



ATIS-1000036

NEXT GENERATION NETWORK (NGN) OPERATOR SERVICES STANDARD

TECHNICAL REPORT



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ATIS-1000036, Next Generation Network (NGN) Operator Services Standard

Is an ATIS Standard developed by the **Signalling, Architecture, and Control (SAC) Subcommittee** under the **ATIS Packet Technologies and Systems Committee (PTSC)**.

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Next Generation Network (NGN) Operator Services Standard

Alliance for Telecommunications Industry Solutions

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Abstract

This standard describes signaling support for Operator Services when the application providing the services resides in the Next Generation Network (NGN). The scope of this standard includes: (1) SS7 to SIP interworking for a call originating into the PSTN and directed toward the Operator Services System (OSS) in the NGN and (2) SIP signaling for a call originating into the NGN and directed toward the Operator Services System (OSS) in the NGN. In addition, this standard describes the signaling for establishing a call/session from the OSS to a further destination, noting the similarities to normal call/session setup signaling.

Interconnection is assumed to be between trusted domains; the information and parameters required for this service are only available either within a trust domain or between trusted domains. In addition, in the absence of trust, and depending on exactly which information is not trusted, basic call setup may not function as expected.

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes a optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PTSC, which was responsible for its development, had the following leadership:

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ATIS Standard on –

Next Generation Network (NGN) Operator Services Standard

Introduction

Network signaling in support of Operator Services in the SS7 PSTN are standardized in ANSI ATIS-1000666.1999 (R2009). That standard describes a network capability. That is, it describes processes for setting up a call from the caller to the Operator Services System (OSS), transferring the call from one OSS to another, and completing a call from an OSS to a called party. Additional specialized functions are described for connection hold (preventing the calling party from disconnecting), network control of a coin phone, and network service recall (returning control of the call to the OSS after the end of a call to a called party.) ATIS-1000666.1999 (R2009) does not provide a detailed service description from the perspective of the calling party.

This standard similarly does not provide detailed service descriptions for Operator Services. Rather, high-level descriptions of the basic set of Operator Services are given and the signaling protocol elements required to support those services at an NGN-based OSS are described in detail. Illustrative examples are provided to show the transfer of needed information from the calling party to the OSS and from the OSS to called parties (e.g., for Call Completion service.) The service logic associated with determining which Operator Service to provide to the calling party and the service-specific interactions (e.g., access to databases) is not described in detail.

1 Scope and Purpose

This standard describes signaling support for Operator Services when the application providing the services resides in the NGN. The services within the scope of this standard include:

1. Directory Assistance/Directory Assistance Call Completion
2. Operator Assistance
3. Transfer to an Interexchange Carrier
4. Call Completion
5. Person Handling (Person-to-Person Call)
6. Calling Card and Commercial Credit Card (Alternate Billing)
7. Collect or Reverse Charging (Alternate Billing)
8. Third Number Billing (Alternate Billing)

Network control of a coin phone is out of the scope of this standard.

Intercept Service (providing information to the calling party regarding a number which has been disconnected or changed) is for further study.

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Busy Line Verification Service (verifying for the calling party that a called number is not out of service but is busy) and Interrupt Service (bridging an operator onto an existing call between two parties to alert one of the parties that the calling party is attempting to reach them) are also for further study.

The scope of this standard includes:

1. SS7 to SIP interworking for a call originating in the PSTN and directed toward the Operator Services System (OSS) in the NGN.
2. SIP signaling for a call originating in the NGN and directed toward the Operator Services System (OSS) in the NGN.

In addition, this standard describes the signaling for establishing a call/session from the OSS to a further destination, noting the similarities to normal call/session setup signaling.

Interconnection is assumed to be between trusted domains; the information and parameters required for this service are only available either within a trust domain or between trusted domains. In addition, in the absence of trust, and depending on exactly which information is not trusted, basic call setup may not function as expected.

Signaling to establish a call/session from the NGN to an OSS in the PSTN is not addressed in this standard, though ISUP/SIP interworking as described in ANSI ATIS-1000679.2004 (R2010) should provide the needed functionality

2 Normative References

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

[ATIS-1000005] ATIS-1000005, *Technical Report: Service Description of ETS*¹

[ATIS-1000018] ATIS-1000018, *Technical Report on NGN Architecture*¹

[ATIS-1000009] ATIS-1000009.2006(R2011), *IP Network-to-Network Interface (NNI) Standard for VoIP*¹

[ATIS-1000011] ATIS-1000011, *Technical Report: ETS Packet Priority for IP NNI Interfaces – Use of Existing DiffServ Per Hop Behaviors*¹

[ATIS-1000666] ATIS-1000666.1999 (R2009), *Signalling System No. 7 – Operator Services Network Capabilities Technical Report: Operator Services in a Next Generation Network*¹

[ATIS-1000679] ATIS-1000679.2004 (R2010), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control or ISDN User Part*¹

[ATIS-1000027] ATIS-1000027-2008, *Technical Report: Operator Services in a Next Generation Network*¹

[P-Charge-Info] draft-york-sipping-p-charge-info-07²

¹ This document is available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G Street N.W., Suite 500, Washington, DC 20005. < <https://www.atis.org/docstore/default.aspx> >

[SIP-History-Info] RFC 4244, An Extension to the Session Initiation Protocol (SIP) for Request History Information³

[3GPP 24.229] 3GPP TR 24.229, IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3⁴

3 Definitions and Abbreviations

ACM	(SS7) Address Complete Message
AS	Application Server
BGCF	Border Gateway Control Function
CIP	(SS7) Carrier Information Parameter
CSCF	Call Session Control Function
CSI	(SS7) Carrier Selection Information
EO	End Office
HSS	Home Subscriber Server
IAM	(SS7) Initial Address Message
ISUP	(SS7) ISDN User Part
MIME	Multipurpose Internet Mail Extensions
MGCF	Media Gateway Control Function
MS	Media Server
NGN	Next Generation Network
OBCI	(SS7) Optional Backward Call Indicators
OLI	(SS7) Originating Line Information
OSI	Open Systems Interconnection
OSS	Operator Services System
PAI	P-Asserted Identity
PSTN	Public Switched Telephone Network
RFC	(IETF) Request For Comments
SAP	(SS7) Service Activation Parameter
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SS7	(ANSI) Signaling System 7
TNS	(SS7) Transit Network Selection
UE	User Equipment
URI	Uniform Resource Identifier

² At the time of publication of this Technical Report, the listed reference is as yet unpublished by the Internet Engineering Task Force (IETF). See < <http://www.ietf.org> > for any updated information pertaining to this document.

³ This document is available from the Internet Engineering Task Force (IETF). < <http://www.ietf.org> >

⁴ This document is available from the Third Generation Partnership Project (3GPP) at < <http://www.3gpp.org/specs/specs.htm> >.

4 Service Definitions

The definitions in this section are based on specific PSTN operator services and describe these services from the calling party perspective. The definitions also apply to a mixed PSTN/NGN environment, where the calling party may be in an NGN and where the operator services system (OSS) may be in an NGN.

4.1 Directory Assistance/Directory Assistance Call Completion

The telephone number for a given name and address is provided to the calling party by an OSS. Call completion to the telephone number is an option available to the calling party. The OSS can complete a call for a calling party to the directory listing number that is found. Operator recall returns the calling party to the OSS when signaled by the calling party. In the PSTN, Operator Recall may be signaled by depressing the flash-hook. NGN terminals are not expected to signal the hook-flash. Therefore, NGN signaling for Operator Recall is assumed to be network-specific and is not described in this standard.

4.2 Operator Assistance

Dialing instructions and rate information are provided to the calling party by an OSS.

4.3 Transfer to Inter-exchange Carrier

Based on the interaction with the calling party, the OSS progresses the call into another network. The call appears to the other network as an incoming call to that network's OSS.

4.4 Call Completion

The OSS completes a call for the calling party. The called number may be determined by an operator service (e.g., Directory Assistance), or by the calling party alone (by providing the called number to the OSS.) This service includes an optional feature (Operator Recall) to return the calling party to an operator when signaled by the calling party. In the PSTN, Operator Recall may be signaled by depressing the flash-hook. NGN terminals are not expected to signal the hook-flash. Therefore, NGN signaling for Operator Recall is assumed to be network-specific and is not described in this standard.

4.5 Person Handling (Person-to-Person Call)

The OSS completes a call for the calling party. In addition, before the calling and called parties are connected, the OSS confirms that a specific individual has been contacted, rather than just the requested end user equipment.

4.6 Calling Card and Commercial Credit Card (Alternate Billing)

The calling party requests that the call be billed to a calling card or commercial credit card. After verification of the card, the OSS completes the call for the calling party.

4.7 Collect or Reverse Charging (Alternate Billing)

The calling party requests that the call be billed to the called party. The OSS completes the call for the calling party. In addition, before the calling and called parties are connected, the OSS confirms that the called party will accept the charges for the call.

4.8 Third Number Billing (Alternate Billing)

The calling party requests that the call be billed to a third party. The OSS places a call to the third party to confirm that the third party will accept the charges for the call. After this verification, the OSS completes the call for the calling party.

5 Architecture Model

The network architecture that provides the context for Operator Services is taken from [ATIS-1000018] with the assumption that the Operator Services System will be a part of a Voice over IP network. For completeness in describing the PSTN/NGN gateway, a PSTN-based OSS is also shown.

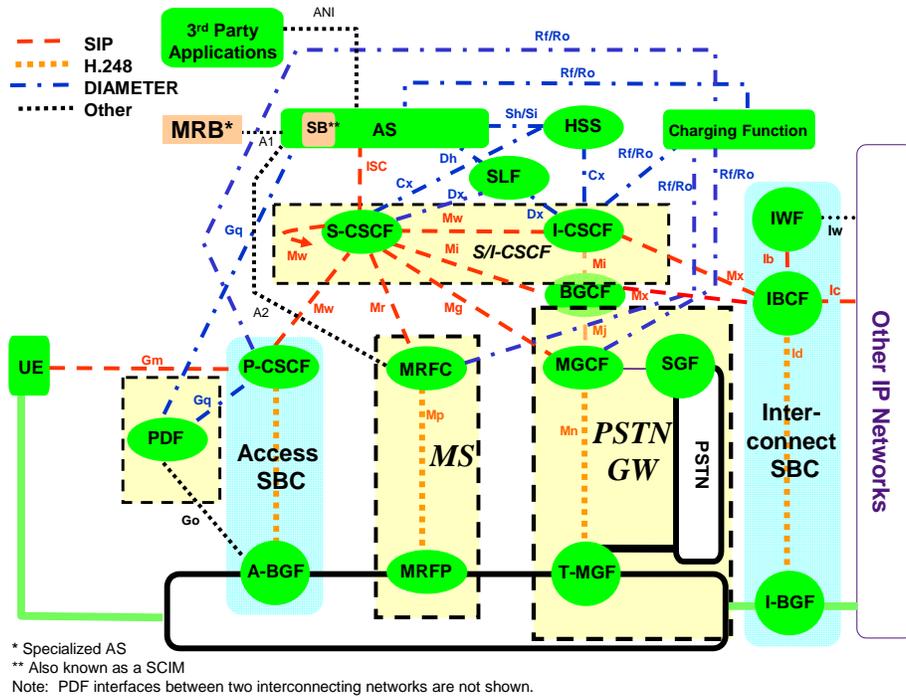


Figure 1: ATIS NGN Functional Architecture

With its focus on the operator services AS, this standard does not address the Access SBC, Charging Function and HSS. Therefore, the effective functional architecture for purposes of this document is as shown in Figure 2.

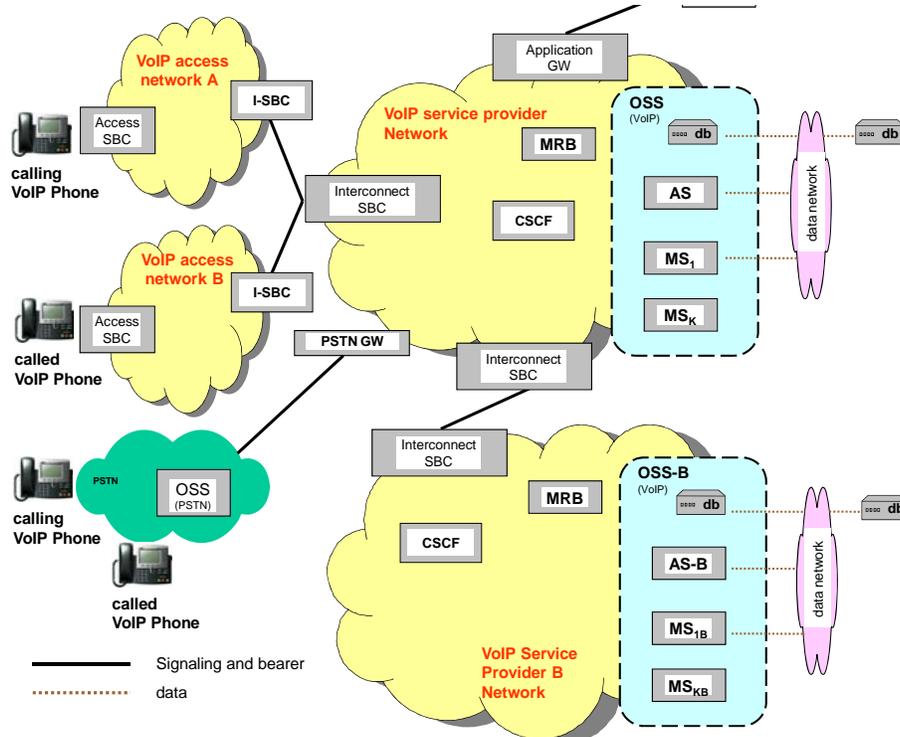


Figure 2- Operator Services Architecture

In Figure 2:

- Either “VoIP access network X” or the PSTN is the network where the call to be directed to the Operator Services System originates.
- Either “VoIP access network Y” or the PSTN is the network where a call is delivered from the Operator Services System (e.g., for Call Completion service.)
- “VoIP Service Provider A” supplies the initial contact with operator services.
- “VoIP Service Provider B” provides operator services to the calling party if and when the call is transferred there from VoIP Service Provider A.

Internal functionality within the Operator Services System is described in [ATIS-1000018].

6 Information Elements

This section describes the information elements needed for operator services processing which are conveyed between the OSS and other network entities. These information elements are carried in the SS7 ISUP and NGN SIP messaging described in this standard.

For PSTN originations, information elements mapped to SIP headers will be included in any encapsulated ISUP. Per ATIS-1000679, information in the SIP headers has precedence over encapsulated content when there is any conflict between the two.

6.1 Information Elements in Forward Direction Toward the OSS

6.1.1 Dialed Digits

The Dialed Digits information element is applicable for PSTN and NGN originations.

For PSTN originations, the MGCF recognizes the call as directed to Operator Services (e.g., by incoming trunk group or by referring to the received Operator Services Information parameter for send paid calls) and populates the Request-URI accordingly. Thus, the Request-URI may be populated with an operator-services specific domain name (e.g., for dialed digits "NPA-NXX-YYYY" recognized as destined for operator services based on the incoming trunk group, "sip:NPA-NXX-YYYY@OperatorServices.provider.com; phone-context=+1;user=phone" or "AlternateBilling@OperatorServices.provider.com"). The original dialed digits (including any prefix digits) are provided in SIP History-Info and Diversion headers. The information is also passed in the Called Party Number parameter in the encapsulated ISUP.

Alternatively, the Request-URI may be populated such that if the IAM includes an indication that the user dialed an initial "0" (Operator Services Information parameter with a Information Value Type = 001 (original access prefix) and Information Value = 0010 (0+ or 01+) or Information Value = 0010 (0-), then the "0" is included as the first (or only) digit of the called number.

For NGN originations, the caller's AS, on recognizing that the dialed digits reference an operator service will retarget the call, providing a Request-URI that is specific to the particular Operator Service (e.g., "sip:NPA-NXX-YYYY@OperatorServices.provider.com; phone-context=+1;user=phone" or "AlternateBilling@OperatorServices.provider.com"), and providing the original dialed digits (including any prefix digits) in SIP History-Info [SIP-History-Info] and Diversion [SIP Diversion] headers.

Alternatively, the caller's P-CSCF may populate the Request-URI, with the user-dialed digits as for any other call, i.e., as received in the INVITE from the UE. Note that for operator services calls that are dialed with an initial "0", "00", or "01", the prefix digits are then required to be included as the first digits of the called number.

6.1.2 Nature of Connections

The ISUP Nature of Connections Indicator information includes the Satellite Indicator, Continuity Check Indicator, and Echo Indicator. These indicators should receive normal treatment. No Operator Services-specific processing is needed.

The Nature of Connections information element is currently only applicable for PSTN originations or PSTN Interworking scenarios. Consistent with normal processing, the Ingress MGCF will not pass the Continuity Check Indicator or Echo Indicator information into the NGN except as part of encapsulated ISUP. Per ATIS-1000679.2004 (R2010), the Ingress MGCF will increment the Satellite Indicator and encapsulate the ISUP message for operator services calls as for other calls.

6.1.3 Calling Party's Category

The Calling Party's Category information element identifies the type of call origination.

The Calling Party's Category information element is only applicable for PSTN originations. Priority for emergency calls in the NGN is addressed in [ATIS-1000005] and [ATIS-1000011].

The ISUP Calling Party's Category parameter may identify the call as an emergency call, test call, or normal call. The Ingress MGCF does not separately pass the ISUP Calling Party's Category information into the NGN; it is passed in the encapsulated ISUP.

6.1.4 User Service Information

The User Service Information information element contains information regarding the bearer capabilities requested for the call.

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The bearer capabilities are applicable to PSTN and NGN originations.

For PSTN originations, the bearer capabilities are contained in the ISUP User Service Information. The Ingress MGCF maps the User Service Information to SDP. The SIP INVITE may also include a MIME-encapsulated version of the IAM.

For NGN originations, the caller's P-CSCF forwards the SDP received in the INVITE from the UE. The SDP provides information analogous to the User Service Information.

6.1.5 Charge Number

The Charge Number information element is a chargeable number for the calling party. It is not the alternate billing number.

The Charge Number is applicable to PSTN and NGN originations. For PSTN originations, the Charge Number, if present, is mapped to a P-Charge-Info SIP header. The mapping is performed per [P-Charge-Info]. In addition, the Charge Number, if present, may be passed in encapsulated ISUP.

For example, an incoming IAM containing an ISUP Charge Number parameter populated as follows:

Odd/even indicator:	Even
Nature of Address indicator:	ANI of the calling party; national number (hex 0x03)
Numbering Plan:	ISDN Numbering Plan
Address Information:	7326996201

would be mapped at an interworking function to a SIP INVITE message containing a P-Charge-Info header coded as follows:

```
P-Charge-Info: <SIP: +17326996201@domain;user=phone>; npi=ISDN; noa=3
```

where the domain reflects the PSTN network associated with the Charge Number.

When the Charge Number is not present in the received ISUP message, if:

- The Calling Party Number parameter is present, and
- The Originating Line Information parameter is present and does not contain a value of 02 (ANI failure)

then a P-Charge-Info SIP header is populated with the value contained in the Calling Party Number parameter.

For example, an incoming IAM without a Charge Number parameter, but containing an ISUP Calling Number parameter populated as follows:

Odd/even indicator:	Even
Nature of Address indicator:	ANI of the calling party; national number (hex 0x03)
Numbering Plan:	ISDN Numbering Plan
Address Information:	7326996201

and an Originating Line Information parameter populated with value 00 (POTS call) would be mapped at a ATIS-1000679.2004 (R2010) interworking function to a SIP INVITE message containing a P-Charge-Info header coded as follows:

```
P-Charge-Info: <SIP: +17326996201@domain;user=phone>; npi=ISDN; noa=3
```

The SIP INVITE may also include a MIME-encapsulated version of the IAM.

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For NGN originations, an Application Server in the NGN home network populates the charge number information in a P-Charge-Info SIP header.

For example, for a Charge Number of 732-699-6201, the Application Server would add a P-Charge-Info SIP header to the outgoing INVITE as follows:

P-Charge-Info: <sip:+17326996201@domain;user=phone>

where the domain reflects the NGN network associated with the Charge Number⁵.

6.1.6 Originating Line Information

The Originating Line Information (OLI) information element contains the class of line and originating line characteristics.

The OLI is applicable to PSTN and NGN originations.

For PSTN originations, the OLI parameter is mapped to an “oli” SIP URI parameter in the P-Asserted-Identity header. The ISUP OLI parameter may also be passed in encapsulated ISUP.

For example, an incoming IAM containing an ISUP OLI parameter coded as “07” would be mapped at a ATIS-1000679.2004 (R2010) interworking function to a SIP INVITE message containing a P-Asserted-Identity parameter encoded as a SIP URI and containing an “oli” parameter as follows:

P-Asserted-Identity: sip:+1-732-699-6201@domain; user=phone; oli=7.

Alternatively, the information could also be included encoded as a tel URI as follows:

P-Asserted-Identity: <tel:+1-732-699-6201>; oli=7.

The SIP INVITE may also contain a MIME-encapsulated version of the IAM.

For NGN originations, the entity that adds the P-Asserted Identity header (P-CSCF or Caller-AS) adds a SIP oli parameter with the corresponding OLI value to the P-Asserted-Identity header. For example:

P-Asserted-Identity: “Cell Block A” <[sip: cell-block-a@rahway-federal-pen.us](mailto:sip:cell-block-a@rahway-federal-pen.us); oli=7>

If both a sip and a tel URI are present in the P-Asserted Identity header, the oli should be added to both.

The use of the oli parameter is described in [3GPP 24.229], clause 7.2A.12. Annex B provides a further description of the semantics and use of the oli parameter.

6.1.7 Original Access Prefix

The Original Access Prefix information element indicates any prefix (e.g., ‘0’ or ‘1’) which was dialed. Operator Services process needs to know whether ‘0’ was dialed, but these services do not need to know whether ‘1’ was dialed.

The Original Access Prefix is applicable to PSTN and NGN originations.

For PSTN originations, the Ingress MGCF recognizes the call as directed to Operator Services (e.g., by incoming trunk group, or by referring to the received Operator Services Information parameter for sent-paid calls) and populates the Request-URI accordingly. Thus, the Request-URI may be populated with an operator-services specific domain name (e.g., if the OSI parameter indicates that an initial ‘0’ was dialed, “sip: NPA-NXX-YYYY;phone-context=+1@OperatorServices.provider.com;user=phone”,).

⁵ SS7 implementations always send Originating Line Information when a Charge Number is sent. Therefore, it would be appropriate to include an oli parameter (see section 7.1.6) in the P-Asserted Identity header whenever the P-Charge-Info header is sent.

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Alternatively, the Ingress MGCF may populate the digits field in the called party number in the Request-URI, based, in part, on the ISUP Operator Services Information parameter. If the OSI parameter indicates that a '0' was dialed, this is prepended to the other dialed digits in the Request-URI. Per ATIS-1000679.2004 (R2010), information in the SIP headers has precedence over encapsulated content. Prepending the '0' allows the OSS to ignore the encapsulated ISUP OSI parameter and Called Party Address parameter.

For NGN originations, the home network provides one of the following treatments:

1. The P-CSCF includes all of the user-dialed digits (including any '0', '00', or '01' prefix, or '101XXXX' carrier code) with the other dialed digits in the Request-URI. The Caller's AS, recognizing the initial dialed '0', will forward the call to the appropriate domain, e.g., "sip: NPA-NXX-YYYY;phone-context=+1@OperatorServices.provider.com;user=phone", or
2. The Caller's AS recognizes an initial dialed '0' and retargets the call to a domain that identifies that the '0' was dialed. For example, a call dialed "0+NPA-NXX-XXXX" could be retargeted to "sip:AlternateBilled@OperatorServices.provider.com;user=phone". The initial '0' and the NPA-NXX-XXXX dialed digits will be included in SIP History-Info and Diversion headers.

6.1.8 Transit Network Selection (TNS)

The TNS information element contains the Carrier Identification Code of the Carrier network that is required to be in the initial call setup path.

The TNS information element is applicable to PSTN and NGN originations.

For PSTN originations, the Ingress MGCF populates the host portion of the Request URI with a domain name that will cause SIP routing to terminate in the network specified in the received ISUP Transit Network Selection parameter.

For NGN originations, in the originating network, the Caller's AS, the BGCF, or other system populates the host portion of the sip Request URI with a domain name that will cause SIP routing to terminate in the network required to handle the call. The identity of this network may depend on a service invoked by the Caller (e.g., the specific operator service required), or a dialed or presubscribed Carrier Identification Code.

One consequence of this method of carrying the TNS information is that the PSTN/NGN interworking function will always use a sip URI in the Request URI. Further, an SS7 TNS parameter received at an Egress MGCF in encapsulated ISUP will be ignored; if any SS7 TNS parameter is generated, it will be based on the domain name in the host portion of the received sip Request URI.

In addition, since the information about the required intermediate network (whether dialed, presubscribed, or otherwise determined) is carried in the URI domain name, SIP routing should not be based on the "cic=" parameter, even if it is present in the Request URI.

6.1.9 Carrier Identification

The Carrier Identification information element includes the Carrier Identification Code of the Carrier that is either presubscribed by the calling party or dialed by the calling party. For operator services, this is the Carrier that will handle call completion from the OSS. Thus, for operator services, the Carrier Identification information element may not have the same value as the TNS information element prior to reaching the OSS, but is used by the OSS when determining the TNS information for call completion.

The Carrier Selection information element is applicable to PSTN and NGN originations.

For PSTN originations, the Carrier Identification information is mapped from the received ISUP Carrier Identification Parameter (CIP) and passed in the "cic=" parameter in the Request URI. It may also be carried in the CIP in encapsulated ISUP.

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For NGN originations, the Caller's AS populates the Carrier Identification information in the "cic=" parameter in the Request URI.

Note that RFC 4694 includes a requirement to remove the cic parameter before exiting the network identified in that parameter. Since the parameter is not used for routing in North American networks, its removal is optional.

6.1.10 Carrier Selection

The Carrier Selection information element identifies how the Carrier identified in the Carrier Identification information element was obtained from the user (i.e., user input, presubscribed, both, or unknown).

The Carrier Selection information element is applicable to PSTN and NGN originations.

For PSTN originations, the Carrier Identification the Carrier Selection information is mapped from the received ISUP Carrier Selection Information (CSI) parameter and is carried in the "dai=" parameter of the Request URI. It may also be carried in the CSI in encapsulated ISUP.

For NGN originations, the Carrier Selection Information is carried in the "dai" parameter of the Request URI.

Since removal of the "cic=" parameter by the network identified in that parameter (as described in RFC 4694) is not required, removal of the "dai=" parameter is also not required. However, the two parameters should be treated the same; they should either both be removed or both should be retained in the message.

6.1.11 Calling Party Number/Privacy Indicator

The Calling Party Number and Privacy Indicator information elements are applicable to PSTN and NGN originations.

For PSTN originations, per ATIS-1000679.2004 (R2010), the Ingress MGCF populates the From header and P-Asserted Identity header with the Calling Party Number received in the ISUP Calling Party Number parameter. In addition, per ATIS-1000679.2004 (R2010) the Ingress MGCF populates the Privacy header based on the information in the ISUP Calling Party Number parameter. ATIS-1000679.2004 (R2010) due to the precedence of SIP content over encapsulated content, the OSS ignores the encapsulated Calling Party Number and Privacy Indicator parameters.

For NGN originations, the P-CSCF populates the user identity in the P-Asserted-Identity header and forwards the corresponding user-provided information in the From header. In addition, the P-CSCF forwards the Privacy header as populated by the user.

6.1.12 Jurisdiction Information

The Jurisdiction Information information element contains information sent in the forward direction indicating the geographic origination of the call. The Jurisdiction Information Parameter is applicable to PSTN and NGN originations.

For PSTN originations, the Jurisdiction Information parameter is included in the encapsulated ISUP.

For NGN originations, the entity that adds the P-Asserted Identity (P-CSCF or Caller-AS) populates the Jurisdiction Information in the rn parameter in the P-Asserted Identity (e.g., sip:+1NPANXXXXXX;rn=+1NPANXXXXXX@host;user=phone.) See [ATIS-1000009].

6.1.13 Incoming Trunk Group

The Incoming Trunk Group information element identifies the SS7 trunk group that carries the Operator Services call to connect to the Ingress MGCF.

The incoming trunk group is applicable to PSTN originations.

For PSTN originations, the Ingress MGCF, populates the incoming trunk group/trunk group context in the tel-URI parameter in the Contact header based on the incoming PSTN trunk group. The Ingress MGCF uses the “tgrp” Tel URI parameter as defined in [RFC 4904].

6.1.14 Home Provider

The Home Provider information element identifies the service provider that directly serves the calling party.

The Home Provider is applicable to NGN originations.

For NGN originations, the P-CSCF populates the P-Asserted-Identity header with the SIP URI associated with the UE. The right hand side (domain) of this SIP URI identifies the Home Provider.

6.1.15 Intermediate Provider

The Intermediate Provider information element includes the provider which has agreements with home providers to handle OSS requests, and with OSS Providers which actually provide the requested services. Note that some home providers will have direct relationships with OSS Providers, rather than using an Intermediate Provider. Intermediate Providers are the targets of SIP requests from Home Providers since they are involved when a home provider does not have a direct relationship with an OSS Provider. Intermediate Providers perform retargeting of received SIP requests toward the OSS Provider.

The Intermediate Provider is applicable to NGN originations.

For NGN originations, a system in an Intermediate Provider network retargets the received SIP request toward the OSS Provider. In doing so, it populates the retargeted-from Request-URI into a SIP History-Info header. The SIP domain of the retargeted-from URI identifies the Intermediate Provider.

6.1.16 Charging Information

The Charging Information information element includes correlation information for charging purposes. It contains a globally-unique IMS Charging Identity (icid). The orig-ioi represents the IMS Inter Operator Identifier for the home provider.

Charging Information is applicable to NGN originations.

For NGN Originations, this information is carried in the P-Charging-Vector defined in [RFC 3455]. This is inserted by the caller’s P-CSCF.

6.1.17 User Equipment Capabilities and Characteristics

The User Equipment Capabilities and Characteristics information element identifies the capabilities of the user equipment of the calling party. It is applicable to NGN originations.

For NGN originations, the P-CSCF forwards the Contact header, which may include capability feature tags, received in the INVITE from the UE. Such capability feature tags are defined in [RFC 3840] to identify the calling UE’s capabilities and characteristics.

6.2 Information Elements in the Backward Direction from OSS System

6.2.1 User Network Interaction

The User Network Interaction information element informs the network elements in the backward direction that the voice path should be cut through in both directions prior to receipt of an answer indication.

The OSS System sends a 18x message with an SDP with an attribute that indicates “sendrecv.”

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For PSTN originations, the ingress MGCF sends the Optional Backward Call Indicators (OBCI) parameter with the User Network Interaction indicator field set to “user network interaction, cut through in both directions.”

For NGN originations, the 18x message with the SDP is passed back to previous SIP-based network elements.

6.3 Information Elements Summary Table

Table 1- Information Elements Summary Table

Information Element	Original Source	Source of SIP Content	Sink	Orig Type	Direction	Conveyance Mechanism
1. Dialed Digits	User	PSTN – Ingress MGCF NGN- User Equipment	OSS	PSTN and NGN	Forward	Digits of Request-URI (or SIP History-Info and Diversion for retargeting) For PSTN, defined in ATIS-1000679.2004 (R2010)
2. Nature of Connections Indicator	PSTN - Calling EO	Ingress MGCF	OSS	PSTN origination and PSTN interworking	Forward	Encapsulation defined in ATIS-1000679.2004 (R2010)
3. Calling Party's Category	Calling EO	N/A, since it is not sent to OSS	N/A	PSTN	Forward	Encapsulation defined in ATIS-1000679.2004 (R2010)
4. User Service Information	NGN – User Equipment PSTN – Calling EO	NGN – User Equipment PSTN- Ingress MGCF	OSS	PSTN and NGN	Forward	SDP
5. Charge Number	NGN – P-CSCF PSTN – Calling EO	NGN – P- CSCF PSTN – Ingress MGCF	OSS	PSTN and NGN	Forward	P-Charge Info Header
6. Originating Line Information	NGN – P-CSCF PSTN – Calling EO	NGN – P- CSCF PSTN – Ingress MGCF	OSS	PSTN and NGN	Forward	oli parameter in P-Asserted ID

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Information Element	Original Source	Source of SIP Content	Sink	Orig Type	Direction	Conveyance Mechanism
7. Original Access Prefix	User	NGN – User Equipment PSTN – Ingress MGCF	OSS	PSTN and NGN	Forward	Domain name or digits of Request-URI (or SIP- History-Info for retargeting)
8. Transit Network Selection	PSTN – Calling EO or user NGN- P-CSCF or S-CSCF or AS or user	PSTN – Ingress MGCF NGN- P-CSCF or S-CSCF or AS	OSS	PSTN and NGN	Forward	Hostport component of SIP Request-URI
9. Carrier ID	PSTN – Calling EO or user NGN- P-CSCF or S-CSCF or AS or user	Ingress MGCF NGN- P-CSCF or S-CSCF or AS	OSS	PSTN and NGN	Forward	cic parameter in Request URI
10. Carrier Selection	PSTN – Calling EO NGN - P-CSCF or S-CSCF or AS	PSTN - Ingress MGCF NGN - P-CSCF or S-CSCF or AS	OSS	PSTN and NGN	Forward	dai parameter in Request-URI
11. Calling Party Number/ Privacy	PSTN- Calling EO/User Equipment NGN- User Equipment (From header) P-CSCF (PAI header)	PSTN- Ingress MGCF NGN- User Equipment (From header) P-CSCF (PAI header)	OSS	PSTN and NGN	Forward	From P-Asserted ID Privacy

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Information Element	Original Source	Source of SIP Content	Sink	Orig Type	Direction	Conveyance Mechanism
12. Jurisdiction Information	PSTN- Calling EO NGN - P-CSCF or S-CSCF	PSTN- Ingress MGCF NGN - P-CSCF or S-CSCF or AS	OSS	PSTN and NGN	Forward	encapsulated ISUP. rn parameter in the P-Asserted Identity
13. Incoming Trunk Group	PSTN – Ingress MGCF	PSTN – MGCF	OSS	PSTN	Forward	trgp and trunk-context parameters in Contact Header
14. Home Provider	NGN – P-CSCF	NGN-P-CSCF	OSS	NGN	Forward	Hostport component of SIP URI in P-Asserted Identity
15. Intermediate Provider	NGN – System in Intermediate Provider network	NGN – System in Intermediate Provider network	OSS	NGN	Forward	Hostport component of SIP URI in SIP History-Info and Diversion
16. Charging Information	NGN – P-CSCF	NGN- P-CSCF	OSS	NGN	Forward	P-Charging Vector
17. User Equipment Capability and Characteristics	NGN – User Equipment	NGN – User Equipment	OSS	NGN	Forward	Contact header capability feature tags
18. User Network Interaction	OSS	OSS	PSTN – Ingress MGCF NGN - P-CSCF or S-CSCF or AS	PSTN and NGN	Backward	SDP attribute

7 Security Considerations

The information and parameters required for this service are only available within a trust domain or between trusted domains as defined by the network operators. In addition, in the absence of trust, and depending on exactly which information is not trusted, basic call setup may not function as expected.

Annex A

(informative)

A: Service Call Flows

A.1 Directory Assistance/Directory Assistance Call Completion

A.1.1 NGN call origination, NGN operator services system, Calling Party Dials 4-1-1

This flow is divided into two diagrams. The first (Figure A.1) shows the INVITE as it originates inside the caller's home provider. The elements within the home provider network which influence the formation of the initial INVITE message are shown. To simplify the representation of the call flows, the second diagram (Figure A.2) depicts the caller's home provider as a single entity and does not include elements (such as border elements and CSCFs) which simply forward messages.

A.1.1.1 Processing within the Caller's Home Provider

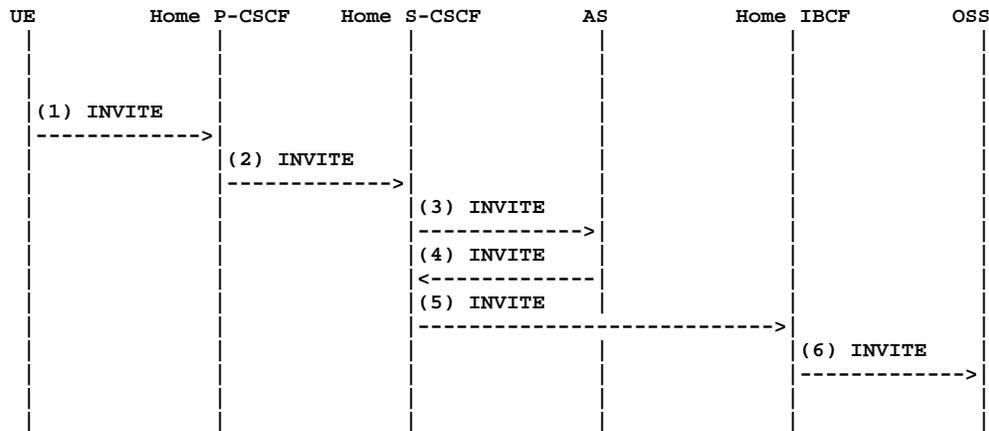


Figure A.1 – Processing within the caller's home provider

1. INVITE UE -> P-CSCF

INVITE tel:411;phone-context=provider-a.com SIP/2.0

Via: SIP/2.0/UDP client.provider-a.com:5060;branch=z9hG4bK74bf9

Max-Forwards: 70

From: <tel:+17327585735@provider-a.com>;tag=1234567

To: <tel:411;phone-context=provider-a.com>

CSeq: 101 INVITE

Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2

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Contact: <sip:7327585735@provider-a.com> ;audio;video;text
;actor="principle";automata;mobility="fixed"
;methods="INVITE,BYE,OPTIONS,ACK,CANCEL,REFER,MESSAGE"

Content-Type: application/sdp

Content-Length: ...

This INVITE is generated by the UE, and sent to the P-CSCF. It contains:

- a. Request-URI – indicates the user or service to which this request is being addressed. It could contain a SIP URI, a SIPS URI, or a tel URI. In this example, the Request-URI contains a “tel” URI indicating the digits “411” as dialed by the user. Since this is a local number as opposed to a global number, the “phone-context” parameter is included in order to show the context in which the digits should be interpreted, per RFC 3966, *The tel URI for Telephone Numbers*. Note that this could also be a SIP or SIPS URI.
- b. Via – The Via header field indicates the path taken by the request so far and indicates the path that should be followed in routing responses. The branch ID parameter in the Via header field values serves as a transaction identifier, and is used by proxies to detect loops.
- c. Max-Forwards – The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination. It consists of an integer that is decremented by one at each hop. If the Max-Forwards value reaches 0 before the request reaches its destination, it will be rejected with a 483(Too Many Hops) error response. The default initial value is 70 per RFC 3261.
- d. From – The From header field indicates the logical identity of the initiator of the request, possibly the user's address-of-record. Like the To header field, it contains a URI and optionally a display name. The contents of the From header being generated by the UE is not trusted; rather the network asserted identity in P-Asserted-Identity header is trusted.
- e. To – The To header field first and foremost specifies the desired "logical" recipient of the request, or the address-of-record of the user or resource that is the target of this request. This may or may not be the ultimate recipient of the request.
- f. CSeq – The CSeq header field serves as a way to identify and order transactions. It consists of a sequence number and a method.
- g. Call-ID – The Call-ID header field acts as a unique identifier to group together a series of messages.
- h. Contact - The Contact header field provides a SIP or SIPS URI that can be used to contact that specific instance of the UA for subsequent requests. In this example, the Contact header also includes capability feature tags as defined in RFC 3840 *Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)* in order to identify the calling UE's capabilities and characteristics. In our example, it supports audio, video, and text media types, and supports the “principle” actor type and methods as identified in RFC 3840.
- i. Content-Type — The Content-Type header field indicates the media type of the message-body sent to the recipient. The "media-type" element is defined in RFC 2046, *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types*.

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- j. Content-Length – The Content-Length header field indicates the size of the message-body, in decimal number of octets, sent to the recipient.
- k. Session Description – The INVITE contains a message body including a Session Description (SDP) offer per RFC 2327, *SDP: Session Description Protocol*.
- l. Other headers and message bodies may potentially be present, but are not included here.

2. INVITE P-CSCF-> S-CSCF

INVITE tel:411;phone-context=provider-a.com SIP/2.0

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bf9

Via: SIP/2.0/UDP client.provider-a.com:5060;branch=z9hG4bK74bf9

Max-Forwards: 69

Record-Route: <sip:p-cscf-1.provider-a.com>

From: <tel:+17327585735@provider-a.com>;tag=1234567

To: <tel:411;phone-context=provider-a.com>

CSeq: 101 INVITE

Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2

Contact: <sip:7327585735@provider-a.com> ;audio;video;text

 ;actor="principle";automata;mobility="fixed"

 ;methods="INVITE,BYE,OPTIONS,ACK,CANCEL,REFER,MESSAGE"

P-Asserted-Identity: "+17327585735" <sip:+17327585735@provider-a.com;user=phone>

P-Charging-Vector: icid-value=1234bc9876e;

 icid-generated-at=192.0.6.8;

 orig-voi=provider-a.com

Content-Type: application/sdp

Content-Length: ...

In addition to information received from the user, the P-CSCF adds the P-Asserted-Identity header. It also adds the P-Charging-Vector header which provides correlation information. The P-CSCF also adds to the Via header and adds a Record-Route header, which are of no special consequence to Operator Services. It also decrements the Max-Forwards value.

Therefore, this INVITE contains:

- a. Request-URI – The Request-URI indicates the user or service to which this request is being addressed. It could contain a SIP URI, a SIPS URI, or a tel URI. In this example, the Request-URI contains a “tel” URI indicating the digits “411” as dialed by the user. Since this is a local number as opposed to a global number, the “phone-context”

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parameter is included in order to show the context in which the digits should be interpreted, per RFC 3966, *The tel URI for Telephone Numbers*. Note that this could also be a SIP or SIPS URI. This was not modified from the information received from the user.

- b. Via – The Via header field indicates the path taken by the request so far and indicates the path that should be followed in routing responses. The branch ID parameter in the Via header field values serves as a transaction identifier, and is used by proxies to detect loops. The P-CSCF adds a Via header identifying itself to the top of the Via header stack.
- c. Max-Forwards – The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination. It consists of an integer that is decremented by one at each hop. If the Max-Forwards value reaches 0 before the request reaches its destination, it will be rejected with a 483(Too Many Hops) error response. The default initial value is 70 per RFC 3261. The value is decremented by one.
- d. Record-Route – The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy. The P-CSCF inserts itself into the Record-Route header.
- e. From - The From header field indicates the logical identity of the initiator of the request, possibly the user's address-of-record. Like the To header field, it contains a URI and optionally a display name. The contents of the From header being generated by the UE is not trusted; rather the network asserted identity in P-Asserted-Identity header is trusted. This was not modified from the information received from the user.
- f. To – The To header field first and foremost specifies the desired "logical" recipient of the request, or the address-of-record of the user or resource that is the target of this request. This may or may not be the ultimate recipient of the request. This was not modified from the information received from the user
- g. CSeq – The CSeq header field serves as a way to identify and order transactions. It consists of a sequence number and a method. This was not modified from the information received from the user.
- h. Call-ID – The Call-ID header field acts as a unique identifier to group together a series of messages. This was not modified from the information received from the user.
- i. Contact - The Contact header field provides a SIP or SIPS URI that can be used to contact that specific instance of the UA for subsequent requests. In this example, the Contact header also includes capability feature tags as defined in RFC 3840 *Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)* in order to identify the calling UE's capabilities and characteristics. In our example, it supports audio, video, and text media types, and supports the "principle" actor type and methods as identified in RFC 3840. This was not modified from the information received from the user.
- j. P-Asserted-Identity - The P-Asserted-Identity header field is used among trusted SIP entities (typically intermediaries) to carry the identity of the user sending a SIP message as it was verified by authentication. This header is defined in RFC 3325, *Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity Within Trusted Network*. It is inserted by the P-CSCF, and could also be inserted by other trusted network elements such as S-CSCF, or possibly by an AS or IBCF. It is possible for a SIP message to contain up to two P-Asserted-Identity headers; if so, one of them shall be a SIP or SIPS URI and the other shall be a tel URI. The P-Asserted-Identity header can contain a SIP, SIPS, or tel URI. In our example the P-Asserted-Identity provides several important pieces of information:
 - i. It identifies the Calling Party Number
 - ii. The right hand side of the SIP URI identifies the caller's home provider.

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- k. P-Charging-Vector: This is defined in RFC 3455, *Private Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)*. It provides correlation information for charging purposes. It contains a globally-unique IMS Charging Identity (icid). The orig-ioi represents the IMS Inter Operator Identifier for the home provider.
- l. Content-Type: The Content-Type header field indicates the media type of the message-body sent to the recipient. The "media-type" element is defined in RFC 2046, *Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types*. This was not modified from the information received from the user.
- m. Content-Length: The Content-Length header field indicates the size of the message-body, in decimal number of octets, sent to the recipient. This was not modified from the information received from the user.
- n. Session Description: The INVITE contains a message body including a Session Description (SDP) offer per RFC 2327, *SDP: Session Description Protocol*. This was not modified (proxies never modify message bodies) from the information received from the user.
- o. Other headers and message bodies may potentially be present, but are not included here.

3. INVITE S-CSCF-> AS

The S-CSCF performs any processing based on initial trigger criteria, which results in sending the INVITE request to an Application Server.

4. INVITE AS->S-CSCF

The AS analyses the dialed digits along with user profile information, and determines that this is a Directory Assistance call, and identifies the provider where this is to be sent. It thus retargets the request from "tel:411" to "sip:dacc@ossp-b.net". (Note that other routing choices could be made. For example, the AS could route using the "tel-411" directly.)

The AS logic also identifies that the Originating Line Information needs to be sent, so it appends an "oli" parameter to the P-Asserted-Identity. The logic also identifies that additional data needs to be included, which in this example includes Charge Number (encoded per section 7.1.5 of this document), Carrier Identification information (encoded per section 7.1.9 of this document), and Carrier Selection information (encoded per section 7.1.10 of this document.)

```
INVITE sip:dacc;cic=0123;dai=presub@ossp-b.net SIP/2.0
Via: SIP/2.0/UDP as-1.provider-a.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Record-Route: <sip:as-1.provider-a.com>
From: <tel:+17327585735@provider-a.com>;tag=1234567
To: <tel:411;phone-context=provider-a.com>
CSeq: 101 INVITE
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf7@192.168.1.3
```

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```
Contact: <sip:as-1@provider-a.com> ;audio;video;text
        ;actor="principle";automata;mobility="fixed"
        ;methods="INVITE,BYE,OPTIONS,ACK,CANCEL,REFER,MESSAGE"
P-Asserted-Identity: "+17327585735" <sip:+17327585735;oli=29@provider-
a.com;use=phone>
P-Charge-Info: <sip:+17327585735@provider-a.com;user=phone>
P-Charging-Vector: icid-value=1234bc9876e;
                  icid-generated-at=192.0.6.8;
                  orig-ioi=provider-a.com
History-Info: <tel:411;phone-context=provider-a.com>; index=1
Diversion: <tel:411;phone-context=provider-a.com>
Content-Type: application/sdp
Content-Length: ...
[SDP body not shown]
```

The AS, acting as a B2BUA, initiates a new call leg, sending an INVITE to the S-CSCF. The INVITE includes (but is not limited to):

- a. Request-URI: The Request-URI has been retargeted to identify the requested service, it now specifies "sip:dacc@ossp-b.net", which identifies the "dacc" service at the provider "ossp-b.net". A "cic" parameter, identifying the carrier to be used for any call completion attempts, as well as a "dai" parameter, indicating how that carrier was selected, are included in the userinfo of the SIP URI.
- b. History-Info: This header is added, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- c. Diversion: This header is added with the History-Info header, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- d. Via, Max-Forwards, Record-Route, Call-ID, Cseq, Contact, etc.: These are populated as would normally be done for an AS acting as a B2BUA, initiating a new outgoing leg. There is nothing unique to operator services here.
- e. P-Asserted-Identity: The AS inserts an "oli" parameter identifying the Originating Line Information value for the caller.
- f. P-Charge-Info: The AS inserts a "P-Charge-Info" header, identifying the Charge Number associated with the caller
- g. SDP: The AS copies the SDP as received in the incoming INVITE.

Other information may also be present. Content-Type and Content-Length reflect this.

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5. INVITE S-CSCF-> IBCF

The S-CSCF receives the INVITE from the AS, and routes it toward the operator services provider (via the IBCF) based on the domain in the Request-URI. It makes no changes to the INVITE other than those a proxy would normally make in forwarding an INVITE, such as adding itself into Via and Record-Route headers, and decrementing Max-Forwards.

INVITE sip:dacc;cic=0123;dai=presub@ossp-b.net SIP/2.0

Via: SIP/2.0/UDP as-1.provider-a.com:5060;branch=z9hG4bK74bf92

Via: SIP/2.0/UDP client.provider-a.com:5060;branch=z9hG4bK74bf91

Max-Forwards: 69

Record-Route: <sip:s-cscf-1.provider-a.com>, <sip: as-1.provider-a.com>

From: <tel:+17327585735@provider-a.com>;tag=1234567

To: <tel:411;phone-context=provider-a.com>

CSeq: 101 INVITE

Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf7@192.168.1.3

Contact: <sip:7327585735@provider-a.com> ;audio;video;text

 ;actor="principle";automata;mobility="fixed"

 ;methods="INVITE,BYE,OPTIONS,ACK,CANCEL,REFER,MESSAGE"

P-Asserted-Identity: "+17327585735" <sip: +17327585735;oli=29@provider-a.com;user=phone>

P-Charge-Info: <sip:17327585735@provider-a.com>

P-Charging-Vector: icid-value=1234bc9876e;

 icid-generated-at=192.0.6.8;

 orig-ioi=provider-a.com

History-Info: <tel:411;phone-context=provider-a.com>; index=1

Diversion: <tel:411;phone-context=provider-a.com>

Content-Type: application/sdp

Content-Length: ...

[SDP body not shown]

[MIME body not shown]

This INVITE contains:

- a. Request-URI – Contains the value from the INVITE as received from the AS.
- b. Via – The S-CSCF adds a Via header identifying itself to the top of the Via header stack.

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- c. Max-Forwards – The value is decremented by one.
- d. Record-Route – The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy. The S-CSCF inserts itself into the Record-Route header.
- e. From - This was not modified.
- f. To – This was not modified
- g. CSeq – This was not modified.
- h. Call-ID – This was not modified.
- i. Contact - This was not modified.
- j. P-Asserted-Identity - This was not modified
- k. P-Charging-Info
- l. P-Charging Vector. –This was not modified.
- m. History-Info – This header is added, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- n. Diversion – This header is added with the History-Info header, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- o. Content-Type – This was not modified.
- p. Content-Length – This was not modified.
- q. Session Description – This was not modified (proxies never modify message bodies).
- r. Other headers and message bodies may potentially be present, but are not included here.

6. INVITE IBCF-> OSS

The IBCF is a border element and is optionally used to perform any desired border control functions, which could include topology hiding, protocol Interworking, protocol “correction” or “fixup”, etc. In our example it does not perform any functions significant to operator services. Since the OSS is within the trust domain with the home provider, certain headers such as P-Asserted-Identity, etc. which might be removed when leaving trust boundaries are retained. The IBCF in this example essentially looks like a proxy and thus does things a proxy would do such as insert itself into Via and Record-Route, Since the IBCF is not a proxy, the MIME body is passed transparently.

INVITE sip:dacc;cic=0123;dai=presub@ossp-b.net SIP/2.0

Via: SIP/2.0/UDP ibcf-1.provider-a.com:5060;branch=z9hG4bK74bf94

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bf93

Via: SIP/2.0/UDP as-1.provider-a.com:5060;branch=z9hG4bK74bf92

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Max-Forwards: 68

Record-Route: <sip: ibcf-1.provider-a.com>, <sip:s-cscf-1.provider-a.com>, <sip: as-1.provider-a.com>

From: <tel:+17327585735@provider-a.com>;tag=1234567

To: <tel:411;phone-context=provider-a.com>

CSeq: 101 INVITE

Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2

Contact: <sip: as-1@provider-a.com> ;audio;video;text

 ;actor="principle";automata;mobility="fixed"

 ;methods="INVITE,BYE,OPTIONS,ACK,CANCEL,REFER,MESSAGE"

P-Asserted-Identity: "+17327585735" <sip:+17327585735;oli=29@provider-a.com;user=phone>

P-Charge-Info: <sip:+17327585735@provider-a.com;user=phone>

History-Info: <tel:411;phone-context=provider-a.com>; index=1

Diversion: <tel:411;phone-context=provider-a.com>

P-Charging-Vector: icid-value=1234bc9876e;

 icid-generated-at=192.0.6.8;

 orig-ioi=provider-a.com

Content-Type: application/sdp

Content-Length: ...

[SDP body not shown]

[MIME body not shown]

This INVITE contains:

- a. Request-URI – Contains the value from the INVITE as received from the IBCF.
- b. Via – The IBCF adds a Via header identifying itself to the top of the Via header stack.
- c. Max-Forwards – The value is decremented by one.
- d. Record-Route – The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy. The IBCF inserts itself into the Record-Route header.
- e. From - This was not modified.
- f. To – This was not modified
- g. CSeq – This was not modified.

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- h. Call-ID – This was not modified.
- i. Contact - This was not modified.
- j. P-Asserted-Identity - This was not modified.
- k. P-Charge-Info – This was not modified.
- l. P-Charging-Vector. This was not modified.
- m. History-Info – This header is added, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- n. Diversion – This header is added with the History-Info header, and contains the original retargeted-from Request-URI value. In this example, it effectively preserves the original dialed digits.
- o. Content-Type – This was not modified.
- p. Content-Length – This was not modified.
- q. Session Description – This was not modified (proxies never modify message bodies).
- r. Other headers and message bodies may potentially be present, but are not included here.

A.1.1.2 OSS Processing

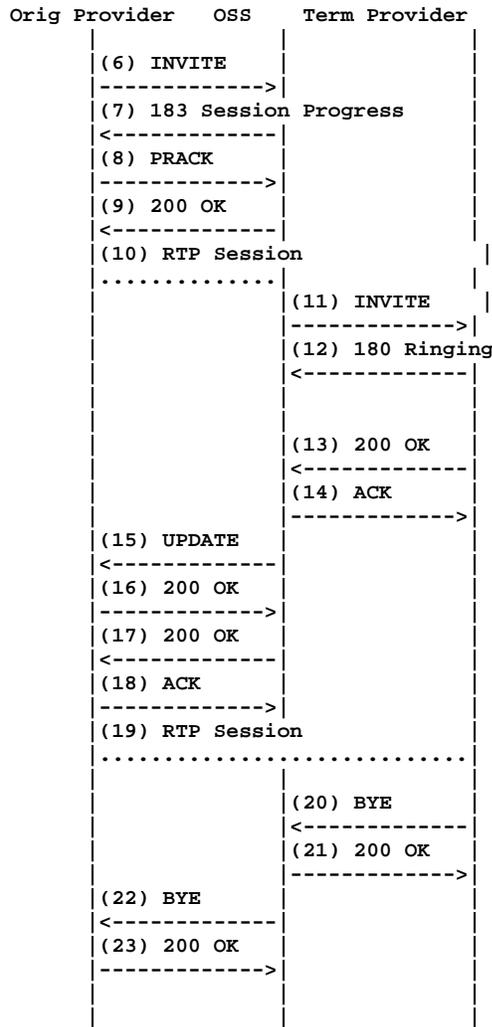


Figure A.2– OSS Processing

6. INVITE Orig Provider -> OSS

This is the same INVITE described in step 6 above. Received INVITEs are routed to the OSS, which examines the contents of the INVITE, determines processing, and performs third party call control, as needed. In this capacity it acts as the endpoint of the SIP dialog with the home provider, and acts as a SIP endpoint for the purposes of originating sessions, e.g., for call completion service.

When the OSS receives this incoming INVITE from the caller’s home provider, it decides in this example, based on the received information (e.g. request-uri, calling party identity, caller’s home provider identity, charge number, OLI, calling UE capabilities and characteristics) on the appropriate interactions with the caller.

7. 183 Session Progress OSS -> Orig Provider

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The OSS sends a 183 Session Progress message with the OSS-MS SDP toward the originating provider. This allows establishment of a media path between the caller and the OSS so that announcements can be played and so that the caller can provide responses. This message is sent as a reliable provisional response. The most significant information in this message is the SDP answer. Akin to final responses, much of the content of this message, other than the Contact, Content, and SDP are copies from the INVITE to which this is being sent in response.

8. PRACK Orig Provider -> OSS-AS

The “reliable” part of “reliable provisional” requires that provisional responses receive acknowledgement, thus the need to for a Provisional Response ACKnowledgement (PRACK). So the caller’s home provider responds to the 183 Session Progress message with a PRACK.

9. 200 OK OSS -> Orig Provider

The OSS sends a 200 OK in response to the PRACK. This message does not convey information of specific relevance to operator services.

10. Once this exchange is complete, then the caller has a media session with the OSS, and the OSS may play announcements, prompts, etc and receive and process audio input from the user.

11. INVITE OSS-> Terminating Provider

Based on the information received from the caller, the OSS decides to initiate an outgoing call completion call. In this example, this is to a SIP destination, and the simplified diagram shows the OSS communicating directly with the terminating domain. Other elements would typically be included, but would not be expected to include any operator services specific information.

This INVITE contains:

- a. Request-URI: Specifies the called party.
- b. P-Asserted-Identity: Specifies the caller, as identified via the initial incoming INVITE to the OSS.
- c. SDP: The SDP of the caller, as received by the OSS. Note that if home provider egress border elements or OSS provider ingress border elements modified the SDP as part of topology hiding then this is what will be used. It’s also possible that OSS provider egress border elements could modify this information. Such topology hiding functions are not specific to OSS and are not expected to have OSS specific impacts on this flow.
- d. Other headers and bodies may be present, but are not included here, since no operator services specific processing is required for this message.

12. 180 Ringing Terminating Provider -> OSS

The terminating provider would return some non final response such as 180 Ringing.

13. 200 OK Terminating Provider -> OSS

When the called party answers, the terminating provider would return a 200 OK final response. If the called party’s SDP has not yet been sent, it will be sent at this point.

14. The OSS acknowledges this 200 OK with an ACK message.

15. The OSS sends the SDP received from the terminating provider towards the originating provider.

16. The originating provider acknowledges the updated SDP by sending a 200 OK response.

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17. Since the called party has answered, the OSS sends a final 200 OK response towards the calling party, which contains the SDP of the called party.
18. The calling party acknowledges the 200 OK with an ACK.
19. There is now an RTP session between calling and called parties. The OSS, performing third party call control, remains in the signaling path as the SIP endpoint of both the incoming and outgoing legs.
20. The OSS responds with 200 OK.
21. The OSS initiates clearing toward the caller's home provider by sending BYE.
22. The caller's home provider responds with 200 OK.

A.1.1.3 PSTN Origination, Calling Party Dials 4-1-1

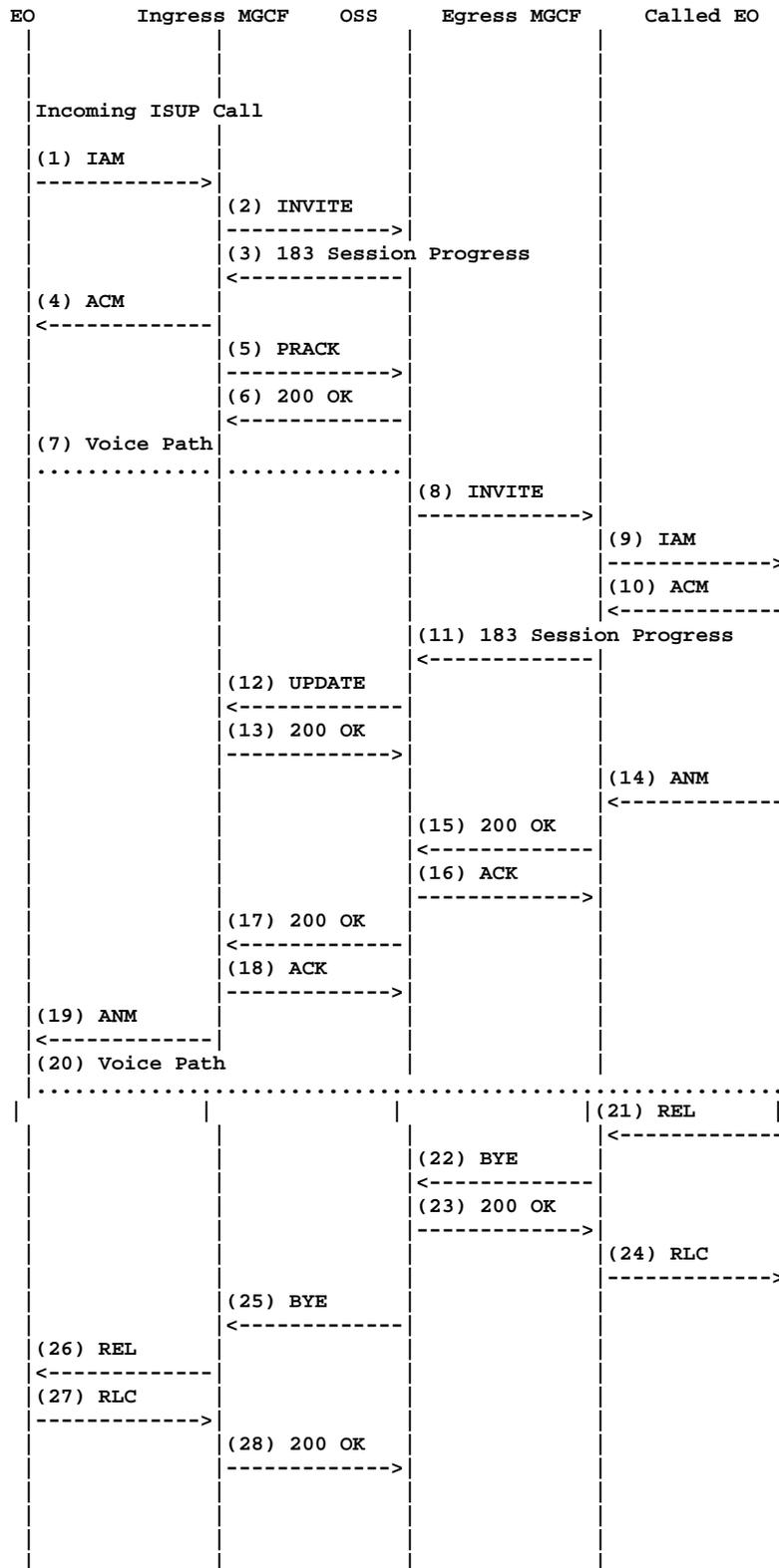


Figure A.3 – Calling Party Dials 411

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1. The IAM is sent by the calling end office to the Ingress GW (I-GW). The IAM includes the mandatory parameters identified in ATIS-1000113 and may include the optional parameters identified in ATIS-1000666.1999 (R2009).

The mandatory parameters include:

- a. Message Type
- b. Nature of Connections Indicator
- c. Forward Call Indicators
- d. Calling Party's Category
- e. User Service Information
- f. Called Party Number

As discussed in ANSI ATIS-1000666.1999 (R2009), the following optional parameters may be included for operator services origination parameters:

- a. Charge Number
- b. Originating Line Information
- c. Operator Services Information/Original Access Prefix
- d. Operator Services Information/Access signaling
- e. Transit Network Selection (TNS) – not needed for 411-based DA/DACC, but may be received for international operator services
- f. Carrier Identification Parameter – not needed for 411-based DA/DACC, but may be received for calls involving IXCs, such as 0+interLATA calls
- g. Carrier Selection Parameter – not needed for 411-based DA/DACC, but may be received for calls involving IXCs, such as 0+interLATA calls

As indicated in ATIS-1000666.1999 (R2009), the Called Party Number/Nature of Address will be set to an "operator requested" codepoint appropriate for the address.

The Jurisdiction Information and Calling Party Number parameters may also be received.

2. The Ingress MGCF receives the IAM message from the calling EO and generates an INVITE. The consumer of the operator-services specific information in the message is the OSS in the operator service provider network. The INVITE includes:

Request-URI – indicates the user or service to which this request is being addressed. It could contain a SIP URI, a SIPS URI, or a tel URI. In this example, the Request-URI contains a SIP URI indicating the digits "411" as dialed by the user. Since this is a local number as opposed to a global number, the "phone-context" parameter is included in order to show the context in which the digits should be interpreted, per RFC 3966, The tel URI for Telephone Numbers. Since the URI represents a phone number, the "user=phone" parameter is also appended. This request URI thus appears as follows: "sip:411;phone-context=+1@ossp-b.net;user=phone". The domain of the Request-URI may be populated with a value representing the specified carrier if a TNS value is to be conveyed. This is not applicable in this example.

- a. Via – The Via header field indicates the path taken by the request so far and indicates the path that should be followed in routing responses. The branch ID parameter in the Via header field values serves as a transaction identifier, and is used by proxies to detect loops.

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- b. From - The From header field indicates the logical identity of the initiator of the request, possibly the user's address-of-record. Like the To header field, it contains a URI and optionally a display name. In this example, the From header identifies the Ingress MGCF.
 - c. Contact - The Contact header field provides a SIP or SIPS URI that can be used to contact that specific instance of the UA for subsequent requests. In this example, the Contact header provides a SIP or SIPS URI that can be used to contact the Ingress MGCF. It also contains a "tgrp" parameter identifying the ingress trunk group on which the IAM was received, and a "trunk-context" parameter representing the gateway.
 - d. Content-Type – The Content-Type header field indicates the media type of the message-body sent to the recipient. The "media-type" element is defined in RFC 2046, Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types.
 - e. Content-Length – The Content-Length header field indicates the size of the message-body, in decimal number of octets, sent to the recipient.
 - f. Session Description – The INVITE contains a message body including a Session Description (SDP) offer per RFC 2327, SDP: Session Description Protocol. The SDP identifies the Ingress MGF, which is controlled by the MGCF.
 - g. P-Asserted-Identity - The P-Asserted-Identity header field is used among trusted SIP entities (typically intermediaries) to carry the identity of the user sending a SIP message as it was verified by authentication. This header is defined in RFC 3325, Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity, within Trusted Network. It is possible for a SIP message to contain up to two P-Asserted-Identity headers; if so, one of them shall be a SIP or SIPS URI and the other shall be a tel URI. The P-Asserted-Identity header can contain a SIP, SIPS, or tel URI. In this example, the Ingress Gateway populates the P-Asserted-Identity with several important pieces of information:
 - 1- The Calling Party Number from the IAM is included
 - 2- The right hand side of the SIP URI identifies the OSS provider network, which is the network where the Ingress MGCF resides
 - 3- An "oli" parameter is included conveying the Originating Line Information value received in the IAM.
 - h. P-Charge-Info: This header field conveys the Charge Number value received in the IAM
 - i. Message Body (application/ISUP) – contains the complete received ISUP message.
 - j. Other headers and message bodies may be present, but are not included here.
- 3. The OSS sends a 183 Session Progress to the Ingress MGCF. It includes the SDP of the MRFC with an attribute that identifies the send/receive mode. In this example, the attribute is set to "sendrecv," which indicates that the sender is willing to both send and receive media. Headers and other message bodies may be present, but are not included here, since no additional operator services-specific information is required for this message.
 - 4. Upon receipt of the 183 Session Progress message, the Ingress MGCF returns an ACM for user network interaction as defined in ATIS-1000113.4, 4A6.
 - 5. The Ingress GW then sends a PRACK to the OSS.
 - 6. The OSS sends a 200 OK to the Ingress MGCF. Headers and message bodies may be present, but are not included here, since no operator services-specific information is required for this message.

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7. An RTP session is established between the Ingress MGCF and the OSS, completing the voice path is established between the calling party and the OSS. The calling party interacts with the OSS.
8. The OSS, having determined the need to initiate call completion, sends an Invite to the Egress MGCF. The OSS includes the headers and message bodies normally included for call processing. No operator services specific headers or message bodies are needed for the call completion.
9. The Egress MGCF sends an IAM message to the called EO. No operator services specific ISUP parameters are needed for the call completion.
10. The called EO sends an ACM message back to the Egress MGCF. No operator services specific ISUP parameters are needed for the call completion.
11. The Egress MGCF sends a 183 Session Progress with the Egress MGCF SDP.
12. The OSS sends the Ingress MGCF an Update message with the Egress MGCF SDP. No operator services specific IAM parameters are needed for the call completion. Headers and message bodies may be present, but are not included here, since no operator services-specific information is required for this message.
13. The Ingress MGCF sends a 200 OK to the OSS. The RTP path between the Ingress MGCF and Egress MGCF is now established in the backward direction or in both directions, depending on the SDP attributes.
14. The called EO sends an ANM to the Egress MGCF. No operator services specific ISUP parameters are needed for the call completion.
15. The Egress MGCF sends a 200 OK to the OSS. Headers and message bodies may be present, but are not included here, since no operator services-specific information is required for this message.
16. The OSS sends an ACK to the Egress MGCF.
17. The OSS sends a 200 OK to the Ingress GW. Headers and message bodies may be present, but are not included here, since no operator services-specific information is required for this message.
18. The Ingress GW sends an ACK to the OSS. The RTP path between the Ingress MGC and Egress MGF is established in both directions. The voice path is established in both directions between the calling party and called party.
19. The OSS sends an ANM to the calling EO. This does not add the establishment of the voice path in the forward direction, since the voice path was established in the forward direction when the ACM was returned for user network interaction.
20. Conversation takes place.
21. The called EO initiates clearing by sending REL to the egress GW.
22. The egress GW sends BYE to the AS.
23. The AS responds with 200 OK.
24. The egress GW responds to the called EO with RLC, completing the clearing of this call.
25. The AS initiates clearing toward the caller by sending BYE toward the ingress GW.
26. The ingress GW sends REL toward the originating EO.

27. The originating EO responds with RLC, completing the clearing of this call.

28. The ingress GW responds to the AS's BYE with 200 OK.

A.2 Operator Assistance, with Call Completion and Return to Operator Options

A.2.1 Call to the OSS

A.1.1.1 NGN call origination, NGN operator services AS

The processing within the home provider's network is identical to that in A.1.1.1, with the following differences:

- In step 1, the caller dials "0", and the UE sends an INVITE with a Request-URI "tel:0; phone-context=provider-a.com".
- In step 4, the AS retargets the request to "sip:operator-assistance; cic=0123;dai=presub@ossp-b.net" rather than to "sip:dacc; cic=0123;dai=presub@ossp-b.net".

The flow for OSS processing is identical to that in A.1.1.2, with the following differences:

- Steps 11 through 17 only occur if the interaction with the operator leads to the decision to provide Call Completion service.
- The OSS may insert itself into the media path. Alternatively, the OSS media resources may be excluded from the media path until the called party initiates call clearing and may be added at that time.
- After the requested information is provided, processing proceeds as described in A.2.2.1.

A.1.1.2 PSTN call origination, NGN operator services system

Initial processing is identical to steps 1 through 7 in A.1.1.3, with the following differences:

- In step 1, the user dials "0", and the IAM contains a Called Party Number parameter with no digits included. The IAM also contains an Operator Services Information parameter, with the Original Access Prefix indicating that zero was dialed.
- In step 2, the INVITE includes a Request-URI coded as "sip:0;phone-context=+1@ossp-b.net; user=phone".
- At step 7, the caller and OSS have a two way voice path, and interaction occurs. The requested information is provided.
- The OIS system may arrange to have the MGCF monitor for the # signal. The SIP KPML package, which makes use of the SIP SUBSCRIBE/NOTIFY mechanism, would be used for this. Note that hook flash, which is often used for this in the PSTN, is not signaled inband and thus cannot be detected by the MGCF.
- After the requested information is provided, processing proceeds as described in A.2.2.2.

A.2.1 Return to Operator Functionality

A.2.1.1 NGN call origination, NGN operator services AS

After conversation has ended, the called party initiates call clearing by sending a BYE to the OSS, as in step 18 of A.1.1.2. The OSS processes the BYE to clear the outbound call leg, but leaves the incoming

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call leg up. If the OSS media resources were not bridged on during the call, the OSS could send a SIP UPDATE toward the caller in order to move the caller's media session onto an OSS media resource.

The OSS media resource, which now has a media session with the caller, monitors for the predetermined signal, in this example, the "#", sent inband. Note that while hook flash is often used for this in the PSTN, this stimulus signaling is not represented in SIP, so a signal such as "#" must be used instead.

The caller wants to connect back to the OSS, and therefore the UE emits the appropriate signal.

If the OSS media resource detects this signal in the media stream, it notifies the OSS application server. While this is an internal interface for the purpose of this document, KPML would be an appropriate mechanism. If the operator services application server receives notification of the agreed upon signal within a designated time interval, interaction with the caller can be re-established, so that another call may be made.

If notification of the signal is not received within the designated interval, the OSS clears the call toward the caller. Alternatively, the caller may clear the call towards the OSS.

A.2.1.2 PSTN call origination, NGN operator services AS

After conversation has ended, the called EO initiates call clearing by sending a REL to the OSS as in step 21 of A.1.1.3. The OSS processes the corresponding SIP BYE to clear the outbound call leg, but leaves the incoming call leg up.

The caller wants to connect back to the OSS, and therefore signals this by pressing the "#" key.

The MGCF detects this signal in the media stream and it notifies the OSS with a SIP NOTIFY message. If the OSS receives notification of the agreed upon signal within a designated time interval, interaction with the caller can be re-established, so that another call may be made. Such re-establishment would take place using UPDATE from the OSS to the ingress GW as in steps 12 and 13 of A.1.1.3.

If notification of the signal is not received within the designated interval, the OSS clears the call toward the caller. Alternatively, the caller may clear the call toward the OSS.

A.3 Transfer to Inter-exchange Carrier

A.3.1 NGN operator services system to NGN operator services system

Initial processing within the home provider's network is identical to steps 1 through 6 in A.1.1.1, with the following differences:

- In step 1, the caller dials "0", and the UE sends an INVITE with a Request-URI "tel: 0; phone-context=provider-a.com."
- In step 4, the AS retargets the request to "sip:operator-assistance@ossp-b.net" rather than to "sip:dacc@ossp-b.net".

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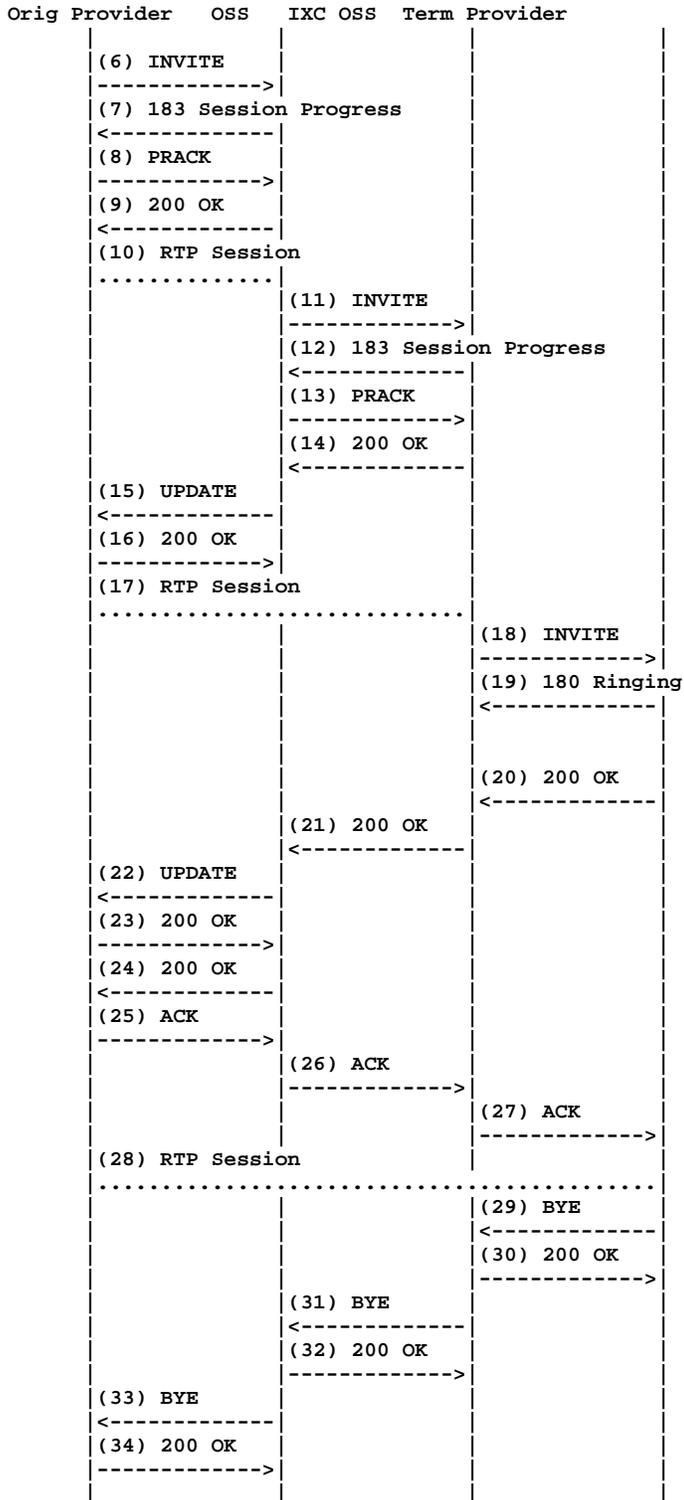


Figure A.4– OSS Processing for Transfer to IXC

Steps 6 through 10 are identical to those in A.1.1.2, except that the incoming INVITE has been retargeted to "sip:operator-assistance@ossp-b.net".

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In step 10, the OSS interacts with the caller, and determines that the call needs to be sent to the OSS of the IXC. The identification of the IXC could have been from the received cic parameter in the Request URI, or it could have been from interactions with the caller.

In step 11, the OSS forms an INVITE and sends it toward the IXC's OSS. The INVITE includes (but is not limited to):

- Request-URI identifies the IXC's OSS: "sip:operator-assistance@ixc-c.net"
- The History-Info header contains the information from the received History-Info, and adds the OSS in the most recent retargeted-from URI.
- If the carrier was identified via interaction with the caller, the cic parameter and dai parameter are added or updated to reflect this.
- The P-Asserted-Identity is unchanged, so it is clear to the IXC OSS that this call originated from the calling user.

In steps 12 through 17, the OSS performs 3PCC functions to establish a voice path between the caller and the IXC OSS. The IXC OSS interacts with the caller to determine the desired called party number.

In step 18, the IXC OSS performs 3PCC to set up a call to the called party.

In steps 19 through 28, both OSSs are performing 3PCC to facilitate establishment of the call and voice path between the calling and called users.

In steps 29 through 34, clearing is initiated by the called party and the call is cleared toward the caller. Alternatively, the caller may clear the call toward the OSS.

A.4 Person Handling (Person-to-Person Call)

Person Handling calls are processed similarly to Directory Assistance Call Completion Calls (see Section A.1) with the following exceptions and additions:

1. The calling party dials 0+10 digits, rather than 411. This is reflected in the Request-URI in the initial INVITE.
2. After completion of the Call Completion call, the operator interacts with the called party to determine whether the requested person is available to accept the call. If the requested person is available, the operator connects the calling and called parties; if not, the Call Completion call is ended, the calling party is informed, and the operator disconnects from the calling party.

No protocol or signaling procedure enhancements are required to support Person Handling.

A.5 Alternate Billing

Alternate Billing calls (Calling Card and Commercial Credit Card, Collect or Reverse Charging, and Third Number Billing) are processed similarly to Directory Assistance Call Completion Calls (see Section A.1) with the following exceptions and additions:

1. The calling party dials 0+10 digits, rather than 411. This is reflected in the Request-URI in the initial INVITE.
2. After the initial call reaches the OSS, the operator gathers information regarding how the call is to be billed before proceeding.
3. Verification of the requested billing may take the form of a database query (for Calling Card and Commercial Credit Card) or of a call. In the latter case, call processing closely follows that for Call Completion. For Collect or Reverse Charging, call processing parallels Person Handling; a call is set up to the called party and verification of the billing (rather than the availability of a

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particular person) determines whether the call proceeds. For Third Number Billing, a separate call (following the Call Completion procedures) connects the operator to the proposed billing number. If the billing arrangement is verified, call processing proceeds as for Call Completion.

No protocol or signaling procedure enhancements are required to support Alternate Billing.

Annex B

(normative)

B: oli Parameter

B.1 Semantics and Usage of the oli Parameter

The SS7 ISUP OLI parameter is an ANSI ISUP parameter which conveys the “class of service” characterizing the calling party. Examples include: 07, *Special Operator Handling Required*, 29, *prison/inmate service* 34, *Telco Operator Handled Call*. Note that the values for this ANSI ISUP parameter are two-digit numeric values and are the same values as conveyed by the ANI II digits used in North American MF signaling. Operator Services and other services make use of the OLI value in determining how to service calls. For example, the types of calls offered by Operator Services systems may be restricted when the OLI value indicates: 29, *prison/inmate service*.

B.1.1 Semantics

When a SIP URI contains an oli parameter, the intended semantic is that the “class of service” and the oli parameter value is attributed to the originating access, for the purpose of handling the specific session. For example, if a P-Asserted Identity header contains a SIP URI with an oli parameter with value “29” indicating *prison/inmate service*, this means that the call/session is characterized as originating from a prison/inmate access and handling of the session within the network may take this into consideration.

Population and interpretation of “oli-value” is taken from the ANI II values registry maintained by the North American Numbering Plan Administration (NANPA) at http://www.nanpa.com/number_resource_info/ani_ii_assignments.html. That is, the value is in the format of exactly two digits, with the interpretation as given in the registry.

B.1.2 Intended Usage

For interworking from ISUP to SIP, if the incoming IAM includes an ISUP OLI parameter, then the gateway will include a P-Asserted-Identity header with the oli parameter populated with the same numeric value of the received ISUP OLI. In addition, if the gateway populates the From header of the corresponding SIP INVITE message with a SIP URI based on the ISUP Calling Party Number parameter, then any included oli parameter shall be populated with the numeric value of the received ISUP OLI parameter.

Recognizing the need for backward compatibility with gateways that do not populate the P-Asserted-Identity header as specified in this Annex, if:

- The gateway does not include a P-Asserted-Identity header, then a downstream element within the same administrative domain as the gateway may insert a P-Asserted-Identity header containing a SIP URI, with an oli parameter based on the oli parameter in the From header.
- The gateway does include a P-Asserted-Identity header with a SIP URI, but does not include an oli parameter in the P-Asserted-Identity header, then, a downstream element within the same administrative domain as the gateway may insert an oli parameter into the P-Asserted-Identity value based on the oli value in the From header.

For interworking from SIP to ISUP,

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- If the incoming INVITE contains a P-Asserted-Identity header containing a SIP URI with an oli parameter (whether the INVITE does or does not include a From header with an oli parameter), then the gateway includes an OLI parameter in the outgoing IAM containing the value received in the P-Asserted-Identity header.
- If the incoming INVITE contains a From header with an oli parameter but
 - The INVITE does not contain a P-Asserted-Identity header, or
 - The P-Asserted-Identity header in the INVITE does not contain a SIP URI with an oli parameter,Then the gateway includes an OLI parameter containing the received oli value in the outgoing IAM⁶.

Since it is desirable for the network to assert the class of service associated with a call/session from a SIP based originating endpoint and since the oli is a parameter, it is sent in the P-Asserted-Identity header, which also carries the network asserted identity. The oli parameter is not expected to be sent in the From header for calls originated from a SIP endpoint.

While this standard defines oli as a SIP URI parameter, it also notes the further need to support its use as a tel URI parameter. The syntax, semantics, and usage of the oli parameter when used as a tel URI parameter are identical to those when it is used as SIP URI parameter. In addition, the ATIS procedures for use of the oli SIP URI parameter are also applicable for use with tel URIs.

B.2 oli SIP URI Parameter in a From Header

Some existing implementations of PSTN-SIP gateways populate an “oli” SIP URI parameter in the From header to convey in SIP the information corresponding to a received SS7 ISUP Originating Line Information (OLI) parameter⁷. Thus, if the incoming IAM contains an OLI parameter, and if the gateway populates the From header of the corresponding SIP INVITE message with a SIP URI based on the ISUP Calling Party Number parameter, then the gateway also includes the oli parameter populated with the numeric value of the received ISUP OLI parameter.

For example, if one of these gateways receives an incoming IAM message with calling party number 732-758-5735, and OLI value of “29”, then the gateway includes in the outgoing SIP INVITE message the following From header:

From: sip:+1-732-758-5735@example.com;user=phone;oli=29.

Note that these existing implementations will populate an outgoing ISUP OLI parameter with the value of an oli parameter received in the From header. Therefore, the NGN with such a SIP-PSTN gateway should assure the value in the oli parameter in the From header is the same as the value in the oli parameter in the P-Asserted Identity header.

⁶ Recognize that the gateway is trusting information in a From header (e.g., when that information comes from another gateway which has put the oli parameter in the From header based on current implementation.)

⁷ The use of the oli parameter is described in [3GPP 24.229], clause 7.2A.12.