5DIGIT

NOTE: The syntax for "operator-specific-GI" header field parameter is defined as gen-value (token / host / quoted-string) in [RFC 7315]. In this Standard, token is applied.

4.3.4.4.2.3. Network provided identifier ("network-provided" header field parameter)

An originating IMS network shall include "network-provided" header field parameter into the P-Access-Network-Info header field as an indication that the header field is populated by the IMS network.

4.3.4.4.2.4. Other header field parameters

An originating IMS network shall not include any parameters other than "access-type", "access-class", "operator-specific-GI", and "network-provided" header field parameter into the P-Access-Network-Info header field.

4.3.4.4.2.5. Other considerations

An IMS network receives a SIP request outside an existing dialog from the preceding IMS network and forwards the SIP request to the succeeding IMS network, based on the operator policy, the IMS network determines whether forwards the P-Access-Network-Info header field received from the preceding IMS network to the succeeding IMS network or not.

4.3.4.4.2.6. Example of P-Access-Network-Info header field

Example of a P-Access-Network-Info header field is shown in Table 4.3.4.4.2.6-1.

Table 4.3.4.4.2.6-1 / JJ-90.30: Example of the P-Access-Network-Info header field

SIP message	Example of the P-Access-Network-Info header field
SIP request outside an existing dialog	P-Access-Network-Info: IEEE-802.3ah;operator-specific-GI=32000;network-provided

4.3.4.5. Contractor number (P-Charge-Info header field)

This Subclause describes the procedure for delivering contractor number of originating user included in the SIP request outside an existing dialog over the II-NNI.

4.3.4.5.1. Trust relationship

Trust relationship with respect to the P-Charge-Info header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in accordance with Subclause 9.1 of [RFC 8496] and this Standard.

The IMS network shall not send the P-Charge-Info header field to user equipment via UNI, and shall remove the P-Charge-Info header field received from international networks (overseas).

4.3.4.5.2. Setting conditions of the P-Charge-Info header field

An originating IMS network shall populate a single P-Charge-Info header field in a SIP request outside an existing dialog destined for any 00XY format number, +81AB0 format number or the number which is agreed to set originating contractor number in the SIP request between the originating and the terminating operators, if the calling party number (global-number-digits in the tel URI of P-Asserted-Identity header field) specified in Subclause 4.3.4.1 of this Standard is different from the contractor number.

If the called party number is a number other than above numbers, based on the operator policy, the IMS network may populate a single P-Charge-Info header field in a SIP request outside an existing dialog.

The format of P-Charge-Info header field is shown in Table 4.3.4.5.2-1. The addr-spec of name-addr part of the P-Charge-Info header field shall be tel URI format. The telephone-subscriber part of tel URI shall be global number format, and shall not include display-name part and any tel URI parameters. The maximum number of DIGIT (number from 0 to 9) in the global-number-digits of the telephone-subscriber part of tel URI in the P-Charge-Info header field is 16.

Table 4.3.4.5.2-1 / JJ-90.30: Syntax of P-Charge-Info header field

P-Charge-Info = "P-Charge-Info" HCOLON (name-addr / addr-spec) name-addr =LAQUOT addr-spec RAQUOT

4.3.4.5.3. Example of the P-Charge-Info header field

An examples with respect to the P-Asserted-Identity and P-Charge-Info header field of a SIP request outside an existing dialog in case where the calling party number is different from the contract number are shown below:

P-Asserted-Identity: <tel:+81311111112;cpc=ordinary>

P-Charge-Info: <tel:+81311111111>

4.3.4.6. Inter-operator accounting (P-Charging-Vector header field)

This Subclause specifies the interface conditions for delivering the information for inter-operator accounting using the P-Charging Vector header field over the II-NNI. The P-Charging-Vector header field is defined in [RFC 7315] as a SIP header field for transferring correlation information such as a correlation ID (ICID) for inter-operator accounting and inter-operator identifiers (IOI). The ICID value identifies a SIP transaction outside an existing dialog and a SIP dialog. The IOI value identifies the network from which a SIP request was sent or the network to which the SIP request was sent.

4.3.4.6.1. Trust relationship

Trust relationship with respect to the P-Charging-Vector header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in accordance with Subclause 6.1.1.3.1 of [TS 29.165] in order to deliver information for inter-operator accounting. The IMS network shall not send the P-Charging-Vector header field to user equipment via UNI.

4.3.4.6.2. Setting conditions of the P-Charging-Vector header field

An originating IMS network shall populate a P-Charging-Vector header field into a SIP request outside an existing dialog.

A terminating IMS network shall populate a P-Charging-Vector header field in the 18x and 200 (OK) responses to the SIP request outside an existing dialog. Also, the terminating IMS network shall not populate a P-Charging-Vector header field in the 100 (Trying) responses to the SIP request outside an existing dialog and may populate a P-Charging-Vector header field in a 4xx-6xx response to the SIP request outside an existing dialog.

4.3.4.6.2.1. Correlation ID ("icid" header field parameter)

An originating IMS network shall generate an ICID value and populate the ICID value in the "icid-value" header field parameter of a P-Charging-Vector header field in a SIP request outside an existing dialog in accordance with Subclause 5.6 of [RFC 7315]. The syntax of an "icid-value" header field parameter shall conform to the syntax as described in Table 4.3.4.6.2.1-1.

A terminating IMS network shall populate the "icid-value" header field parameter in a P-Charging-Vector header field of a SIP response to the SIP request outside an existing dialog. The populated "icid-value" header field parameter shall be identical to the header field parameter received in the SIP request outside an existing dialog.

Table 4.3.4.6.2.1-1 / JJ-90.30: Syntax of "icid-value" header field parameter

icid-value = "icid-value" EQUAL token

NOTE: The syntax for "icid-value" header field parameter is defined as gen-nvalue (token / host / quoted-string) in [RFC 7315]. In this Standard, token is applied.

4.3.4.6.2.2. Inter-operator identifier ("orig-ioi" and "term-ioi" header field parameters)

An originating IMS network shall populate the "orig-ioi" header filed parameter containing the inter-operator identifier of the originating IMS network in a P-Charging-Vector header field of a SIP request outside an existing dialog.

Also, a terminating IMS network shall populate the "term-ioi" header field parameter containing the inter-operator identifier of the terminating IMS network in a P-Charging-Vector header field in a SIP response to the SIP request outside an existing dialog, in addition to the "orig-ioi" header field parameter received in the P-Charging-Vector header field of

the SIP request outside an existing dialog from the originating IMS network.

Based on the requirements for inter-operator accounting in Japan, the IOI value defined for "orig-ioi" and "term-ioi" header field parameters in this Standard identifies both a SIP domain name as an inter-operator identifier and additional information for inter-operator accounting. An IMS network shall populate the "orig-ioi" and the "term-ioi" header field parameter in accordance with the syntax described in Table 4.3.4.6.2.2-1. Note that the additional information not listed in Table 4.3.4.6.2.2-1 may be used only when the originating and the terminating operators agree to use.

Table 4.3.4.6.2.2-1 / JJ-90.30: Syntax of "orig-ioi" and "term-ioi" header field parameter

orig-ioi = "orig-ioi" EQUAL ioi-name
term-ioi = "term-ioi" EQUAL ioi-name
ioi-name = [additional-info "."] sip-domain-name
additional-info = "IEEE-802.3ah" / "3GPP-E-UTRAN-FDD" / "GSTN" / "050-IP-Phone" / "PHS" / "SAT-Type1" / "SAT-Tyape2" / "SAT-Type3"
sip-domain-name = *(domainlabel ".") toplabel ["."]
domainlabel = alphanum / alphanum *(alphanum / "-") alphanum
toplabel = ALPHA / ALPHA *(alphanum / "-") alphanum

If there are several alternatives for additional information that can be set for a single SIP domain name, an IMS network using the SIP domain name may omit one of the additional information.

If there are several alternatives for additional information that can be set for a single SIP domain name, the IMS network may omit the setting of additional information in IOI value.

The IMS network generating a 4xx-6xx response may omit the setting of additional information in IOI value.

> NOTE: [TS 24.229], [RFC 7315] and [RFC 7913] do not specify the IOI value to be set. Also, [TS 29.165] specifies that the IOI value to use needs to be determined in the inter-operator agreements. Therefore, the IOI value and whether or not additional information is omitted shall be determined in the inter-operator agreements. {Appendix Table i.4-21 Item 1}

4.3.4.6.2.3. Other header field parameters

Any header field parameters other than "icid-value", "orig-ioi" and "term-ioi" shall not be populated in the P-Charging-Vector header field.

4.3.4.6.2.4. Other considerations

An IMS network which receives a SIP request outside an existing dialog from the preceding IMS network and forwards the SIP request to a succeeding IMS network shall populate a P-Charging-Vector header field in a SIP request outside an existing dialog to the succeeding IMS network. The P-Charging-Vector header field shall contain the "icid-value" and the "orig-ioi" header filed parameter set to the IMS network's own inter-operator identifier.

Also, the IMS network which receives a SIP response to the SIP request outside an existing dialog from the succeeding IMS network and forwards the SIP response to the preceding IMS network shall populate a P-Charging-Vector header field in the SIP response. The P-Charging-Vector header field shall contain the "term-ioi" header field parameter set to the IMS network's own inter-operator identifier, in addition to the "icid-value" header field parameter and the "orig-ioi" header field parameter received in the P-Charging-Vector header field of the SIP request outside an existing dialog from the preceding IMS network. Note that the setting condition of the P-Charging-Vector header field in a SIP response conforms to Subclause 4.3.4.6.2 of this Standard.

4.3.4.6.2.5. Example of the P-Charging-Vector header field

Table 4.3.4.6.2.5-1 shows the coding examples of the P-Charging-Vector header field.

Table 4.3.4.6.2.5-1 / JJ-90.30: Examples of the P-Charging-Vector header field

SIP message	Example of the P-Charging-Vector header field
SIP request outside an existing dialog	P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
SIP response to SIP request outside an existing dialog	P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.example2.ne.jp

4.3.4.7. History of translation from logical number to actual number (History-Info header field)

When a service in which a logical number is used as a called party number (e.g., the number for toll free services beginning with "0120" or "0800", the number for unified service beginning with "0570") is invoked, the logical number is translated into an actual number used for routing of a call to the terminating user equipment (e.g., the number "0AB-J", the number beginning with "0A0").

In a network serving the terminating user equipment, the supplementary services (e.g., notification of an original called logical number to the terminating user equipment, call rejection when an original called number is an actual number of a service which needs to be invoked by the logical number) can be provided. Therefore, the history information of number translation from a logical number to an actual number needs to be delivered through the II-NNI.

This Subclause provides the procedures for delivering the history information of number translation over the II-NNI.

4.3.4.7.1. History information of number translation from logical number to actual number

An IMS network uses History-Info header field in accordance with [RFC 7044] and [RFC 7131], and can include two history information of number translation from logical number to actual number (at most two translations) in the initial INVITE request.

4.3.4.7.2. Trust relationship

Trust relationship with respect to the History-Info header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in accordance with Subclause 6.1.1.3.1 of [TS 29.165] in order to deliver information for history of number translation from logical number to actual number between the IMS networks. The IMS network shall not send the History-Info header field to user equipment via UNI, and shall remove the History-Info header field received from international networks (overseas).

4.3.4.7.3. Setting conditions of the History-Info header field

The history of translation from logical number to actual number created at a service provider network consists of:

- the entry including the called party number before performing number translation ("mapped-from-entry") and;

- the entry including the called party number after performing number translation ("mapped-to-entry").

The setting conditions of the History-Info header field are specified in the following Subclauses.

4.3.4.7.3.1. hi-targeted-to-uri

Since the URIs populated in the hi-targeted-to-uri of the "mapped-from-entry" and the "mapped-to-entry" are derived from the Request-URIs according to [RFC 7044], the SIP URI containing the called party number as a global E.164 number specified in Subclause 4.3.2.2 of this Standard shall be applied. The maximum number of DIGIT (number from 0 to 9) in the global-number-digits of the SIP URI in the hi-targeted-to-uri of the History-Info header field is 26. In the last entry of the History-Info header field, the hostport part of the SIP URI shall be the SIP domain name identifying the network to which the SIP request is sent specified in Subclause 4.3.2.3 of this Standard. In the entries other than the last entry, the hostport part of the SIP URI should be the SIP domain name identifying the network from which the SIP request was sent. In this Standard, display-name part shall not be populated into the hi-targeted-to-uri.

NOTE: [RFC 7044] Clause 5 specifies the syntax that name-addr can be populated into the hi-targeted-to-uri.

4.3.4.7.3.1.1. Privacy header field

A service provider's network may populate the Privacy header field ("Privacy=none" or "Privacy=history") into the hi-targeted-to-uri of the "mapped-from-entry" to indicate whether the presentation of the history information of number translation to the terminating user (equipment) is allowed ("presentation allowed") or restricted ("presentation restricted").

When indicating "presentation allowed", the service provider's network shall populate either "Privacy=none" or no Privacy header field into the hi-targeted-to-uri of the "mapped-from-entry". When indicating "presentation restricted", the service provider's network shall populate "Privacy=history" into the hi-targeted-to-uri of the "mapped-from-entry".

If "presentation allowed" is indicated in the hi-targeted-to-uri of the "mapped-from-entry", then the terminating IMS network may send the information of the entry in the header field or parameter other than the History-Info header field to the terminating user (equipment). If "presentation restricted" is indicated in the hi-targeted-to-uri of the "mapped-from-entry", then the terminating IMS network shall not send the information of the entry to the user (equipment).

4.3.4.7.3.1.2. Reason header field

A service provider's network shall not populate the Reason header field into the hi-targeted-to-uri of the "mapped-from-entry" and the "mapped-to-entry"; however, there is a possibility that the SIP header field not used for the history information of number translation is populated into the hi-targeted-to-uri when sharing the entry(ies) with the other services.

4.3.4.7.3.1.3. tel URI parameter

The tel URI parameters such as "isub", "npdi" and "rn" shall not be populated into the hi-targeted-to-uri.

4.3.4.7.3.1.4. SIP URI parameter

The "user" SIP URI parameter with the value of "phone" shall be set in the SIP URI in accordance with Subclause 4.3.2.4.1 of this Standard, since the global number format is used for the called party number.

The "cause" SIP URI parameter defined in [RFC 4458] shall be populated into the hi-targeted-to-uri of the "mapped-to-entry". The value of "cause" SIP URI parameter shall be 380 indicating service number translation defined in [RFC 8119]. The terminating IMS network recognizes the entry whose value of "cause" SIP URI parameter is 380 as "mapped-to-entry". The SIP URI parameters other than the above SIP URI parameters shall not be populated into the entries.

4.3.4.7.3.2. History-Info header parameter

4.3.4.7.3.2.1. Index value (hi-index)

The index value (i.e., hi-index value) in the "mapped-from-entry" or the "mapped-to-entry" is set in accordance with [RFC 7044].

4.3.4.7.3.2.2. "mp" parameter (mp-param)

The "mp" header field parameter shall be populated into the "mapped-to-entry" containing the "cause" SIP URI parameter with the value of 380. The index-val defined in [RFC 7044] of "mp" header field parameter shall be set to the index value of the "mapped-from-entry". The terminating IMS network shall consider a received entry as the "mapped-from-entry" when the received entry has an index value equivalent to the index-val of the "mp" header field parameter populated in another entry containing the "cause" SIP URI parameter with the value of 380.

4.3.4.7.3A.   Procedures when the number translations from logical number to actual number are performed more than the upper limit.

The number of number translations from logical number to actual number is equal to the number of hi-entries containing "cause" SIP URI parameter with the value of 380. When the number of numter translations exceeds 2 that is the upper limit, the call shall be released with a 480 (Temporarily Unavailable) response.

4.3.4.7.4. Interaction with other services

This Subclause describes the interaction with other services and the signalling requirements when providing the other services simultaneously.

4.3.4.7.4.1. Communication Diversion (CDIV)

The History-Info header field is used for delivering the history information of number translation from logical number to actual number specified in this Standard and/or the history information of communication diversion specified in [JJ-90.27]. If both history information is included in the History-Info header field, then the entries of the History-Info header field are populated in the order Request-URIs were changed according to [RFC 7044].

NOTE 1: In the ISUP specification, no parameter indicating the ordering of the history information of number translation and that of communication diversion; therefore, the information related to the ordering of the history information will be discarded when the interworking between SIP and ISUP is invoked.

NOTE 2: The SIP domain name of the hi-targeted-to-uri included in the the History-Info header field may not indicate the IMS network serving the number included in the corresponding entry when the interworking from ISUP to SIP was invoked.

NOTE 3: In accordance with Annex b of [JJ-90.27], if the redirection counter is equal to 3 or more when the interworking from ISUP to SIP is invoked, placeholder History-Info hi-entry is set in the History-Info header field.

4.3.4.7.4.2. Upper limit of History-Info header field and procedures when the number of the entries in the History-Info header field exceeds the upper limit

The upper limit of the number of the entries in the History-Info header field is 8.

The procedures when the number of the entries in the History-Info header field exceeds the upper limit are shown below:

- when the number of the entries in the History-Info header field exceeds the upper limit due to the number translation from logical number to actual number, the procedures defined in the Subclause 4.3.4.7.3A of this Standard shall be applied.

- when the number of the entries in the History-Info header field exceeds the upper limit due to Communication DIVersion, the procedure defined in in the Subclause 3.2.3 of [JJ-90.27] shall be applied.

4.3.4.8. SIP session timer (timer)

SIP session timer shall be applied for all sessions. An originating IMS network shall populate the "timer" in the Supported header field in the initial INVITE request, and a terminating IMS network shall populate the "timer" in the Require header field in the 200 (OK) response to the initial INVITE request.

The session timer duration specified in the Session-Expires header field shall be selected from 180 to 300 seconds by each IMS network operator, and the selected value shall be exchanged between IMS network operators. {Appendix Table i.4-7 Item 1}

4.3.5. SIP message body

4.3.5.1. Session Description Protocol (SDP)

The SDP message body shall be included in the INVITE request, except for a specific service (e.g. 3rd party call control) requiring the usage of the INVITE request without SDP message body, in accordance with [TS 29.165]. Also, the SDP message body shall not be included in the ACK request except for a specific service (e.g. 3rd party call control) requiring the usage of the ACK request with SDP message body.

The SDP message body shall not be included in the PRACK request unless the negotiation of resource reservation {Appendix Table i.4-7 Item 3} is used between the IMS networks based on the inter-operator agreement.

4.3.5.1.1. Session level description

The IMS network shall be prepared to receive "v=" line (protocol version), "o=" line (origin), "s=" line (session name), "c=" line (connection data), "b=" line (bandwidth) and "a=" line (attributes), in accordance with [TS 29.165].

4.3.5.1.1.1. "v=" line (protocol version)

As specified in [TS 29.165], the protocol version shall be set to "0" in accordance with [RFC 4566].

4.3.5.1.1.2. "o=" line (origin)

As specified in [TS 29.165], "o=" line shall be set as described in [RFC 4566].

4.3.5.1.1.3. "s=" line (session name)

As specified in [TS 29.165], "s=" line shall be set as described in [RFC 4566].

4.3.5.1.1.4. "c=" line (connection data)

As specified in [TS 29.165], "c=" line shall be set as described in [RFC 4566].

> NOTE: If a "c=" line is included in a media description, the "c=" line in the media description is applied to the corresponding media.

4.3.5.1.1.5. "b=" line (bandwidth)

The applicable type(s) of "b=" line between the IMS networks are determined in the inter-operator agreements. {Appendix Table i.4-10 Item 2, 3}

IMS operators shall control the sending rate of total amount of the outgoing media stream(s) for the session so as not to exceed the bandwidth specified in the last received "b=AS" parameter from the receiver of the outgoing media stream(s) or the pre-defined bandwidth for the session. IMS operators shall control the sending rate of total amount of the outgoing RTCP packets for the session so as not to exceed the bandwidth specified in the last received or sent "b=RR"/"b=RS" parameters included in the SDP answer from the receiver of the outgoing RTCP packets or the pre-defined bandwidth decided in the inter-operator agreements.

4.3.5.1.1.6. "a=" line (attributes)

The applicable "a=" line(s) between the IMS networks are determined in the inter-operator agreements. {Appendix Table i.4-10 Item 4}

> NOTE: If an "a=" line of the session description is also included in the media description, the "a=" line of the media description is applied to the corresponding media.

4.3.5.1.2. Time description

The IMS network shall be prepared to receive "t=" line (timing), in accordance with [TS 29.165].

4.3.5.1.2.1. " t=" line (timing)

As specified in [TS 29.165], "t=" line is set as described in [RFC 4566].

4.3.5.1.3. Media description

The IMS network shall be prepared to receive "m=" line (media descriptions), "c=" line (connection data), "b=" line (bandwidth) and "a=" line (attributes), in accordance with [TS 29.165].

4.3.5.1.3.1. "m=" line (media descriptions)

The applicable media types other than audio and the conditions applied to each media are determined in the inter-operator agreements, in accordance with [TS 29.165]. {Appendix Table i.4-10 Item 1}
{Appendix Table i.4-11 Item 1, 2, 3, 4, 5 and 6}

The port number to which the RTP is sent shall be set to an even port number in the <port> sub-field, in accordance with [RFC 3550].

> NOTE: The next higher (odd) port number of the RTP port number is used to receive the RTCP corresponding to the RTP.

4.3.5.1.3.2. "c=" line (connection data)

As specified in [TS 29.165], "c=" line shall be set as described in [RFC 4566].

NOTE: When an originating IMS network sends the initial INVITE request including "c=" line containing IPv6 address in the SDP towards a terminating UE not supporting IPv6 in the circumstances where both IPv4 and IPv6 are applied between IMS networks, the 488 (Not Acceptable Here) response containing the Warning header field set to either warn-code "300" indicating "Incompatible network protocol" or warn-code "301" indicating "incompatible network address format" will be returned to the originating IMS network. In this case, the originating IMS network or an originating UE can perform the IPv4 fallback procedure.

4.3.5.1.3.3. "b=" line (bandwidth)

The applicable type(s) of "b=" line between the IMS networks and the bandwidth value corresponding to a codec are determined in the inter-operator agreements. {Appendix Table i.4-10 Item 2, 3}{Appendix Table i.4-17 Item 3}

IMS operators shall control the sending rate of the outgoing media stream so as not to exceed the bandwidth for the media stream specified in the last received "b=AS" parameter from the receiver of the outgoing media stream or the pre-defined bandwidth for the media stream. IMS operators shall control the sending rate of the outgoing RTCP packets so as not to exceed the bandwidth specified in the last received or sent "b=RR"/"b=RS" parameters included in the SDP answer from the receiver of the outgoing RTCP packets or the pre-defined bandwidth decided in the inter-operator agreements.

When all the following conditions are fulfilled, setting of "b=" line in the SDP is able to be omitted.

- the corresponding "m=" line in the SDP is speech media ("m=audio").

- the speech codec list contains only the codec which is constant bit rate and the bandwidth for the codec is uniquely determined, or contains only the codec which is constant bit rate and the bandwidth for the codec is uniquely determined and the DTMF codec (telephone-event).

NOTE 1: [JT-Y1221] specifies the media bandwidth when 20 ms is applied as paketization period for G.711 μ-law. In the case jitter is not considered, the RTP bandwidth are 96kbps (IPv4/IPv4 header length=60byte) and 88kbps (IPv4) respectively, excluding the RTCP bandwidth. Taking both RTP bandwidth and RTCP bandwidth into consideration, the recommended bandwidth for G.711 μ-law is 105kbps.

NOTE 2: [TS 26.114] specifies the setting condition of "b=AS" line for mobile speech codecs (e.g., AMR).

NOTE 3: The example codecs with constant bit rate and uniquely determined bandwidth are G.711 μ-law, G.711 A-law and G.722. Codecs such as AAC and EVS do not comply with the conditions.

4.3.5.1.3.4. "a=" line (attributes)

The applicable "a=" line(s) between the IMS networks are determined in the inter-operator agreements. {Appendix Table i.4-10 Item 4}

4.3.5.1.4. Codec

4.3.5.1.4.1. Speech codec

G.711 μ-law shall be contained in the codec list of speech media for an SDP offer sent over the II-NNI except the II-NNI between two mobile IMS networks. If "a=ptime" is not contained in an SDP answer, 20 ms shall be used as paketization period for G.711 μ-law. In case of the II-NNI between two mobile IMS networks, AMR(-NB) shall be contained in the codec list of speech media for an SDP offer. Other speech codec(s) shall be determined in the inter-operator agreements. {Appendix Table i.4-11 Item 1}

The speech codec(s) that is neither the mandatory speech codec specified in this Standard nor the applicable codec determined in the inter-operator agreements shall not be set in an SDP offer and answer.

4.3.5.1.4.2. Video codec

If video media is used between two IMS networks, video codec(s) to use shall be determined based on the inter-operator agreements. {Appendix Table i.4-11 Item 2}

The video codec(s) that is not included in the codec list of video media determined in the inter-operator agreemtns shall not be set in an SDP offer.

4.3.5.1.4.3. Transcoding

Refer to Annex T of [TS 24.229] for the procedures and rules of transcoding.

4.3.5.1.5. Dual Tone Multi Frequency (DTMF)

DTMF is transported between the IMS networks as RTP packets using either "telephone-event" payload format defined in [RFC 4733] or audio tones.

> NOTE 1 DTMF transport using SIP INFO mechanism is specified as an option item in [TS 29.165], however, the mechanism is outside the scope of this Standard.

Either one of the above two methods is used for DTMF transport in the session based on the SDP negotiation between the originating and the terminating IMS networks. If "telephone-event" is included in the SDP answer, "telephone-event" based DTMF transport is used for the session. Otherwise, the DTMF is transported as audio tones. In order to guarantee DTMF transport, the IMS network shall be prepared to receive and send DTMF as audio tones. Also, the sampling rate of DTMF (specified in the <clock rate> component of the "a=rtpmap" attribute line) shall be same as the sampling rate of the used speech codec in the session.

> NOTE 2 When G.711 μ-law is used as a speech codec in the session, <clock rate> component of the "a=rtpmap" attribute line for "telephone-event" payload format in the SDP is set to "8000", since 8000 Hz is applied for G.711 μ-law as sampling rate according to [RFC 3551].

4.3.5.1.6. Media change

4.3.5.1.6.1. Codec change in an established media

The applicability of the codec change in an established media between the IMS networks shall be determined in the inter-operator agreements. If applicable, the allowed pattern of the codec change shall be determined. {Appendix Table i.4-12 Item 1, 2}

4.3.5.1.6.2. Media direction attribute change

When stopping to send RTP packets e.g., due to hold, an SDP offer containing either "a=inactive" or "a=recvonly" in the UPDATE request or re-INVITE request shall be sent over the II-NNI.

> NOTE: Note that the call could be released at the IMS network monitoring for extinction of RTP packets, if no RTP packet is sent in a certain time without the media direction attribute change.

When resuming to send RTP packets, an SDP offer containing either "a=sendrecv" or "a=sendonly" in the UPDATE request or re-INVITE request shall be sent over the II-NNI.

4.3.5.2. Other MIME

The applicability of MIME type other than SDP (MIME type is "application/sdp") over the II-NNI shall be determined in the inter-operator agreements in accordance with Subclause 6.1.4 of [TS 29.165]. {Appendix Table i.4-13 Item 1}

4.3.6. Provision of announcement/talkie

This Subclause specifies the requirements and procedures for providing announcement and talkie.

4.3.6.1. Announcements

This Subclause specifies the requirements and procedures for providing announcement.

An announcement during the establishment of the session is sent as early media in accordance with Subclause 4.3.6.1.1 of this Standard, and an announcement after the establishment of the session is sent as established media on the confirmed dialog in accordance with Subclause 4.3.6.1.2 of this Standard. An announcement for rejecting the communication request is sent as either early media or established media on the confirmed dialog.

> NOTE 1: [TS 24.628] specifies the method using Call-Info header field in addition to the method using established media on the confirmed dialog for providing an announcement after the establishment of the session by IMS network. In this Standard, only the method using established media on the confirmed dialog is allowed.

NOTE 2: [TS 24.628] specifies the method using established media on the confirmed dialog, the method using early media and the method using Error-Info header field in 3xx -6xx responses for providing an announcement by an IMS network when rejecting the communication request. In this Standard, the method using established media on the confirmed dialog or the method using early media are allowed.

4.3.6.1.1. Early media

For providing early media, either the gateway model specified in [TS 24.628] Annex G or the method using an SDP answer for an early media in the 18x provisional response not containing the SIP option tag "100rel" in the Require header field shall be used over the II-NNI.

In case of II-NNI between mobile IMS networks, the multiple early dialog method specified in [TS 24.628] Annex D can also be applied in addition to the above method.

The terminating IMS network providing early media shall send a 18x response in accordance with Subclause 4.3.6.1.1.1 and Subclause 4.3.6.1.1.2 of this Standard.

NOTE 1: [TS 24.628] specifies the gateway model, the multiple early dialog method and the method using the Alert-Info header field to provide early media.

NOTE 2: Appendix vi in this Standard describes the procedure for converting multiple early dialogs to a single dialog.

For the gateway model, after providing an early media, in order to replace media between the originating UE and the audio source with media between the originating UE and the terminating UE, the IMS network sends an UPDATE request with an SDP offer indicating replaced media IP address and port number. If there is no change in the IP address and the port number of media over the II-NNI, the IMS network providing early media can omit sending an UPDATE request.

4.3.6.1.1.1. Authorization of early media (P-Early-Media header field)

When sending early media (e.g. announcement during the establishment of a session) from an IMS network, the P-Early-Media header field defined in [RFC 5009] is populated into a 18x response in order to indicate that the SDP in the 18x response is a network-generated SDP in accordance with the relevant 3GPP specifications (e.g., [TS 24.628]).

4.3.6.1.1.1.1. Trust relationship

Trust relationship with respect to the P-Early-Media header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in order to deliver information for authorization of early media in accordance with Subclause 6.1.1.3.1 of [TS 29.165].

4.3.6.1.1.1.2. Setting conditions of the P-Early-Media header field

The IMS network shall follow the setting conditions of the P-Early-Media header field described as follow:

- the P-Early-Media header filed containing "supported" parameter may be populated into an initial INVITE request;

- when an IMS network starts sending early media, the P-Early-Media header field containing either "sendrecv" or "sendonly" parameter shall be populated into a 18x response containing an SDP answer generated by the IMS network to an initial INVITE request regardless of whether or not the IMS network had received the P-Early-Media header field containing "supported" parameter in the initial INVITE request. In addition to "sendrecv" or "sendonly" parameter, the IMS network may populate "gated" parameter if the IMS network has already performed gating control for media path based on the value of the direction parameter in the P-Early-Media header field (NOTE 1);

- when the IMS network stops sending early media, the P-Early-Media header field containing "inactive" parameter may be populated into an UPDATE request or a 18x response (NOTE 2);

- when the IMS network restarts sending early media, the P-Early-Media header field containing either "sendrecv" or "sendonly" parameter may be populated into an UPDATE request or a 18x response (NOTE 2); and

- otherwise, the P-Early-Media header field shall not be populated into any SIP requests and responses on early dialog.

NOTE 1: Upon receiving the P-Early-Media header field from an international network (overseas) in the untrusted domain, then an IMS network needs to remove the P-Early-Media header field as specified in [TS 24.229] Subclause 4.4.11. Also, an terminating IMS network needs to populate the P-Early-Media heder field set to either "sendrecv" or "sendonly" parameter in a 18x response including an SDP containing either "a=sendrecv" (the omission of the direction attribute) or "a=sendonly", if the terminating IMS network forwards early media from a international network (overseas) in the untrusted domain.

NOTE 2: Subclause 13.2.1 of [RFC 3261] specifies that once an SDP answer has been sent to an initial SDP offer, setting of SDP offers in any subsequent SIP responses on the same dialog is prohibited. Therefore, if the IMS network has sent the first 18x response with the SDP answer and the Require header field containing "100rel", the IMS network needs to stop or restart sending early media on the same dialog by an UPDATE request or a 18x response not containing SDP.

NOTE 3: Subclause 10.2.11 of [TS 29.162] specifies that H.248 specification (gate control procedure) between IBCF and TrGW for controlling media path of early media based on the value of the P-Early-Media header field and/or SDP answer.

NOTE 4: Subclause 4.7.2.1 of [TS 24.628] specifies that whether the originating UE plays locally generated ring back tone or received early media from IMS network is decided based on the presence of P-Early-Media header field containing either "sendrecv" or "sendonly" parameter.

4.3.6.1.1.2.   Setting conditions of SDP when providing early media

When sending early media using the method other than forking model (multiple early dialog method), the SDP in the 18x responses and the SDP in the 200 (OK) response to the initial INVITE request shall be same. If the session which provides early media applies "100rel", subsequent 18x responses after the first reliable provisional response and 200 (OK) response to the initial INVITE request basically do not contain any SDP, since the SDP answer has sent in the first reliable provisional response to the initial INVITE request and the SDP offer/answer has been completed.

4.3.6.1.1.3.   Setting condition of SIP timer when early media is provided

When the terminating IMS network wishes to provide an early media beyond 170 seconds, the terminating IMS network shall send a 18x response within the duration from 120 to 160 seconds (for the case applying "100rel") or the duration from 60 to 160 seconds (for the case not applying "100rel") to refresh Timer C in the originating IMS network. The terminating IMS network may release the call based on the timer determined by the operator policy, regardless of Timer C expiration.

Based on the operator policy, the originating IMS network may terminate the early dialog by sending a CANCEL request regardless of Timer C expiration, if the duration of the early dialog exceeds 170 seconds and the originating IMS network has not refreshed Timer C by receiving a 18x response from the terminating IMS network.

NOTE: The originating IMS network can terminate the early dialog after the excess of the minimum time to maintain early dialog (determined by the operator), even if Timer C is refreshed by receiving 18x responses and does not expire. The specific value of the minimum time to maintain early dialog is FFS.

4.3.6.1.2. Established media on the confirmed dialog

The IMS network can provide an announcement using established media on the confirmed dialog.

4.3.6.2.   Communication progress information

This Subclause describes the function and procedure required for IMS networks when providing communication progress information toward an originating UE.

Communication progress information shall be provided by one of the following procedures:

1) if a terminating IMS network does not to provide a communication progress information to an originating IMS network, the terminating IMS network sends a 180 (Ringing) provisional response not containing SDP answer towards the originating IMS network. If the originating IMS network determines to send early media as a communication progress information to an originating UE, the originating UE will play early media as a communication progress information. Otherwise, the originating UE generates a local communication progress information; or

2) if a terminating IMS network does provide a communication progress information to an originating IMS network, the terminating IMS network sends either a 180 (Ringing) or a 183 (Session Progress) provisional response with SDP answer as specified in Subclause 4.3.6.1.1.1 of this Standard, an originating UE will play early media as a communication progress information.

NOTE: According to [IR.92] and [JT-Q3402], the originating UE continues playing early media even if receiving a 180 (Ringing) or a 183 (Session Progress) provisional response without indication of early media information.

4.3.6.3. Unallocated (unassigned) number talkie

This Subclause describes the functions and procedures required for IMS networks when providing an unallocated (unassigned) number talkie service.

In principle, the following conditions shall be followed when connecting to an unallocated (unassigned) number talkie:

- A terminating IMS network returns a SIP response indicating an unallocated number to an originating IMS network, and a connection to the unallocated (unassigned) number talkie is established inside the originating IMS network.

- When a terminating IMS network is unable to guarantee an unallocated (unassigned) number, the terminating IMS network notifies the originating IMS network of a status other than "unallocated (unassigned) number" in order to avoid a talkie connection inside the originating IMS network.

NOTE: An originating IMS network may establish unallocated (unassigned) number talkie when the RCODE field set to 3 in the ENUM answer is received.

4.3.6.3.1. Trust relationship

Trust relationship with respect to the Reason header field shall exist between the two interconnected networks which implement an interconnection interface compliant to this Standard, in accordance with Subclause 6.1.1.3.1 of [TS 29.165] in order to deliver SIP response indicating unallocated number. The IMS network shall not send the Reason header field to user equipment via UNI.

4.3.6.3.2. Required functionality on terminating IMS network side

When the called party number is an unallocated number, a terminating IMS network sends back a 404 (Not Found) response containing a Reason header field. When receiving a 404 (Not Found) response containing a Reason header field from terminating user equipment, the terminating IMS network shall verify whether the SIP response can be guaranteed then returns the SIP response only when the called party number is actually an unallocated number.

When an unallocated number is detected, the Reason header field is encoded as follows:

Reason: Q.850;cause=1;text="unallocated number"

(The setting of text="unallocated number" can be omitted.)

NOTE: If a terminating IMS network can not guarantee that the called party number is actually an unallocated number (e.g., terminating calls from international networks (overseas)), the Reason header field specified in this Subclause will not be included in a 404 (Not Found) response.

4.3.6.3.3. Required functionality on originating IMS network side

When receiving a 404 (Not Found) response containing a Reason header field specified in Subclause 4.3.6.3.2 of this Standard from the terminating IMS network, the originating IMS network takes into account that the called party number is an unallocated number then connects to an unallocated (unassigned) number talkie.

4.3.7. Supplementary services

The applicability of supplementary services over the II-NNI and its details shall be determined in the inter-operator agreements. {Appendix Table i.4-15 Item 1, 3 and 4} This Standard regards supplementary service(s) as "applied" only if the particular SIP message(s) required for the service is sent over the II-NNI. Therefore, the applicability is not depending on whether an operator provides the service inside the operator's network.

4.3.7.1. Communication HOLD

When the Communication HOLD service is applied over the II-NNI, the following signals specific to this supplementary service are sent over II-NNI.

- During bidirectional communication, an SDP offer to change direction attribute (a=sendonly) is provided by one network (and the session state transits to unidirectional HOLD).

- During unidirectional HOLD, an SDP offer to change direction attribute (a=inactive) is provided by the network which is not on HOLD (and the state transits to bidirectional HOLD).

- During bidirectional HOLD, an SDP offer to change direction attribute (a=recvonly) is provided by one network (and the state transits to unidirectional HOLD).

- During unidirectional HOLD, an SDP offer to change direction attribute (a=sendrecv) is provided by the network which is on HOLD (and the state transits to bidirectional communication).

If the network which selects this supplementary service "Yes" interconnects with the network which selects this supplementary service "No", and the latter network which selects "No" is not able to accept SDP offers to change direction attribute which are mentioned above, the network which selects "No" may either send a 488 (Not Acceptable Here) response and maintain the call, or send an error response and terminate the call according to its policy.

In the same way, if the network which selects this supplementary service "Yes" but does not allow the change of directional attribute except for Communication HOLD service, interconnects with the network which applies the method to change of directional attribute other than Communication HOLD at the II-NNI irrespective of applicability of this supplementary service, and if the former network which does not allow the change of directional attribute except for Communication HOLD is not able to accept SDP offers except for changing directional attribute which are mentioned above, the former network may either send a 488 (Not Acceptable Here) response and maintain the call, or send an error response and terminate the call according to its policy.

4.3.8. Maximum SIP message length

The maximum value and the minimum value of the SIP message elements shall conform to Table 4.3.8-1 and Table 4.3.8-2. Appendix viii describes a recommended method for IMS operators to avoid failures when they send SIP messages compliant to this Subclause.

Table 4.3.8-1 / JJ-90.30: Limitation on the maximum acceptable value of SIP message

Elements	Acceptable maximum value
Maximum length per line of a SIP message	More than/equal to 255 bytes (including CRLF)
Maximum length of a SIP message (exclude message body)	More than/equal to 3,000 bytes
Maximum length of a SIP message body	More than/equal to 999 bytes
NOTE: 1 byte is equal to 1 octet in this Standard.	

Table 4.3.8-2 / JJ-90.30: Limitation on the entries of the same SIP header field for a SIP message

SIP header field	Maximum entry of the same SIP header field
Via	1 entry
Record-Route	0 entry
Route	Emergency call specified in [JJ-90.28] : 1 entry Other SIP requests : 0 entry

4.3.8.1. SIP header field which is able to include multiple entries

When a SIP header field is able to have multiple entries, the entries can be described in a line and separated with a delimiter such as "," within the receiving operator's limit of acceptable maximum length per line.

If the line (SIP header field) with multiple entries exceeds the limitation of the maximum length per line described in Table 4.3.8-1, the entries in the SIP header field shall be divided into multiple lines per entry and every line of the SIP header field shall be started from the SIP header field name.

- the example of the SIP header field exceeding the acceptable maximum length per line for the operator receiving the SIP message (257 byte including the CRLF at the end of the line);

```
History-Info: <sip:+81800DEFGHJK@free.example1.ne.jp;user=phone?Privacy=history>;index=1,<sip:+81ABCDEF
GHJ@example2.ne.jp;user=phone;cause=380?Privacy=none>;index=1.1;mp=1,<sip:+81A0CDEFGHJK@ims.mncXX
X.mcc440.3gppnetwork.org;user=phone;cause=302>;index=1.1.1;mp=1.1[CR][LF]
```

- the example of the SIP header field divided into multiple lines per entry in the SIP message, in order not to exceed the acceptable maximum length per line for the operator receiving the SIP message;

```
History-Info: <sip:+81800DEFGHJK@free.example1.ne.jp;user=phone?Privacy=history>;index=1[CR][LF]
History-Info: <sip:+81ABCDEFGHJ@example2.ne.jp;user=phone;cause=380?Privacy=none>;index=1.1;mp=1[CR][
LF]
History-Info: <sip:+81A0CDEFGHJK@ims.mncXXX.mcc440.3gppnetwork.org;user=phone;cause=302>;index=1.1.1
;mp=1.1[CR][LF]
```

4.3.8.2. Maximum SIP URI length

The maximum value of the SIP URI in the SIP message shall conform to Table 4.3.8.2.

Table 4.3.8.2 / JJ-90.30: Limitation on the maximum acceptable value of SIP URI

Elements	Acceptable maximum value
Maximum length of a SIP URI in a Reqest-URI	More than/equal to 128 bytes
Maximum length of a host part	More than/equal to 44 bytes
NOTE: 1 byte is equal to 1 octet in this Standard.	

Annex a (Normative): Clarification of TS 29.165

a.1. General

[TS 29.165] defines items that are recommended to be selected by inter-operator agreements as "option items", and then option items are classified and listed as "the list of option items". For the purpose of achieving smooth and easy inter-operator agreements, this Annex defines the applicability of the option items specified in [TS 29.165] based on consensus between IMS network operators within Japan.

a.2. Methodology

For the purpose of clarifications on the specification, applicability of option items specified in Annex C of [TS 29.165] are selected, and correspondence relation between clarifications in this Standard and option items are described explicitly. Note that the selection of applicability of option item and the result of selection of the option item are called "option item selection" and "option item selection table", respectively.

a.3. Format of option item selection table and definition

The format of the option item selection table is equivalent to that of [TS 29.165] Annex C. The example format and definition of option item selection table are shown in Annex Table a.3.1 and Annex Table a.3.2, respectively.

Annex Table a.3-1 / JJ-90.30: Example format [TS 29.165] Table C.x.y.z (NOTE)

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	-	-	Yes	*<The clarifications in this Standard are described here.>*
			No	
NOTE: In the title of each table, the table number of the option item table in Annex C of [TS 29.165] is described.				

Annex Table a.3-2 / JJ-90.30: Definitions relating to option item selection table

column	Title of option item selection table		Definition
2	Option item		This column shows the option item.
3	References		This column shows the relevant Clause(s) and/or table number(s) in the main body of [TS 29.165].
4	Applicability at the II-NNI	Hatching "Yes"	This type of item shows the capability(ies) and/or service(s) relating to the option item are provided over the II-NNI. In order to ensure the inter-connectivity, each side IMS operator's network of II-NNI shall support the capability(ies) in accordance with this Standard and normative reference(s).
		Hatching "No"	This type of item is the option item which is outside the scope of this Standard.
		No hatching	This type of item is the option item in this Standard as with [TS 29.165].
5	Details for operator choice		This column shows the details for the inter-operator agreements if the corresponding pattern is chosen in the "Applicability at the II-NNI". The clarifications in this Standard are described between angled brackets (< >). This column intends to be used to exchange each detailed conditions of the II-NNI by means of filling the conditions in the columns corresponding to the selected pattern. Blank row intends to be used as a place holder for describing additional detailed conditions specific to the operator network not associated with the explicit descriptions in this Annex.

a.4. Option item selection table

Annex C of [TS 29.165] specifies option item tables, which are common to roaming and non-roaming II-NNI, specific to roaming II-NNI, and specific to non-roaming II-NNI, respectively. In subsequent Subclauses, the results with respect to the selection of option items in this Standard are specified.

> NOTE: Note that there are possible changes to the result with respect to the selection of option items based on the demand from IMS network operators within Japan.

a.4.1. Supported II-NNI traversal scenarios

This Subclause describes the results with respect to the selection of option items specified in Subclause C.3.0 of [TS 29.165].

Annex Table a.4.1-1 / JJ-90.30: Supported II-NNI traversal scenarios   [TS 29.165] Table C.3.0.1

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Roaming II-NNI support	entire present specification	Yes	Option items in Subclauses C.3.1 and C.3.2.
				Applicability of "Roaming Architecture for Voice over IMS with Local Breakout" (Clause 19).
				If "Roaming Architecture for Voice over IMS with Local Breakout" is applicable: + option items in Subclause C.3.3; and + the default TRF address to use when not included in the g.3gpp.trf feature-capability indicator.
			No	
2	Non-roaming II-NNI support	entire present specification	Yes	Option items in Subclauses C.3.1 and C.3.3. *<Select the option items in Subclause a.4.2 and Subclause 4.4>*
			No	

a.4.2. Option item selection tables common to roaming and non-roaming II-NNI

This Subclause describes the results with respect to the selection of option items specified in Subclause C.3.1 of [TS 29.165].

Annex Table a.4.2-1 / JJ-90.30: SIP method   [TS 29.165] Table C.3.1.1

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	INFO method	table 6.1/5A table 6.1/5B	Yes	Info package name to use.
			No	
2	MESSAGE method	table 6.1/9A table 6.1/9B	Yes	Inside or outside existing dialog, and content of MESSAGE request.
			No	
3	REFER method	table 6.1/16 table 6.1/17	Yes	Inside or outside existing dialog.
			No	

Annex Table a.4.2-2 / JJ-90.30: SIP overload control　[TS 29.165] Table C.3.1.1A

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	SIP overload control	table 6.1.3.1/106 clause 21.1	Yes	Mechanisms to be used.
				Whether to exempt MPS from SIP overload controls.
			No	
2	Feedback control	table 6.1.3.1/107 clause 21.2	Yes	Algorithm to be used if not default (see Subclause 21.2).
			No	
3	Event control	table 6.1.3.1/108 clause 21.3	Yes	Addresses to targets that can be supervised.
			No	

Annex Table a.4.2-3 / JJ-90.30: Negotiation of resource reservation [TS 29.165] Table C.3.1.2

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Negotiation of resource reservation (precondition)	table 6.1.3.1/20	Yes	
			No	*<As specified in Subclause 4.3.4.3 of this Standard, apply "No" over the II-NNI between an IMS network using precondition and an IMS network not using precondition.>*

Annex Table a.4.2-4 / JJ-90.30: Periodic refresh of SIP sessions　[TS 29.165] Table C.3.1.2A

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	SIP session timer (timer)	table 6.1.3.1/45	Yes	Use conditions. (Possible restriction on range of times and whether SIP session timer is applied in all sessions.) *<As specified in Subclause 4.3.4.8 of this Standard, Session timer duration shall be selected from 180 to 300 seconds by each IMS network operator, and the value shall be exchanged between IMS network operators.>* *<SIP session timer shall be applied for all sessions.>*

Annex Table a.4.2-5 / JJ-90.30: Replacing of SIP dialogs　[TS 29.165] Table C.3.1.3

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Replacing of SIP dialogs (replaces)	table 6.1.3.1/47	Yes	
			No	

Annex Table a.4.2-6 / JJ-90.30: Session participation   [TS 29.165] Table C.3.1.4

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Session participation (join)	table 6.1.3.1/48	Yes	
			No	

Annex Table a.4.2-7 / JJ-90.30: Conveying capabilities of UE   [TS 29.165] Table C.3.1.5

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Conveying capabilities of UE	table 6.1.3.1/49	Yes	
			No	

Annex Table a.4.2-8 / JJ-90.30: Authorization of early media   [TS 29.165] Table C.3.1.5A

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Authorization of early media	table 6.1.3.1/69	Yes	Whether to use authorization request of early media generated by the entity outside the IMS network. *<For use of authorization request generated by the entity outside the IMS network is not applied as specified in Subclause 4.3.6.1.1.1 of this Standard.>* *<For use of P-Early-Media header field, follow the signalling requirements specified in Subclause 4.3.6.1.1.1 of this Standard.>*
			No	

Annex Table a.4.2-9 / JJ-90.30: Asserting the service of authenticated users   [TS 29.165] Table C.3.1.6

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Managing the indication of the asserted service (P-Asserted-Service header field)	table 6.1.3.1/77	Yes	Service identifier values to use.
			No	

Annex Table a.4.2-10 / JJ-90.30: Mode of signalling   [TS 29.165] Table C.3.1.7

No.	Option item		References	Applicability at the II-NNI	Details for operator choice
1	Overlap signalling	In-dialog method	clause 6.1.1.5	Yes	
				No	
		Multiple-INVITE method		Yes	
				No	

Annex Table a.4.2-11 / JJ-90.30: SIP message bodies　[TS 29.165] Table C.3.1.7A

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	MIME type	clause 6.1.4	Yes	MIME types to use. *<Follow the signalling requirements for MIME type specified in Subclause 4.3.5.2 of this Standard.>*
				Applicable characteristics of the SIP message body MIMEs (i.e. the value(s) of Content-Disposition header field and Content-Language header field) if necessary. *<Content-Disposition header field and Content-Language header field shall not be included in the SIP message at the II-NNI.>*

Annex Table a.4.2-12 / JJ-90.30: SIP message body size　[TS 29.165] Table C.3.1.7B

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Limitation on maximum length of a SIP message body	clause 6.1.4	Yes	Maximum length accepted. *<For the limitations of the maximum length related to the SIP message, follow the requirements specified in Subclause 4.3.8 of this Standard.>*
			No	

Annex Table a.4.2-13 / JJ-90.30: Control plane transport　[TS 29.165] Table C.3.1.8

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	TCP	clause 6.2	Yes	Use conditions. (e.g. port number to accept, number of simultaneous connections in case of reuse of the existing connections)
			No	
2	UDP	clause 6.2	Yes	Use conditions. (e.g. port number to accept) *<For specifications related to transport protocol for control plane such as the port number to accept, follow the signalling requirements specified in Subclause 4.2 of this Standard.>*
			No	
3	SCTP	clause 6.2	Yes	Use conditions. (e.g. port number to accept)
			No	

Annex Table a.4.2-14 / JJ-90.30: User plane transport, media, and codec [TS 29.165] Table C.3.1.9

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Speech media (m=audio)	clause 7.1	Yes	Names of speech codecs to use. (NOTE 1, NOTE 2) *<For the speech codecs, follow the signalling requirements specified in Subclause 4.3.5.1.4.1 of this Standard.>*
2	Video media (m=video)	clause 7.1	Yes	Names of video codecs to use. (NOTE 1, NOTE 2) *<For the video codecs, follow the signalling requirements specified in Subclause 4.3.5.1.4.2 of this Standard.>*
			No	
3	Other media	clause 7.1	Yes	Media type (m=line of SDP) to use. (e.g. application, image, message)
			No	
4	RTP/AVPF	clause 7.2	Yes	Media type (m=line of SDP) that uses the protocol.
			No	
5	Transmission Control Protocol	clause 6.1.2.1 clause 7.2	Yes	Media type (m=line of SDP) that uses the protocol.
			No	
6	Other user plane protocols	clause 7.2	Yes	Protocols to use (e.g. udptl, TCP/MSRP) and media types (m=line of SDP) that describe the protocols.
			No	
NOTE 1: Codecs which are included in the applicable codec list made by inter-operator agreements are guaranteed to be supported at the II-NNI by the peer operators.				
NOTE 2: Whether it is allowed to offer codecs which are not included in the applicable codec list made by inter-operator agreements over the II-NNI is also determined by the inter-operator agreement if necessary.				

Annex Table a.4.2-15 / JJ-90.30: Dual tone multi frequency (DTMF) [TS 29.165] Table C.3.1.10

No.	Option item		References	Applicability at the II-NNI	Details for operator choice
1	DTMF transport	"telephone-event" based DTMF transport	clause 7.1 table 6.1/5A table 6.1/5B clause 12.14 clause 12.15	Yes	Sampling rate to use. *<For the DTMF transport, follow the signalling requirements specified in Subclause 4.3.5.1.5.>*
				No	
		The SIP INFO mechanism		Yes	
				No	

Annex Table a.4.2-16 / JJ-90.30: Numbering, naming and addressing   [TS 29.165] Table C.3.1.10A

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Subaddress ("isub" tel URI parameter)	clause 8.1	Yes	*<For use of subaddress, follow the signalling requirements specified in Annex b of this Standard.>*
			No	

Annex Table a.4.2-17 / JJ-90.30: IP version   [TS 29.165] Table C.3.1.11

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	IPv4	clause 9	Yes	Use conditions (e.g. for control plane, for user plane). *<For used IP version, follow the requirements specified in Subclause 4.1.1 of this Standard.>*
			No	
2	IPv6	clause 9	Yes	Use conditions (e.g. for control plane, for user plane). *<For used IP version, follow the requirements specified in Subclause 4.1.1 of this Standard.>*
			No	

Annex Table a.4.2-18 / JJ-90.30: Supplementary services　[TS 29.165] Table C.3.1.12

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Malicious Communication IDentification (MCID)	clause 12.2	Yes	Minimum information exchanged over the II-NNI.
			No	
2	Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR)	clause 12.3	Yes	*<For the calling party number presentation and restriction, follow the signalling requirements specified in Subclause 4.3.4.1 of this Standard.>*
			No	
3	Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR)	clause 12.4	Yes	
			No	
4	Anonymous Communication Rejection (ACR)	clause 12.5	Yes	
			No	
5	Communication DIVersion (CDIV)	clause 12.6	Yes	*<For the communication diversion, follow the signalling requirements specified in [JJ-90.27].>*
			No	
6	Communication Waiting (CW)	clause 12.7	Yes	
			No	
7	Communication HOLD (HOLD)	clause 12.8	Yes	
			No	
8	Message Waiting Indication (MWI)	clause 12.9	Yes	
			No	
9	Incoming Communication Barring (ICB)	clause 12.10.1	Yes	
			No	
10	Completion of Communications to Busy Subscriber (CCBS)	clause 12.11	Yes	
			No	
11	Completion of Communications by No Reply (CCNR)	clause 12.12	Yes	
			No	
12	Explicit Communication Transfer (ECT)	clause 12.13	Yes	Type of explicit communication transfer to support.
			No	
13	Customized Alerting Tone (CAT)	clause 12.14	Yes	CAT model to use. (Gateway model, forking model and early session model)
				Media type (m=line of SDP) applicable to CAT.
			No	
14	Customized Ringing Signal (CRS)	clause 12.15	Yes	
			No	
15	Closed User Group (CUG)	clause 12.16	Yes	
			No	
16	Personal Network Management (PNM)	clause 12.17	Yes	
			No	
17	Three-Party (3PTY)	clause 12.18	Yes	
			No	
18	Conference (CONF)	clause 12.19	Yes	
			No	
19	Flexible Alerting (FA)	clause 12.20	Yes	
			No	

No.	Option item		References	Applicability at the II-NNI	Details for operator choice
20	Announcements	During the establishment of a session	clause 12.21.2	Yes	Methods for sending announcement. *<For the methods for sending early media, follow the signalling requirements specified in Subclause 4.3.6.1.1 of this Standard.>*
				No	
		During an established communication session	clause 12.21.3	Yes	Methods for sending announcement. *<For sending announcement, follow the signalling requirements specified in Subclause 4.3.6.1.2 of this Standard.>*
				No	
		Providing announcements when communication request is rejected	clause 12.21.4	Yes	Methods for sending announcement. *<For sending announcement, follow the signalling requirements specified in Subclause 4.3.6.1.1 and Subclause 4.3.6.1.2 of this Standard.>*
				No	
21	Advice of Charge (AOC)		clause 12.22	Yes	
				No	
22	Completion of Communications on Not Logged-in (CCNL)		clause 12.23	Yes	
				No	
23	Presence service		clause 15	Yes	Presence services to use (3GPP TS 24.141, OMA 1.1 or OMA 2.0)
				No	
24	Messaging service	Page-mode messaging	clause 16.2	Yes	
				No	
		Session-mode messaging	clause 16.4	Yes	
				No	
		Session-mode messaging conferences	clause 16.5	Yes	
				No	
24A	Delivery of original destination identity		clause 22	Yes	Services to apply the function.
				No	
24B	Service access number translation		table 6.1.3.1/52A clause 27	Yes	*<For the history of number translation, follow the signalling requirements specified in Subclause 4.3.2.4.2 and Subclause 4.3.4.7 of this Standard.>*
				No	
25	Other additional service using other SIP extensions		clause 6.1.1.3.2 clause 12.6	Yes	The SIP extensions to use and the service that uses the extensions.
				No	
26	Multi-Identity (MiD)			Yes	Types of non-native identities to support.

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
		clause 6.1.1.3.1 table 6.1.3.1/126 clause 12.26.2	No	

Annex Table a.4.2-19 / JJ-90.30: Additional functions   [TS 29.165] Table C.3.1.13

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Optimal Media Routeing	clause 17	Yes	
			No	
2	Applying forking (NOTE 1)	table 6.1.3.1/5	Yes	Usage of Request-Disposition header field with value "no-fork". (NOTE 3)
			No (NOTE 2)	*<Note that forking can be applicable at the II-NNI between mobile IMS networks as described in Subclause 4.3.6.1.1 of this Standard.>*
3	Transfer of IP multimedia service tariff information	clause 11.3	Yes	The value of the Content-Disposition header field.
			No	
4	Telepresence using IMS	clause 23	Yes	The value of "+sip.clue" media feature tag in Contact header field.
			No	
5	Dynamic services interactions	table 6.1.3.1/125	Yes	The identities of the services in the Service-Interact-Info header field.
			No	
6	MTSI Data channel	clause 33	Yes	The value "webrtc-datachannel" of "+sip.app-subtype" media feature tag in Contact header field.
			No	
NOTE 1:	Support of handling of several responses due to forking and of the Request-Disposition header field, is mandated on the II-NNI.			
NOTE 2:	In case the operator interconnects with IMS non-compliant networks, and wishes to use the II-NNI anyway, then the operators might want to negotiate if the forking procedures are applicable.			
NOTE 3:	The Request-Disposition header field with value "no-fork" can be used to suppress that forking occurs. However, a peer operator might require the usage of forking e.g. to implement certain services.			

Annex Table a.4.2-20 / JJ-90.30: SDP lines   [TS 29.165] Table C.3.1.14

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	m=line	clause 6.1.2	Yes	Used static RTP payload type numbers.
2	b=line	clause 6.1.2	Yes	Used bandwidth modifier types. *<"b=" line is able to be omitted if the conditions specified in Subclause 4.3.5.1.3.3 of this Standard are fulfilled.>*
3	a=line	clause 6.1.2	Yes	Used attributes. For the "rtpmap" attribute, used "encoding names".

Annex Table a.4.2-21 / JJ-90.30: Emergency services   [TS 29.165] Table C.3.1.15

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Public Safety Answering Point (PSAP) Callback	table 6.1.3.1/110	Yes	Used PSAP callback indicator.
			No	*<This option item is outside the scope of this Standard. Refer to [JJ-90.28] for emergency calls.>*

Annex Table a.4.2-22 / JJ-90.30: Indicating the II-NNI traversal scenario   [TS 29.165] Table C.3.1.16

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Support of the "iotl" SIP URI parameter indicating the II-NNI traversal scenario	table 6.1.3.1/112 clause 5.3	Yes	
			No	

Annex Table a.4.2-23 / JJ-90.30: Mission critical services   [TS 29.165] Table C.3.1.17

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Support of mission critical services	clause 28.2.2	Yes	MBMS transmission usage and location procedures support.
			No	

Annex Table a.4.2-24 / JJ-90.30: Calling number verification using signature verification and attestation information   [TS 29.165] Table C.3.1.18

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Calling number verification using signature verification and attestation information	table 6.1.3.1/119 table 6.1.3.1/120 table 6.1.3.1/123 table 6.1.3.1/124 clause 29	Yes	
			No	

a.4.3. Option item selection tables specific to roaming II-NNI

The present version of this Standard does not specify any signalling requirements on the roaming II-NNI; thus, the option item selection tables specific to the roaming II-NNI specified in Subclause C.3.2 of [TS 29.165] are also not specified.

a.4.4. Option item selection tables specific to non-roaming II-NNI

This Subclause describes the results with respect to the selection of option items specific to the non-roaming II-NNI specified in Subclause C.3.3 of [TS 29.165].

Annex Table a.4.4-1 / JJ-90.30: SIP method   [TS 29.165] Table C.3.3.1

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	NOTIFY method	table 6.1/10 table 6.1/11	Yes	Event package name to use.
			No	
2	SUBSCRIBE method	table 6.1/20 table 6.1/21	Yes	Event package name to use.
			No	
3	PUBLISH method	table 6.1/15A table 6.1/15B	Yes	Event package name to use.
			No	

Annex Table a.4.4-2 / JJ-90.30: Charging   [TS 29.165] Table C.3.3.2

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Inter-operator accounting	table 6.1.3.1/38 clause 11.2	Yes	Operator network identifiers populated in the type 2 "orig-ioi" and "term-ioi" header field parameters of the P-Charging-Vector header field. *<For use of the P-Charging-Vector header field, follow the signalling requirements specified in Subclause 4.3.4.6 of this Standard.>*
			No	
2	Inter-operator accounting for the transit scenario	clause 11.2	Yes	Operator network identifiers populated in the "transit-ioi" header field parameters of the P-Charging-Vector header field.
			No	

Annex Table a.4.4-3 / JJ-90.30: Globally routable user agent URIs   [TS 29.165] Table C.3.3.3

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Globally Routable User Agent URIs (gruu)	table 6.1.3.1/56	Yes	
			No	

Annex Table a.4.4-4 / JJ-90.30: Media feature tags   [TS 29.165] Table C.3.3.4

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Media feature tags	table 6.1.3.1/76 clause 12 clause 13 clause 14 clause 16 clause 18	Yes	Names of media feature tags to use.
			No	

Annex Table a.4.4-5 / JJ-90.30: ISDN interworking   [TS 29.165] Table C.3.3.5

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	User to User Call Control Information in SIP for ISDN Interworking (uui)	table 6.1.3.1/79	Yes	
			No	

Annex Table a.4.4-6 / JJ-90.30: Corporate network   [TS 29.165] Table C.3.3.6

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Private network traffic (P-Private-Network-Indication header field)	table 6.1.3.1/80	Yes	*<For the private traffic using this SIP header field, follow the signalling requirements specified in [TS-1018].>*
			No	

Annex Table a.4.4-7 / JJ-90.30: Numbering, naming and addressing   [TS 29.165] Table C.3.3.7

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
0	SIP URI	clause 8.1	Yes	Any local numbers that are allowed in addition to global numbers. *<Follow the signalling requirements specified in Subclause 4.3.2.2 of this Standard.>*
				Domain name(s) and/or IP address(es) of hostportion to accept.
				Applicability of PSI to use. *<The PSI shall not be set in the Request-URI in accordance with Subclause 4.3.2 of this Standard.>*
				Applicability of "Unavailable User Identity" in the P-Asserted-Identity header filed. *<The "Unavailable URI" is set to the P-Asserted-Identity header field in accordance with Subclause 4.3.4.1.2 of this Standard.>*
0A	tel URI	clause 8.1	Yes	Any local numbers that are allowed in addition to global numbers. *<Follow the signalling requirements specified in Subclause 4.3.4.1 of this Standard.>*
				Applicability of the URI in P-Asserted-Identity header field and/or Request-URI. *<The tel URI shall not be set in the Request-URI in accordance with Subclause 4.3.2 of this Standard.> <The tel URI shall apply to the P-Asserted-Identity in accordance with Subclause 4.3.4.1 of this Standard.>*
			No	
0B	IM URI	clause 8.1	Yes	
			No	
0C	PRES URI	clause 8.1	Yes	
			No	
1	Number Portability Routing Number ("rn" and "npdi" tel URI parameter)	clause 8.1	Yes	*<The parameters shall be applied in accordance with Subclause 4.3.2 of this Standard.>*
			No	
2	Calling Party's Category ("cpc" tel URI parameter)	clause 8.1	Yes	cpc-values to use. *<For use of calling party's category, follow the signalling requirements specified in Subclause 4.3.4.1.3 of this Standard.>*
			No	
3	Originating Line Information ("oli" tel URI parameter)	clause 8.1	Yes	oli-values to use.
			No	

Annex Table a.4.4-8 / JJ-90.30: Additional functions   [TS 29.165] Table C.3.3.8

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Support of out-of-dialog OPTIONS method	table 6.1/12 table 6.1/13	Yes	The purpose of the method. *<In accordance with Subclause 4.3.1 and Annex d of this Standard, out-of-dialog OPTIONS method is applicable only for fault detection and restoration detection.>*
			No	

Annex Table a.4.4-9 / JJ-90.30: IMS emergency session traversal scenario   [TS 29.165] Table C.3.3.9

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	IMS emergency session traversal scenario	table 6.1.3.1/55 clause 30.1 clause 30.2	Yes	The emergency service URN(s) to use.
				The PSAP URI to use in the Route header field, if applicable.
			No	*<This option item is outside the scope of this Standard. Refer to [JJ-90.28] for emergency calls.>*
2	Next-Generation Pan-European eCall emergency service	clause 30.3	Yes	
			No	

Annex b (Normative): Subaddress

b.1. General

The services equivalent to those provided by the transfer of subaddress information available in the ISUP network through the interconnection interface defined in [JJ-90.10] may be provided for served users in the IMS network. Application of subaddress information at the II-NNI is determined in the inter-operator agreements in accordance with Subclause 8.1 of [TS 29.165]. {Appendix Table i.4-5 Item 4}

When the originating IMS operator selects "Yes" to the "Subaddress" option item, the originating IMS operator can set calling party subaddress and/or called party subaddress. When the terminating IMS operator selects "No" to the "Subaddress" option item, the terminating IMS operator may ignore the calling party subaddress information and called party subaddress information in the received SIP message.

This Annex specifies the signalling requirements of subaddress information in SIP messages as a complementary of [TS-1008] to properly deliver subaddress information at the II-NNI. If subaddress information is delivered at the II-NNI between the IMS networks, the signalling requirements in this Annex shall be followed. Except for the signalling requirements described in this Annex, the IMS networks shall follow the signalling requirements for NNI defined in [TS-1008].

b.2. Content of subaddress information

The content of subaddress information is 19 decimal digits or less strings using numbers 0 to 9. The other details are based on [RFC 4715] and [TS-1008].

b.3. Formats of subaddress information

Subaddress information can be included in any SIP requests/responses. The subaddress information can be populated into the tel URI or "telephone-subscriber" part of SIP URI which includes user parameter set to "phone" of the SIP header fields and/or Request-URI. Subaddress is represented as a tel URI parameter (which is described after the semicolon (;) appended after global-number-digits) and the subaddress information is populated as decimal digits strings appended after "isub=".

As described in the following Subclauses, the IMS operator shall treat the "isub" tel URI parameter as calling party subaddress/called party subaddress only when the subaddress information is attached to the URI representing the calling party number/called party number, since subaddress information is the accompaniment information with calling party number or called party number.

The "isub-encoding" tel URI parameter shall not be populated into the tel URI.

b.3.1. Calling party subaddress

Calling party subaddress is treated as the accompaniment information with calling party number. The subaddress information in the tel URI of the P-Asserted-Identity header field is treated as "calling party subaddress" only when the information is obtained from the "TEL_URI" information element of Table 4.3.4.1.2-1 in accordance with Subclause 4.3.4.1.2,

    NOTE: Subaddress information can not be set into "TEL_DISPLAYNAME".

b.3.2. Called party subaddress

Called party subaddress is treated as the accompaniment information with called party number. The subaddress information set into the Request-URI of the SIP request outside an existing dialog is treated as "called party subaddress" since the called party number is set into the Request-URI as described in Subclause 4.3.2.

Annex c (Normative):
Bandwidth control

c.1. General

This Annex specifies the signalling procedure for bandwidth control functions and its relationship with the transport layer protocol by referring to [JT-Y1221].

The following descriptions make the assumption that the bandwidth is controlled by Resource and Admission Control Functions (RACF) shown in [TR-1014]; however, the implementation inside the network may be realized by other mechanisms. Note that the bandwidth control functions shall conform to this Annex if the function is applied for calls over the II-NNI; therefore, the bandwidth requested by this function shall be guaranteed inside the network, even if a mechanism other than RACF is used.

c.2. Bandwidth control mechanism in IMS network

The bandwidth control mechanism in IMS network shall conform to Annex a of [JT-Y1221]. Supplementary specifications and option items when applying Annex a of [JT-Y1221] at the II-NNI are as follows.

- When the token bucket size is configured without applying the proportional relationship specified in Subclause a.2.3 of [JT-Y1221], the configured value is determined in the inter-operator agreements. {Appendix Table i.4-17 Item 1}

- With regard to the values of rate factors at the II-NNI, QoS class α defined in [JT-Y1221] conforms to Subclause a.2.5.2 of [JT-Y1221]. The values applied for the other QoS classes are determined in the inter-operator agreements. {Appendix Table i.4-17 Item 2}

c.3. SIP/SDP signalling requirements

SIP/SDP signalling requirements shall conform to Annex a of [JT-Y1221]. Supplementary specifications and option items to apply Annex a of [JT-Y1221] at the II-NNI are as follows.

- In accordance with Subclause a.2.2 of [JT-Y1221], the applied token bucket rate is the value specified in "b=" line of the SDP. Only for speech media, it is possible to apply individual designation of the token bucket rate for particular codec(s) instead of the rate indicated in "b=" line sent from user equipment. {Appendix Table i.4-17 Item 3}

- When both "b=RR" line and "b=RS" line are not specified, the RTCP bandwidth shall be less than 5% of the RTP bandwidth.

Annex d (Normative):
Restoration detection using OPTIONS method

d.1. General

In this Standard, the OPTIONS method is applicable only for fault detection and restoration detection of an opposing IBCF. According to Subclause iii.5 of this Standard, the IMS operator can select to send an OPTIONS request over the II-NNI as a method to detect fault/restoration at SIP layer.

The OPTIONS method can be used for detecting restoration of an opposing IBCF after detecting fault at the IBCF by expiration of initial INVITE transaction timeout timer (Timer B). Also, the OPTIONS method can be used for restoration detection after the peer's failure detection by receiving a 503 (Service Unavailable) response, if the terminating IMS network has the ability to send 200 (OK) response to the OPTIONS request only when the terminating IMS network is able to process the initial INVITE request from the originating IMS network to which the terminating IMS network has sent the 503 (Service Unavailable) response. {Appendix Table i.4-20 Item 3}

The IMS operator who applies restoration detection using OPTIONS method shall determine the interval time to sending OPTIONS requests within the range from 10 seconds to 600 seconds and shall share the determined interval time value between the IMS operators. {Appendix Table i.4-20 Item 2}

NOTE: An originating IMS network shall recognize a terminating IMS network is restored based on the 200 (OK) response to the OPTIONS request from a terminating IMS network.

In accordance with Subclause 4.3.1 of this Standard, the IMS network shall be able to send a 200 (OK) response to the OPTIONS request regardless whether the IMS operator applies restoration detection using OPTIONS method or not, in order to notify the restoration of the IBCF which has been detected a fault.

d.2. Setting of OPTIONS request/response

The URI format for the Request-URI of the OPTIONS request is SIP URI. The SIP URI includes the IP address and the port number of the opposing IBCF where a fault has been detected. The port number can be omitted if the port number is 5060 and userinfo part is not set in the SIP URI.

The SIP header fields shall be set in the OPTIONS request and the response to the OPTIONS request in accordance with Annex Table d.2-1 and SIP message body shall not be set in the OPTIONS request and the response to the OPTIONS request.

Annex Table d.2-1 / JJ-90.30: SIP header field setting for OPTIONS request/response

SIP header field	Setting conditions at the II-NNI (NOTE 1)	
	Request	**Response**
Accept	Optional (NOTE 2)	Optional (NOTE 2)
Allow	Optional (NOTE 2)	Optional (NOTE 2)
Call-ID	Mandatory	Mandatory
Contact	Mandatory (NOTE 3)	Optional (NOTE 2)
Content-Length	Mandatory, set to 0	Mandatory, set to 0
CSeq	Mandatory	Mandatory
From	Mandatory (NOTE 4)	Mandatory
Max-Forwards	Mandatory	Not allowed
P-Charging-Vector	Optional (NOTE 5)	Optional (NOTE 2)
Require	Not allowed	Not allowed
Supported	Not allowed	Optional (NOTE 2)
To	Mandatory (NOTE 6)	Mandatory
Via	Mandatory (NOTE 7)	Mandatory
Other header fields	Not allowed	Not allowed

NOTE 1: The detailed setting conditions are not described in this Subclause are set in accordance with normative part of this Standard.

NOTE 2: The IMS network ignores the contents of the received SIP header field if there is no inter-operator agreements.

NOTE 3: The IP address and the port number of the IBCF in the opposing IMS network are set in the SIP URI. The port number can be omitted if the port number is 5060 and userinfo part can be set in the SIP URI.

NOTE 4: The IP address of the IBCF in the opposing IMS network is set in the SIP URI. The port number and userinfo part is not set in the SIP URI.

NOTE 5: If there is no inter-operator agreements, the IMS network received the SIP header field ignores the contents of the header field or set the SIP header field in the response to the request using the received information.

NOTE 6: Same IP address of the Request-URI is set in the SIP URI. The port number and userinfo part is not set in the SIP URI.

NOTE 7: Only one line is set.

Appendix i (Informative): List of option items

i.1. Introduction

For improving the inter-connectivity between IMS networks within Japan, this Appendix provides the list of option items that are recommended to be selected by inter-operator agreements in this Standard.

If there remains item(s) that should be determined for an option item by the inter-operator agreements in "Details for operator choice" in Annex a of this Standard, then the option item is also listed here, even if "Applicability at the II-NNI" of the option item is already selected.

Note that any interaction among the option items is not described in the list of option items, as with [TS 29.165]. Note also that if there are misalignments between [TS 29.165] and this Appendix, or misalignments between the normative part of this Standard and this Appendix, [TS 29.165] and the normative part of this Standard take precedence.

i.2. Format of option item table and definition

The example format and definition of option item table is shown in Appendix Table i.2-1 and Appendix Table i.2-2, respectively.

Appendix Table i.2-1 / JJ-90.30: Example format

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	-	-	Yes	
			No	

Appendix Table i.2-2 / JJ-90.30: Definitions relating to option item table

column	Title of option item table		Definition	
2	Option item		This column shows the option item.	
3	References		This column shows the relevant Clause(s) and/or table number(s) in the main body of [TS 29.165] and/or this Standard.	
4	Applicability at the II-NNI	Hatching "Yes"	This type of item shows that there are the details that are recommended to be selected in "Details for operator choice" although "Yes" is selected in the main body and/or Appendix of this Standard.	
		No hatching	This type of item is option item.	
			In case "Yes" is selected in the inter-operator agreements.	This means the capability(ies) and/or service(s) relating to the option item are provided over the II-NNI. In this case, each side IMS operator's network of II-NNI support the capability(ies) in accordance with this Standard and normative reference(s).
			In case "No" is selected in the inter-operator agreements.	This means the capability(ies) and/or service(s) relating to the option item are not provided over the II-NNI. In this case, each side IMS operator's network of II-NNI will not guarantee the provision of the capability(ies) in accordance with this Standard and normative reference(s).
5	Details for operator choice		This column shows the details for the inter-operator agreements if the corresponding pattern is chosen in the "Applicability at the II-NNI". The clarifications in this Standard are described between angled brackets (< >). This column intends to be used to exchange each detailed conditions of the II-NNI by means of filling the conditions in the columns corresponding to the selected pattern. Blank row intends to be used as a place holder for describing additional detailed conditions specific to the operator network not associated with the explicit descriptions in this Appendix.	

i.3. Supported II-NNI traversal scenarios

There are no option items relating to supported II-NNI traversal scenarios.

i.4. Option item tables specific to non-roaming II-NNI

This Subclause describes the option item tables for the non-roaming II-NNI.

NOTE 1: The item related to Annex a is extracted from Subclause a.4.2 and Subclause a.4.4.

NOTE 2: Appendix Table i.4-2, Appendix Table i.4-3 item 5, Appendix Table i.4-5 item 2 and item 3, Appendix Table i.4-8, Appendix Table i.4-9, Appendix Table i.4-14, Appendix Table i.4-15 item 1 and item 2, Appendix Table i.4-16 are voided in this version.

Appendix Table i.4-1 / JJ-90.30: IP version

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	IPv6	[TS 29.165] Clause 9 [JJ-90.30] Subclause 4.1.1 Table a.4.2-17/2	Yes	Use conditions (e.g. for control plane, for user plane). *<For used IP version, follow the requirements specified in Subclause 4.1.1 of this Standard.>*
			No	

Appendix Table i.4-3 / JJ-90.30: SIP methods

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	MESSAGE method	[TS 29.165] Table 6.1/9A Table 6.1/9B [JJ-90.30] Subclause 4.3.1 Table a.4.2-1/2	Yes	Inside or outside existing dialog, and content of MESSAGE request.
			No	
2	REFER method	[TS 29.165] Table 6.1/16 Table 6.1/17 [JJ-90.30] Subclause 4.3.1 Table a.4.2-1/3	Yes	Inside or outside existing dialog.
			No	
3	NOTIFY method	[TS 29.165] Table 6.1/10 Table 6.1/11 [JJ-90.30] Subclause 4.3.1 Table a.4.4-1/1	Yes	Event package name to use.
			No	
4	SUBSCRIBE method	[TS 29.165] Table 6.1/20 Table 6.1/21 [JJ-90.30] Subclause 4.3.1 Table a.4.4-1/2	Yes	Event package name to use.
			No	

Appendix Table i.4-4 / JJ-90.30: Carrier ENUM interface

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Carrier ENUM interface	[JJ-90.30] Subclause 4.3.2 [JJ-90.31]	Yes	*<Determine IP address(es) of ENUM query to accept. (Refer to Item 1 in [JJ-90.31].)>*
				<Determine the setting values of the ORDER/PREFERENCE field in NAPTR resource records. (Refer to Item 3 and Item 4 in [JJ-90.31].)>
2	Attempt to connect to the donor carrier's IMS network when receiving no NAPTR resource record	[JJ-90.30] Subclause 4.3.2	Yes	Domain name(s) and/or IP address(es) of donor carrier's IMS network to accept.
			No	

Appendix Table i.4-5 / JJ-90.30: Numbering, naming and addressing

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	SIP URI	[TS 29.165] Subclause 8.1 [JJ-90.30] Subclause 4.3.2 Table a.4.4-7/0	Yes	Domain name(s) and/or IP address(es) of hostportion to accept.
4	Subaddress ("isub" tel URI parameter)	[TS 29.165] Subclause 8.1 [JJ-90.30] Table a.4.2-16/1 Annex b	Yes	*<For use of subaddress, follow the signalling requirements specified in Annex b.>*
			No	

Appendix Table i.4-6 / JJ-90.30: Method for selection of the terminating IBCF

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Provision of the inter-operator DNS interface for SIP domain name resolution	[JJ-90.30] Subclause 4.3.3	Yes	*<Determine IP address(es) of DNS query to accept. (Refer to Item 1 in [JJ-90.32].)>*
				<Determine a method for selection of IP address when receiving multiple A/AAAA records. (Refer to Item 2 in [JJ-90.32].)>
				<Determine the TTL values applied for NAPTR resource records, SRV resource records and A/AAAA records respectively. (Refer to Item 4, Item 5, Item 7 and Item 8 in [JJ-90.32].)>
				<Determine the maximum number of SRV resource records. (Refer to Item 6 in [JJ-90.32].)>
				<Determine the condition whether continue processing a call or reject a call when an originating IMS network can not obtain any normal DNS answer from DNS servers.>
			No	*<Determine IP address(es) of SIP to accept.>*
				<Determine a method for selection of the terminating IBCF when there are multiple IP addresses of SIP to accept.>

Appendix Table i.4-7 / JJ-90.30: SIP option tags

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	SIP session timer (timer)	[TS 29.165] Table 6.1.3.1/45 [JJ-90.30] Subclause 4.3.4.8 Table a.4.2-4/1	Yes	Use conditions. (Possible restriction on range of times and whether SIP session timer is applied in all sessions.) *<As specified in Subclause 4.3.4.8 of this Standard, Session timer duration shall be selected from 180 to 300 seconds* by each IMS network operator*, and the value shall be exchanged between IMS network operators.>* *<SIP session timer shall be applied for all sessions.>*
2	Reliability of provisional response (100rel)	[JJ-90.30] Subclause 4.3.4.2	Yes	Use conditions. (Possible restriction on whether 100rel is applied in all sessions.)
3	Negotiation of resource reservation (precondition)	[TS 29.165] Table 6.1.3.1/20 [JJ-90.30] Subclause 4.3.4.3 Table a.4.2-3/1	Yes	
			No	*<As specified in Subclause 4.3.4.3 of this Standard, apply "No" over the II-NNI between an IMS network using precondition and an IMS network not using precondition.>*
4	Replacing of SIP dialogs (replaces)	[TS 29.165] Table 6.1.3.1/47 [JJ-90.30] Table a.4.2-5/1	Yes	
			No	
5	Conveying capabilities of UE (pref)	[TS 29.165] Table 6.1.3.1/49 [JJ-90.30] Table a.4.2-7/1	Yes	
			No	

Appendix Table i.4-10 / JJ-90.30: SDP lines

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	m=line	[TS 29.165] Subclause 6.1.2 [JJ-90.30] Subclause 4.3.5.1.3.1 Table a.4.2-20/1	Yes	Used static RTP payload type numbers.
2	b=line	[TS 29.165] Subclause 6.1.2 [JJ-90.30] Subclause 4.3.5.1.1.5 Subclause 4.3.5.1.3.3 Table a.4.2-20/2	Yes	Used bandwidth modifier types. *<"b=" line is able to be omitted if the conditions described in Subclause 4.3.5.1.3.3 of this Standard are fulfilled.>*
3	Negotiation of RTCP bandwidth by using b=RR/b=RS	[JJ-90.30] Subclause 4.3.5.1.1.5 Subclause 4.3.5.1.3.3	Yes / No	
4	a=line	[TS 29.165] Subclause 6.1.2 [JJ-90.30] Subclause 4.3.5.1.1.6 Subclause 4.3.5.1.3.4 Table a.4.2-20/3	Yes	Used attributes. For the "rtpmap" attribute, used "encoding names".

Appendix Table i.4-11 / JJ-90.30: User plane transport, media, and codec

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Speech media (m=audio)	[TS 29.165] Subclause 7.1 [JJ-90.30] Subclause 4.3.5.1.3.1 Subclause 4.3.5.1.4.1 Table a.4.2-14/1	Yes	Names of speech codecs to use. (NOTE 1, NOTE 2) *<For the speech codecs, follow the signalling requirements specified in Subclause 4.3.5.1.4.1 of this Standard.>*
2	Video media (m=video)	[TS 29.165] Subclause 7.1 [JJ-90.30] Subclause 4.3.5.1.3.1 Subclause 4.3.5.1.4.2 Table a.4.2-14/2	Yes	Names of video codecs to use. (NOTE 1, NOTE 2) *<For the video codecs, follow the signalling requirements specified in Subclause 4.3.5.1.4.2 of this Standard.>*
			No	
3	Other media	[TS 29.165] Subclause 7.1 [JJ-90.30] Subclause 4.3.5.1.3.1 Table a.4.2-14/3	Yes	Media type (m=line of SDP) to use. (e.g. application, image, message)
			No	
4	RTP/AVPF	[TS 29.165] Subclause 7.2 [JJ-90.30] Subclause 4.3.5.1.3.1 Table a.4.2-14/4	Yes	Media type (m=line of SDP) that uses the protocol.
			No	
5	TCP	[TS 29.165] Subclause 6.1.2.1 Subclause 7.2 [JJ-90.30] Subclause 4.3.5.1.3.1 Table a.4.2-14/5	Yes	Media type (m=line of SDP) that uses the protocol.
			No	
6	Other user plane protocols	[TS 29.165] Subclause 7.2 [JJ-90.30] Subclause 4.3.5.1.3.1 Table a.4.2-14/6	Yes	Protocols to use (e.g. udptl, TCP/MSRP) and media types (m=line of SDP) that describe the protocols.
			No	

NOTE 1: Codecs which are included in the applicable codec list made by inter-operator agreements are guaranteed to be supported at the II-NNI by the peer operators.
NOTE 2: Whether it is allowed to offer codecs which are not included in the applicable codec list made by inter-operator agreements over the II-NNI is also determined by the inter-operator agreement if necessary.

Appendix Table i.4-12 / JJ-90.30: Media change

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Codec change of an established speech media (m=audio)	[JJ-90.30] Subclause 4.3.5.1.6.1	Yes	*<Change pattern of speech codec.>*
			No	
2	Codec change of an established video media (m=video)	[JJ-90.30] Subclause 4.3.5.1.6.1	Yes	*<Change pattern of video codec.>*
			No	

Appendix Table i.4-13 / JJ-90.30: SIP message bodies

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	MIME type	[TS 29.165] Subclause 6.1.4 [JJ-90.30] Subclause 4.3.5.2 Table a.4.2-11/1	Yes	MIME types to use other than SDP (application/sdp).

Appendix Table i.4-15 / JJ-90.30: Supplementary services

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
3	Communication HOLD (HOLD)	[TS 29.165] Subclause 12.8 [JJ-90.30] Table a.4.2-18/7	Yes	
			No	
4	Private network traffic (P-Private-Network-Indication header field)	[TS 29.165] Table 6.1.3.1/80 [JJ-90.30] Table a.4.4-6/1	Yes	*<For the private traffic using this SIP header field, follow the signalling requirements specified in [TS-1018].>*
			No	

Appendix Table i.4-17 / JJ-90.30: Bandwidth control

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Individual designation of token bucket size	[JJ-90.30] Subclause c.2	Yes	Upper and lower limits.
			No	
2	Rate factor	[JJ-90.30] Subclause c.2	Define the values of rate factors for each QoS class	Used values of the rate factors.
			Define a specific value of rate factor for all QoS classes	Used value of the rate factor.
3	Individual designation of token bucket rate for particular codec(s)	[JJ-90.30] Subclause 4.3.5.1.3.3 Subclause c.3	Yes	Use conditions for each codec.
			No	

Appendix Table i.4-18 / JJ-90.30: Maximum number of sessions

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Limitation of the maximum number of outgoing simultaneous SIP sessions	[JJ-90.30] Subclause iii.1	Yes	Use conditions (the maximum number of outgoing SIP sessions at each operator's network, the method for reserving bandwidth (i.e., non-shared reservation method and shared reservation method), and the shared bandwidth if the operators apply the shared reservation method).

Appendix Table i.4-19 / JJ-90.30: Monitoring for extinction of RTP/RTCP packets

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Monitoring for extinction of RTP/RTCP packets	[JJ-90.30] Subclause iii.4	Yes	Monitoring condition(s) is exchanged between two operators.
			No	

Appendix Table i.4-20 / JJ-90.30: Fault detection and restoration detection

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	Detouring when detecting fault	[JJ-90.30] Subclause iii.5.2	Yes	Determine the conditions for detouring when detecting fault
			No	
2	Method for restoration detection for SIP layer	[JJ-90.30] Subclause d.1 Subclause iii.5.3	Yes	Determine the method applied for restoration detection (select both of following methods or one of them.) - Method using the OPTIONS method - Method using the pilot INVITE request *<Determine the interval time to sending OPTIONS requests within the range from 10 seconds to 600 seconds and share the determined interval time value between the IMS operators, in accordance with Subclause d.1 of this Standard.>* *<Determine the timer value for the method using the pilot INVITE request within the range from 30 seconds to 900 seconds and share the timer value between IMS operators, in accordance with Subclause iii.5.3 of this Standard.>*
			No	
3	The function to send 200 (OK) response to the received OPTIONS request for restoration detection, only when the IBCF is able handle the received initial INVITE request from the peer, after having sent the 503 (Service Unavailable) response to the initial INVITE request	[JJ-90.30] Subclause d.1	Yes	
			No	

Appendix Table i.4-21 / JJ-90.30: Inter-operator accounting

No.	Option item	References	Applicability at the II-NNI	Details for operator choice
1	IOI value	[JJ-90.30] Subclause 4.3.4.6.2.2	Yes	IOI value (s) is exchanged between two operators. (Including whether or not additional information is omitted)

Appendix ii (Informative):
SIP-ISUP interworking

ii.1. General

This Appendix describes the interworking of the information related to the calling party number and number portability between SIP and ISUP.

ii.2. SIP-ISUP interworking of calling party number information

Interworking between an IMS network and TTC ISUP network related to network asserted user identity information and presentation/restriction information are described in the subsequent Subclauses.

ii.2.1. General

The terms used in this Appendix are aligned with the definition in the main body of this specification.

ii.2.2. Applicable model

An applicable model is shown in Appendix Figure ii.2.2-1. The interconnection interface between IMS and TTC ISUP networks conforms to TTC ISUP. When the interconnected IMS and TTC ISUP networks are managed by two different operators, the interconnection interface conforms to [JJ-90.10]. It is assumed that a trust relationship between interconnected networks exists.

The SIP-ISUP interworking of network asserted user identity information at the MGCF in this model is described in the subsequent Subclauses.
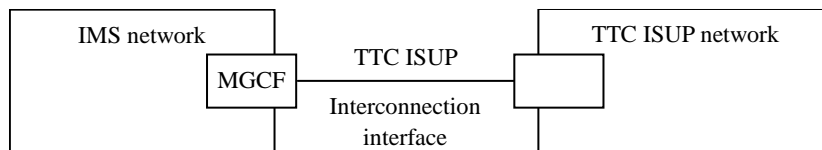
Appendix Figure ii.2.2-1 / JJ-90.30: Interconnection model between IMS and TTC ISUP networks

ii.2.3. ISUP and SIP messages

This Appendix describes the mapping between the ISUP initial address message (IAM) and SIP initial INVITE request.

The mapping of the ISUP initial address message (IAM) to the SIP initial INVITE request is described in Subclause ii.2.4, and the mapping of the SIP initial INVITE request to the ISUP initial address message (IAM) is described in Subclause ii.2.5.

ii.2.4. Mapping of IAM to initial INVITE request

ii.2.4.1. Presentation/restriction information

If both the valid parameters of the generic number (see Subclause ii.2.4.2) and the valid parameters of the calling party number (see Subclause ii.2.4.2) are present in the ISUP initial address message (IAM), then the address presentation restriction indicator of the generic number is interworked as follows:

- If the received address presentation restriction indicator contains "presentation allowed", then the presentation/restriction information is "presentation".

- If the received address presentation restriction indicator contains the value other than "presentation allowed", then the presentation/restriction information is "restriction".

NOTE: The case in which the valid generic number parameters are present but the valid calling party number parameters are not does not normally occur. In this case, the setting of the presentation/restriction information is based on operator policy.

If the valid parameters of the generic number are not present but the valid parameters of the calling party number are in the ISUP initial address message (IAM), then the address presentation restriction indicator of the calling party number is interworked as follows:

- If the received address presentation restriction indicator contains "presentation allowed", then the presentation/restriction information is "presentation".

- If the received address presentation restriction indicator contains the value other than "presentation allowed", then the presentation/restriction information is "restriction".

If neither the valid parameters of the generic number nor the valid parameters of the calling party number are present, the presentation/restriction information is "restriction".

ii.2.4.2. Network asserted user identity information

The definitions of the valid generic number parameters, valid calling party number parameters, and main number are described as follows:

1) Valid generic number parameters

   - The values listed in Appendix Table ii2.4.2-1 constitute the conditions for valid generic number parameters, which provide the elements for network asserted user identity information.

Appendix Table ii.2.4.2-1 / JJ-90.30: Conditions for valid generic number parameters

Parameter field	Value	Meaning
Number qualifier indicator	00000110	additional calling party number
Nature of address indicator	0000011	national (significant) number
Number incomplete indicator	0	number complete
Numbering plan indicator	001	ISDN (telephony) numbering plan (ITU-T Recommendation E.164)
Address presentation restriction indicator	00 or 01	presentation allowed or presentation restricted
Screening indicator	01 or 11	user provided, verified and passed or network provided
Address signal	Max 16 digits	

2) Valid calling party number parameters

   - The values listed in Appendix Table ii.2.4.2-2 constitute the conditions for valid calling party number parameters, which provide the elements for network asserted user identity information.

Appendix Table ii.2.4.2-2 / JJ-90.30: Conditions for valid calling party number parameters

Parameter field	Value	Meaning
Nature of address indicator	0000011 0000100 1111110	national (significant) number (national use) international number network specific number
Number incomplete indicator	0	complete
Numbering plan indicator	001	ISDN (Telephony) numbering plan (ITU-T Recommendation E.164)
Address presentation restriction indicator	00 or 01	presentation allowed or presentation restricted
Screening indicator	01 or 11	user provided, verified and passed or network provided
Address signal	Max 16 digits	

3) Main number

   - The main number is determined as follows:

a) If the valid generic number parameters are present, then the main number is obtained from part of the generic number parameters (nature of address indicator and address signal).

b) If the valid generic number parameters are not present but the valid calling party number parameters are, then the main number is obtained from some of the calling party number parameters (nature of address indicator and address signal).

c) If neither the valid generic number parameters nor the valid calling party number parameters are present, the main number is null.

NOTE: The case in which the valid generic number parameters are present but the valid calling party number parameters are not does not normally occur. In this case the setting of the main number is based on operator policy.

ii.2.4.3. Mapping to information components

ii.2.4.3.1. SIP_URI

If the presentation/restriction information is "presentation", then the SIP_URI can be omitted. If the presentation/restriction information is "restriction", then the setting of the SIP_URI is mandated. When the SIP_URI is populated, the telephone-subscriber part of the SIP URI is derived from the nature of address indicator and address signal in the main number applying the mapping rule in Appendix Table ii.2.4.3.1-1. The hostport part of the SIP URI is populated based on operator policy with the "user" SIP URI parameter that equals "phone".

If the main number is null, then the SIP_URI is set to anonymous URI or unavailable URI.

Appendix Table ii.2.4.3.1-1 / JJ-90.30: Mapping of ISUP nature of address indicator to SIP_URI

Usage	Nature of address indicator	Address signal	SIP_URI
Originating call on international network (oversea)	international number	[Country code] [National number]	sip:+[Country code][National number]@example.ne.jp;user=phone
Originating call on mobile/PHS network	national (significant) number	A0CDEFGHJK	sip:+81A0CDEFGHJK@example.ne.jp;user=phone
Originating call on local fixed telephone network	national (significant) number	ABCDEFGHJ	sip:+81ABCDEFGHJ@example.ne.jp;user=phone
Operator-originating call	network specific number	operator specific digits	sip:operator specific digits;phone-context=+81@example.ne.jp;user=phone

ii.2.4.3.2. SIP_DISPLAYNAME

If the presentation/restriction information is "presentation", then the SIP_DISPLAYNAME is either omitted or the value equivalent to the TEL_DISPLAYNAME. If the presentation/restriction information is "restriction", then the SIP_DISPLAYNAME is derived from the cause of no ID parameter applying the mapping rule in Appendix Table ii.2.4.3.2-1. The value of the SIP_DISPLAYNAME is case sensitive.

NOTE: Double quotation marks (DQUOTE) can be added in front and behind of SIP_DISPLAYNAME.

Appendix Table ii.2.4.3.2-1 / JJ-90.30: Mapping of ISUP cause of no ID parameter to SIP_DISPLAYNAME

Value	Meaning	SIP_DISPLAYNAME
No parameter	-	Unavailable
0000001	No caller ID: rejected by user	Anonymous
0000010	No caller ID: service conflict	Interaction with other service
0000011	No caller ID: call from payphone	Coin line/payphone

ii.2.4.3.3. TEL_URI

If the valid parameters of the calling party number are present, then the TEL_URI is derived from the calling party number parameters applying the mapping rule in Appendix Table ii.2.4.3.3-1. If the valid parameters of the calling party number are not present, then the tel URI is omitted. The mapping of the ISUP nature of address indicator and address signal to the TEL_URI and the tel URI is given in Appendix Table ii.2.4.3.3-1.

Appendix table ii.2.4.3.3-1 / JJ-90.30: Mapping of ISUP nature of address indicator and address signal to TEL_URI and tel URI

Usage	Nature of address indicator	Address signal	TEL_URI	tel URI
Originating call on international network (overseas)	international number	[Country code] [National number]	+[Country code] [National number]	tel:+[Country code] [National number] ;verstat=No-TN-Validation
		81A0CDEFGHJK	+81A0CDEFGHJK	tel:+81A0CDEFGHJK;ve rstat=No-TN-Validation
		81ABCDEFGHJ	+81ABCDEFGHJ	tel:+81ABCDEFGHJ;ver stat=No-TN-Validation
		operator specific digits	operator specific digits	tel: operator specific digits;verstat=No-TN-Validation
Originating call on mobile/PHS network	national (significant) number	A0CDEFGHJK	+81A0CDEFGHJK	tel:+81A0CDEFGHJK
Originating call on local fixed telephone network	national (significant) number	ABCDEFGHJ	+81ABCDEFGHJ	tel:+81ABCDEFGHJ
Operator-originating call	network specific number	operator specific digits	operator specific digits	tel:operator specific digits;phone-context=+81

ii.2.4.3.4. TEL_DISPLAYNAME

If the presentation/restriction information is "presentation", then the TEL_DISPLAYNAME is derived from address signal of the main number applying the mapping rule in Appendix Table ii.2.4.3.4-1. If the presentation/restriction information is "restriction", then the TEL_DISPLAYNAME is either omitted or derived from address signal of the main number applying the mapping rule in Appendix Table ii.2.4.3.4-1.

The mapping of the ISUP nature of the address indicator and the address signal to the TEL_DISPLAYNAME based on the ordinary dialing plan is given in Appendix Table ii.2.4.3.4-1.

Appendix Table ii.2.4.3.4-1 / JJ-90.30: Mapping of ISUP nature of address indicator and address signal to TEL_DISPLAYNAME

Usage	Nature of address indicator	Address signal	TEL_DISPLAYNAME
Originating call on mobile/PHS network	national (significant) number	A0CDEFGHJK	0A0CDEFGHJK
Originating call on local fixed telephone network	national (significant) number	ABCDEFGHJ	0ABCDEFGHJ
Logical number	national (significant) number	AB0~	0AB0~
Operator-originating call	network specific number	operator specific digits	operator specific digits

ii.2.4.3.5. PRIVACY

If the presentation/restriction information is "presentation", then the PRIVACY is either "none" or omitted. If the presentation/restriction information is "restriction", then the PRIVACY is "id".

The mapping of presentation/restriction information to PRIVACY is given in Appendix Table ii.2.4.3.5-1.

Appendix Table ii.2.4.3.5-1 / JJ-90.30: Mapping of presentation/restriction information to PRIVACY

Presentation/restriction information	PRIVACY
Presentation	"none" or omit the Privacy header field
Restriction	"id"

ii.2.4.4.   ISUP to SIP interworking conditions table

The ISUP IAM to the SIP initial INVITE request interworking is summarized in Appendix Table ii.2.4.4-1.

Appendix Table ii.2.4.4-1 / JJ-90.30: ISUP IAM to SIP initial INVITE request interworking condition

IAM					initial INVITE request				
Generic number		**Calling party number**		**cause of no ID**		**SIP URI**		**tel URI**	
Has the valid parameters been received?	**Address presentation restriction indicator**	**Has the valid parameters been received?**	**Address presentation restriction indicator**	**Has the parameter been received?**	**PRIVACY**	**SIP_URI**	**SIP_DISPLAYNAME**	**TEL_URI**	**TEL_DISPLAYNAME**
Yes	Presentation allowed	Yes	Presentation allowed / Other value than presentation allowed	Yes/No	Presentation (Appendix Table ii.2.4.3.5-1)	Derived from generic number or omitted (Appendix Table ii.2.4.3.1-1)	Derived from generic number or omitted (Appendix Table ii.2.4.3.2-1)	Derived from calling party number (Appendix Table ii.2.4.3.3-1)	Derived from generic number (Appendix Table ii.2.4.3.4-1)
		No	-		Not generally considered; configuration depends on operator policy (Subclause ii.2.4.1)				
	Other value than presentation allowed	Yes	Presentation allowed	Yes	Restriction (Appendix Table ii.2.4.3.5-1)	Derived from generic number (Appendix Table ii.2.4.3.1-1)	Derived from cause of no ID (Appendix Table ii.2.4.3.2-1)	Derived from calling party number (Appendix Table ii.2.4.3.3-1)	Derived from generic number or omitted (Appendix Table ii.2.4.3.4-1)
				No			"Unavailable" (Appendix Table ii.2.4.3.2-1)		
			Other value than presentation allowed	Yes			Derived from cause of no ID (Appendix Table ii.2.4.3.2-1)		
				No			"Unavailable" (Appendix Table ii.2.4.3.2-1)		
		No	-	Yes/No	Not generally considered; configuration depends on operator policy (Subclause ii.2.4.1)				
No	-	Yes	Presentation allowed	Yes/No	Presentation (Appendix Table ii.2.4.3.5-1)	Derived from calling party number or omitted (Appendix Table ii.2.4.3.1-1)	Derived from calling party number or omitted (Appendix Table ii.2.4.3.2-1)	Derived from calling party number (Appendix Table ii.2.4.3.3-1)	Derived from calling party number (Appendix Table ii.2.4.3.4-1)
			Other value than presentation allowed	Yes	Restriction (Appendix Table ii.2.4.3.5-1)	Derived from calling party number (Appendix Table ii.2.4.3.1-1)	Derived from cause of no ID (Appendix Table ii.2.4.3.2-1)		Derived from calling party number or omitted (Appendix Table ii.2.4.3.4-1)
				No			"Unavailable" (Appendix Table ii.2.4.3.2-1)		
		No	-	Yes		Anonymous URI or unavailable URI (NOTE) (Appendix Table ii.2.4.3.1-1)	Derived from cause of no ID (Appendix Table ii.2.4.3.2-1)	Omitted (Appendix Table ii.2.4.3.3-1)	Omitted (Appendix Table ii.2.4.3.4-1)
				No			"Unavailable" (Appendix Table ii.2.4.3.2-1)		

NOTE: While [TS 29.163] defines the unavailable URI is set to the P-Asserted-Identity header filed when populating the P-Asserted-Identity header filed into the outgoing Initial INVITE request after the reception of the IAM message not containing a calling party number with complete E.164 number and Screening Indicator equal to either UPVP or NP, this Standard allows the anonymous URI in addition to the unavailable URI for backward compatibility with the previous versions of [JJ-90.30].

ii.2.5. Mapping of initial INVITE request to IAM

ii.2.5.1. Presentation/restriction information

If the presentation/restriction information is "presentation", then the address presentation restriction indicator is populated as follows:

- If only the calling party number parameters are populated after performing the procedures described in Subclause ii.2.5.2, then the address presentation restriction indicator of the calling party number is set to "presentation allowed".

- If both the calling party number parameters and generic number parameters are populated after performing the procedures described in Subclause ii.2.5.2, then the address presentation restriction indicator of the calling party number is set to "presentation restricted" and the address presentation restriction indicator of the generic number is set to "presentation allowed".

If the presentation/restriction information is "restriction", then the address presentation restriction indicator is populated as follows:

- If the calling party number parameters are populated after performing the procedures described in Subclause ii.2.5.2, then the address presentation restriction indicator of the calling party number is set to "presentation restricted".

- If the generic number parameters are populated after performing the procedures described in Subclause ii.2.5.2, then the address presentation restriction indicator of the generic number is set to "presentation restricted".

ii.2.5.2. Mapping to information components

ii.2.5.2.1. Calling party number

If the TEL_URI is present, then the calling party number parameter is derived from the value of the TEL_URI applying the mapping rule in Appendix Table ii.2.5.2.1-1.

1) When the tel URI contains verstat=No-TN-Validation, the nature of address indicator is set to international number, and:

    a) if the value of the TEL_URI starts from "+", the remaining digits after removing the leading "+" from the TEL_URI is set to the address signal; or

    b) if the value of the TEL_URI starts from the character string other than "+", the TEL_URI is set to the address signal.

2) When the tel URI does not contain verstat=No-TN-Validation, and:

    a) if the value of the TEL_URI starts from "+81", the nature of address indicator is set to "national (significant) number" and the remaining digits after removing the leading "+81" from the TEL_URI is set to the address signal; or.

    b) if the value of the TEL_URI starts from "+" other than "+81", the nature of address indicator is set to international number and the remaining digits after removing the leading "+" from the TEL_URI is set to the address signal; or

    c) if the value of the TEL_URI starts from the character string other than "+", the nature of address indicator is set to "network specific number" and the TEL_URI is set to the address signal.

When the calling party number is populated, the screening indicator is set to "user provided, verified and passed". Setting of the calling party number parameters other than nature of address indicator, address signal, and screening indicator conforms to the specification of [JJ-90.10].

If the TEL_URI is not present, then the calling party number parameters are not populated.

Appendix Table ii.2.5.2.1-1 / JJ-90.30: Mapping of tel URI to ISUP nature of address indicator and address signal of calling party number

tel URI	Usage	Nature of address indicator	Address signal
tel:+Country code national (significant) number;verstat=No-TN-Validation	Originating call on international network (overseas)	international number (NOTE)	[Country code] [National number]
tel:+81A0CDEFGHJK	Originating call on mobile/PHS network	national (significant) number	A0CDEFGHJK
tel:+81A0CDEFGHJK;verstat=No-TN-Validation	Originating call on international network (overseas)	international number (NOTE)	81A0CDEFGHJK
tel:+81ABCDEFGHJ	Originating call on local fixed telephone network	national (significant) number	ABCDEFGHJ
tel:+81ABCDEFGHJ;verstat=No-TN-Validation	Originating call on international network (overseas)	international number (NOTE)	81ABCDEFGHJ
tel:operator specific digits;phone-context=+81	Operator-originating call from operator	network specific number	operator specific digits
NOTE: While [TS 29.163] defines the procedure for interworking the value of the "verstat" tel URI parameter to "Screening indicator", this Standard defines the procedure for interworking the value of the "verstat" tel URI parameter not to "Screening indicator" but to "nature of address indicator" for consistency with [JJ-90.10].			

ii.2.5.2.2. Generic number

If the TEL_DISPLAYNAME is present and the value of the TEL_DISPLAY_NAME is not equivalent to that of TEL_URI, then a generic number is populated. The equivalency between the TEL_DISPLAYNAME and TEL_URI is determined from Appendix Table ii.2.5.2.2-1 based on the general dialling plan for local fixed telephone networks and mobile/PHS networks.

Appendix Table ii.2.5.2.2-1 / JJ-90.30: Equivalency between TEL_URI and TEL_DISPLAYNAME

TEL_URI	TEL_DISPLAYNAME
+81A0BCDEFGHJK	0A0CDEFGHJK
+81ABCDEFGHJ	0ABCDEFGHJ
+81ABCDEFGH	0ABCDEFGH

If the TEL_DISPLAYNAME starts from "0" other than the character strings "00", then the nature of the address indicator of the generic number is set to "national (significant) number (national use)". The address signal of the generic number is derived from the remaining digits after removing the leading "0" from the TEL_DISPLAYNAME. In other cases, the generic number parameters are not populated.

If the generic number parameter is populated, then the screening indicator of the generic number parameter is set to "user provided, verified and passed". Setting of the generic number parameters other than the nature of address indicator, address signal, and screening indicator conforms to the specification of [JJ-90.10].

ii.2.5.2.3. Cause of no ID

If the presentation/restriction information is "restriction" and either the calling party number parameters or the generic number parameters are populated, the cause of no ID is derived from the value of the SIP_DISPLAYNAME. The value of cause of no ID is populated, as described in Appendix Table ii.2.5.2.3-1. If the presentation/restriction information is "presentation" or the value of the SIP_DISPLAYNAME is null, then the cause of no ID is not populated.

Appendix Table ii.2.5.2.3-1 / JJ-90.30: Mapping of SIP_DISPLAYNAME to cause of no ID

SIP_DISPLAYNAME	Cause of no ID	Meaning
Unavailable	Omitted	-
Anonymous	0000001	No caller ID: rejected by user
Interaction with other service	0000010	No caller ID: service conflict
Coin line/payphone	0000011	No caller ID: call from payphone
Other than the above values	0000001	No caller ID: rejected by user

ii.2.5.3. SIP to ISUP interworking conditions table

The SIP initial INVITE request to the ISUP IAM interworking is summarized in Appendix Table ii.2.5.3-1.

Appendix Table ii.2.5.3-1 / JJ-90.30: SIP initial INVITE request to ISUP IAM interworking condition

	initial INVITE				IAM				
	TEL_URI	TEL_DISPLAYNAME		SIP_DISPLAYNAME	Calling party number		Generic number		Cause of no ID
PRIVACY	Has TEL_URI been received?	Has TEL_DISPLAYNAME been received?	Equivalency with TEL_URI	Has SIP_DISPLAYNAME been received?	Address signal, etc.	Address presentation restriction indicator	Address signal, etc.	Address presentation restriction indicator	
Presentation	Yes	Yes	Equivalent	Yes/No	Derived from TEL_URI (Appendix Table ii.2.5.2.1-1)	Presentation allowed	Not set	-	Not set
		Yes	Not equivalent			Presentation restricted	Derived from TEL_DISPLAYNAME	presentation allowed	
		No	-			Presentation allowed	Not set	-	
	No	-	-		Not set (Appendix Table ii.2.5.2.1-1)	-	Not set	-	
Restriction	Yes	Yes	Equivalent	Yes	Derived from TEL_URI (Appendix Table ii.2.5.2.1-1)	Presentation restricted	Not set	-	SIP_DISPLAYNAME (Appendix Table ii.2.5.2.3-1)
				No					"No caller ID: rejected by user" or omitted (Appendix Table ii.2.5.2.3-1)
			Not equivalent	Yes			Derived from TEL_DISPLAYNAME	presentation restricted	SIP_DISPLAYNAME (Appendix Table ii.2.5.2.3-1)
				No					"No caller ID: rejected by user" or omitted (Appendix Table ii.2.5.2.3-1)
		No	-	Yes			Not set	-	SIP_DISPLAYNAME (Appendix Table ii.2.5.2.3-1)
				No					"No caller ID: rejected by user" or omitted (Appendix Table ii.2.5.2.3-1)
	No	-	-	Yes	Not set (Appendix Table ii.2.5.2.1-1)	-	Not set	-	Not set
				No					

ii.3. Interworking of number portability information from ISUP to SIP

ii.3.1. General

For the interworking of the number portability information obtained within PSTN and/or ENUM specified in [JJ-90.31] between PSTN and IMS network, the mapping of ISUP called party number and called directory number in the IAM (defined in [JT-Q763]) to SIP Request-URI of the initial INVITE is described. Note that this Appendix does not specify the II-NNI requirements with regard to the number portability.

ii.3.2. SIP-ISUP interworking of number portability (MNP/LNP) information

ii.3.2.1. ISUP to SIP

In the redirection mechanism used between PSTNs, the IAM redirected and redirection REL are used to transfer the following address information:

- Network routing number

- Called directory number

NOTE: The network routing number can be included in the address signal of the called party number parameter in the IAM or included in the address signal of the redirection number parameter in the REL.

NOTE: Called directory number is included in the address signal of the called directory number parameter.

When an MGCF receives an IAM that was redirected, the global-number-digits of the Request-URI in the initial INVITE request is derived from the called directory number. Then the "npdi" tel URI parameter and the "rn" parameter containing the network routing number that starts from "81" are populated.

When an MGCF receives a REL that triggers the redirection, the global-number-digits of the SIP URI in the Contact header field of 3xx response is derived from the called directory number. Then the "npdi" tel URI parameter and the "rn" parameter containing the network routing number that starts from "81" are populated.

NOTE: MGCF can terminate a redirection REL and generate a redirection IAM.

ii.3.2.1.1. SIP message coding example

An example Request-Line of the initial INVITE request created from a number portability information of a redirection IAM in PSTN is shown below.

INVITE sip:+81312345678;npdi;rn=+8134512345@example.ne.jp;user=phone SIP/2.0

ii.3.2.2. SIP to ISUP

A Request-URI of an initial INVITE is set in accordance with Subclause 4.3.2 after accessing to ENUM specified in [JJ-90.31]. An example for Request-Line is shown below.

1) Coding example when network routing number is populated.

   INVITE sip:+81312345678;npdi;rn=+8134512345@example.ne.jp;user=phone SIP/2.0

2) Coding example when network routing number is not populated

   INVITE sip:+81312345678;npdi@example.ne.jp;user=phone SIP/2.0

When the received Request-URI of initial INVITE contains a called party number with a valid number of digits, "npdi" tel URI parameter and "rn" tel URI parameter containing a network routing number with a valid number of digits, a MGCF generates an IAM in accordance with Appendix Table ii.3.2.2-1 as an interworking for separate directory number addressing method.

NOTE: A network routing number with a valid number of digits conforms to Subclause 4.3.2.2 of this Standard.

Appendix Table ii.3.2.2-1 / JJ-90.30: Interwork from SIP to ISUP when network routing number is available

initial INVITE→	IAM→	
	ISUP parameter [JT-Q763]	**Procedures**
Global-rn or local-rn in "rn" tel URI parameter of Request-URI	Called party number	
	Odd/even indicator	Set according to [JT-Q763].
	Nature of address indicator	Set "0000011: national number" (NOTE 1).
	Internal Network Number indicator (INN)	Set "0: routing to internal network number allowed" (NOTE 2).
	Numbering plan indicator	Set "001: ISDN (Telephony) numbering plan (ITU-T Recommendation E.164)".
	Address signal	Set global-rn or local-rn in "rn" tel URI parameter according to the format permitted in [JT-Q763].
	Filler	Set according to [JT-Q763]
Called party number (global-number-digits or local number-digits) in telephone-subscriber part of Request-URI	Called directory number	
	Odd/even indicator	Set according to [JT-Q763]
	Nature of address indicator	Set "0000011: national number".
	Internal Network Number indicator (INN)	Set "1: routing to internal network number not allowed".
	Numbering plan indicator	Set "001: ISDN (Telephony) numbering plan (ITU-T Recommendation E.164)".
	Address signal	Set called party number (global-number-digits or local-number-digits) in the format allowed in [JT-Q763].
	Filler	Set according to [JT-Q763]

NOTE 1: While both "0000110: network routing number in national (significant) number format (national use)" and "0000111: network routing number in network-specific number format (national use)" can be set according to Subclause 7.2.3.1.2A.1.1 of [TS 29.163], "0000011: national number" allowed in Subclause 4.3 of [JJ-90.10] is set.

NOTE 2: While the procedure for setting "1: routing to internal network number not allowed" is specified in Subclause 7.2.3.1.2A.1.1 of [TS 29.163], "0:routing to internal network number allowed" is set according to Subclause 4.3 of [JJ-90.10].

When the received Request-URI of initial INVITE contains a called party number with a valid number of digits and "npdi" tel URI parameter, but not contain "rn" tel URI parameter containing a network routing number with a valid number of digits, a MGCF generates an IAM in according to Appendix Table ii.3.2.2-2

Appendix Table ii.3.2.2-2 / JJ-90.30: Interwork from SIP to ISUP when network routing number is not available

initial INVITE→	IAM→	
	ISUP parameter [JT-Q763]	Procedures
Called party number (global-number-digits or local number-digits) in telephone-subscriber part of Request-URI	Called party number	
	Odd/even indicator	Set according to [JT-Q763].
	Nature of address indicator	Set "0000011: national number".
	Internal Network Number indicator (INN)	Set "0: routing to internal network number allowed" (NOTE).
	Numbering plan indicator	Set "001: ISDN (Telephony) numbering plan (ITU-T Recommendation E.164)".
	Address signal	Set called party number (global-number-digits or local-number-digits) in the format allowed in [JT-Q763].
	Filler	Set according to [JT-Q763].
NOTE: While the procedure of the set of "1: routing to internal network number not allowed" is specified in Subclause 7.2.3.1.2A.1.1 of [TS 29.163], "0: routing to internal network number allowed" is set according to Subclause 4.3 of [JJ-90.10].		

ii.4. SIP-ISUP interworking of calling party's category

The correspondences of the "cpc" tel URI parameter to the ISUP calling party's category defined in [JT-Q763] are listed in Appendix Table ii.4-1. Each "cpc" tel URI parameter should be handled in the same manner as its corresponding calling party's category defined in [JT-Q763]. If the "cpc" tel URI parameter is not specified, the call is handled as if "ordinary" is specified.

The calling party's category is populated in the userinfo part (appended after ";" and right before "@") of the SIP URI in the P-Asserted-Identity header field using the syntax of "cpc" described in Table 4.3.4.1.3.2-1, if the SIP URI in the P-Asserted-Identity header field is not including global-number-digits. The correspondences described in Appendix Table ii.4-1 is also applied to the value of the parameter representing calling party's category in the userinfo part of the SIP URI in the P-Asserted-Identity header field.

Appendix Table ii.4-1 / JJ-90.30:
Correspondences of cpc header field parameter value with calling party's category defined in JT-Q763

cpc parameter value	Calling party's category defined in [JT-Q763]	
operator	00001001	national operator
ordinary	00001010	ordinary calling subscriber
priority	00001011	calling subscriber with priority
test	00001101	test call
payphone	00001111	payphone

ii.5. SIP-ISUP interworking of originating area information

ii.5.1. General

This Subclause describes the SIP-ISUP interworking conditions for originating area information.

ii.5.2. ISUP to SIP interworking

The ISUP charge area information parameter ([JT-Q763] Subclause 3.103) is interworked to the P-Access-Network-Info header field specified in Subclause 4.3.4.4 of this Standard. Appendix Table ii.5.2-1 shows the interworking conditions.

Appendix Table ii.5.2-1 / JJ-90.30: Interworking condition for originating area information from ISUP to SIP

IAM→	INVITE→	
	SIP header field	Procedures
CA code of charge area information parameter (NOTE)	P-Access-Network-Info	Set into the operator-specific-GI header field parameter.
NOTE: Information type indicator field is set to "0000001", if the charge area information is CA code.		

ii.5.3. SIP to ISUP interworking

The P-Access-Network-Info header field specified in Subclause 4.3.4.4 of this Standard is interworked to the ISUP charge area information parameter ([JT-Q763] Subclause 3.103). Appendix Table ii.5.3-1 shows the interworking conditions.

Appendix Table ii.5.3-1 / JJ-90.30: Interworking condition for originating area information from SIP to ISUP

Initial INVITE→	IAM→	
	ISUP parameter [JT-Q763]	Procedures
P-Access-Network-Info header field	Charge area information	
	Odd / even indicator	Set according to [JT-Q763].
	Information type indicator	Set "0000001: CA code".
	Charge area information	Set CA code in operator-specific-GI parameter of P-Access-Network-Info header field in accordance with the format allowed in [JT-Q.763].
	Filler	Set according to [JT-Q763].

ii.6. SIP-ISUP interworking of contractor number

ii.6.1. General

This Subclause describes the SIP-ISUP interworking conditions for contractor number information.

ii.6.2. ISUP to SIP interworking conditions

The ISUP contractor number parameter ([JT-Q763] Subclause 3.106) is interworked to the P-Charge-Info header field specified in Subclause 4.3.4.5 of this Standard. Appendix Table ii.6.2-1 shows the interworking conditions.

Appendix Table ii.6.2-1 / JJ-90.30: Interworking condition for contractor number from ISUP to SIP

IAM→	INVITE→	
	SIP header field	Procedures
Address information of contractor number parameter	P-Charge-Info	global-number-digits and local-number-digits in telephone-subscriber part.

ii.6.3. SIP to ISUP interworking conditions

The P-Charge-Info header field specified in Subclause 4.3.4.5 of this Standard is interworked to the ISUP contractor number parameter ([JT-Q763] Subclause 3.106). Appendix Table ii.6.3-1 shows the interworking conditions.

Appendix Table ii.6.3-1 / JJ-90.30: Interworking condition for contractor number from SIP to ISUP

initial INVITE→	IAM→	
	ISUP parameter [JT-Q763]	**Procedures**
P-Charge-Info header field	Contractor number	
	Odd / even indicator	Set according to [JT-Q763].
	Nature of address indicator	Set "0000011: national number".
	Numbering plan indicator	Set "001: ISDN (Telephony) numbering plan (ITU-T Recommendation E.164)".
	Address signal	Set contractor number (global-number-digits or local-number-digits) in the P-Charge-Info header field in accordance with the format allowed in [JT-Q.763].
	Filler	Set according to [JT-Q763].

ii.7. SIP-ISUP interworking of history information of number translation

Refer to the Annex L of [TS 29.163] for the SIP-ISUP interworking between the history information of number translation specified in the Subclause 4.4.2 of this Standard, and the "called IN number" defined in [JT-Q763] Subclause 3.73 and the "original called IN number" defined in [JT-Q763] Subclause 3.87.

Appendix iii (Informative): Considerations on network operation and maintenance

iii.1. Controlling number of simultaneous SIP sessions

iii.1.1. General

This Subclause describes the mechanism to control the number of simultaneous SIP sessions between interconnected operator's networks.

iii.1.2. Basic principle

When some kinds of event (e.g., network congestion) occurs, in order to prevent the network resources for the control plane from being occupied by unidirectional session establishment attempts from one side of the interconnected network, the limitation of the number of outgoing simultaneous SIP sessions (called unidirectional session management) shall be applied.

The number of simultaneous SIP sessions is incremented when a SIP request outside an existing dialog (e.g., an initial INVITE request) is sent out, and decremented when a SIP session is terminated.

When the number of simultaneous SIP sessions is controlled by conditions other than the maximum number of SIP sessions, the detailed condition is determined in the inter-operator agreements.

iii.1.3. Controlling number of outgoing SIP sessions by unidirectional session management

The maximum number of outgoing SIP sessions generated by the two operator's networks (, which represents the bandwidth needs to be reserved between the two operators) is determined in the inter-operator agreements. There are two methods for reserving the bandwidth, non-shared and shared reservation methods. Non-shared reservation method allocates bandwidth that is equal to the sum of the declared bandwidth from each operator. Shared reservation method allocates bandwidth based on less than the sum of the declared bandwidth. The two operators determine which reservation method they will apply based on the inter-operator agreement. {Appendix Table i.4-18 Item 1}

If the control of the reserved number of SIP sessions for calling subscribers with priority is required in addition to the control of the number of outgoing SIP sessions, each IMS network configures the number of reserved SIP sessions based on the operator policy and applies the control of the reserved SIP sessions. In this case, the number of the reserved SIP sessions is included in the maximum number of outgoing SIP sessions.

iii.2. Connection test

iii.2.1. General

This Subclause describes test systems for connectivity check between the interconnected operator's networks.

1) The purpose of the test system includes preventive maintenance, failure search in the event of fault occurrence, restoration confirmation, and functional checks when installing new equipment.

2) The following test is possible between the interconnected operator's networks:

   - Normality test for IP routing and normality test for signalling (SIP) / bearer (RTP) / ENUM database access as a normality test

   - Normality test for routing configuration of SIP messages / destination configuration of ENUM queries as a connection test

3) In principle, each operator is responsible for testing its own equipment, and an operator can not perform tests on another operator's equipment.

iii.2.2. Types of tests

1) Normality test

   Each operator performs the normality tests per POI.

a) Normality test for IP routing

The function of checking the normality of IP routing using "ping" program is provided.

b) Normality test for SIP signalling / RTP bearer / ENUM database access

The function of checking the normality by confirming that the media bearer is connected to the media resource corresponding to telephone numbers for the tests, is provided.

2) Connection test

The function of checking the normality of routing configurations corresponding to telephone numbers for tests assigned to each telephone number range (e.g., area code). The normality of routing configurations with respect to a telephone number for the test is confirmed by the speech media from the media resource or sending out of a 180 (Ringing) response or a 487 (Request Terminated) response from the terminating network.

iii.2.3. Numbering configuration for connection tests

This Subclause describes the numbering configuration for connection tests. Regarding the contents related to telephone number information in the SIP message, the requirements are specified in the main body of this Standard.

1) Testing by PSTN and IP telephone (Category A) number

Testing with an operator having IP telephone (Category A) numbers is performed, in principle, with the following number format.

area code + local exchange number + XXXX

area code: Typical area code used by operator.

XXXX: Subscriber number allocated for normality and connection tests (e.g., 0060, 2599).

2) Testing by mobile phone, PHS, and pager service identification numbers

Testing with an operator having mobile phone, PHS, and pager service identification numbers is performed, in principle, with the following number format.

A0 + CDE + 12X

A: 2 (pager), 7 (PHS), 7 to 9 (mobile phone).

CDE: Operator identification code (one required for each operator; C excludes 0).

3) Testing by IP telephone (Category B) service identification numbers

Testing with an operator having IP telephone (Category B) service identification numbers is performed, in principle, with the following number format.

50 + CDEF + XXXX

CDEF: Operator identification code (one required for each operator; C excludes 0).

XXXX: Subscriber number for the tests between the operator's different networks. The numbers are allocated for each connection.

iii.2.4. Indication of test call

When the calling party's category is applied to indicate that the call is "test call", a "cpc" tel URI parameter with the value "test" in the P-Asserted-Identity header field of the initial INVITE request is used based on the signalling requirement specified in Subclause 4.3.4.1.3 of this Standard.

iii.2.5. Call flows of tests for interconnection between operator's network

1) Normality test

The call flows for normality tests of signalling (SIP) / bearer (RTP) are equivalent to those provided in Subclause vi.2.1.1 and Subclause vi.2.2.

2) Connection test

The call flow for connection test is equivalent to that provided in Subclause vi.2.3.

iii.3. Maintenance blocking system

iii.3.1. General

This Subclause describes the required functions for enforcement of maintenance blocking between interconnected operator's networks.

iii.3.2. Required functions

Pre-blocking function and blocking function required for maintenance blocking are defined as follows.

1) Pre-blocking function

The pre-blocking function restricts the sending out of a SIP request outside an existing dialog from its own network to the opposite network based on the IP addresses belonging to the opposite network. Established sessions are not restricted. The operator responsible for maintenance can confirm or be notified that all sessions are terminated.

2) Blocking function

The blocking function restricts the sending out of a SIP request outside an existing dialog from its own network to the opposite network, but does not restrict the receiving of a SIP request outside an existing dialog from the opposite network to its own network.

When implementing a function that sends a test call in the network, the network can send a test call to the opposite network, even if blocking is being enforced in the network.

iii.4. Monitoring for extinction of RTP/RTCP packets

iii.4.1. General

[TS 24.229] specifies a functionality to monitor extinction of RTP/RTCP packets for a media negotiated in an existing dialog and release the dialog when no RTP/RTCP packet is available in an operator defined time. In the inter-connection between different operator's networks, call failure can occur due to the conditions applied in the inter-connected operator's network (e.g., the inter-connected operator applies the timer for monitoring extinction of RTP/RTCP packets shorter than the extinction time which frequently occurs in its own network or at the UE). Therefore, it is preferable that the monitoring condition(s) should be exchanged between the operators.

This Appendix describes the monitoring method for extinction of RTP/RTCP packets for the inter-connection between different operator's networks.

iii.4.2. Basic principle

An IMS network can release a call when no RTP/RTCP packet is available for a certain time. However, the monitoring for extinction of RTP/RTCP packets is basically applied at the II-NNI, and the monitoring condition(s) (e.g., trigger timing of monitoring start, timer value of monitoring for extinction, call type of monitoring target, direction of monitoring target, action when the extinction is detected) is   exchanged between two operators.
{Appendix Table i.4-19 Item 1}

Appendix Table iii.4.2-1 describes the examples of monitoring conditions for extinction of RTP/RTCP.

Appendix Table iii.4.2-1 / JJ-90.30: Examples of monitoring conditions for extinction of RTP/RTCP

Monitoring condition	Example
Trigger of monitoring start	Reception of 200 (OK) response to the initial INVITE request or reception of a 18x response containing SDP
Timer value of monitoring for extinction	Over 10 sec
Call type of monitoring target	All audio session
Direction of monitoring target	Media packets from its own network to the opposite network Media packets from opposite network to own network Both directions
Action when the extinction is detected (During the establishment of a session)	Release a call by sending CANCEL request
Action when the extinction is detected (During an established session)	Release a call by sending BYE request

iii.5. Fault detection and restoration detection for SIP layer

iii.5.1. General

This Subclause describes fault detection and restoration detection at SIP layer between the two interconnected operator's networks.

iii.5.2. Conditions for fault detection and detouring

For fault detection at SIP layer, "Expiration of initial INVITE transaction timeout timer (Timer B)" and "reception of 503 response to an initial INVITE request" are applied as the basic conditions and detailed conditions of detouring (selection of another IBCF for detouring) are determined by the inter-operator agreements. {Appendix Table i.4-20 Item 1}

iii.5.3. Conditions for restoration detection for SIP layer

The applicability of the restoration method at the II-NNI and the method(s) applied for restoration detection (selecting both "method using the OPTIONS method" and "method using the pilot INVITE request" or one of them) are determined by the inter-operator agreements. {Appendix Table i.4-20 Item 2}

"A restration detection using the OPTIONS method" is specified in Annex d of this Standard.

The method using the pilot INVITE request detects the restoration of the peer server as follows:

1) Start a timer (the timer value determined by originating IMS network or the value of the Retry-After header field in the 503 (Service Unavailable) response to the initial INVITE request) when the originating IMS network detects a failure in the peer server.

2) Send an initial INVITE request to the peer server in failure, after the expiration of the timer.

3) Detect the restoration of peer server and exit the method, if receiving a successful SIP response to the initial INVITE request.

4) Re-start the timer at the end of initial INVITE transaction and continue the method from step 2, if receiving a non-successful final SIP response to the initial INVITE request.

When the method using pilot INVITE request is applied with the timer value determined by the originating IMS network operator, the operator shall determine the timer value within the range from 30 seconds to 900 seconds and shall be shared between the operators.

iii.5.4. Void

Appendix iv (Informative): Void

Appendix v (Informative): Procedure for termination of the precondition mechanism

v.1 General

This Appendix describes the procedures for termination of the precondition mechanism for an originating IMS network to enable IMS network interconnection between an originating IMS network applying the precondition mechanism and a terminating IMS network not applying the precondition mechanism. The procedure in this Appendix is described as an implementation reference, and other procedures are allowed to be implemented.

v.2. Termination procedure

This Subclause describes the SIP signaling procedure to apply the precondition mechanism between an originating IMS network and an originating IBCF, and not to apply the precondition mechanism between the originating IBCF and a terminating IMS network by sending sends an initial INVITE request from the originating IBCF to the terminating IMS network.

v.2.1. Procedure when initial INVITE request is received

When receiving an initial INVITE request containing the SIP option tag "precondition" in the Supported header field, an IBCF in the originating IMS network applying the precondition mechanism stores the initial INVITE request, and does not forward the initial INVITE request. Subsequently, the IBCF sends a 18x response containing the SIP option tag "precondition" in the Require header field and an SDP answer to the originating IMS network. The SDP answer includes an SDP attribute ("a=curr:qos local sendrecv") indicating the completion of resource reservation on the terminating IMS network.

v.2.2. Procedure when the precondition is met

When receiving a PRACK request or an UPDATE request including an SDP offer from the originating IMS network, the IBCF sends 200 (OK) response to the received PRACK request or the UDPATE request including an SDP answer generated by the IBCF. If the preconditions for the resource reservation status on both the originating IMS network and the IBCF are met, before forwarding the stored initial INVITE request to the terminating IMS network, the IBCF deletes the parameters related to the precondition (i.e. SIP option tag "precondition" in the Supported header field) and include an SDP offer generated from the last sent/received SDP answer to/from the originating IMS network. The SDP offer includes the media direction attribute of the "a=sendrecv" and does not include precondition related SDP attributes ("a=curr:", "a=des:" and "a=conf:").

v.2.3. Procedure when a 18x response for the initial INVITE request is received

When receiving a 18x response for the initial INVITE request from the terminating IMS network,

   1) when the 18x response includes an SDP answer:

      a) if a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter is included in the received 18x response, the IBCF sends an UPDATE request including an SDP offer generated from the received SDP offer, the SIP option tag "precondition" in the Supported header field, and the received P-Early-Media header field to the originating IMS network on the existing dialog. The IBCF does not forward the 200 (OK) response to the UPDATE request to the succeeding entity, and if the terminating IMS network supports 100rel sends a PRACK request without an SDP offer to the terminating IMS network on the existing dialog.

      b) if a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter is not included in the received 18x response,

         i) if there are any difference between the last sent/received SDP answer to/from the originating IMS network and the received SDP answer, the IBCF sends an UPDATE request including an SDP offer generated from the received SDP offer and the SIP option tag "precondition" in the Supported header field to the originating IMS network on the existing dialog. The IBCF does not forward the 200 (OK) response to the UPDATE request to the succeeding entity, and if the terminating IMS network supports

100rel sends a PRACK request without an SDP offer to the terminating IMS network on the existing dialog.

    ii) if there are no difference between these SDPs, the IBCF deletes the SDP answer, and sends the 18x response to the originating IMS network on the existing dialog.

  2) when the 18x response does not include an SDP answer, the IBCF forward the 18x response to the originating IMS network on the existing dialog.

v.2.4. Procedure when an UPDATE request including SDP offer is received from the terminating IMS network

When receiving an UPDATE request including SDP offer from the terminating IMS network, the IBCF sends an UPDATE request including an SDP offer generated from the received SDP offer to the originating IMS network on the existing dialog. Subsequently, the IBCF sends a 200 (OK) response to the received UPDATE request on the existing dialog.

v.2.5. Procedure when a 200 (OK) response for the initial INVITE request is received

When receiving a 200 (OK) response for the initial INVITE request from the terminating IMS network,

  1) when the 200 (OK) response includes an SDP answer:

    a) if there are any difference between the last sent/received SDP answer to/from the originating IMS network and the received SDP answer, the IBCF sends an UPDATE request including an SDP offer generated from the received SDP offer and the SIP option tag "precondition" in the Supported header field to the originating IMS network on the existing dialog. The IBCF does not forward the 200 (OK) response to the UPDATE request to the succeeding entity, and sends a 200 (OK) response for the initial INVITE request without an SDP offer to the terminating IMS network on the existing dialog.

    b) if there are no difference between the last sent/received SDP answer to/from the originating IMS network and the received SDP answer, the IBCF either proceeds step a) or sends 200 (OK) response to the initial INVITE request without an SDP answer to the originating IMS network on the existing dialog.

  2) when the 200 (OK) response does not include an SDP answer, the IBCF forward the 200 (OK) response to the originating IMS network on the existing dialog.

v.3. Callflow examples

v.3.1. Callflow example for when early media is not provided

The procedure for termination of the precondition mechanism described in this Appendix for the interconnection scenario between an originating IMS network applying the precondition mechanism and a terminating IMS network not applying the precondition mechanism is shown in the Appendix Figure v.3.1-1. The call flow shows that the terminating IMS network does not send early media.

Originating IMS network

| CSCFs | IBCF | Terminating IMS network |

F1: Initial INVITE [SDP offer]
Supported: precondition
a=curr:qos local none
a=curr:qos remote none

F2: 183 Session Progress [SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote none

Early dialog#0

F3: PRACK

F4: 200 OK (PRA)

F5: UPDATE [SDP offer]
Supported: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

F6: 200 OK (UPD) [SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

F7: Initial INVITE [SDP offer]

F9: 180 Ringing

F8: 180 Ringing

Early dialog#0

Early dialog#1

F10: PRACK

F11: PRACK

F13: 200 OK (PRA)

F12: 200 OK (PRA)

F14: 200 OK (INV) [SDP answer]

F15: UPDATE [SDP offer]
Supported: precondition
a=curr:qos local sendrecv
a=curr:qos remote none

F16: 200 OK (UPD) [SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

F17: 200 OK (INV)

F18: ACK

F19: ACK

Appendix Figure v.3.1-1 / JJ-90.30: The example for when early media is not provided

v.3.2. Callflow example for when early media is provided using gateway model

The procedure for termination of the precondition mechanism described in this Appendix for the interconnection scenario between an originating IMS network applying the precondition mechanism and a terminating IMS network not applying the precondition mechanism is shown in the Appendix Figure v.3.2-1. The call flow shows that the terminating IMS network sends early media using gateway model.
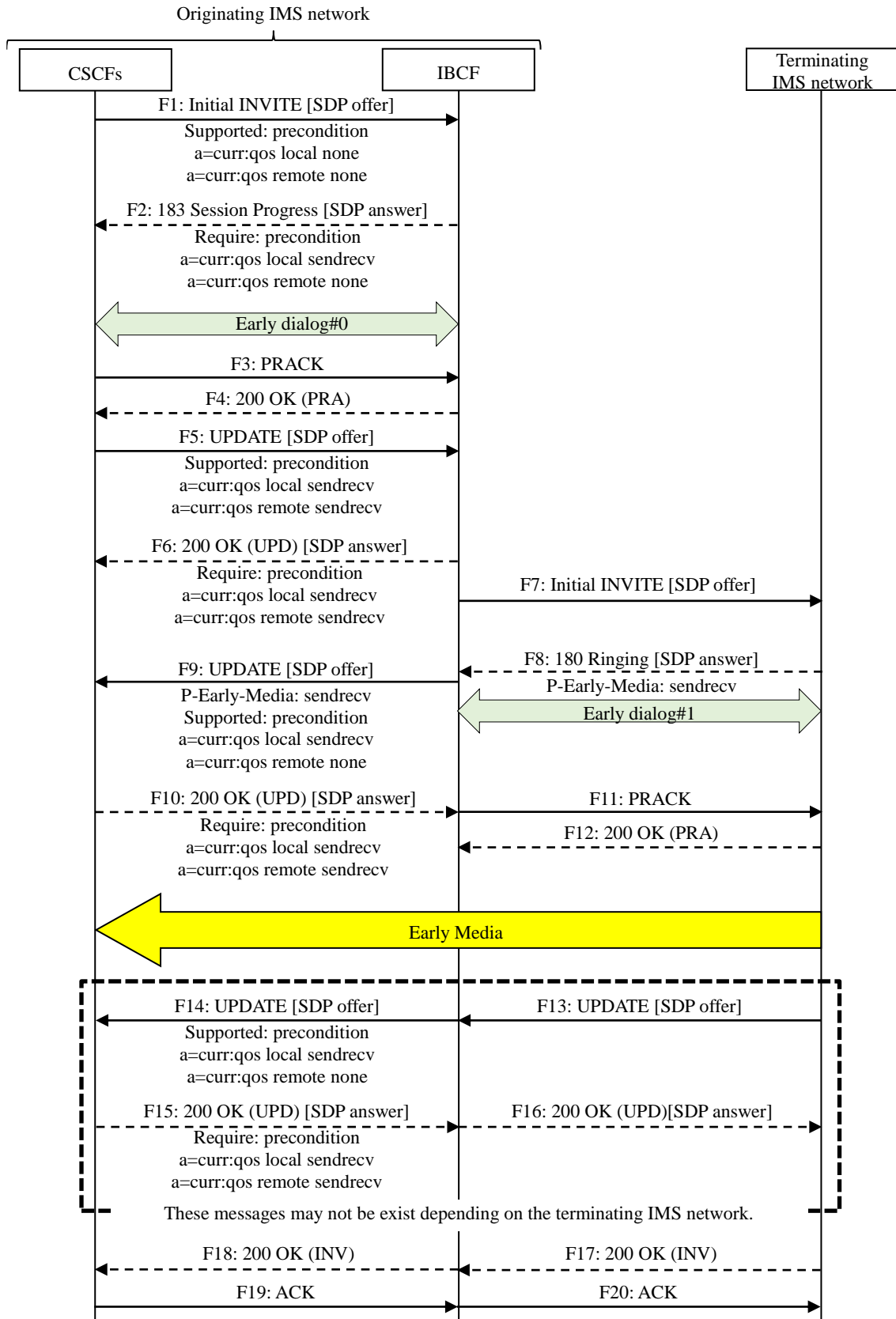
Appendix Figure v.3.2-1 / JJ-90.30: The example for when early media is provided using gateway model

v.3.3. Callflow example for when early media is provided not using 100rel

The procedure for termination of the precondition mechanism described in this Appendix for the interconnection scenario between an originating IMS network applying the precondition mechanism and a terminating IMS network not applying the precondition mechanism is shown in the Appendix Figure v.3.3-1. The call flow shows that the terminating IMS

network sends early media not using 100rel.

Originating IMS network

| CSCFs | IBCF | Terminating IMS network |

F1: Initial INVITE [SDP offer]
Supported: precondition
a=curr:qos local none
a=curr:qos remote none

F2: 183 Session Progress [SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote none

Early dialog#0

F3: PRACK

F4: 200 OK (PRA)

F5: UPDATE [SDP offer]
Supported: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

F6: 200 OK (UPD) [SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

F7: Initial INVITE [SDP offer]

F8: 180 Ringing [SDP answer]

F9: UPDATE [SDP offer]
P-Early-Media: sendrecv
Supported: precondition
a=curr:qos local sendrecv
a=curr:qos remote none

P-Early-Media: sendrecv
Early dialog#1

F10: 200 OK (UPD)[SDP answer]
Require: precondition
a=curr:qos local sendrecv
a=curr:qos remote sendrecv

Early Media

F12: 200 OK (INV)

F11: 200 OK (INV) [SDP answer]

F13: ACK

F14: ACK

Appendix Figure v.3.3-1 / JJ-90.30: The example for when early media is provided not using 100rel

Appendix vi (Informative):
Procedure for converting multiple early dialogs to a single dialog

vi.1. General

This Subclause describes the SIP signaling procedure for conversion from multiple early dialogs to a single dialog, to enable IMS network interconnection between IMS networks if a terminating IMS network applies forking in the network. The procedure in this Appendix is described as an implementation reference, and other procedures are allowed to be implemented.

vi.2. Conversion procedure

This Subclause describes the SIP signaling procedure at an IBCF in the terminating IMS network where forking is applied for converting multiple early dialogs established on the terminating IMS network side to a single dialog for the originating IMS network side.

vi.2.1. Procedure when the first 18x response is received

When receiving the first 18x response corresponding to an initial INVITE request, from an entity in the terminating IMS network, and:

1) if the 18x response includes an initial SDP answer, the IBCF:

   a) stores the SDP answer as the last received SDP answer on the established dialog (early dialog#1 in Appendix Figure vi.2.1-1); and

   b) sends a 18x response including an SDP answer generated from the received SDP answer to the originating IMS network. At the same time, the IBCF stores the SDP answer sent to the originating IMS network in association with the dialog established on the terminating IMS network side (early dialog#1 in Appendix Figure vi.2.1-1); or

2) if the 18x response does not include an initial SDP answer, the IBCF:

   a) sends the received 18x response to the originating IMS network.

In this conversion procedure, the IBCF needs to terminate 18x responses received from the terminating IMS network side, therefore the IBCF also needs to terminate PRACK requests and responses, by sending a PRACK request generated by the IBCF to an entity in the terminating IMS network and sending a 200 (OK) response to the received PRACK request generated by the IBCF to the originating IMS network.
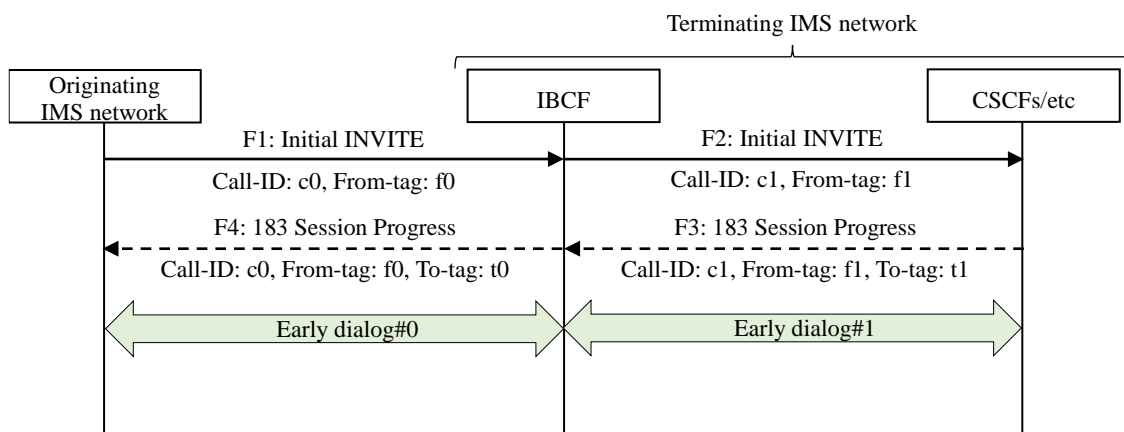
Appendix Figure vi.2.1-1 / JJ-90.30: Procedure when an IBCF receives the first 18x response

vi.2.2. Procedure after receiving the first 18x response before receiving 200 (OK) response to the initial INVITE request

This Subclause describes the procedure at the IBCF after establishing an early dialog on the terminating IMS network

side and an early dialog with the originating IMS network before receiving 200 (OK) response to the initial INVITE request.

If the IBCF has not sent the first SDP answer to the originating IMS network, the IBCF sends the SDP answer generated from the received SDP answer from the terminating IMS network side to the originating IMS network on the existing dialog.

If the IBCF has sent the first SDP answer to the originating IMS network, and if the IBCF receives an SDP offer/answer from the terminating IMS network for sending an early media, the IBCF sends the SDP answer generated from the received SDP answer to the originating IMS network on the existing dialog for modification of the media. Also, when receiving an SDP offer/answer from the originating IMS network, then the IBCF sends a SDP generated from the received SDP on the existing dialog in which the IBCF receives the SDP offer/answer from which the last SDP sent to the originating IMS network was generated.

The procedures per received SIP message are described in the following Subclauses.

vi.2.2.1. Procedure when a subsequent 18x response is received

When receiving a subsequent 18x response containing an SDP answer to the initial INVITE request, from an entity in the terminating IMS network, and:

1) if the IBCF has not sent the first SDP answer to the originating IMS network, the IBCF:

   a) stores the SDP answer as the last received SDP answer on the established dialog (early dialog#2 in Appendix Figure vi.2.2.1-1); and

   b) sends a 18x response containing an SDP answer generated from the received SDP answer to the originating IMS network on an existing dialog (early dialog#0 in Appendix Figure vi.2.2.1-1). At the same time, the IBCF stores the SDP answer sent to the originating IMS network in association with the dialog on the terminating network side (early dialog#2 in Appendix Figure vi.2.2.1-1); or

2) if the IBCF has sent an SDP answer to the originating IMS network, the IBCF:

   a) stores the SDP answer as the last received SDP answer on the dialog on the terminating network side (early dialog#2 in Appendix Figure vi.2.2.1-2), and:

      i) if a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter is included, sends an UPDATE request containing an SDP offer generated from the received SDP answer to the originating IMS network on the existing dialog (early dialog#0 in Appendix Figure vi.2.2.1-2). At the same time, the IBCF stores the SDP offer sent to the originating IMS network in association with the dialog on the terminating network side (early dialog#2 in Appendix Figure vi.2.2.1-2). After that the IBCF does not forward the SIP response to the UPDATE request to a succeeding entity.

      ii) if a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter is not included, does not forward the received 18x response to the originating IMS network or forwards the 18x response not containing the SDP on the existing dialog (see Appendix Figure vi.2.2.1-3).

When receiving a subsequent 18x response to the initial INVITE request from the terminating IMS network side, and:

1) if the received 18x response contains the P-Early-Media header field, then the IBCF forwards the received 18x response to the originating IMS network on the existing dialog (see Appendix Figure vi.2.2.1-4).

2) if the received 18x response does not contain the P-Early-Media header field, then the IBCF does not forward the received 18x response or forwards the received 18x response to the originating IMS network on the existing dialog (see Appendix Figure vi.2.2.1-5).

The IBCF needs to terminate a PRACK request received from the originating IMS network by sending a 200 (OK) response to the received PRACK request generated by the IBCF to the originating IMS network, and sending a PRACK request generated by the IBCF to the terminating IMS network side.
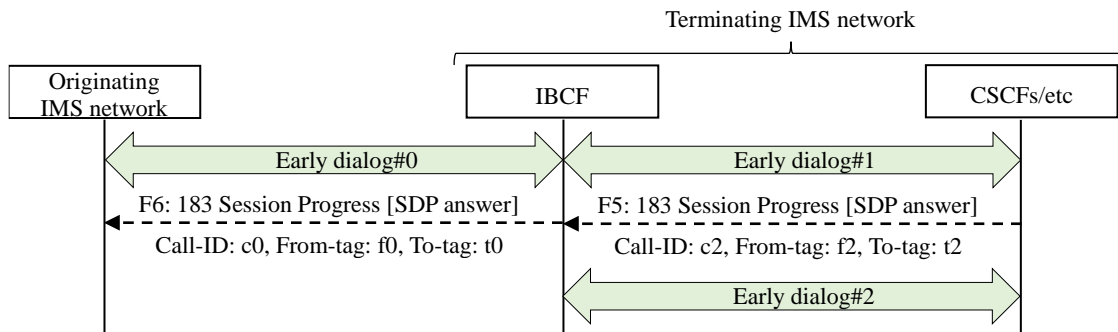
Appendix Figure vi.2.2.1-1 / JJ-90.30: Procedure when an IBCF receives a subsequent 18x response (IBCF receives an SDP answer when the IBCF has not sent an SDP answer to the originating IMS network)
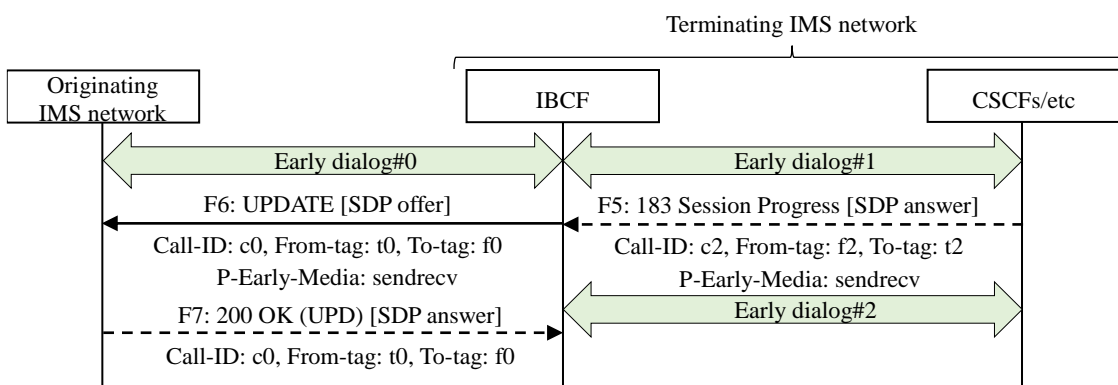
Appendix Figure vi.2.2.1-2 / JJ-90.30: Procedure when an IBCF receives a subsequent 18x response (IBCF receives an SDP answer and a P-Early-Media header field when the IBCF has sent an SDP answer to the originating IMS network)
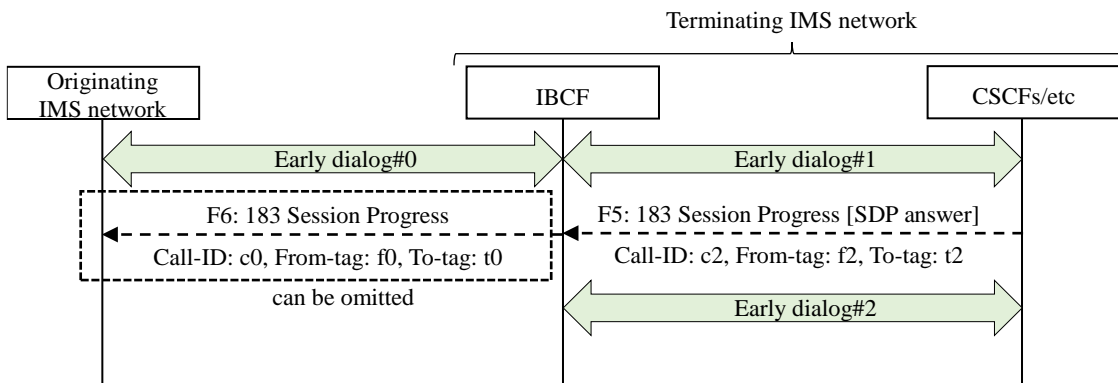
Appendix Figure vi.2.2.1-3 / JJ-90.30: Procedure when an IBCF receives a subsequent 18x response (IBCF receives an SDP answer and no P-Early-Media header field when the IBCF has sent an SDP answer to the originating IMS network)
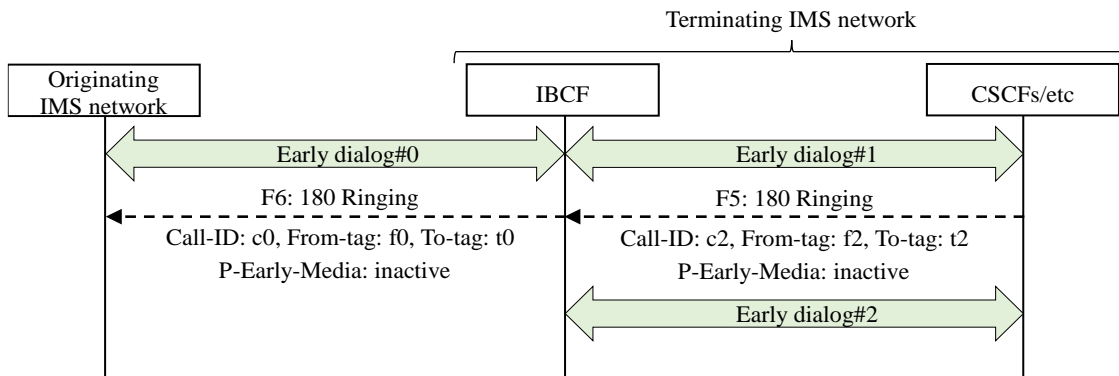
Appendix Figure vi.2.2.1-4 / JJ-90.30: Procedure when an IBCF receives a subsequent 18x response (No SDP answer and a P-Early-Media header field)
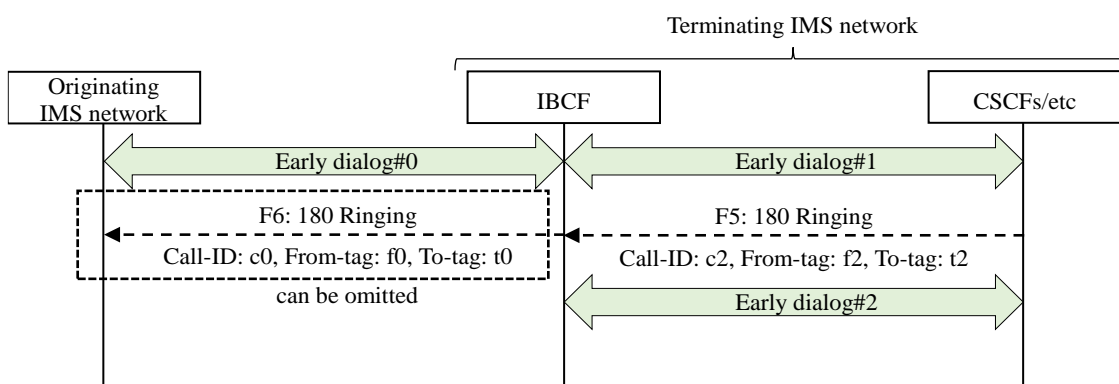
Appendix Figure vi.2.2.1-5 / JJ-90.30: Procedure when an IBCF receives a subsequent 18x response (No SDP answer and no P-Early-Media header field)

vi.2.2.2. Procedure when an UPDATE request from the terminating IMS network side

When receiving an UPDATE request containing an SDP offer from the terminating IMS network side, and:

1) if the received UPDATE request contains a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter, then the IBCF sends an UPDATE request containing an SDP offer generated from the received SDP offer to the originating IMS network on the existing dialog (early dialog#0 in Appendix Figure vi.2.2.2-1). At the same time, the IBCF stores the SDP offer sent to the originating IMS network in association with the dialog on the terminating IMS network side (early dialog#2 in Appendix Figure vi.2.2.2-1). After that, the IBCF forwards the SIP response to the UPDATE request on the existing dialog on which the IBCF receives the UPDATE request from the terminating IMS network side; or

2) if the received UPDATE request does not contain a P-Early-Media header field containing either "sendrecv" or "sendonly" parameter, then the IBCF does not send the UPDATE request to the originating IMS network side, sets an SDP answer generated by the IBCF to a 200 (OK) response to the UPDATE request, and sends the response to the terminating IMS network side (see Appendix Figure vi.2.2.2-2).

When receiving an UPDATE request not containing an SDP offer from the terminating IMS network side, and:

1) if the received UPDATE request contains a P-Early-Media header field, then the IBCF sends an UPDATE request on the existing dialog (see Appendix Figure vi.2.2.2-3); or

2) if the received UPDATE request does not contain a P-Early-Media header field, then the IBCF does not send the UPDATE request to the originating IMS network side, and sends a 200 (OK) response to the UPDATE request not containing a SDP answer to the terminating IMS network side (see Appendix Figure vi.2.2.2-4).
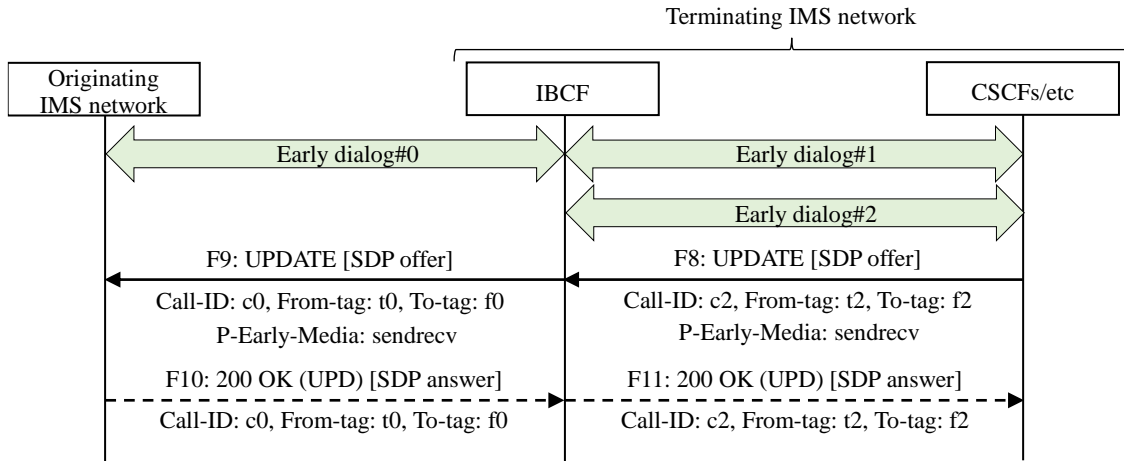
Appendix Figure vi.2.2.2-1 / JJ-90.30: Procedure when an IBCF receives an UPDATE request containing an SDP offer (with P-Early-Media header field)
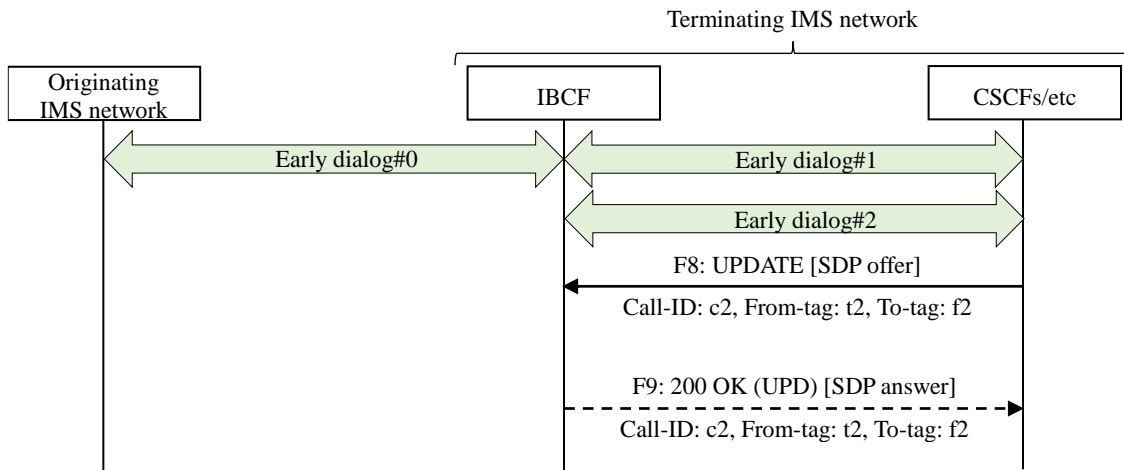
Appendix Figure vi.2.2.2-2 / JJ-90.30: Procedure when an IBCF receives an UPDATE request containing an SDP offer (without P-Early-Media header field)
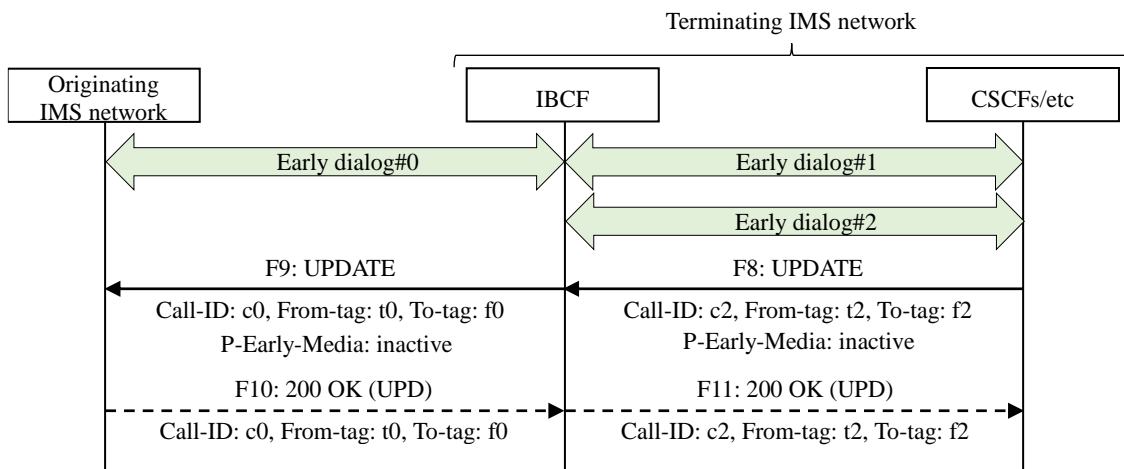
Appendix Figure vi.2.2.2-3 / JJ-90.30: Procedure when an IBCF receives an UPDATE request not containing an SDP offer (with P-Early-Media header field)
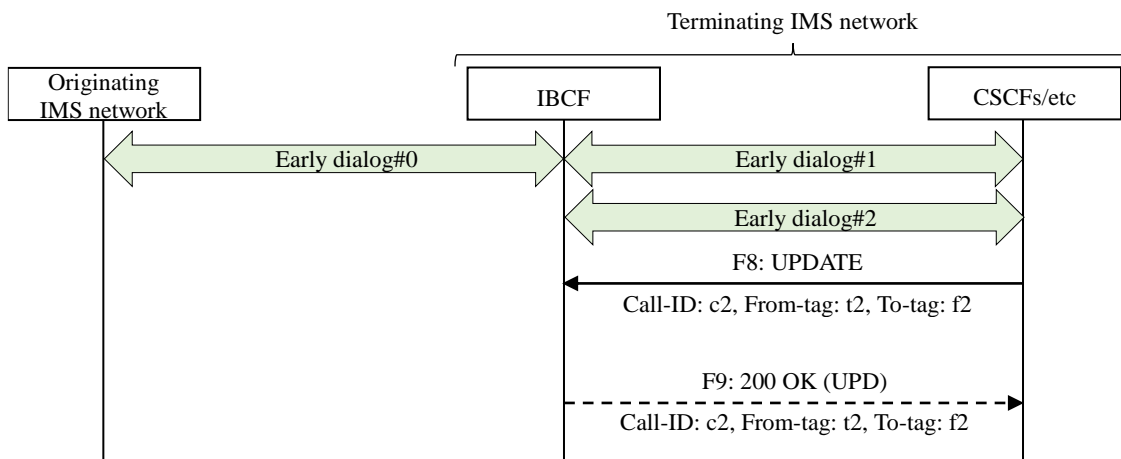
Appendix Figure vi.2.2.2-4 / JJ-90.30: Procedure when an IBCF receives an UPDATE request not containing an SDP offer (without P-Early-Media header field)

vi.2.2.3. Procedure when an UPDATE request is received from the originating IMS network

When receiving an UPDATE request from the originating IMS network side, and

1) if the received UPDATE request contains an SDP offer, then the IBCF regards the UPDATE request as the request to change the media on the existing dialog established with the originating IMS network, and the IBCF sends an UPDATE request containing an SDP offer generated from the received SDP offer to the terminating IMS network side on the dialog in which the IBCF receives the SDP offer/answer from which the last SDP sent to the originating IMS network was generated (See Appendix Figure vi.2.2.3-1); or

2) if the received UPDATE request does not contain an SDP offer, then the IBCF sends an UPDATE request not containing an SDP offer to the terminating IMS network side on the dialog in which the IBCF receives the SDP offer/answer from which the last SDP sent to the originating IMS network was generated (See Appendix Figure vi.2.2.3-2).
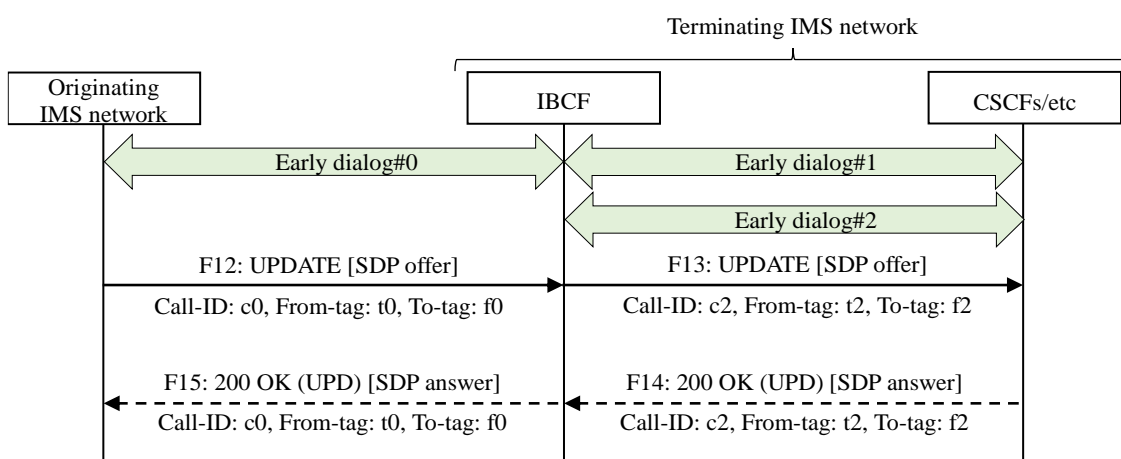
Appendix Figure vi.2.2.3-1 / JJ-90.30: Procedure when an IBCF receives an UPDATE request containing an SDP offer from the originating IMS network
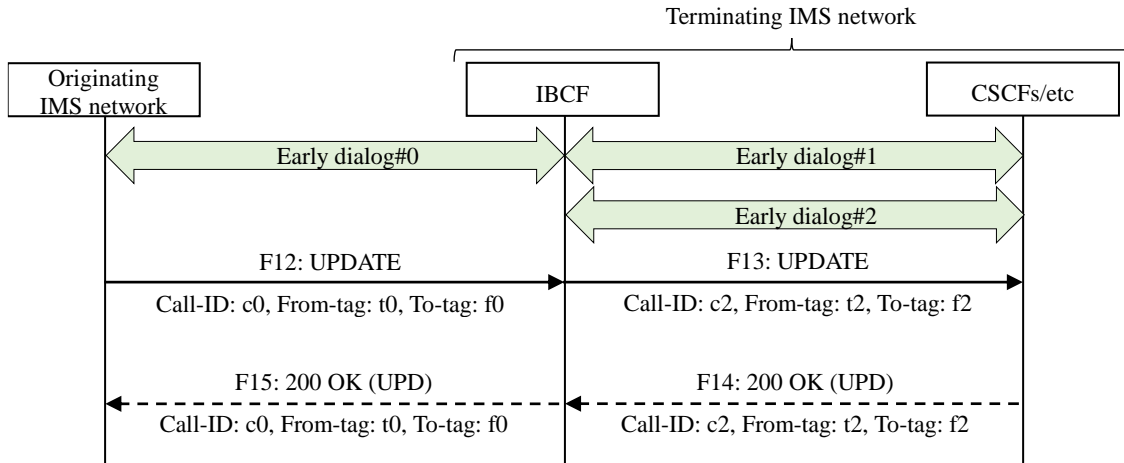
Appendix Figure vi.2.2.3-2 / JJ-90.30: Procedure when an IBCF receives an UPDATE request not containing an SDP offer from the originating IMS network

vi.2.3. Procedure when a 200 (OK) response to the initial INVITE request is received

When receiving a 200 (OK) response to the initial INVTE request, and:

1) if the 200 (OK) response contains an SDP answer, and:

    a) if the IBCF has not sent the first SDP answer to the originating IMS network, sends 200 (OK) response containing an SDP answer generated from the received SDP answer to the originating IMS network on the existing dialog (Appendix Figure vi.2.3-1); or

    b) if the IBCF has sent/received an SDP answer to/from the originating IMS network, compares the SDP answer generated from the received SDP answer and the last sent/received SDP answer to the originating IMS network, and:

        i) if there are any differences between these SDPs, sends an UPDATE request including an SDP offer generated from the received SDP answer to the originating IMS network on existing dialog. After that, when receiving the SIP response to the UPDATE request, then the IBCF does not forward the SIP response to the succeeding entity, and sends 200 (OK) response to the initial INVITE request not containing an SDP to the originating IMS network on the existing dialog (Appendix Figure vi.2.3-2); or

        ii) if there are no difference between these SDPs, either proceeds step i) or sends 200 (OK) response to the initial INVITE request not containing SDP to the originating IMS network on the existing dialog (Appendix Figure vi.2.3-3); or

2) if the 200 (OK) response does not contain an SDP answer:

    a) compares the SDP answer generated from the last received/sent SDP answer on the dialog on which 200 (OK) response is received and the last sent SDP answer to the originating IMS network; and

        i) if there are any differences between these SDPs, sends an UPDATE request containing an SDP offer generated from last received/sent SDP answer on the dialog on which 200 (OK) response is received to the originating IMS network on existing dialog. After that, when receiving the SIP response to the UPDATE request, then the IBCF does not forward the SIP response to the succeeding entity, and sends 200 (OK) response to the initial INVITE request not containing SDP to the originating IMS network on the existing dialog (Appendix Figure vi.2.3-4); or

        ii) if there are no difference between these SDPs, either proceeds step i) or sends 200 (OK) response to the initial INVITE request not containing an SDP to the originating IMS network on the existing dialog (Appendix Figure vi.2.3-5).

Terminating IMS network

Originating IMS network		IBCF		CSCFs/etc

Early dialog#0   Early dialog#1

Early dialog#2

F16: 200 OK (INV) [SDP answer]   F15: 200 OK (INV) [SDP answer]

Call-ID: c0, From-tag: f0, To-tag: t0   Call-ID: c1, From-tag: f1, To-tag: t1

Confirmed dialog#0   Confirmed dialog#1

F17: ACK   F18: ACK
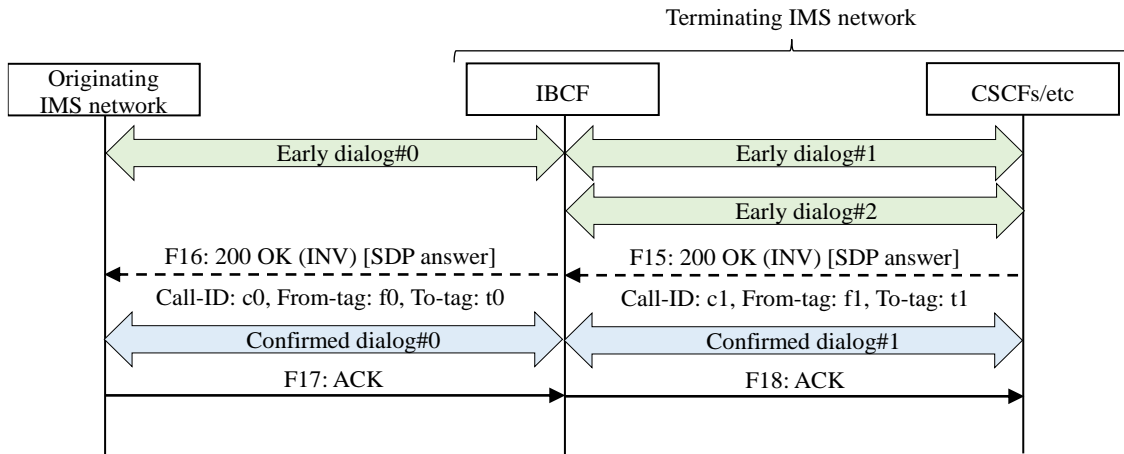
Appendix Figure vi.2.3-1 / JJ-90.30: Procedure when an IBCF receives a 200 (OK) response containing an SDP answer to the initial INVITE request (the IBCF has not sent the first SDP answer to the originating IMS network)

Terminating IMS network

Originating IMS network		IBCF		CSCFs/etc

Early dialog#0   Early dialog#1

Early dialog#2

F16: UPDATE [SDP offer]   F15: 200 OK (INV) [SDP answer]

Call-ID: c0, From-tag: t0, To-tag: f0   Call-ID: c1, From-tag: f1, To-tag: t1

F17: 200 (UPD) [SDP answer]   Confirmed dialog#1

Call-ID: c0, From-tag: t0, To-tag: f0

F18: 200 OK (INV)

Call-ID: c0, From-tag: f0, To-tag: t0

Confirmed dialog#0

F19: ACK   F20: ACK
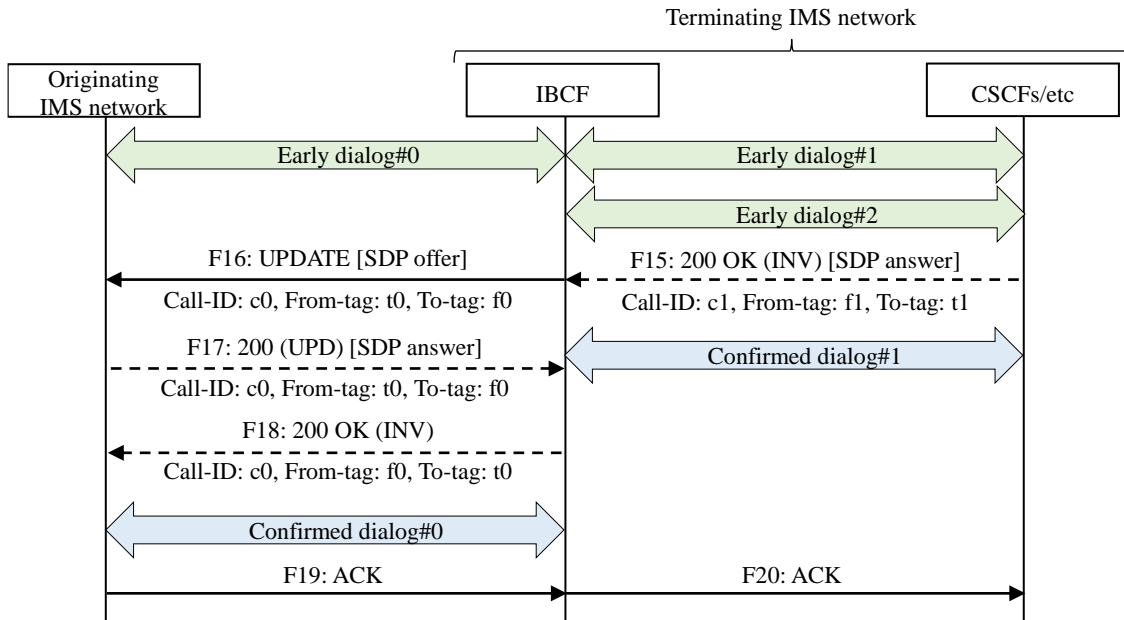
Appendix Figure vi.2.3-2 / JJ-90.30: Procedure when an IBCF receives a 200 (OK) response containing an SDP to the initial INVITE request
(There are differences between the received SDP answer and the last sent/received SDP answer to/from the originating IMS network)
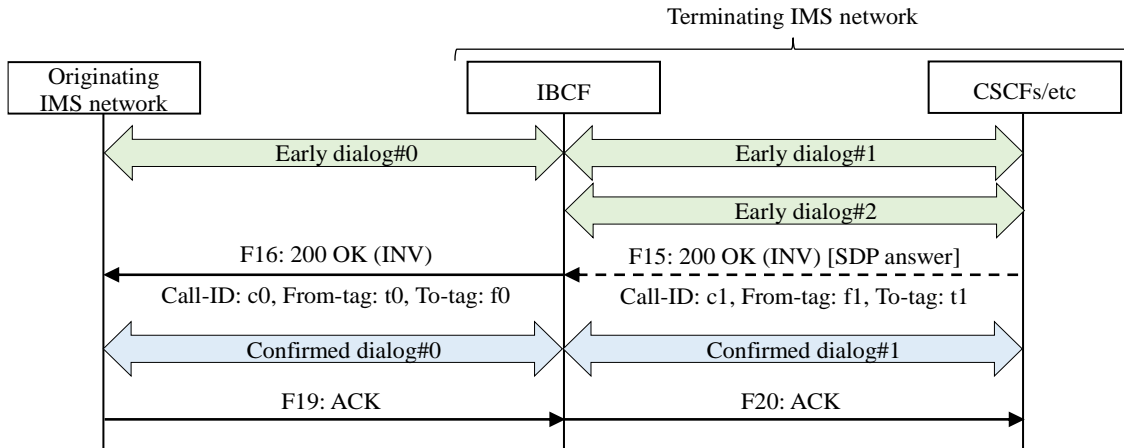
Appendix Figure vi.2.3-3 / JJ-90.30: Procedure when an IBCF receives a 200 (OK) response containing an SDP to the initial INVITE request
(No difference between the received SDP answer and the last sent/received SDP answer to/from the originating IMS network)
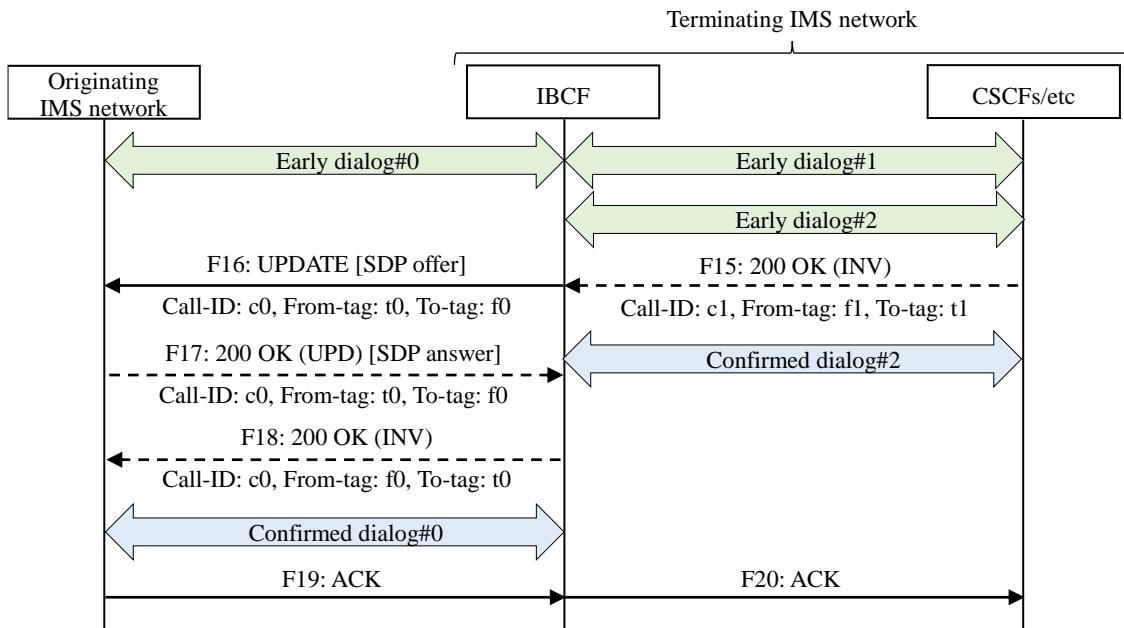
Appendix Figure vi.2.3-4 / JJ-90.30: Procedure when an IBCF receives a 200 (OK) response to the initial INVITE request not containing SDP
(There are differences between the last sent/received SDP answer on the dialog on which 200 (OK) response is received and the last sent/received SDP answer to/from the originating IMS network)
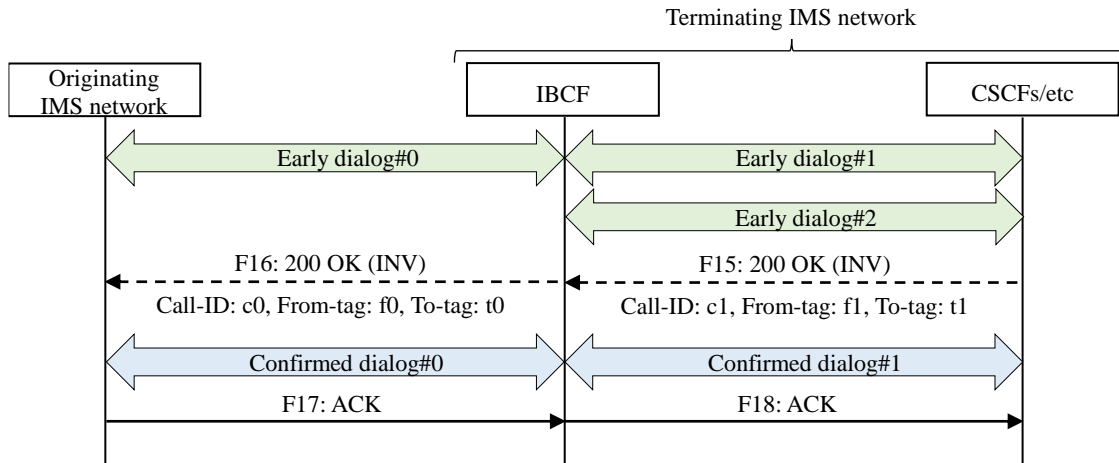
Appendix Figure vi.2.3-5 / JJ-90.30: Procedure when an IBCF receives a 200 (OK) response to the initial INVITE request not containing SDP
(No difference between the last sent/received SDP answer on the dialog on which 200 (OK) response is received and the last sent/received SDP answer to/from the originating IMS network)

vi.2.4. Procedure after receiving 200 (OK) response to the initial INVITE request

After receiving the 200 (OK) response to the initial INVITE request from the terminating IMS network side and sends the 200 (OK) response to the originating IMS network, one of the following procedures is applied to releases the dialogs except the confirmed dialog.

1) The IBCF sends a CANCEL request.

2) The IBCF sends an ACK request after receiving a 200 (OK) response to the initial INVITE request and releases the dialog by sending a BYE request.

3) The IBCF receives a 199 (Early Dialog Terminated) response.

Appendix vii (Informative):
Examples of callflow and message coding

vii.1. General

This Appendix provides examples of call flow and message coding at the II-NNI.

Note that the call flow and message coding examples described in this Appendix are intended to help with system implementation. Actual call flows and message coding may be different from the examples described in this Appendix depending on the selection of "option items". Note also that the contents of these call flow and message coding examples do not guarantee any connectivity or quality of calls.

vii.2. Call flow examples

The call flow examples described in this Appendix and corresponding Subclauses are listed in Appendix Table vii.2-1.

Appendix Table vii.2-1 / JJ-90.30: List of call flow examples

						Subclause
					Call flow scenario	**Subclause**
1	Basic call (A call is released by originating side)	1	Originating from fixed IMS network and terminating in fixed IMS network	1	no early media	vii.2.1.1.1
				2	early media not using "100rel"	vii.2.1.1.2
				3	early media using gateway model	vii.2.1.1.3
		2	Originating from mobile IMS network and terminating in mobile IMS network (precondition mechanism applied)	1	Initial condition: resources of both originating and terminating side are not reserved	vii.2.1.2.1
				2	Initial condition: resource of originating side is not reserved, resource of terminating side is reserved	vii.2.1.2.2
				3	Initial condition: resource of originating side is reserved, resource of terminating side is not reserved	vii.2.1.2.3
		3	Originating from fixed IMS network and terminating in mobile IMS network			vii.2.1.3
		4	Originating from mobile IMS network and terminating in fixed IMS network			vii.2.1.4
2	Basic call (A call is released by terminating side)					vii.2.2
3	Call cancellation (A call is cancelled before establishment)					vii.2.3
4	Call failure (Destination identity is unallocated number)					vii.2.4
5	Number translation	1	Presentation of "mapped-from" number is allowed			vii.2.5.1
		2	Presentation of "mapped-from" number is restricted			vii.2.5.2
		3	Presentation of two "mapped-from" numbers is allowed			vii.2.5.3
		4	Presentation of both "mapped-from" number and diverting number is allowed			vii.2.5.4
		5	Presentation of both "mapped-from" number and diverting number is restricted			vii.2.5.5
6	International calls	1	Early media is provided			vii.2.6.1
		2	Early media is not provided			vii.2.6.2
7	Restoration detection using OPTIONS					vii.2.7

vii.2.1. Basic call (Call is released by originating side)

vii.2.1.1. Originating from fixed IMS network and terminating in fixed IMS network

This Subclause describes the examples of call flow and message coding when the call is established between originating and terminating fixed IMS network, and the established call is released by the originating fixed IMS network. The preconditions applied for the examples are listed in Appendix Table vii.2.1.1-1.

Appendix Table vii.2.1.1-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	IP version	Use IPv4.	-	-
2	Control plane transport	Use UDP.	-	-
3	SIP Session timer (timer)	Set "300" in the Session-Expires header field.	Appendix Table i.4-7 Item 1	Apply the condition described in the 3rd column of this table.
4	Reliability of provisional response (100rel)	Use.	-	-
5	Calling party number presentation and restriction	Presentation is allowed.	-	-
6	Originating area information	Set.	-	-
7	Speech codec	Use G.711 μ-law.	Appendix Table i.4-11 Item 1	Select that speech codecs other than G.711 μ-law is not use in the 3rd column of this table.
8	DTMF (telephone-event)	Use.	-	-

vii.2.1.1.1.　No early media

This Subclause describes the examples when early media is not provided from the terminating IMS network.

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 03-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

Originating IMS network		Terminating IMS network
	F1: INVITE →	
	← F2: 100 Trying	
	← F3: 180 Ringing	
	F4: PRACK →	
	← F5: 200 OK (PRACK)	
	← F6: 200 OK (INVITE)	
	F7: ACK →	
	Call in progress	
	F8: UPDATE →	
	← F9: 200 OK (UPDATE)	
	F10: BYE →	
	← F11: 200 OK (BYE)	
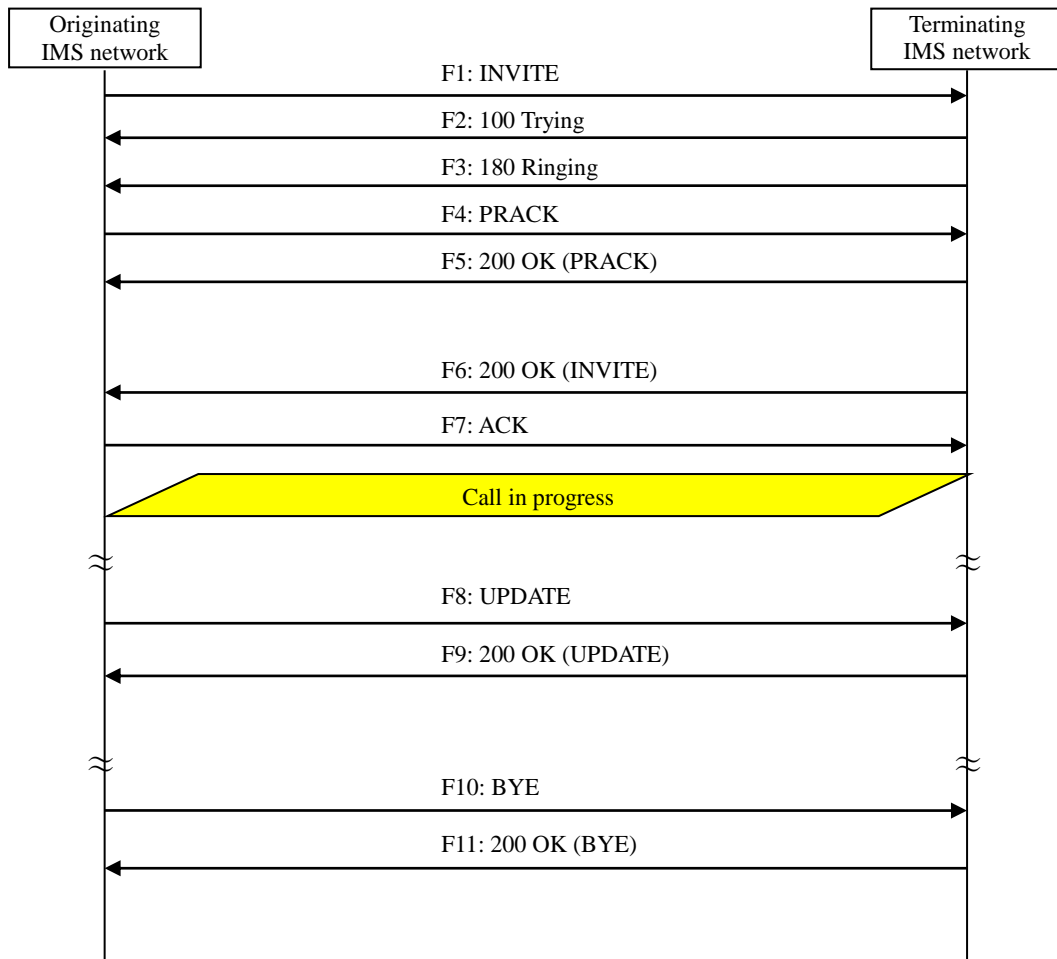
Appendix Figure vii.2.1.1.1-1 / JJ-90.30: Basic call (Call is released by originating side)
<Originating from fixed IMS network and terminating in fixed IMS network, no early media>

F1: INVITE

```
INVITE  sip:+81322222222;npdi@example2.ne.jp;user=phone  SIP/2.0
Via: SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+81322222222@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  IEEE-802.3ah;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
```

```
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81322222222@example2.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.2.234:5060;transport=tcp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 1
Content-Length: 0
```

F4: PRACK

```
PRACK sip:192.0.2.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
Content-Length: 0
```

F6: 200 OK (INVITE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.2.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 300;refresher=uac
Content-Type: application/sdp
Content-Length: 207


v=0
o=- 82917391739 82917391739 IN IP4 192.0.2.222
s=-
c=IN IP4 192.0.2.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F7: ACK

```
ACK sip:192.0.2.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK34567890cdefghij
Max-Forwards: 70
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 ACK
Content-Length: 0
```

F8: UPDATE

```
UPDATE sip:192.0.2.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijk
Max-Forwards: 70
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Supported: timer
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Length: 0
```

F9: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijk
To: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 UPDATE
```

```
Contact:  <sip:192.0.2.234:5060;transport=udp>
Require:  timer
Session-Expires:  300;refresher=uac
Content-Length:  0
```

F10: BYE

```
BYE  sip:192.0.2.234:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK56789012efghijkl
Max-Forwards:  70
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  4  BYE
Content-Length:  0
```

F11: 200 OK (BYE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK56789012efghijkl
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  4  BYE
Content-Length:  0
```

vii.2.1.1.2. Early media not using "100rel"

The preconditions applied for the examples are both those described in Subclause vii.2.1.1 and those described in Appendix Table vii.2.1.1.2-1. The example is described as the case that early media is sent using an SDP answer for an early media in the 18x response not containing the SIP option tag "100rel" in the Require header field.

Appendix Table vii.2.1.1.2-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	Reliability of provisional response (100rel)	Not use	-	-
2	Early media	Use	-	-

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 03-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222
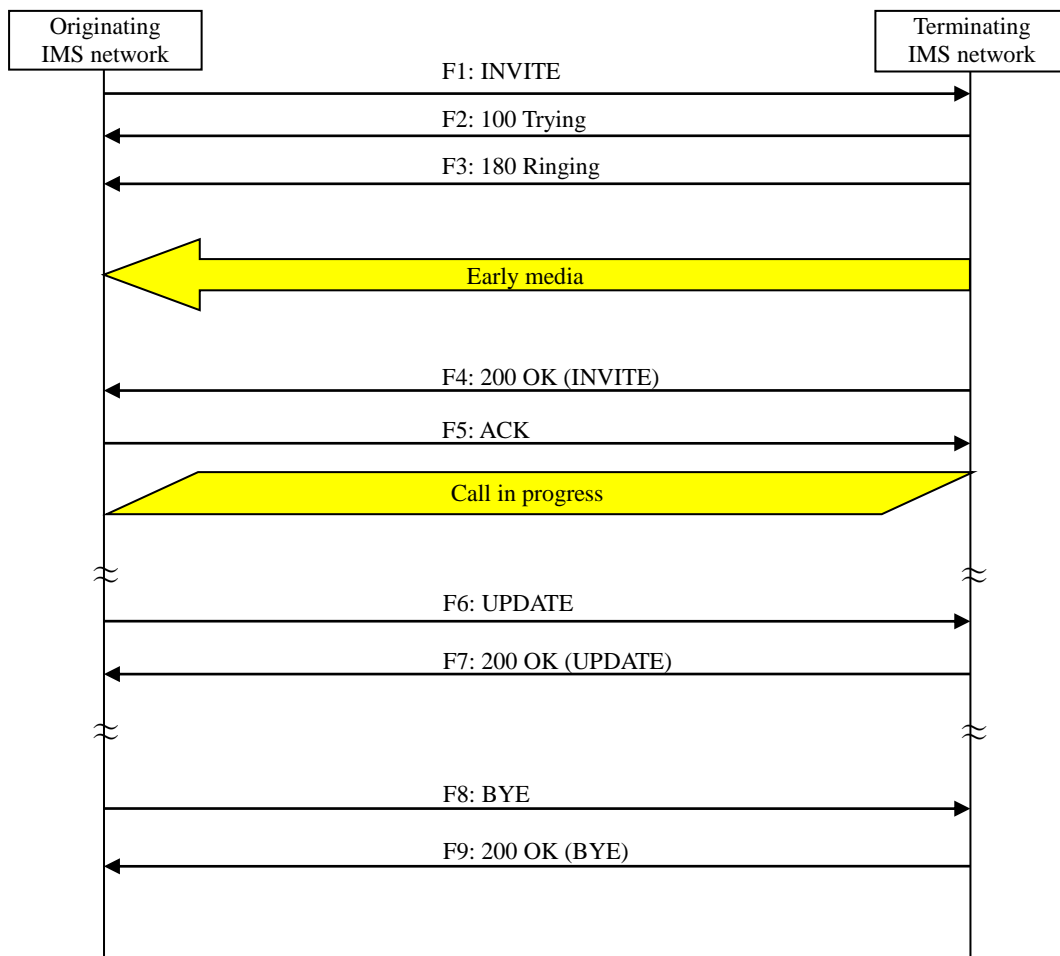
Appendix Figure vii.2.1.1.2-1 / JJ-90.30: Basic call (Call is released by originating side)(early media)
<Originating from fixed IMS network and terminating in fixed IMS network (early media not using "100rel")>

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.1.1.1.

F3: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Early-Media:  sendrecv
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,UPDATE
Content-Type:  application/sdp
Content-Length:  207

v=0
o=-  82917391739  82917391739  IN  IP4  192.0.2.222
s=-
c=IN  IP4  192.0.2.222
t=0  0
```

```
m=audio  20000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

F4: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,UPDATE
Content-Length:  207

v=0
o=-  82917391739  82917391739  IN  IP4  192.0.2.222
s=-
c=IN  IP4  192.0.2.222
t=0  0
m=audio  20000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

The coding examples for F5 - F9 are omitted because they are equivalent to F7 - F11 described in Subclause vii.2.1.1.1.

vii.2.1.1.3.   Early media using gateway model

The preconditions applied for the examples are both those described in Subclause vii.2.1.1 and those described in Appendix Table vii.2.1.1.3-1. The example is described as the case that the early media is provided by using gateway model.

Appendix Table 2.1.1.3-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	Early media	Use	-	-

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 03-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222
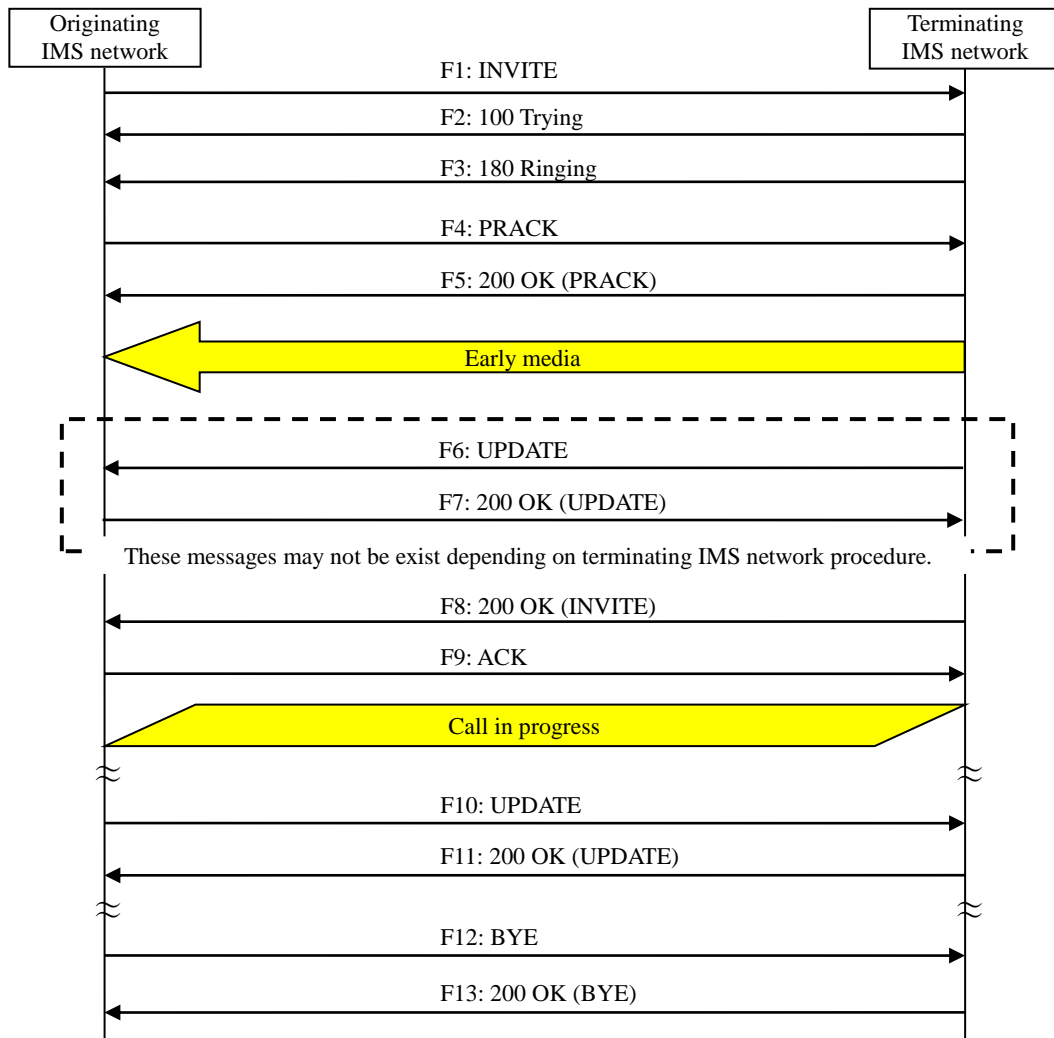
**Appendix Figure 2.1.1.3-1 / JJ-90.30: Basic call (Call is released by originating side)(early media)
<Originating from fixed IMS network and terminating in fixed IMS network (early media using gateway model)>**

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.1.1.1.

F3: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Early-Media:  sendrecv
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100rel
RSeq:  1
Content-Type:  application/sdp
Content-Length:  207
```

```
v=0
o=- 82917391739 82917391739 IN IP4 192.0.2.222
s=-
c=IN IP4 192.0.2.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

The coding examples for F4 - F5 are omitted because they are equivalent to those described in Subclause vii.2.1.1.1.

F6: UPDATE

```
UPDATE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.234:5060;branch=z9hG4bK123456789cdefghi
Max-Forwards: 70
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.2.234:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82917391739 82917391739 IN IP4 192.0.2.222
s=-
c=IN IP4 192.0.2.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F7: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.234:5060;branch=z9hG4bK123456789cdefghi
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82664419472 82664419473 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F8: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81322222222@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=GSTN.example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  timer
Session-Expires:  300;refresher=uac
Content-Length:  0
```

The coding examples for F9 - F13 are omitted because they are equivalent to F7 - F11 described in Subclause vii.2.1.1.1.

vii.2.1.2. Originating from mobile IMS network and terminating in mobile IMS network (precondition mechanism applied)

This Subclause describes the examples of call flow and message coding when the call is established between originating and terminating mobile IMS network, and the established call is released by the originating mobile IMS network. The preconditions applied for the examples are listed in Appendix Table vii.2.1.2-1.

Appendix Table vii.2.1.2-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	IP version	Use IPv4.	-	-
2	Control plane transport	Use UDP.	-	-
3	SIP Session timer (timer)	Set "300" in the Session-Expires header field.	Appendix Table i.4-7 Item 1	Apply the condition described in the 3rd column of this table.
4	Reliability of provisional response (100rel)	Use.	-	-
5	Negotiation of resource reservation (precondition)	Use.	Appendix Table i.4-7 Item 3	Yes
6	Calling party number presentation and restriction	Presentation is allowed.	-	-
7	Originating area information	Set .	-	-
8	Speech codec	Both AMR-WB and AMR(-NB) are offered, then AMR-WB is selected.	Appendix Table i.4-11 Item 1	Apply AMR-WB in the 3rd column of this table.
9	DTMF (telephone-event)	Use.	-	-

vii.2.1.2.1.   Initial condition: resources of both originating and terminating side are not reserved

This Subclause describes the basic call flow and message coding examples under the condition of Appendix Table vii.2.1.2-1 when resources of both originating and terminating side are not initially reserved.

SIP domain name: example3.ne.jp
Calling party number: 080-3333-3333
IP address (control plane): 192.0.3.123
IP address (user plane): 192.0.3.111

SIP domain name: example4.ne.jp
Calling party number: 090-4444-4444
IP address (control plane): 192.0.3.234
IP address (user plane): 192.0.3.222

Originating IMS network		Terminating IMS network

F1: INVITE →

F2: 100 Trying ←

F3: 183 Session Progress ←

F4: PRACK →

F5: 200 OK (PRACK) ←

F6: UPDATE →

F7: 200 OK (UPDATE) ←

F8: 180 Ringing ←

F9: PRACK →

F10: 200 OK (PRACK) ←

F11: 200 OK (INVITE) ←

F12: ACK →

Call in progress

F13: UPDATE →

F14: 200 OK (UPDATE) ←

F15: BYE →

F16: 200 OK (BYE) ←
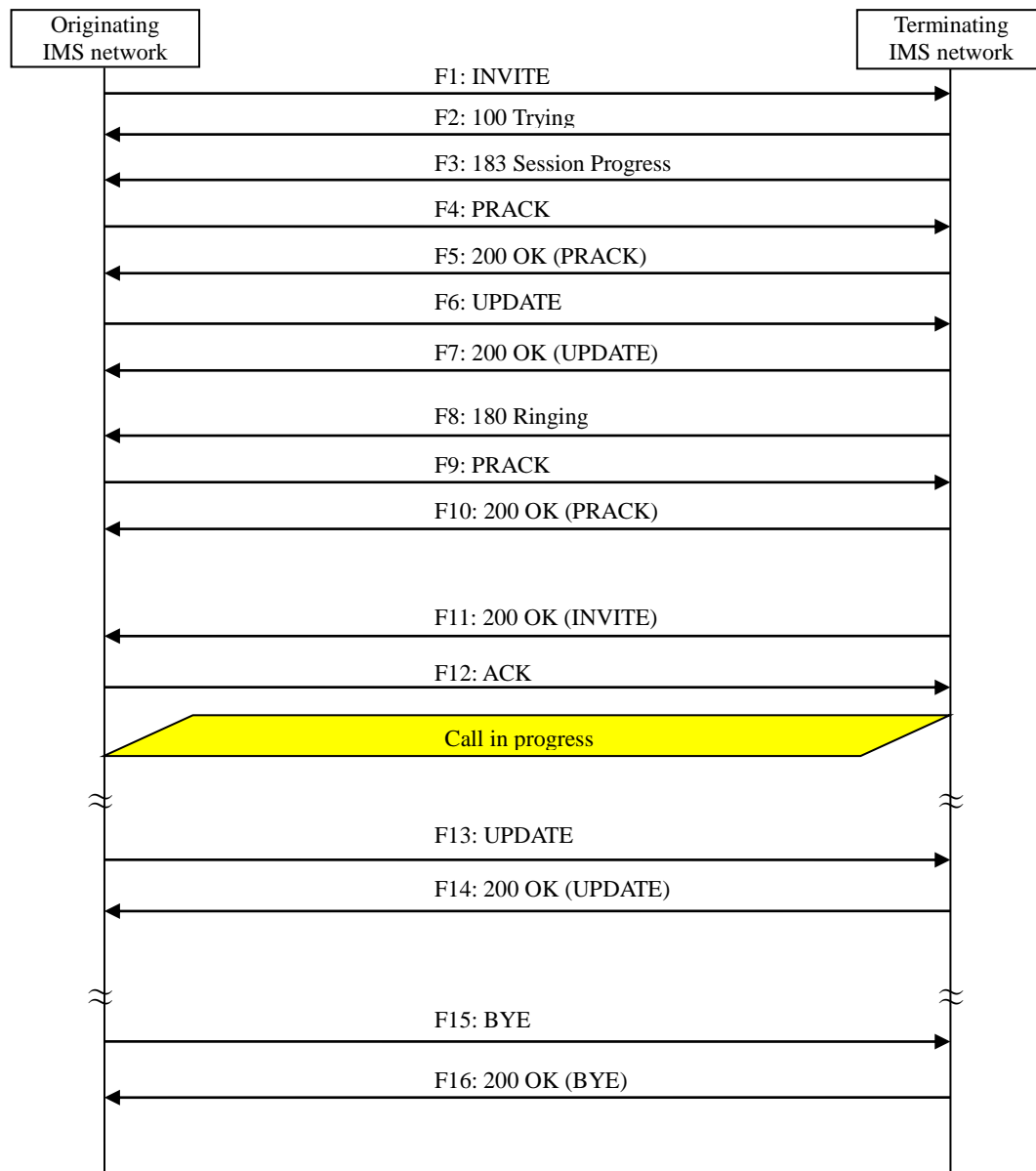
Appendix Figure vii.2.1.2.1-1 / JJ-90.30: Basic call (Call is released by originating side)
< Originating from mobile IMS network and terminating in mobile IMS network
(Initial condition: resources of both originating and terminating side are not reserved)>

F1: INVITE

```
INVITE  sip:+819044444444;npdi@example4.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819044444444@example4.ne.jp;user=phone>
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+818033333333;cpc=ordinary>
```

```
P-Asserted-Identity: <sip:+818033333333;cpc=ordinary@example3.ne.jp;user=phone>
P-Early-Media: supported
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;operator-specific-GI=32000;network-provided
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: 100rel,timer,precondition
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Type: application/sdp
Content-Length: 537

v=0
o=- 82664419472 82664419472 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 96 97 98 99 100 101
b=AS:41
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 octet-align=1
a=rtpmap:98 AMR/8000/1
a=rtpmap:99 AMR/8000/1
a=fmtp:99 octet-align=1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 183 Session Progress

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel,precondition
RSeq: 1
Content-Type: application/sdp
Content-Length: 379

v=0
```

```
o=- 82917391739 82917391739 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 96 100
b=AS:41
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F4: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 2 PRACK
Content-Length: 0
```

F6: UPDATE

```
UPDATE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefghij
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.3.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 355

v=0
o=- 82664419472 82664419473 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 96 100
b=AS:41
a=curr:qos local sendrecv
```

```
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F7: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefghij
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.3.234:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 359

v=0
o=- 82917391739 82917391740 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 96 100
b=AS:41
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F8: 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 2
Content-Length: 0
```

F9: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghijk
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
```

```
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 4 PRACK
RAck: 2 1 INVITE
Content-Length: 0
```

F10: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghijk
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 4 PRACK
Content-Length: 0
```

F11: 200 OK (INVITE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F12: ACK

```
ACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK34567890cdefghijkl
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 ACK
Content-Length: 0
```

F13: UPDATE

```
UPDATE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 5 UPDATE
Contact: <sip:192.0.3.123:5060;transport=udp>
Supported: timer
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Length: 0
```

F14: 200 OK (UPDATE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  5  UPDATE
Contact:  <sip:192.0.3.234:5060;transport=udp>
Require:  timer
Session-Expires:  300;refresher=uac
Content-Length:  0
```

F15: BYE

```
BYE  sip:192.0.3.234:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
Max-Forwards:  70
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  6  BYE
Content-Length:  0
```

F16: 200 OK (BYE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  6  BYE
Content-Length:  0
```

vii.2.1.2.2.   Initial condition: resource of originating side is not reserved, resource of terminating side is reserved

This Subclause describes the basic call flow and message coding examples under the condition of Appendix Table vii.2.1.2-1 when resource of originating side is not initially reserved, and resource of terminating side is initially reserved. Note that the basic call flow and message coding examples after confirmed dialog has been established are omitted because they are equivalent to those described in Subclause vii.2.1.2.1.

SIP domain name: example3.ne.jp
Calling party number: 080-3333-3333
IP address (control plane): 192.0.3.123
IP address (user plane): 192.0.3.111

SIP domain name: example4.ne.jp
Calling party number: 090-4444-4444
IP address (control plane): 192.0.3.234
IP address (user plane): 192.0.3.222

```
 ┌─────────────┐                                          ┌─────────────┐
 │ Originating │                                          │ Terminating │
 │ IMS network │                                          │ IMS network │
 └──────┬──────┘                                          └──────┬──────┘
        │              F1: INVITE                                │
        │───────────────────────────────────────────────────────>│
        │              F2: 100 Trying                            │
        │<───────────────────────────────────────────────────────│
        │              F3: 183 Session Progress                  │
        │<───────────────────────────────────────────────────────│
        │              F4: PRACK                                  │
        │───────────────────────────────────────────────────────>│
        │              F5: 200 OK (PRACK)                        │
        │<───────────────────────────────────────────────────────│
        │              F6: UPDATE                                 │
        │───────────────────────────────────────────────────────>│
        │              F7: 200 OK (UPDATE)                       │
        │<───────────────────────────────────────────────────────│
        │              F8: 180 Ringing                           │
        │<───────────────────────────────────────────────────────│
        │              F9: PRACK                                  │
        │───────────────────────────────────────────────────────>│
        │              F10: 200 OK (PRACK)                       │
        │<───────────────────────────────────────────────────────│
        │              F11: 200 OK (INVITE)                      │
        │<───────────────────────────────────────────────────────│
        │              F12: ACK                                   │
        │───────────────────────────────────────────────────────>│
        │             Call in progress                           │
        │                                                         │
```

Appendix Figure vii.2.1.2.2-1 / JJ-90.30: Basic call (Call is released by originating side)
<Originating from mobile IMS network and terminating in mobile IMS network
(Initial condition: resource of originating side is not reserved, resource of terminating side is reserved)>

F1: INVITE

```
INVITE  sip:+819044444444;npdi@example4.ne.jp;user=phone  SIP/2.0
Via: SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819044444444@example4.ne.jp;user=phone>
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+818033333333;cpc=ordinary>
P-Asserted-Identity:  <sip:+818033333333;cpc=ordinary@example3.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  3GPP-E-UTRAN-FDD;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer,precondition
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
Content-Length:  537

v=0
o=- 82664419472 82664419472 IN IP4 192.0.3.111
```

```
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 96 97 98 99 100 101
b=AS:41
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 octet-align=1
a=rtpmap:98 AMR/8000/1
a=rtpmap:99 AMR/8000/1
a=fmtp:99 octet-align=1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 183 Session Progress

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel,precondition
RSeq: 1
Content-Type: application/sdp
Content-Length: 383

v=0
o=- 82917391739 82917391739 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 96 100
b=AS:41
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
```

```
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F4: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 2 PRACK
Content-Length: 0
```

F6: UPDATE

```
UPDATE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefghij
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.3.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 359

v=0
o=- 82664419472 82664419473 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 96 100
b=AS:41
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F7: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefghij
```

```
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  3  UPDATE
Contact:  <sip:192.0.3.234:5060;transport=udp>
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type:  application/sdp
Content-Length:  359


v=0
o=- 82917391739 82917391740 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 96 100
b=AS:41
a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

The coding examples for F8 – F12 are omitted because they are equivalent to those described in Subclause vii.2.1.2.1.

vii.2.1.2.3. Initial condition: resource of originating side is reserved, resource of terminating side is not reserved

This Subclause describes the basic call flow and message coding examples under the condition of Appendix Table vii.2.1.2-1 when resource of originating side is initially reserved, and resource of terminating side is not initially reserved. Note that the basic call flow and message coding examples after confirmed dialog has been established are omitted because they are equivalent to those described in Subclause vii.2.1.2.1.

SIP domain name: example3.ne.jp
Calling party number: 080-3333-3333
IP address (control plane): 192.0.3.123
IP address (user plane): 192.0.3.111

SIP domain name: example4.ne.jp
Calling party number: 090-4444-4444
IP address (control plane): 192.0.3.234
IP address (user plane): 192.0.3.222

```
┌──────────────┐                                    ┌──────────────┐
│  Originating │                                    │  Terminating │
│  IMS network │                                    │  IMS network │
└──────┬───────┘                                    └──────┬───────┘
       │              F1: INVITE                           │
       │ ─────────────────────────────────────────────────>│
       │              F2: 100 Trying                        │
       │ <─────────────────────────────────────────────────│
       │              F3: 183 Session Progress              │
       │ <─────────────────────────────────────────────────│
       │              F4: PRACK                              │
       │ ─────────────────────────────────────────────────>│
       │              F5: 200 OK (PRACK)                    │
       │ <─────────────────────────────────────────────────│
       │              F6: 180 Ringing                       │
       │ <─────────────────────────────────────────────────│
       │              F7: PRACK                              │
       │ ─────────────────────────────────────────────────>│
       │              F8: 200 OK (PRACK)                    │
       │ <─────────────────────────────────────────────────│
       │              F9: 200 OK (INVITE)                   │
       │ <─────────────────────────────────────────────────│
       │              F10: ACK                              │
       │ ─────────────────────────────────────────────────>│
       │              Call in progress                      │
       │                                                    │
```

Appendix Figure vii.2.1.2.3-1 / JJ-90.30: Basic call (Call is released by originating side)
<Originating from mobile IMS network and terminating in mobile IMS network
(Initial condition: resource of originating side is reserved, resource of terminating side is not reserved)>

F1: INVITE

```
INVITE  sip:+819044444444;npdi@example4.ne.jp;user=phone  SIP/2.0
Via: SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.123:5060;transport=udp>
Privacy: none
P-Asserted-Identity: <tel:+818033333333;cpc=ordinary>
P-Asserted-Identity: <sip:+818033333333;cpc=ordinary@example3.ne.jp;user=phone>
P-Early-Media: supported
P-Access-Network-Info: 3GPP-E-UTRAN-FDD;operator-specific-GI=32000;network-provided
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: 100rel,timer,precondition
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Type: application/sdp
Content-Length: 541

v=0
o=- 82664419472 82664419472 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
```
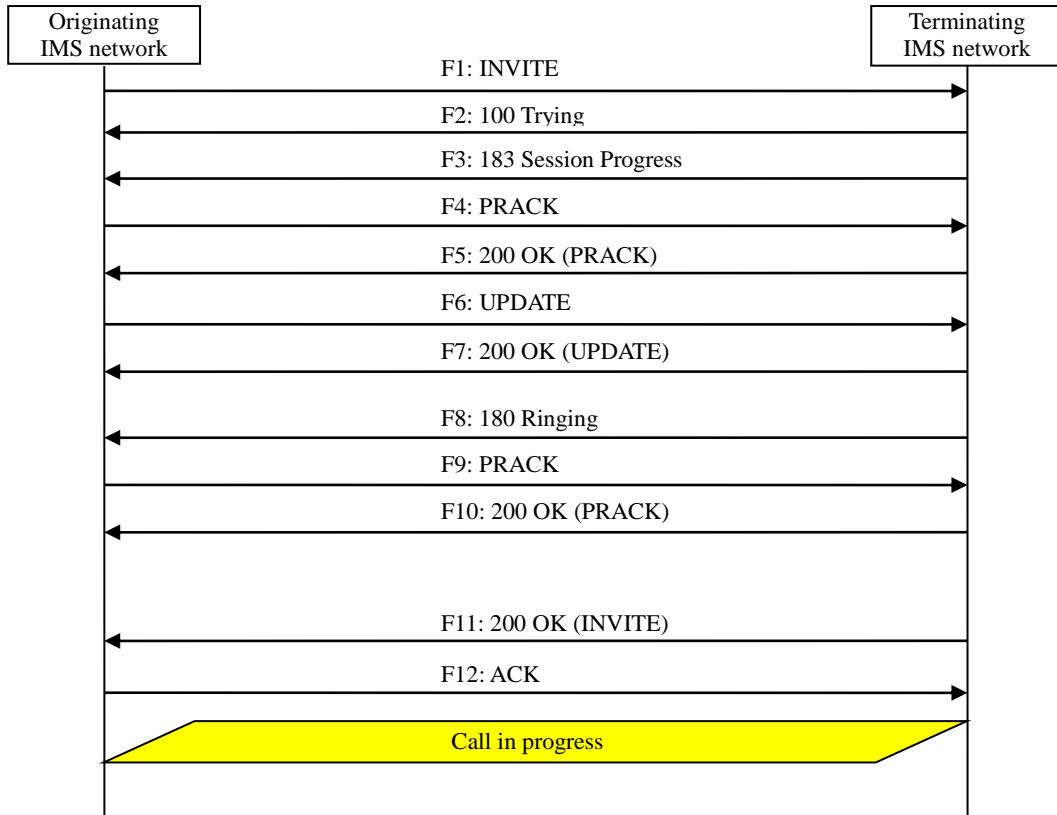
```
m=audio 10000 RTP/AVP 96 97 98 99 100 101
b=AS:41
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:97 AMR-WB/16000/1
a=fmtp:97 octet-align=1
a=rtpmap:98 AMR/8000/1
a=rtpmap:99 AMR/8000/1
a=fmtp:99 octet-align=1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 183 Session Progress

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel,precondition
RSeq: 1
Content-Type: application/sdp
Content-Length: 355

v=0
o=- 82917391739 82917391739 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 96 100
b=AS:41
a=curr:qos local none
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:96 AMR-WB/16000/1
a=rtpmap:100 telephone-event/16000
a=fmtp:100 0-15
a=sendrecv
a=ptime:20
```

F4: PRACK

```
PRACK  sip:192.0.3.234:5060 SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  2  PRACK
RAck:  1  1  INVITE
Content-Length:  0
```

F5: 200 OK (PRACK)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  2  PRACK
Content-Length:  0
```

F6: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100rel
RSeq:  2
Content-Length:  0
```

F7: PRACK

```
PRACK  sip:192.0.3.234:5060 SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghij
Max-Forwards: 70
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  3  PRACK
RAck:  2  1  INVITE
Content-Length:  0
```

F8: 200 OK (PRACK)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghij
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  3  PRACK
```

```
Content-Length: 0
```

F9: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=3GPP-E-UTR
AN-FDD.example4.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  timer
Session-Expires:  300;refresher=uac
Content-Length:  0
```

F10: ACK

```
ACK  sip:192.0.3.234:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK34567890cdefghijk
Max-Forwards:  70
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  ACK
Content-Length:  0
```

vii.2.1.3. Originating from fixed IMS network and terminating in mobile IMS network

This Subclause shows the call flow and message coding examples when the call is originated from the fixed IMS network to the mobile IMS network.

In this example, the originating fixed IMS network offers G.711 μ-law, and then the terminating mobile IMS network answers G.711 μ-law, under the preconditions listed in Appendix Table vii.2.1.3-1.

Appendix Table vii.2.1.3-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	IP version	Use IPv4.	-	-
2	Control plane transport	Use TCP.	Appendix Table i.4-2 Item 2	Yes
3	SIP Session timer (timer)	Set "300" in the Session-Expires header field.	Appendix Table i.4-7 Item 1	Apply the condition described in the 3rd column of this table.
4	Reliability of provisional response (100rel)	Use.	-	-
5	Negotiation of resource reservation (precondition)	Not Use	Appendix Table i.4-7 Item 3	No
6	Calling party number presentation and restriction	Presentation is allowed.	-	-
7	Originating area information	Set .	-	-
8	Early media	Use	-	-
9	Speech codec	Use G.711 μ-law.	Appendix Table i.4-11 Item 1	Apply no speech codec other than G.711μ-law described in the 3rd column of this table.
10	DTMF (telephone-event)	Use.	-	-

NOTE: Subclause 2.4.1 of [IR.92] specifies the procedures at the terminating UE in the mobile IMS network when the originating side does not support a precondition mechanism.

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example4.ne.jp
Calling party number: 090-4444-4444
IP address (control plane): 192.0.3.234
IP address (user plane): 192.0.3.222

```
┌──────────────┐                                      ┌──────────────┐
│ Originating  │                                      │ Terminating  │
│    fixed     │                                      │   mobile     │
│ IMS network  │                                      │ IMS network  │
└──────┬───────┘                                      └──────┬───────┘
       │            F1: INVITE (SDP: G.711 μ-law)            │
       │ ─────────────────────────────────────────────────► │
       │               F2: 100 Trying                        │
       │ ◄───────────────────────────────────────────────── │
       │     F3: 183 Session Progress （SDP: G.711 μ-law)    │
       │ ◄───────────────────────────────────────────────── │
       │                  F4: PRACK                          │
       │ ─────────────────────────────────────────────────► │
       │               F5: 200 OK (PRACK)                    │
       │ ◄───────────────────────────────────────────────── │
       │            Early media (G.711 μ-law)                │
       │ ◄══════════════════════════════════════════════════│
       │               F6: 180 Ringing                       │
       │ ◄───────────────────────────────────────────────── │
       │                  F7 PRACK                           │
       │ ─────────────────────────────────────────────────► │
       │               F8: 200 OK (PRACK)                    │
       │ ◄───────────────────────────────────────────────── │
   ┌ ─ ┼ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┼ ─ ┐
       │        F9: UPDATE (SDP: G.711 μ-law)                │
   │   │ ◄───────────────────────────────────────────────── │   │
       │     F10: 200 OK (UPDATE) (SDP: G.711 μ-law)         │
   │   │ ─────────────────────────────────────────────────► │   │
   └ ─ ┼ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┼ ─ ┘
```

This part may not exist depending on the procedures at the terminating mobile IMS network.

```
       │               F11: 200 OK (INVITE)                  │
       │ ◄───────────────────────────────────────────────── │
       │                  F12: ACK                           │
       │ ─────────────────────────────────────────────────► │
       │          Call in progress (G.711 μ-law)             │
       │ ◄══════════════════════════════════════════════════►│
       │                 F13: UPDATE                         │
       │ ─────────────────────────────────────────────────► │
       │               F14: 200 OK (UPDATE)                  │
       │ ◄───────────────────────────────────────────────── │
       │                  F15: BYE                           │
       │ ─────────────────────────────────────────────────► │
       │               F16: 200 OK (BYE)                     │
       │ ◄───────────────────────────────────────────────── │
```
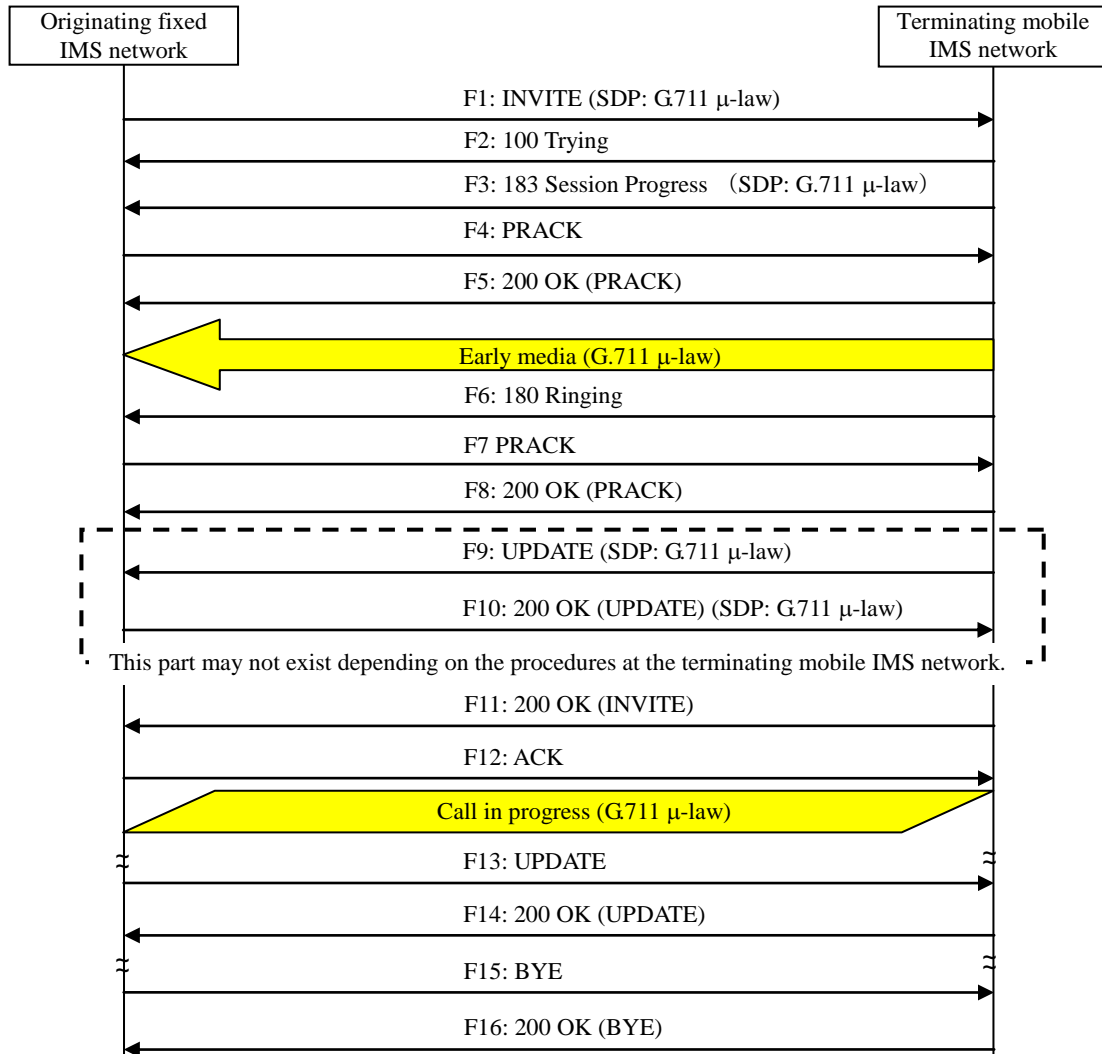
Appendix Figure vii.2.1.3-1 / JJ-90.30: Basic call (Call is released by originating side)
<Originating from fixed IMS network and terminating in mobile IMS network>

F1: INVITE

```
INVITE  sip:+819044444444;npdi@example4.ne.jp;user=phone SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819044444444@example4.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  IEEE-802.3ah;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
Content-Length:  199

v=0
o=-  82664419472  82664419472  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  10000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

. F2: 100 Trying

```
SIP/2.0  100  Trying
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+819044444444@example4.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Content-Length:  0
```

F3: 183 Session Progress

```
SIP/2.0  183  Session  Progress
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+819044444444@example4.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.234:5060;transport=udp>
P-Early-Media:  sendrecv
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.
example4.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100relRSeq:  1
Content-Type:  application/sdp
Content-Length:  199

v=0
```

```
o=- 82917391739 82917391739 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F4: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
RAck: 1 1 INVITE
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
Content-Length: 0
```

F6: 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.
example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 2
Content-Length: 0
```

F7: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghijk
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 PRACK
RAck: 2 1 INVITE
Content-Length: 0
```

F8: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghijk
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 PRACK
Content-Length: 0
```

F9: UPDATE

```
UPDATE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.234:5060;branch=z9hG4bK123456789cdefghi
Max-Forwards: 70
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.3.234:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82917391739 82917391740 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F10: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.234:5060;branch=z9hG4bK123456789cdefghi
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82664419472 82664419473 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F11: 200 OK (INVITE)

```
SIP/2.0 200 OK
```

```
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.
example4.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F12: ACK

```
ACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK34567890cdefghijkl
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 ACK
Content-Length: 0
```

F13: UPDATE

```
UPDATE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijklm
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 4 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Supported: timer
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Length: 0
```

F14: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijklm
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 4 UPDATE
Contact: <sip:192.0.3.234:5060;transport=udp>
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F15: BYE

```
BYE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK56789012efghijklmo
Max-Forwards: 70
To: <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
```

```
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  5  BYE
Content-Length:  0
```

F16: 200 OK (BYE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK56789012efghijklmo
To:  <sip:+819044444444@example4.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  5  BYE
Content-Length:  0
```

vii.2.1.4. Originating from mobile IMS network and terminating in fixed IMS network

This Subclause shows the call flow and message coding examples when the call is originated from the mobile IMS network to the fixed IMS network. Note that the resource of the originating IMS network is initially reserved as with Subclase vii.2.1.2.3.

In this example, the originating mobile IMS network offers G.711 μ-law, and then the terminating fixed IMS network answers G.711 μ-law, under the precondition listed in Appendix Table vii.2.1.4-1.

Appendix Table vii.2.1.4-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	**Condition**	**Option item**	**Option item selection**
1	IP version	Use IPv4.	-	-
2	Control plane transport	Use UDP.	-	-
3	SIP Session timer (timer)	Set "300" in the Session-Expires header field.	Appendix Table i.4-7 Item 1	Apply the condition described in the 3rd column of this table.
4	Reliability of provisional response (100rel)	Use.	-	-
5	Negotiation of resource reservation (precondition)	Not Use	Appendix Table i.4-7 Item 3	No
6	Calling party number presentation and restriction	Presentation is allowed.	-	-
7	Originating area information	Set .	-	-
8	Speech codec	Use G.711 μ-law.	Appendix Table i.4-11 Item 1	Apply no speech codec other than G.711μ-law described in the 3rd column of this table.
9	DTMF (telephone-event)	Use.	-	-

SIP domain name: example3.ne.jp
Calling party number: 080-3333-3333
IP address (control plane): 192.0.3.123
IP address (user plane): 192.0.3.111

SIP domain name: example1.ne.jp
Called party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

Originating mobile IMS network — Terminating fixed IMS network

F1: INVITE (SDP: G.711 μ-law)
F2: 100 Trying
F3: 180 Ringing
F4: PRACK
F5: 200 OK (PRACK)
F6: 200 OK (INVITE) (SDP: G.711 μ-law)
F7: ACK
Call in progress (G.711 μ-law)
F8: UPDATE
F9: 200 OK (UPDATE)
F10: BYE
F11: 200 OK (BYE)
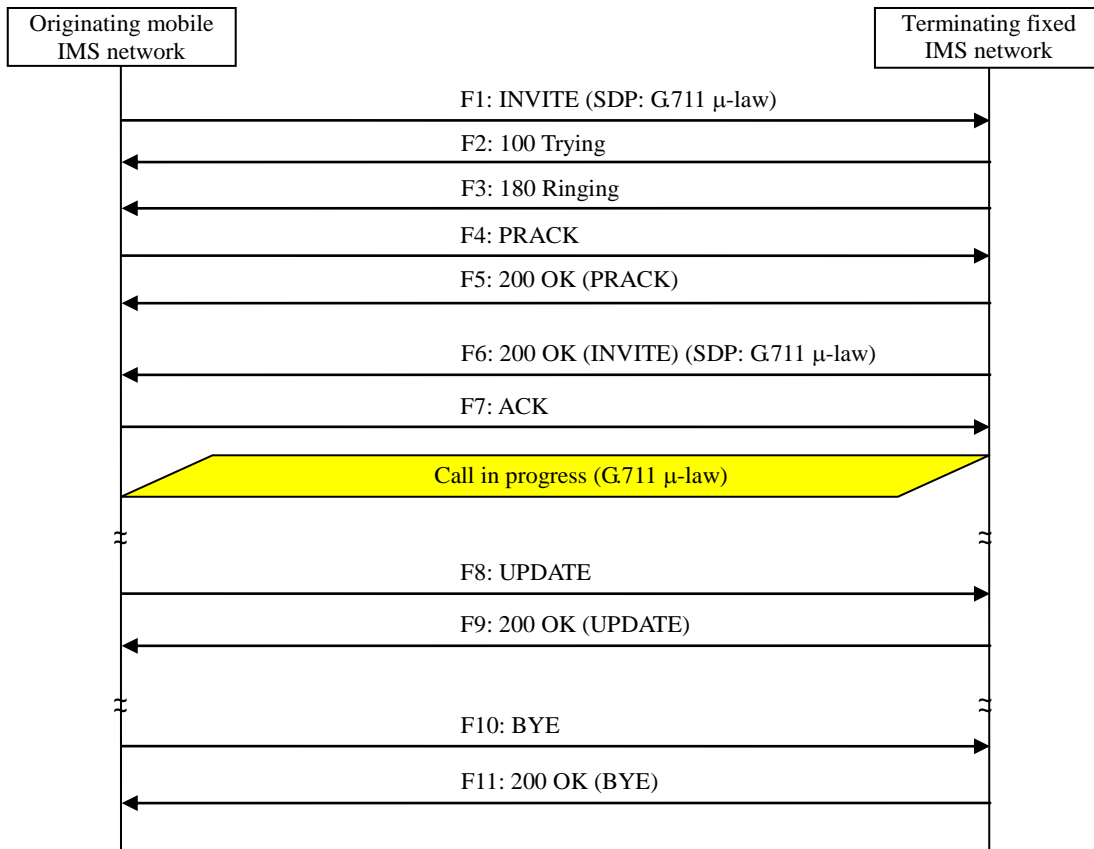
Appendix Figure vii.2.1.4-1 / JJ-90.30: Basic call (Call is released by originating side)
<Originating from mobile IMS network and terminating in fixed IMS network>

F1: INVITE

```
INVITE  sip:+81311111111;npdi@example1.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip: +81311111111@example1.ne.jp;user=phone>
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+818033333333;cpc=ordinary>
P-Asserted-Identity:  <sip:+818033333333;cpc=ordinary@example3.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  3GPP-E-UTRAN-FDD;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
Content-Length:  214

v=0
o=- 82664419472 82664419472 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 101 0
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
```

F2: 100 Trying

```
SIP/2.0  100  Trying
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81311111111@example1.ne.jp;user=phone>
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Content-Length:  0
```

F3: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=IEEE-802.3ah.
example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100rel
RSeq:  1
Content-Length:  0
```

F4: PRACK

```
PRACK  sip:192.0.2.123:5060 SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards:  70
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  2  PRACK
RAck:  1  1  INVITE
Content-Length:  0
```

F5: 200 OK (PRACK)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  2  PRACK
Content-Length:  0
```

F6: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=3GPP-E-UTRAN-FDD.example3.ne.jp;term-ioi=IEEE-802.3ah.
example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  timer
Session-Expires:  300;refresher=uac
Content-Type:  application/sdp
Content-Length:  202

v=0
o=-  82917391739  82917391739  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  20000  RTP/AVP  101  0
a=rtpmap:101  telephone-event/8000
a=fmtp:101  0-15
a=ptime:20
a=rtpmap:0  PCMU/8000
```

F7: ACK

```
ACK  sip:192.0.2.123:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK34567890cdefghij
Max-Forwards:  70
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  ACK
Content-Length:  0
```

F8: UPDATE

```
UPDATE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
Max-Forwards: 70
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.3.123:5060;transport=udp>
Supported: timer
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Length: 0
```

F9: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F10: BYE

```
BYE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
Max-Forwards: 70
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 4 BYE
Content-Length: 0
```

F11: 200 OK (BYE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
To: <sip:+81311111111@example1.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+818033333333@example3.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 4 BYE
Content-Length: 0
```

vii.2.2. Basic call (Call is released by terminating side)

This Subclause describes the call flow and message coding examples when the established call is released by the originating IMS network. The preconditions applied for the examples are equivalent to those described in Subclause vii.2.1.1.1.

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 090-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

```
Originating                              Terminating
IMS network                              IMS network
     |                                        |
     |------------ F1: INVITE --------------->|
     |<----------- F2: 100 Trying ------------|
     |<----------- F3: 180 Ringing -----------|
     |------------ F4: PRACK ---------------->|
     |<----------- F5: 200 OK (PRACK) --------|
     |                                        |
     |<----------- F6: 200 OK (INVITE) -------|
     |------------ F7: ACK ------------------>|
     |////////////  Call in progress  ////////|
     |<----------- F8: BYE -------------------|
     |------------ F9: 200 OK (BYE) --------->|
     |                                        |
```
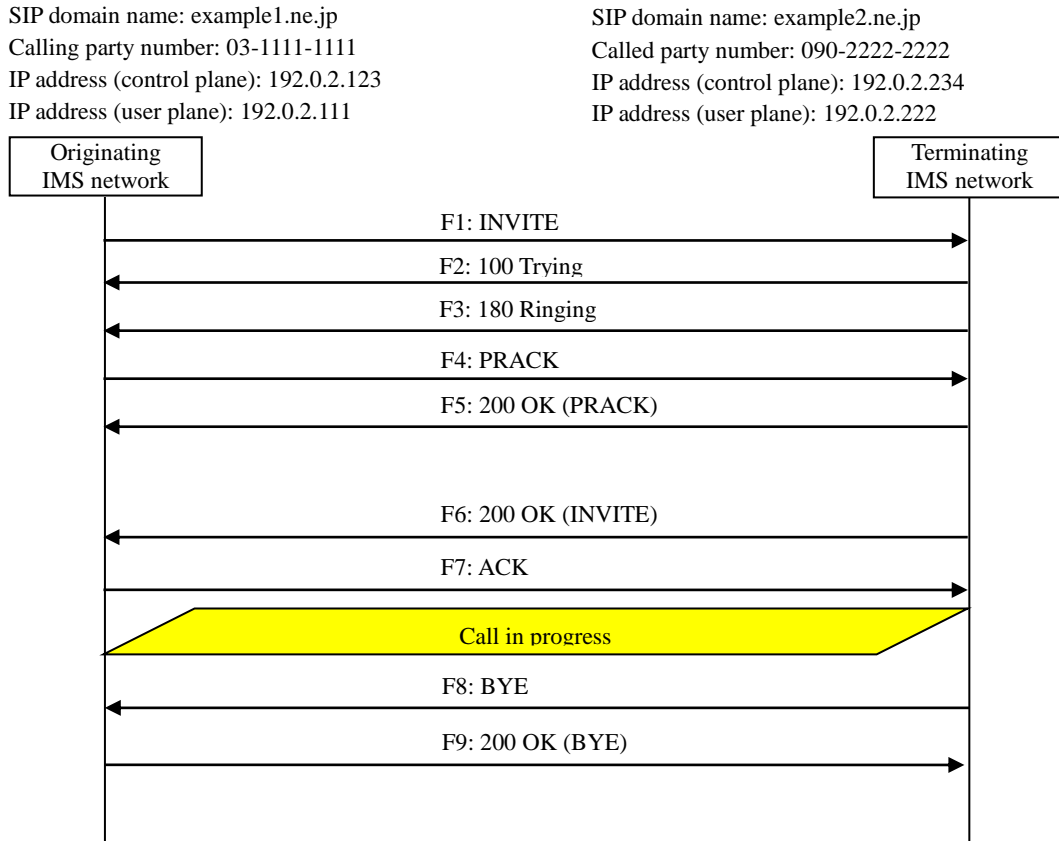
Appendix Figure vii.2.2-1 / JJ-90.30: Basic call (Call is released by terminating side)

The coding examples for F1 - F7 are omitted because they are equivalent to those described in Subclause vii.2.1.1.1

F8: BYE

```
BYE  sip:192.0.2.123:5060  SIP/2.0
Via: SIP/2.0/UDP  192.0.2.234:5060;branch=z9hG4bK98765432stuvwxyz
Max-Forwards:  70
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From:  <sip:+819022222222@example2.ne.jp;user=phone>;tag=9876zyxw
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  100  BYE
Content-Length:  0
```

F9: 200 OK (BYE)

```
SIP/2.0  200  OK
Via: SIP/2.0/UDP  192.0.2.234:5060;branch=z9hG4bK98765432stuvwxyz
To:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
From:  <sip:+819022222222@example2.ne.jp;user=phone>;tag=9876zyxw
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  100  BYE
Content-Length:  0
```

vii.2.3. Call cancellation (Call is cancelled before establishment)

This Subclause describes the call flow and message coding examples when the call is cancelled by the originating IMS network before establishment. The preconditions applied for the examples are equivalent to those described in

Subclause vii.2.1.1.1.

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 090-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

```
Originating                                    Terminating
IMS network                                    IMS network

     |------------------ F1: INVITE ------------------>|
     |<----------------- F2: 100 Trying ---------------|
  ┌ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┐
     |<----------------- F3: 180 Ringing --------------|
  |                                                    |
     |------------------ F4: PRACK ------------------->|
  |                                                    |
     |<----------------- F5: 200 OK (PRACK) -----------|
  |        These messages may not be exist depending   |
  └ ─ ─ ─ ─on the timing of cancellation.─ ─ ─ ─ ─ ─ ─ ┘
     |------------------ F6: CANCEL ------------------>|
     |<----------------- F7: 200 OK (CANCEL) ----------|
     |<----------------- F8: 487 Request Terminated ---|
     |------------------ F9: ACK --------------------->|
```
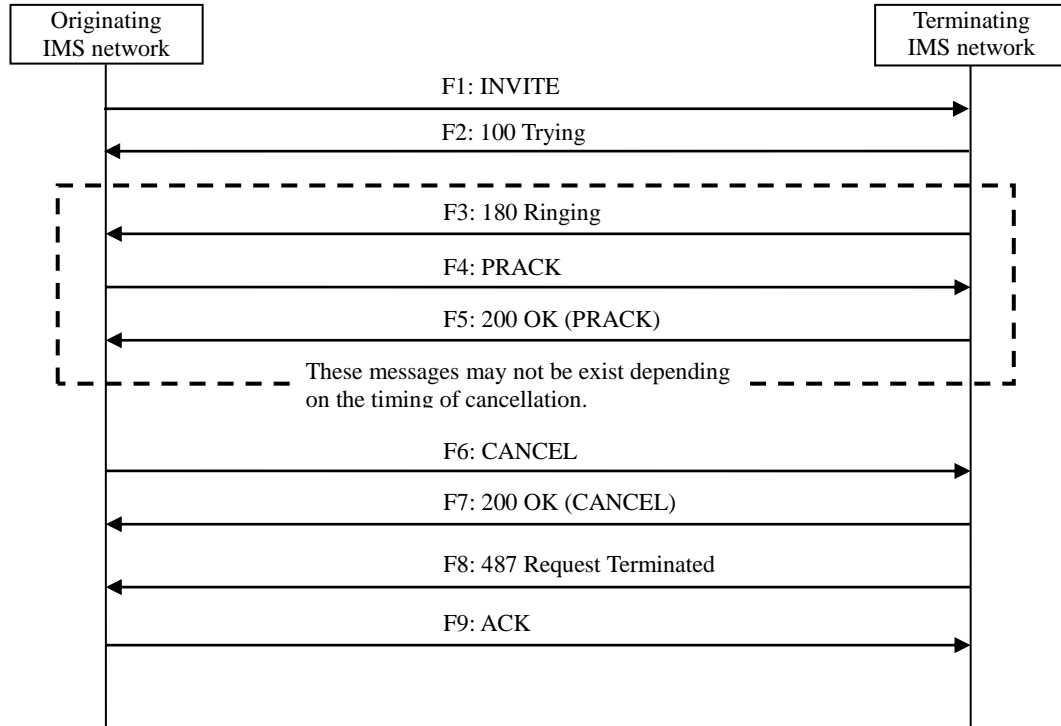
Appendix Figure vii.2.3-1 / JJ-90.30: Call cancellation (Call is cancelled before establishment)

The coding examples for F1 - F5 are omitted because they are equivalent to those described in Subclause vii.2.1.1.1.

F6: CANCEL

```
CANCEL  sip:+819022222222;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819022222222@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  CANCEL
Content-Length:  0
```

F7: 200 OK (CANCEL)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+819022222222@example2.ne.jp;user=phone>;tag=5555eeee
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  CANCEL
Content-Length:  0
```

F8: 487 Request Terminated

```
SIP/2.0  487  Request  Terminated
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
```

```
To:  <sip:+819022222222@example2.ne.jp;user=phone>;tag=5555eeee
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1 INVITE
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.
example2.ne.jp
Content-Length:  0
```

F9: ACK

```
ACK  sip:+819022222222;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819022222222@example2.ne.jp;user=phone>;tag=5555eeee
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  ACK
Content-Length:  0
```

vii.2.4. Call failure (Called party number is unallocated number)

This Subclause describes the call flow and message coding examples when the called party number is an unallocated number for the terminating IMS network. The preconditions applied for the examples are equivalent to those described in Subclause vii.2.1.1.

SIP domain name: example1.ne.jp   SIP domain name: example2.ne.jp
Calling party number: 03-1111-1111   Called party number: 03-0000-0000
IP address (control plane): 192.0.2.123   IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.111   IP address (user plane): 192.0.2.222
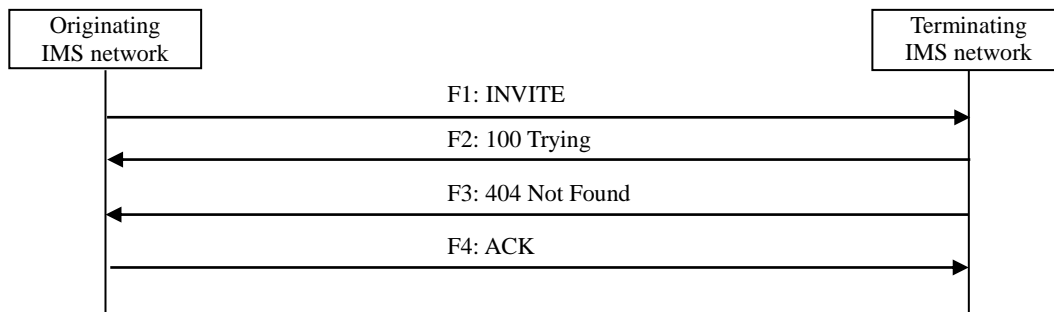
Appendix Figure vii.2.4-1 / JJ-90.30: Call failure (Called party number is unallocated number)

F1: INVITE

```
INVITE  sip:+81300000000;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+81300000000@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  IEEE-802.3ah;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
```

```
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
Content-Length:  207

v=0
o=-  82664419472  82664419472  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  10000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0  100  Trying
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81300000000@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Content-Length:  0
```

F3: 404 Not Found

```
SIP/2.0  404  Not Found
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81300000000@example2.ne.jp;user=phone>;tag=7777gggg
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Reason:  Q.850;cause=1
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=example2.ne.jp
Content-Length:  0
```

F4: ACK

```
ACK  sip:+81300000000;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:+81300000000@example2.ne.jp;user=phone>;tag=7777gggg
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  ACK
Content-Length:  0
```

vii.2.5. Number translation

This Subclause describes the message coding examples when number translation is performed in the service provider network and the history information of number translation is delivered over the II-NNI. The preconditions applied for the examples are both those described in Subclause vii.2.1.1 and the those in Appendix Table vii.2.5-1 and the calling party number is different from the contractor number. In this example, the service provider network removes the contractor number of originating user and the originating area information from the request before forwarding the request.

Appendix Table vii.2.5-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	Service access number translation	Use.	-	-

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
Contractor number: 03-1111-1234
IP address (control plane): 192.0.2.150
IP address (user plane): 192.0.2.155

SIP domain name: example2.ne.jp
Logical number: 0120-012-345
Actual number: 03-2222-2222
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example3.ne.jp
Called party number: 03-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

Appendix Figure vii.2.5-1 / JJ-90.30: Number translation

vii.2.5.1. Presentation of "mapped-from" number is allowed

This Subclause describes the message coding examples when the presentation of the "mapped-from" number is allowed. The "mapped-from" logical number is 0120-012-345 and the "mapped-to" actual number is 03-2222-2222.

F1: INVITE

```
INVITE  sip:+81120012345;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg1
Max-Forwards:  70
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  1  INVITE
Contact:  <sip:192.0.2.150:5060;transport=udp>
```

```
P-Asserted-Identity: <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity: <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media: supported
P-Access-Network-Info: IEEE-802.3ah;operator-specific-GI=32000;network-provided
P-Charge-Info: <tel:+81311111234>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Privacy: none
Supported: 100rel,timer
Session-Expires: 300;refresher=uac
Min-SE: 300
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.155
s=-
c=IN IP4 192.0.2.155
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.150:5060;branch= z9hG4bK12345678abcdefg1
To: <sip:+81120012345@example2.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.150
CSeq: 1 INVITE
Content-Length: 0
```
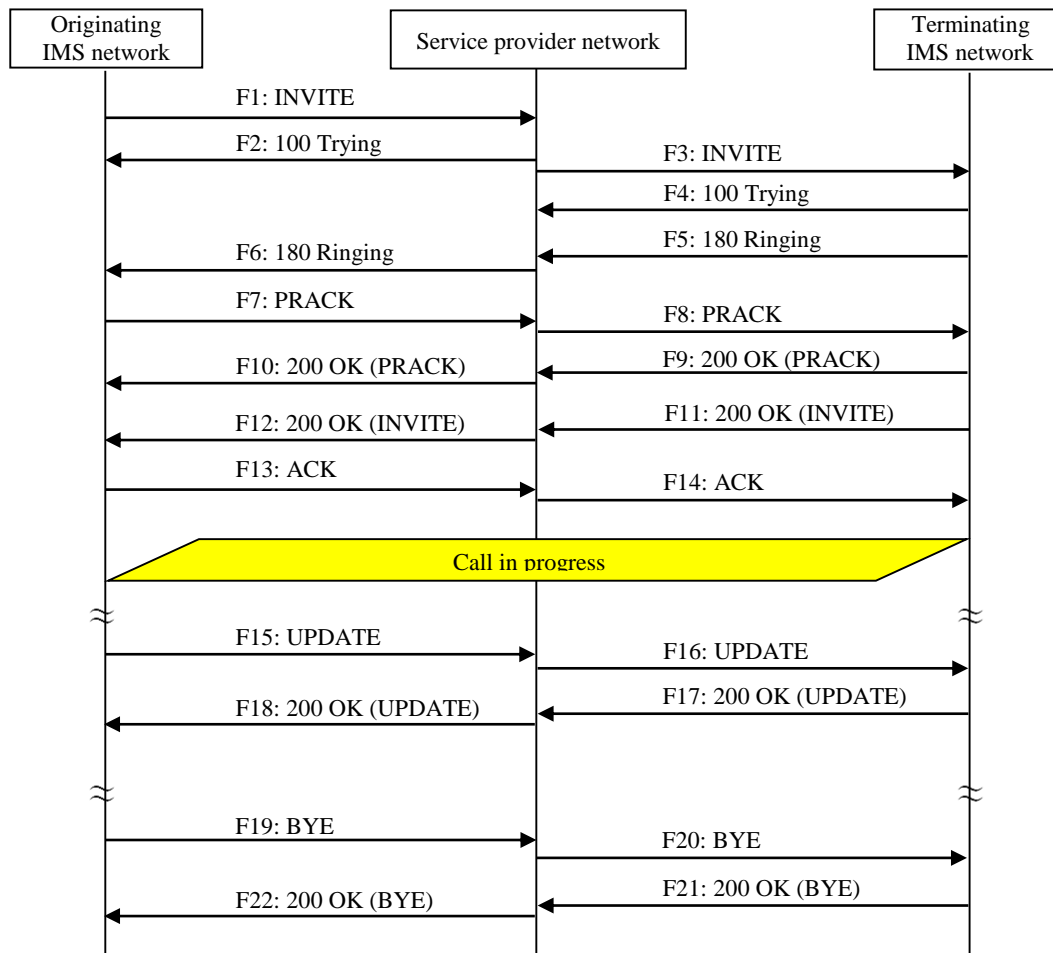
F3: INVITE

```
INVITE sip:+81322222222;npdi@example3.ne.jp;user=phone;cause=380 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
Max-Forwards: 69
To: <sip:+81120012345@example2.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID: poiuytrewq987654@192.0.2.123
CSeq: 21 INVITE
Contact: <sip:192.0.2.123:5060;transport=udp>
Privacy: none
P-Asserted-Identity: <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity: <sip:+81311111111;cpc=ordinary@example2.ne.jp;user=phone>
P-Early-Media: supported
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: 100rel,timer
Session-Expires: 300;refresher=uac
Min-SE: 300
History-Info: <sip:+81120012345@example2.ne.jp;user=phone>;index=1
History-Info: <sip:+81322222222@example3.ne.jp;user=phone;cause=380>;index=1.1;mp=1
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
```

```
t=0  0
m=audio  10000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

F4: 100 Trying

```
SIP/2.0  100  Trying
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  INVITE
Content-Length:  0
```

F5: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example2.ne.jp;term-ioi=GSTN.example3.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100rel
RSeq:  31
Content-Length:  0
```

F6: 180 Ringing

```
SIP/2.0  180  Ringing
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg1
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  100rel
RSeq:  11
Content-Length:  0
```

F7: PRACK

```
PRACK sip:192.0.2.123:5060 SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg2
Max-Forwards:  70
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  2  PRACK
RAck:  11  1  INVITE
Content-Length:  0
```

F8: PRACK

```
PRACK  sip:192.0.2.234:5060 SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh2
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  22  PRACK
RAck:  31  21  INVITE
Content-Length:  0
```

F9: 200 OK (PRACK)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh2
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  22  PRACK
Content-Length:  0
```

F10: 200 OK (PRACK)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg2
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  2  PRACK
Content-Length:  0
```

F11: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  INVITE
Contact:  <sip:192.0.2.234:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example2.ne.jp;term-ioi=GSTN.example3.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  timer
Session-Expires:  300;refresher=uac
Content-Type:  application/sdp
Content-Length:  197

v=0
o=-  82917391739  82917391739  IN  IP4  192.0.2.222
s=-
c=IN  IP4  192.0.2.222
t=0  0
m=audio  20000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

F12: 200 OK (INVITE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg1
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp;term-ioi=.example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require:  timer
Session-Expires:  300;refresher=uac
Content-Type:  application/sdp
Content-Length:  197

v=0
o=-  82917391739  82917391739  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  20000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

F13: ACK

```
ACK  sip:192.0.2.123:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg3
Max-Forwards:  70
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  1  ACK
Content-Length:  0
```

F14: ACK

```
ACK  sip:192.0.2.234:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh3
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  ACK
Content-Length:  0
```

F15: UPDATE

```
UPDATE  sip:192.0.2.123:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg4
Max-Forwards:  70
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  3  UPDATE
Contact:  <sip:192.0.2.150:5060;transport=udp>
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  timer
```

```
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F16: UPDATE

```
UPDATE sip:192.0.2.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefh4
Max-Forwards: 70
To: <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID: poiuytrewq987654@192.0.2.123
CSeq: 23 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F17: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefh4
To: <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID: poiuytrewq987654@192.0.2.123
CSeq: 23 UPDATE
Contact: <sip:192.0.2.234:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F18: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.150:5060;branch=z9hG4bK12345678abcdefg4
To: <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.150
CSeq: 3 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 300;refresher=uac
Content-Length: 0
```

F19: BYE

```
BYE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.150:5060;branch=z9hG4bK12345678abcdefg5
Max-Forwards: 70
To: <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.150
CSeq: 4 BYE
Content-Length: 0
```

F20: BYE

```
BYE  sip:192.0.2.234:5060  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh5
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  24  BYE
Content-Length:  0
```

F21: 200 OK (BYE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh5
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=8765yxwv
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  24  BYE
Content-Length:  0
```

F22: 200 OK (BYE)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg5
To:  <sip:+81120012345@example2.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  4  BYE
Content-Length:  0
```

vii.2.5.2. Presentation of "mapped-from" number is restricted

This Subclause describes the message coding examples when the presentation of the "mapped-from" number is restricted. The "mapped-from" logical number is 0120-012-345 and the "mapped-to" actual number is 03-2222-2222.

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

F3: INVITE

```
INVITE  sip:+81322222222;npdi@example3.ne.jp;user=phone;cause=380  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example2.ne.jp;user=phone>
P-Early-Media:  supported
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
History-Info:  <sip:+81120012345@example2.ne.jp;user=phone?Privacy=history>;index=1
```

```
History-Info:  <sip:+81322222222@example3.ne.jp;user=phone;cause=380>;index=1.1;mp=1
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

The coding examples for F4 - F22 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

vii.2.5.3. Presentation of "mapped-from" number is allowed / number translations are performed twice

This Subclause describes the message coding examples when number translations are performed twice in the service provider network. The presentation of two "mapped-from" numbers are allowed. The first "mapped-from" number is 0120-012-345 and the second "mapped-from" number is 0120-234-567. The last "mapped-to" number is 03-2222-2222.

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

F3: INVITE

```
INVITE sip:+81322222222;npdi@example3.ne.jp;user=phone;cause=380 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
Max-Forwards: 69
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID: poiuytrewq987654@192.0.2.123
CSeq: 21 INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy: none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example2.ne.jp;user=phone>
P-Early-Media: supported
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: 100rel,timer
Session-Expires: 300;refresher=uac
Min-SE: 300
History-Info:  <sip:+81120012345@example2.ne.jp;user=phone>;index=1
History-Info:  <sip:+81120234567@example2.ne.jp;user=phone;cause=380>;index=1.1;mp=1
History-Info:  <sip:+81322222222@example3.ne.jp;user=phone;cause=380>;index=1.1.1;mp=1.1
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

The coding examples for F4 - F22 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

vii.2.5.4. Presentation of both "mapped-from" number and diverting number is allowed

This Subclause describes the message coding examples when one of the CDIV service, Communication Forwarding Unconditionaly (CFU), is performed in the service provider network after number translation. Presentation of both the "mapped-from" logical number and the diverting number is allowed. The "mapped-from" logical number is 0120-012-345, the diverting number is 06-1111-1111, and the diverted-to number is 090-2222-2222. The preconditions applied for the examples are both those described in Subclause vii.2.5 and those in Appendix Table vii.2.5.4-1.

Appendix Table vii.2.5.4-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Selection of option items at the II-NNI	
	Item	Conditions	Option item	Selection of option item
1	Communication DIVersion (CDIV)	Use.	-	-

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

F3: INVITE

```
INVITE  sip:+819022222222;npdi@example3.ne.jp;user=phone;cause=302  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678a  bcdefh1
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  21  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
History-Info:  <sip:+81120012345@example2.ne.jp;user=phone>;index=1
History-Info:  <sip:+81611111111@example2.ne.jp;user=phone;cause=380>;index=1.1;mp=1
History-Info:  <sip:+819022222222@example3.ne.jp;user=phone;cause=302>;index=1.1.1;mp=1.1
Content-Type:  application/sdp
Content-Length:  207

v=0
o=-  82664419472  82664419472  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  10000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

The coding examples for F4 - F22 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

vii.2.5.5. Presentation of both "mapped-from" number and diverting number is restricted

This Subclause describes the message coding examples when one of the CDIV services, Communication Forwarding Unconditional (CFU), is performed in the service provider network after number translation. Presentation of both the "mapped-from" logical number and the diverting number is restricted. The "mapped-from" logical number is 0120-012-345, the diverting number is 06-1111-1111, and the diverted-to number is 090-2222-2222. The preconditions applied for the examples are equivalent to those described in Subclause vii.2.5.4.

The coding examples for F1 - F2 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

F3: INVITE

```
INVITE  sip:+819022222222;npdi@example3.ne.jp;user=phone;cause=302  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
Max-Forwards:  69
To:  <sip:+81120012345@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=2345bcde
Call-ID:  poiuytrewq987654@192.0.2.123
CSeq:  21  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Privacy:  none
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
History-Info:  <sip:+81120012345@example2.ne.jp;user=phone?Privacy=history>;index=1
History-Info:  <sip:+81611111111@example2.ne.jp;user=phone;cause=380?Privacy=history>;index=1.1;mp=1
History-Info:  <sip:+819022222222@example3.ne.jp;user=phone;cause=302>;index=1.1.1;mp=1.1
Content-Type:  application/sdp
Content-Length:  207

v=0
o=-  82664419472  82664419472  IN  IP4  192.0.2.111
s=-
c=IN  IP4  192.0.2.111
t=0  0
m=audio  10000  RTP/AVP  0  96
a=rtpmap:0  PCMU/8000
a=rtpmap:96  telephone-event/8000
a=fmtp:96  0-15
a=ptime:20
```

The coding examples for F4 - F22 are omitted because they are equivalent to those described in Subclause vii.2.5.1.

vii.2.5.6. Communication diversion from an actual number to a logical number

This Subclause describes the message coding examples when number translation is performed after performing the Communication Forwarding Unconditional (CFU) from an actual number to logical number in the service provider network. The diverting number is 06-1111-1111, the diverted-to number is 0120-012-345, and the "mapped-to" number is 06-2222-2222.

SIP domain name: example1.ne.jp
Calling party number: 03-1111-1111
Contractor number: 03-1111-1234
IP address (control plane): 192.0.2.150
IP address (user plane): 192.0.2.155

SIP domain name: example2.ne.jp
Diverting number: 06-1111-1111
Diverted logical number: 0120-012-345
Translated actual number: 03-2222-2222
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example3.ne.jp
Called party number: 03-2222-2222
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

Originating IMS network	Service provider network	Terminating IMS network

F1: INVITE →
F2: 100 Trying ←
F3: INVITE →
F4: 100 Trying ←
F5: 180 Ringing ←
F6: 180 Ringing ←
F7: PRACK →
F8: PRACK →
F9: 200 OK (PRACK) ←
F10: 200 OK (PRACK) ←
F11: 200 OK (INVITE) ←
F12: 200 OK (INVITE) ←
F13: ACK →
F14: ACK →

Call in progress

F15: UPDATE →
F16: UPDATE →
F17: 200 OK (UPDATE) ←
F18: 200 OK (UPDATE) ←

F19: BYE →
F20: BYE →
F21: 200 OK (BYE) ←
F22: 200 OK (BYE) ←

Appendix Figure vii.2.5-1 / JJ-90.30: Number translation

F1: INVITE

```
INVITE  sip:+81611111111;ndpi@example2.ne.jp;user=phone  SIP/2.0
Via: SIP/2.0/UDP  192.0.2.150:5060;branch=z9hG4bK12345678abcdefg1
Max-Forwards:  70
To:  <sip:+81611111111@example2.ne.jp;user=phone>
From:  <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.150
CSeq:  1  INVITE
Contact:  <sip:192.0.2.150:5060;transport=udp>
P-Asserted-Identity:  <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity:  <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media:  supported
P-Access-Network-Info:  IEEE-802.3ah;operator-specific-GI=32000;network-provided
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=IEEE-802.3ah.example1.ne.jp
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Privacy:  none
Supported:  100rel,timer
Session-Expires:  300;refresher=uac
Min-SE:  300
Content-Type:  application/sdp
Content-Length:  207
```

```
v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.155
s=-
c=IN IP4 192.0.2.155
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.150:5060;branch=z9hG4bK12345678abcdefg1
To: <sip:+81611111111@example2.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.150
CSeq: 1 INVITE
Content-Length: 0
```

F3: INVITE

```
INVITE sip:+81622222222;npdi@example3.ne.jp;user=phone;cause=380 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefh1
Max-Forwards: 69
To: <sip:+81611111111@example2.ne.jp;user=phone>
From: <sip:+81311111111@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: poiuytrewq987654@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.2.123:5060;transport=udp>
Privacy: none
P-Asserted-Identity: <tel:+81311111111;cpc=ordinary>
P-Asserted-Identity: <sip:+81311111111;cpc=ordinary@example1.ne.jp;user=phone>
P-Early-Media: supported
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example2.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Supported: 100rel,timer
Session-Expires: 300;refresher=uac
Min-SE: 300
History-Info: <sip:+81611111111@example2.ne.jp;user=phone>;index=1
History-Info: <sip:+81120012345@example2.ne.jp;user=phone;cause=302>;index=1.1;mp=1
History-Info: <sip:+81622222222@example3.ne.jp;user=phone;cause=380>;index=1.1.1;mp=1.1
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```
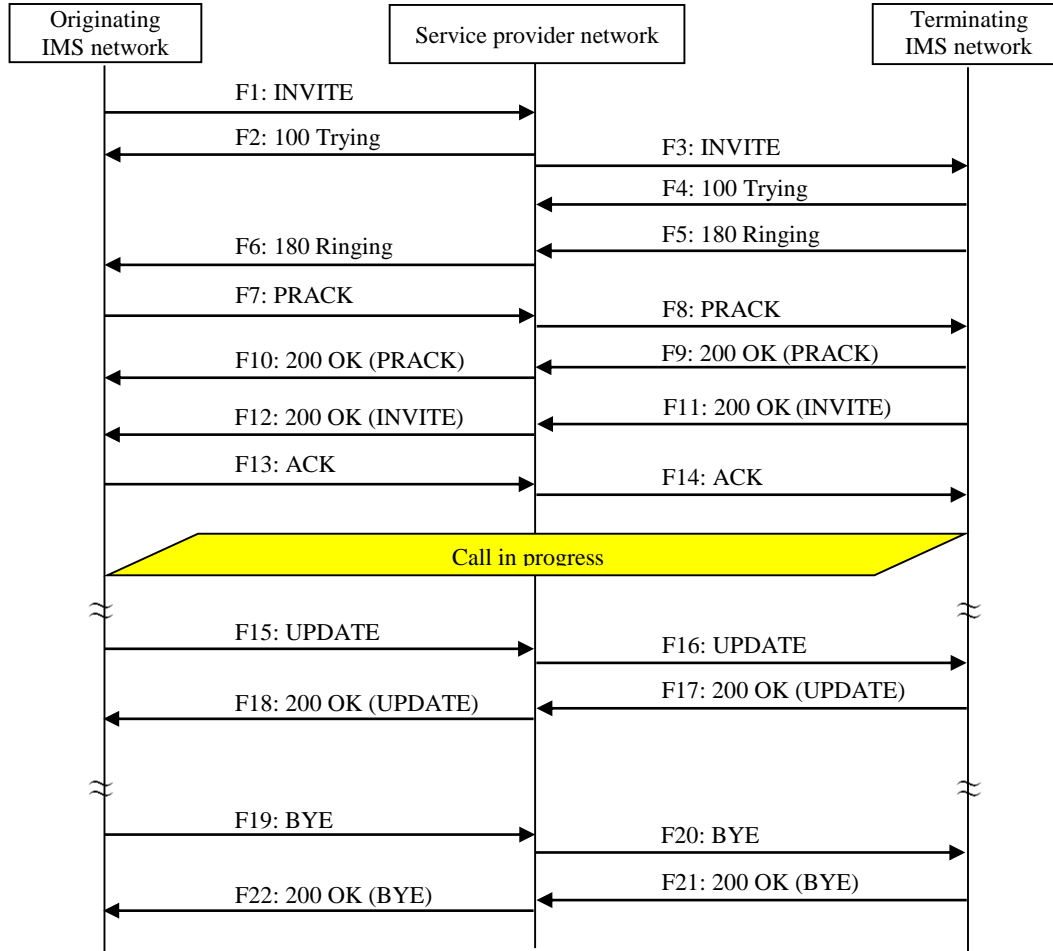
The coding examples for F4 - F22 are omitted.

vii.2.6. International calls

This Subclause shows the call flow and message coding examples when an originating IMS network receives a call from an international network (overseas), then the IMS network initiates a call towards a terminating IMS network.

In this example, the originating IMS network offers G.711 μ-law, and then the terminating IMS network answers G.711 μ-law, under the preconditions listed in Appendix Table vii.2.6-1.

Appendix Table vii.2.6-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Option item selections at the II-NNI	
	Item	Condition	Option item	Option item selection
1	IP version	Use IPv4.	-	-
2	Control plane transport	Use UDP.	-	-
3	SIP Session timer (timer)	Set "180" in the Session-Expires header field.	Appendix Table i.4-7 Item 1	Apply the condition described in the 3rd column of this table.
4	Reliability of provisional response (100rel)	Use.	-	-
5	Negotiation of resource reservation (precondition)	Not Use.	Appendix Table i.4-7 Item 3	No
6	Calling party number presentation and restriction	Presentation is allowed.	-	-
7	Calling party's category	No set.	-	-
8	Originating area information	No set.	-	-
9	Speech codec	Use G.711 μ-law.	Appendix Table i.4-11 Item 1	Apply no speech codec other than G.711 μ-law.
10	DTMF (telephone-event)	Use.	-	-

vii.2.6.1. Early media is provided

This Subclause shows the call flow and message coding examples when early media is provided from the terminating IMS network.

SIP domain name: example1.ne.jp
Calling party number: 1-123-456-7890
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
Called party number: 090-1111-1111
IP address (control plane): 192.0.3.234
IP address (user plane): 192.0.3.222

```
┌─────────────┐                                           ┌─────────────┐
│ Originating │                                           │ Terminating │
│ IMS network │                                           │ IMS network │
└─────────────┘                                           └─────────────┘
       │        F1: INVITE (SDP: G.711 μ-law)                    │
       │ ─────────────────────────────────────────────────────> │
       │        F2: 100 Trying                                   │
       │ <───────────────────────────────────────────────────── │
       │        F3: 183 Session Progress ( SDP: G.711 μ-law)     │
       │ <───────────────────────────────────────────────────── │
       │        F4: PRACK                                        │
       │ ─────────────────────────────────────────────────────> │
       │        F5: 200 OK (PRACK)                               │
       │ <───────────────────────────────────────────────────── │
       │        Early media (G.711 μ-law)                        │
       │ <===================================================    │
       │        F6: 180 Ringing                                  │
       │ <───────────────────────────────────────────────────── │
       │        F7 PRACK                                         │
       │ ─────────────────────────────────────────────────────> │
       │        F8: 200 OK (PRACK)                               │
       │ <───────────────────────────────────────────────────── │
       ┌ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┐
       │        F9: UPDATE (SDP: G.711 μ-law)                    │
       │ <───────────────────────────────────────────────────── │
       │        F10: 200 OK (UPDATE) (SDP: G.711 μ-law)          │
       │ ─────────────────────────────────────────────────────> │
       └ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┘
   This part may not exist depending on the procedures at the terminating mobile IMS network.
       │        F11: 200 OK (INVITE)                            │
       │ <───────────────────────────────────────────────────── │
       │        F12: ACK                                        │
       │ ─────────────────────────────────────────────────────> │
       │        Call in progress (G.711 μ-law)                   │
       │ <===================================================>   │
       │        F13: UPDATE                                     │
       │ ─────────────────────────────────────────────────────> │
       │        F14: 200 OK (UPDATE)                            │
       │ <───────────────────────────────────────────────────── │
       │        F15: BYE                                        │
       │ ─────────────────────────────────────────────────────> │
       │        F16: 200 OK (BYE)                               │
       │ <───────────────────────────────────────────────────── │
```

Appendix Figure vii.2.6.1-1 / JJ-90.30: International call
<Early media is provided>

F1: INVITE

```
INVITE  sip:+819011111111;npdi@example2.ne.jp;user=phone  SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+819011111111@example2.ne.jp;user=phone>
From:  <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  INVITE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Accept:  application/sdp
Supported:  100rel,timer
Session-Expires:  180;refresher=uac
Min-SE:  180
P-Asserted-Identity:  tel:+11234567890;verstat=No-TN-Validation
Privacy:  none
P-Early-Media:  supported
P-Charging-Vector:  icid-value=1234bc9876e;orig-ioi=example1.ne.jp
Content-Type:  application/sdp
```

```
Content-Length: 199

v=0
o=- 82664419472 82664419472 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819011111111@example2.ne.jp;user=phone>
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 183 Session Progress

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819011111111@example2.ne.jp;user=phone>
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Early-Media: sendrecv
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.example2.ne.j
p
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 1
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82917391739 82917391739 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F4: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+819011111111@example2.ne.jp;user=phone>
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
RAck: 1 1 INVITE
```

```
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 2 PRACK
Content-Length: 0
```

F6: 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.example2.ne.j
p
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 2
Content-Length: 0
```

F7: PRACK

```
PRACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghijk
Max-Forwards: 70
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 PRACK
RAck: 2 1 INVITE
Content-Length: 0
```

F8: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK23456789bcdefghijk
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 3 PRACK
Content-Length: 0
```

F9: UPDATE

```
UPDATE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.234:5060;branch=z9hG4bK123456789cdefghi
Max-Forwards: 70
To: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
```

```
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.3.234:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82917391739 82917391740 IN IP4 192.0.3.222
s=-
c=IN IP4 192.0.3.222
t=0 0
m=audio 20000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F10: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.234:5060;branch=z9hG4bK123456789cdefghi
To: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
From: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Content-Type: application/sdp
Content-Length: 199

v=0
o=- 82664419472 82664419473 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 10000 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
```

F11: 200 OK (INVITE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 INVITE
Contact: <sip:192.0.3.234:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp;term-ioi=3GPP-E-UTRAN-FDD.example2.ne.j
p
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 180;refresher=uac
Content-Length: 0
```

F12: ACK

```
ACK sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK34567890cdefghijkl
```

```
Max-Forwards: 70
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 1 ACK
Content-Length: 0
```

F13: UPDATE

```
UPDATE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijklm
Max-Forwards: 70
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 4 UPDATE
Contact: <sip:192.0.2.123:5060;transport=udp>
Supported: timer
Session-Expires: 180;refresher=uac
Min-SE: 180
Content-Length: 0
```

F14: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK45678901defghijklm
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 4 UPDATE
Contact: <sip:192.0.3.234:5060;transport=udp>
Require: timer
Session-Expires: 180;refresher=uac
Content-Length: 0
```

F15: BYE

```
BYE sip:192.0.3.234:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK56789012efghijklmo
Max-Forwards: 70
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 5 BYE
Content-Length: 0
```

F16: 200 OK (BYE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.123:5060;branch=z9hG4bK56789012efghijklmo
To: <sip:+819011111111@example2.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.2.123
CSeq: 5 BYE
Content-Length: 0
```
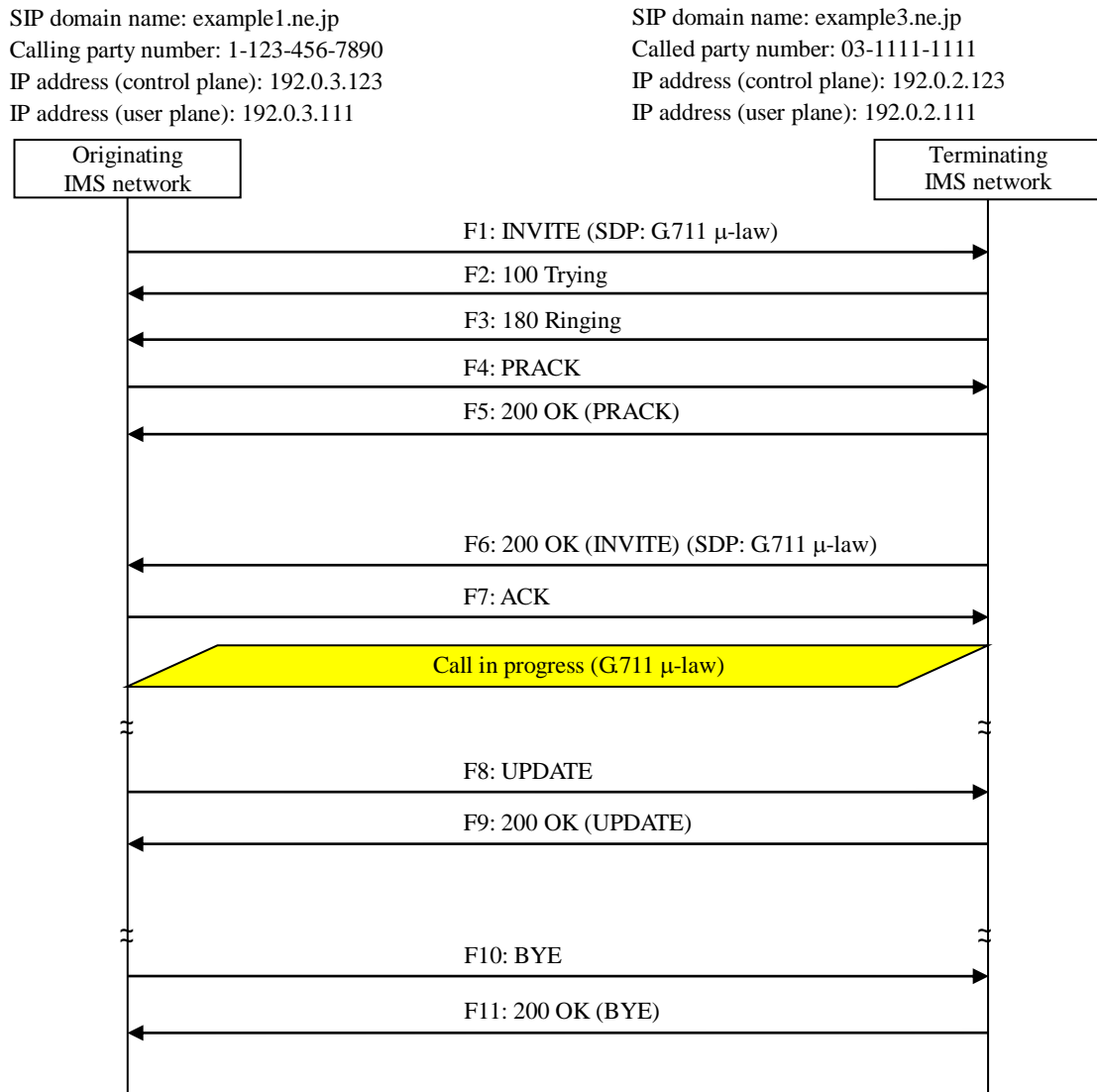
vii.2.6.2. Early media is not provided

This Subclause shows the call flow and message coding examples when early media is not provided from the terminating IMS network.

SIP domain name: example1.ne.jp
Calling party number: 1-123-456-7890
IP address (control plane): 192.0.3.123
IP address (user plane): 192.0.3.111

SIP domain name: example3.ne.jp
Called party number: 03-1111-1111
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

```
Originating                                          Terminating
IMS network                                          IMS network
     |                                                     |
     |------------ F1: INVITE (SDP: G.711 μ-law) --------->|
     |<----------- F2: 100 Trying -------------------------|
     |<----------- F3: 180 Ringing ------------------------|
     |------------ F4: PRACK ----------------------------->|
     |<----------- F5: 200 OK (PRACK) ---------------------|
     |                                                     |
     |<----------- F6: 200 OK (INVITE) (SDP: G.711 μ-law) -|
     |------------ F7: ACK ------------------------------->|
     |           Call in progress (G.711 μ-law)            |
     |                                                     |
     |------------ F8: UPDATE ---------------------------->|
     |<----------- F9: 200 OK (UPDATE) --------------------|
     |                                                     |
     |------------ F10: BYE ------------------------------>|
     |<----------- F11: 200 OK (BYE) ----------------------|
     |                                                     |
```

**Appendix Figure vii.2.6.2-1 / JJ-90.30: International call
<Early media is not provided>**

F1: INVITE

```
INVITE  sip:+81311111111;npdi@example3.ne.jp;user=phone  SIP/2.0
Via: SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:+81311111111@example3.ne.jp;user=phone>
From:  <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  1  INVITE
Contact:  <sip:192.0.3.123:5060;transport=udp>
Allow:  INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Accept:  application/sdp
Supported:  100rel,timer
Session-Expires:  180;refresher=uac
Min-SE:  180
P-Asserted-Identity:  tel:+11234567890;verstat=No-TN-Validation
Privacy:  none
P-Early-Media:  supported
```

```
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp
Content-Type: application/sdp
Content-Length: 214

v=0
o=- 82664419472 82664419472 IN IP4 192.0.3.111
s=-
c=IN IP4 192.0.3.111
t=0 0
m=audio 10000 RTP/AVP 101 0
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
a=rtpmap:0 PCMU/8000
```

F2: 100 Trying

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81311111111@example3.ne.jp;user=phone>
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Content-Length: 0
```

F3: 180 Ringing

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp;term-ioi=IEEE-802.3ah.example3.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: 100rel
RSeq: 1
Content-Length: 0
```

F4: PRACK

```
PRACK sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
Max-Forwards: 70
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 2 PRACK
RAck: 1 1 PRACK
Content-Length: 0
```

F5: 200 OK (PRACK)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK23456789bcdefghi
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
```

```
CSeq: 2 PRACK
Content-Length: 0
```

F6: 200 OK (INVITE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK12345678abcdefgh
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 INVITE
Contact: <sip:192.0.2.123:5060;transport=udp>
P-Charging-Vector: icid-value=1234bc9876e;orig-ioi=example1.ne.jp;term-ioi=IEEE-802.3ah.example3.ne.jp
Allow: INVITE,ACK,BYE,CANCEL,PRACK,UPDATE
Require: timer
Session-Expires: 180;refresher=uac
Content-Type: application/sdp
Content-Length: 202

v=0
o=- 82917391739 82917391739 IN IP4 192.0.2.111
s=-
c=IN IP4 192.0.2.111
t=0 0
m=audio 20000 RTP/AVP 101 0
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=rtpmap:0 PCMU/8000
```

F7: ACK

```
ACK sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK34567890cdefghij
Max-Forwards: 70
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 1 ACK
Content-Length: 0
```

F8: UPDATE

```
UPDATE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
Max-Forwards: 70
To: <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From: <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID: qwertyuiop123456@192.0.3.123
CSeq: 3 UPDATE
Contact: <sip:192.0.3.123:5060;transport=udp>
Supported: timer
Session-Expires: 180;refresher=uac
Min-SE: 180
Content-Length: 0
```

F9: 200 OK (UPDATE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.3.123:5060;branch=z9hG4bK45678901defghijklm
```

```
To:  <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  3  UPDATE
Contact:  <sip:192.0.2.123:5060;transport=udp>
Require:  timer
Session-Expires:  180;refresher=uac
Content-Length:  0
```

F10: BYE

```
BYE sip:192.0.2.123:5060 SIP/2.0
Via: SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
Max-Forwards:  70
To:  <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  4  BYE
Content-Length:  0
```

F11: 200 OK (BYE)

```
SIP/2.0  200  OK
Via: SIP/2.0/UDP  192.0.3.123:5060;branch=z9hG4bK56789012efghijklmo
To:  <sip:+81311111111@example3.ne.jp;user=phone>;tag=9876zyxw
From:  <sip:+11234567890@example1.ne.jp;user=phone>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.3.123
CSeq:  4  BYE
Content-Length:  0
```
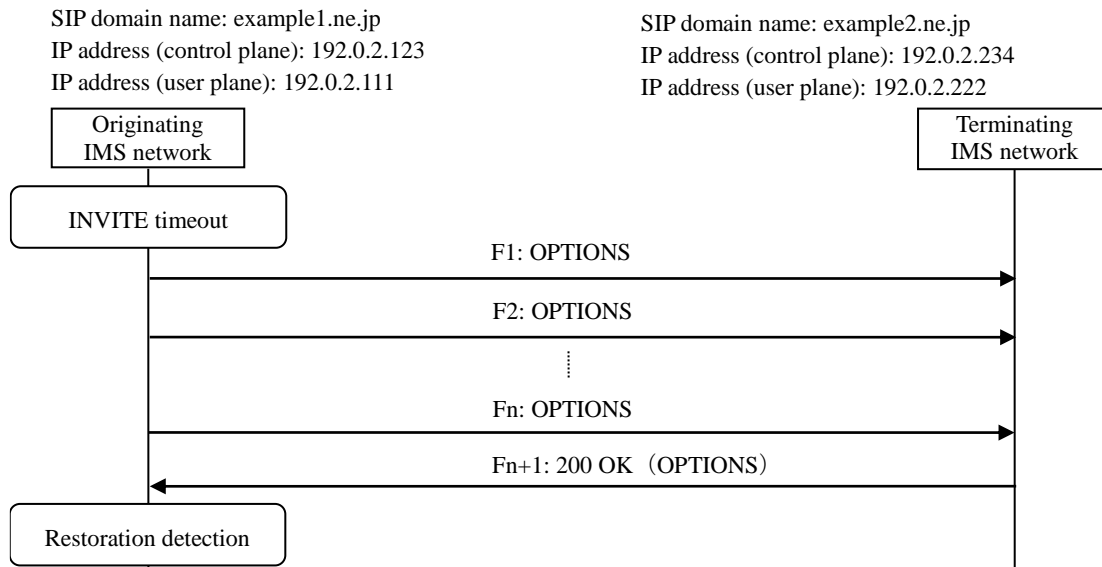
vii.2.7.Restoration detection using OPTIONS

This Subclause describes the examples of call flow and message coding for the case that out-of-dialog OPTIONS method is used for restoration detection, and the preconditions described in Appendix Table vii.2.7-1 are applied.

Appendix Table vii.2.7-1 / JJ-90.30: Preconditions applied for call flow/message coding examples

No.	Applied conditions of SIP message		Selection of option items at the II-NNI	
	Item	Conditions	Option item	Selection of option item
1	Use of out-of-dialog OPTIONS method	Use.	Appendix Table i.4-1 Item 5	Yes

SIP domain name: example1.ne.jp
IP address (control plane): 192.0.2.123
IP address (user plane): 192.0.2.111

SIP domain name: example2.ne.jp
IP address (control plane): 192.0.2.234
IP address (user plane): 192.0.2.222

Appendix Figure vii.2.7-1 / JJ-90.30: Restoration detection using OPTIONS

Fn: OPTIONS

```
OPTIONS  sip:192.0.2.234 SIP/2.0
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
Max-Forwards:  70
To:  <sip:192.0.2.234>
From:  <sip:192.0.2.123>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  OPTIONS
Contact:  <sip:192.0.2.123:5060>
Content-Length:  0
```

Fn+1: 200 OK (OPTIONS)

```
SIP/2.0  200  OK
Via:  SIP/2.0/UDP  192.0.2.123:5060;branch=z9hG4bK12345678abcdefgh
To:  <sip:192.0.2.234>
From:  <sip:192.0.2.123>;tag=1234abcd
Call-ID:  qwertyuiop123456@192.0.2.123
CSeq:  1  OPTIONS
Contact:  <sip:192.0.2.234:5060>
Content-Length:  0
```

Appendix viii (Informative):
SIP message generation considering the maximum length of SIP message

viii.1. General

This Appendix describes the SIP signalling recommendations to avoid the failure of interconnection between operators, even if the operators have different value of the limitation of maximum length based on the maximum SIP message length specified in Subclause 4.3.8. The conditions in this Appendix are described as a reference, then the different implementation is acceptable.

viii.2. SIP header fields set into initial INVITE request

An initial INVITE request, which can possibly exceed the limitation of the maximum length of the SIP message, include the minimum set of SIP header fields from the following SIP header fields:

- Allow, Call-ID, Contact, Content-Length, Content-Type, CSeq, From, History-Info, Max-Forwards, Min-SE, P-Access-Network-Info, P-Asserted-Identity, P-Charge-Info, P-Charging-Vector, P-Early-Media, P-Private-Network-Indication(NOTE 1), Priority(NOTE 2), Privacy, Route(NOTE 2), Session-Expires, Supported, To, User-To-User(NOTE 2), Via.

NOTE 1: The specification for this SIP header field is described in [TS-1018].

NOTE 2: The specification for this SIP header field is described in [JJ-90.28].

viii.3. IP address

If both IPv4 address and IPv6 address are available, IPv4 address is take precedence since IPv4 address is shorter than IPv6 address.

viii.4. SDP

The minimum lines required to establish a session are set into the SDP body in the SIP message. When the session is voice communication using G.711 μ-law, the SDP body include only following lines:

- "v=" line, "o=" line, "c=" line, "t=" line

- "m=" line (only a line for "m=audio")

- "a=" lines (only 5 lines: "a=rtpmap" (for indicating PCMU and telephone-event), "a=fmtp" (for telephone-event), "a=ptime", "a=sendrecv")