



ATIS-1000050.2012(S2022)

**Next Generation Network (NGN) Operator Regular
Intercept Standard**

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ATIS-1000050.2012(S2022), *Next Generation Network (NGN) Operator Regular Intercept Standard*

Is an American National Standard developed by the ATIS **Packet Technologies and Systems Committee (PTSC)**.

Published by

**Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005**

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American National Standard for Telecommunications

Next Generation Network (NGN) Operator Regular Intercept Standard

Alliance for Telecommunications Industry Solutions

Approved August 9, 2012

American National Standards Institute, Inc.

Abstract

This standard describes Next Generation Network (NGN) signaling support for Operator Regular Intercept. The service may be invoked when an NGN caller attempts to set up a call/session to an 'intercepted number' where the address of the called party has changed. In the event that the old address has been replaced with multiple new addresses ('split referral'), interaction with the calling party is required to determine the appropriate new address ('referral number').

The scope of this standard includes:

1. SIP signaling for a call reaching an intercepted number in the NGN where all or part of the Intercept Service is provided by the Application Server associated with other terminating services for this destination.
2. SIP signaling for a call to an intercepted number in the NGN that is directed toward an application in the NGN, or in another NGN, which provides the Intercept Service.

Foreword

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

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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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- M. Dolly, PTSC SAC Chair (AT&T)
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The Signaling, Architecture, and Control (SAC) Subcommittee was responsible for the development of this document.

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American National Standard on –

Next Generation Network (NGN) Operator Regular Intercept Standard

1 Introduction

This standard describes NGN signaling support for Operator Regular Intercept service, analogous to the PSTN Intercept service defined in [ATIS-1000666] for the case where the called party number has changed. When a call termination attempt is addressed to such a destination (the ‘intercepted number’), then Operator Regular Intercept service indicates (e.g., by playing a verbal announcement) the reason the call cannot be completed. Additional information may also be provided to the caller – e.g., providing a new destination address. Operator Regular Intercept service may also offer the caller Call Completion functionality, as defined in [ATIS-1000027].

2 Scope & Purpose

This standard describes NGN signaling support for Operator Regular Intercept service.

The details of the caller interaction with the Operator Regular Intercept service (e.g., the specific announcement that might be provided to the caller) are out of the scope of this standard.

The scope of this standard includes:

1. SIP signaling for a call reaching an intercepted number in the NGN where all or part of the Intercept Service is provided by the Application Server associated with other terminating services for this destination.
2. SIP signaling for a call reaching an intercepted number in the NGN and directed toward an application in the NGN, or in another NGN, which provides the Intercept Service.

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.

3 Normative References

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

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

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> >

[ATIS-1000036] ATIS-1000036, *Technical Report on NGN Operator Services*.¹

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

[ATIS-1000027] ATIS-1000027, *Operator Services Technical Report*.¹

[ATIS-1000666] ATIS-1000666.1999 (R2009), *Signalling System No. 7 – Operator Services Network Capabilities*.¹

4 Definitions & Acronyms

4.1 Definitions

Called Party – The user subscribed to the address that is optionally provided to the Calling Party. NGN Operator Regular Intercept service may offer call completion service to the Called Party. Note that in the case of ‘split referral’, there may be more than one Called Party associated with one Intercepted Number.

Calling Party – The user attempting call/session setup to an address that is subject to NGN Operator Intercept service.

Intercepted Address – The address of a subscribed user that is subject to NGN Operator Regular Intercept service.

Intercepting Network – The network that serves the address that is subject to NGN Operator Regular Intercept service.

NGN Operator Regular Intercept Service Provider Network – The network responsible for providing NGN Operator Regular Intercept service functionality beyond a simple signal or announcement to the Calling Party identifying the reason for the Intercept service. The NGN Operator Regular Intercept Service Provider Network may also provide the simple signal or announcement.

Originating Network – The network where a call/session originates, destined for an address that is subject to NGN Operator Regular Intercept service.

Referral Number – The new address (of the Called Party) optionally provided to the Calling Party.

4.2 Acronyms

AS	Application Server
BGCF	Breakout Gateway Control Function
BGF	Border Gateway Function
CSCF	Call Session Control Function
HSS	Home Subscriber Server
MGCF	Media Gateway Control Function
MGF	Media Gateway Function
MRB	Media Resource Broker
MRFP	Multimedia Resource Function Processor
MS	Media Server
NGN	Next Generation Network
PCRF	Policy and Charging Rules Function
PSTN	Public Switching Telephone Network

RFC	(IETF) Request for Comments
RTP	Real-time Transport Protocol
SBC	Session Border Controller
SCIM	Service Capability Interaction Manager
SDP	Session Description Protocol
SGF	Signaling Gateway Function
SIP	Session Initiation Protocol
SLF	Subscription Locator Function
UDR	Universal Data Repository
UE	User Entity
VoIP	Voice over Internet Protocol

5 Architecture

The network architecture that provides the context for NGN Operator Regular Intercept Service is based on expected revisions of the network architecture found in [ATIS-100018].

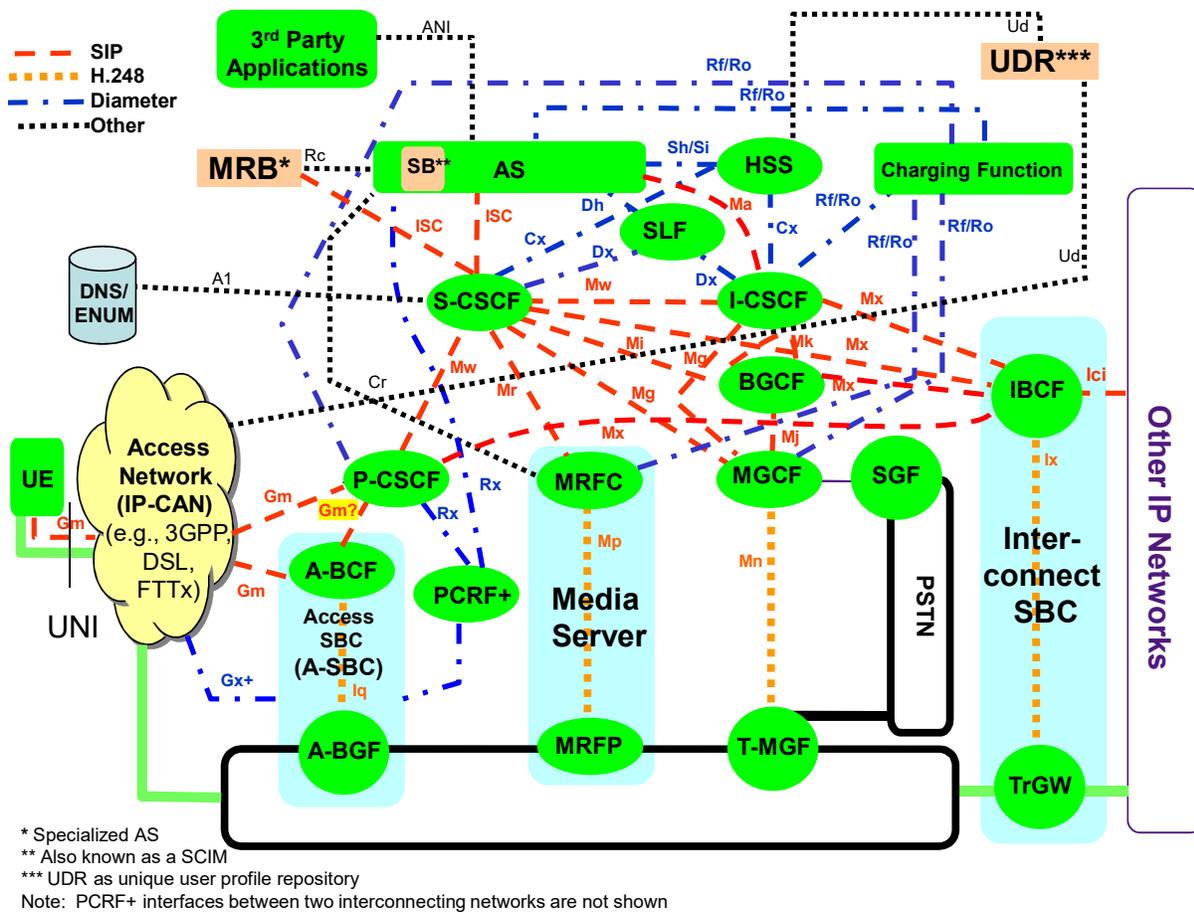


Figure 1: ATIS NGN Functional Architecture

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.

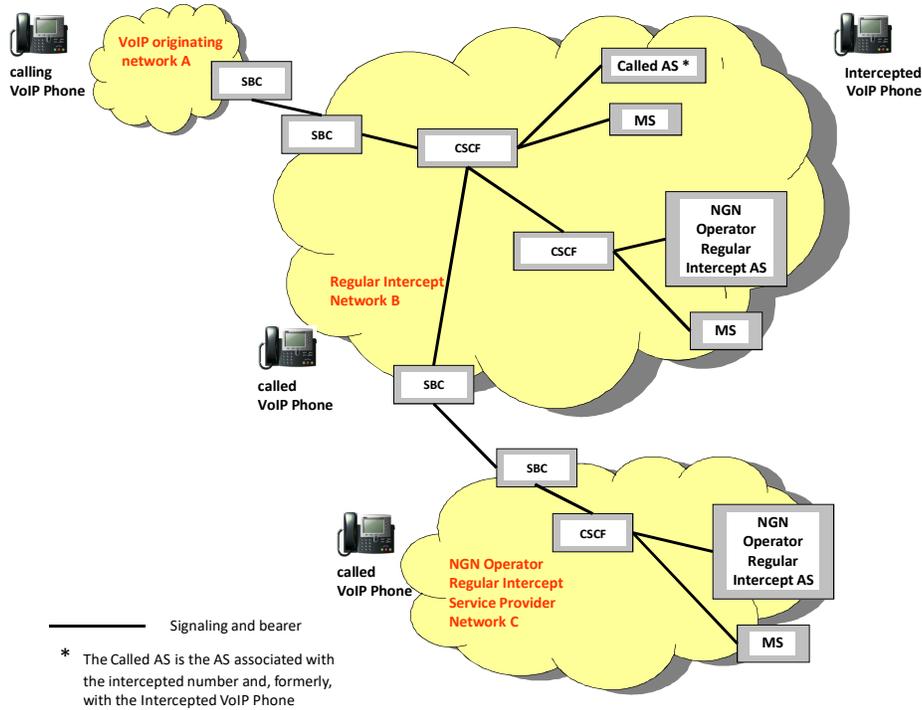


Figure 2: NGN Operator Intercept Architecture

The Calling Party initiates a call/session from the calling VoIP phone², providing the address of the Intercepted VoIP phone as the destination for the call/session. The Called AS provides initial handling of the Intercept Service. This may include providing an announcement from the associated MS. That announcement may or may not include a referral number. That is, the announcement could only provide the information that the called party can no longer be reached at this address or it could provide a new address.

Regular Intercept Network B may alternatively arrange routing at the CSCF to direct call/session setup requests for the Intercepted address directly to the NGN Operator Regular Intercept AS – e.g., because desired service functionality is not available at the Called AS.

If no announcement is provided at the Called AS or if further Intercept Service functions are desired (e.g., if the Called AS does not provide the referral number or if additional functions such as call completion are desired), Intercept Service functionality at the Called AS will include redirecting the call/session to the NGN Operator Regular Intercept AS. The NGN Operator Regular Intercept AS may be in the Regular Intercept Network B or in the NGN Operator regular Intercept Service Provider Network C.

Depending on the handling that is expected to have occurred at the Called AS, the NGN Operator Regular Intercept AS may provide an announcement, with or without a referral number, and additional services, such as call completion.

² Note that the Calling Party could also initiate this as a call from a PSTN phone. In that case, the call would enter the NGN as a call/session request from a PSTN Gateway.

6 Service Requirements

In the PSTN, Operator Regular Intercept service is used for the case where the called party has moved (i.e., the number associated with the called party has changed), but where a service remains at the terminating switch to indicate this fact to the calling party and potentially to provide additional services (e.g., providing the new number to the calling party and to offer call completion service to that new number). The simplest versions of the service (only providing an indication that the number has changed or playing an announcement providing the new number) may be provided at the PSTN terminating switch or may be provided at an operator services switch. More complex versions of the service that require interaction with the calling party (e.g., to offer call completion service) are provided by the operator services switch.

In the NGN, these different categories of versions of the service may be illustrated by two use cases:

1. *Simple Announcement.* In the Simple Announcement use case, the calling party initiates a communication session toward a destination. The terminating Service Provider determines that Operator Regular Intercept service should be invoked and directs the session to the appropriate AS. This may or may not involve redirection of the session setup. The AS directs an MS to play the required announcement, indicating that the address provided by the calling party no longer reaches the called party. The announcement may optionally include a referral number (i.e., a new address that will reach the called party.) How the referral number is derived is outside the scope of this standard.
2. *User Interaction Required.* In the User Interaction Required case, the calling party initiates a communication session toward a destination. The terminating Service Provider determines that Operator Regular Intercept service should be invoked and directs the session to the appropriate AS. This may or may not involve redirection of the session setup. The AS directs an MS to play the required announcement, indicating that the address provided by the calling party no longer reaches the called party. For this use case, the address information originally provided by the calling party is insufficient for the AS to provide all of its services. For example, the AS may require user interaction to determine whether to provide a call completion service. As another example, the mapping of the calling party-provided address information to the referral number may be ambiguous. Therefore, the AS/MS interaction with the calling party includes:
 - a. Any required announcement, indicating that the address provided by the calling party no longer reaches the called party.
 - b. Interaction to obtain the additional required information.
 - c. Additional announcements or services, possibly including providing a referral number or offering a call completion service.

Section 6.1 of this standard describes options for meeting the Simple Use Case of the service at a Terminating AS or at an AS specific to the service.

Section 6.2 of this standard describes options for meeting the User Interaction Required Use Case.

Figure 3 illustrates the service options for routing the call and for interacting with the calling party and maps these to the two use cases.

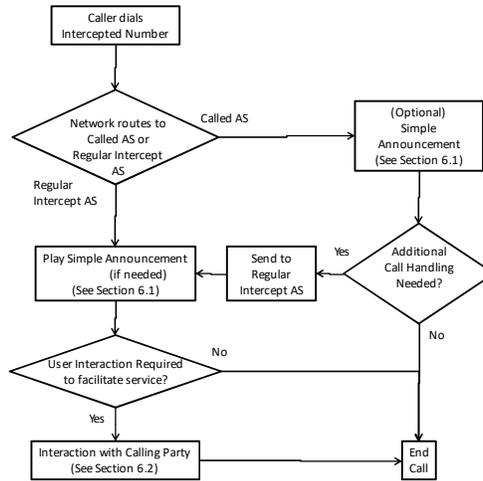


Figure 3: Service Flow

Thus, call signaling may either:

- a. Reach the Called AS, where an announcement is played and the call terminates.
- b. Reach the Called AS then progress to the Regular Intercept AS for call handling, possibly including an announcement and interaction with the user.
- c. Reach the Regular Intercept AS for call handling without encountering the Called AS.

6.1 Simple Announcement

It is a network option whether to provide the information regarding the reason for NGN Operator Regular Intercept Service from an MS controlled by the Terminating AS (i.e., as a terminating feature for calls to this number) or from an MS controlled by an AS in the Operator Services System.

Where NGN Operator Intercept Service is provided from an MS controlled by an AS in the Operator Services System, the AS may be specific to NGN Operator Regular Intercept Service (as shown in Figure 2) or NGN Operator Intercept Service may be integrated with other Operator Services.

6.1.1 Simple Announcement Controlled by Terminating AS

There are no signaling requirements specific to NGN Operator Regular Intercept service for a simple signal or announcement provided to the calling party from the terminating AS. Normal call/session setup procedures set up a call path to the destination. The destination may be the same as before Regular Intercept was invoked (i.e., the terminating AS may be the Called AS). Alternatively, the network may, as part of activation of the service, configure routing to the Intercepted number to terminate at an Intercept-specific AS. In either case, the Terminating AS (Called AS or Intercept-specific AS), recognizing the need for NGN Operator Regular Intercept service, directs the call/session to the MS responsible for other terminating services at this destination and instructs the MS to provide the appropriate information inband to the Calling Party. An illustrative example of the messaging to set up the call is shown in Annex A (section A.1).

6.2 Interaction with Calling Party

6.2.1 Service Description

The NGN Operator Intercept Service includes the option for its service logic to require further input from the Calling Party (beyond the fact that the initial call/session setup was directed to a number that is subject to the service). One use case for interaction with the Calling Party occurs when there are multiple referral numbers. For example, a joint business venture (e.g., a medical or dental practice) may disband, with the different partners moving to different new network addresses. In this case, the Calling Party may know which of the former partners is the intended Called Party. When the call/session is set up to the initial address, an announcement is played to inform the Calling Party that the number has been changed and that additional information is needed to disambiguate the desired destination address. The interaction with the Calling Party may be automated or may use live operator assistance; the mechanism for deriving the appropriate referral address is outside the scope of this document.

Once the desired destination address has been derived, the options for follow-on services (e.g., Call Completion) are similar to the Simple Announcement use cases.

6.2.2 Interaction Provided by NGN Operator Intercept Service AS

As with the simple announcement case, there are no signaling requirements specific to NGN Operator Regular Intercept service delivering the initial call/session setup to a number subject to Regular Intercept service. Normal call/session setup procedures set up a call path to the AS associated with the called number. This may be the Called AS (i.e., the AS associated with the number before Regular Intercept service was activated), or may be the Regular Intercept AS (i.e., an AS that provides Regular Intercept processing). In the former case, the service will redirect the call/session setup to an AS that will play any necessary announcements, interact with the calling party to obtain the information needed to derive the referral number, or offer additional services. The Regular Intercept AS may be in the same network as the Intercepted Number AS (and, therefore, in the same network as the Called AS) or may be in a different network. Annex A contains illustrative examples of the messaging to set up the call using SIP Redirection (section A.2) and using Call Forwarding (section A.3) for the use case where the Intercepted Number AS and the AS that provides Regular Intercept processing are in different networks.

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 Detailed Example Service Call Flows

A.1. Simple Announcement

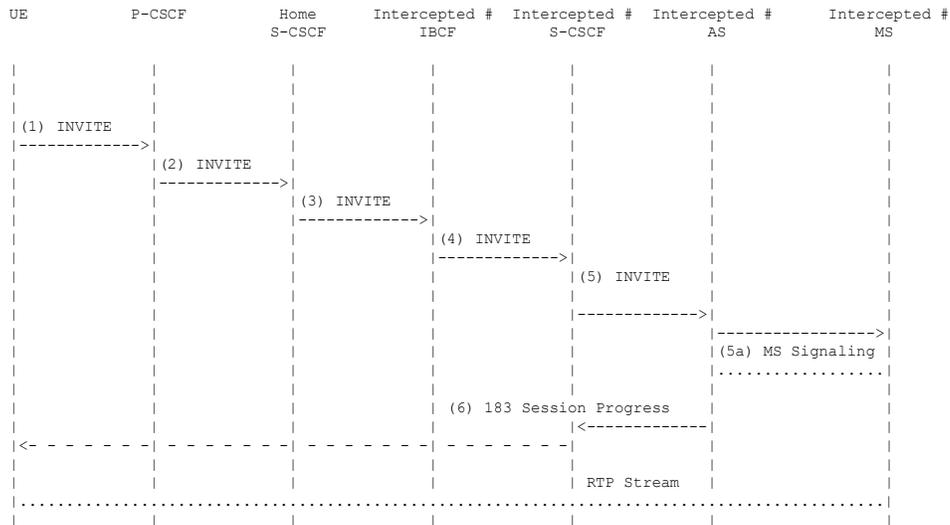


Figure A.1: Call Setup to the Terminating AS

1. INVITE UE -> P-CSCF

INVITE sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone SIP/2.0

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

Max-Forwards: 70

From: <sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>;tag=1234567

To: <sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@ 192.0.4.1:5060>

P-Preferred-Identity: <sip:+1NPA-NXX-ORIG@provider-a.com;user=phone>

Content-Type: application/sdp

Content-Length: ...

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

- a. *Request-URI* – indicates the user to which this request is being addressed. In this example, the Request-URI contains a “SIP” URI indicating the digits “1+NPA-NXX-DIAL” as dialed by the user. The domain identifies the caller’s home provider, and the “user=phone” URI parameter is provided to identify that this represents a phone number.
- 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 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 are not trusted; rather, the network asserted identity in the 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 UE supplies the Address of Record that was used at registration.
- 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*. At a minimum, application/sdp will be indicated.
- 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-> Home S-CSCF

INVITE sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone SIP/2.0

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bg0

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: <sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>;tag=1234567

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To: < sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>CSeq: 101 INVITE
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2
Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060>
P-Asserted-Identity: "+1NPA-NXX-ORIG" < sip:+1NPA-NXX-ORIG@provider-a.com; user=phone>
P-Charging-Vector: icid-value=1234bc9876e;
 icid-generated-at=192.0.6.8;
 orig-ioi=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 has changed:

- a. *Via* – The P-CSCF adds a Via header identifying itself to the top of the Via header stack.
- b. *Max-Forwards* – The P-CSCF decrements the value by one.
- c. *Record-Route* –The P-CSCF inserts itself into the Record-Route header.
- d. *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. The P-CSCF inserts the P-Asserted-Identity header field, in this example, asserting the identity received in the P-Preferred-Identity.
- e. *P-Preferred-Identity* – The P-CSCF deletes the P-Preferred Identity
- f. *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. The P-CSCF inserts the P-Charging-Vector.

3. INVITE Home S-CSCF-> Intercepted Number IBCF

INVITE sip:+1-NPA-NXX-DIAL@provider-b.com; user=phone SIP/2.0
Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1
Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch= z9hG4bK74bg0
Via: SIP/2.0/UDP client.provider-a.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 68
Record-Route: < sip:s-cscf-1.provider-a.com>
Record-Route: < sip:p-cscf-1.provider-a.com>
From: < sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>; tag=1234567
To: < sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>
CSeq: 101 INVITE
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2

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Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060>

P-Asserted-Identity: "+1NPA-NXX-ORIG" < sip:+1NPA-NXX-ORIG@provider-a.com;user=phone>

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

 icid-generated-at=192.0.6.8;

 orig-ioi=provider-a.com

Content-Type: application/sdp

Content-Length: ...

The S-CSCF determines the home provider of the called party, changes the domain in the Request-URI accordingly, and routes the INVITE to the ingress node of this domain. It 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 has changed:

1. *Request-URI* – The S-CSCF replaces the domain with the home domain of called party.
2. *Via* – The S-CSCF adds a Via header identifying itself to the top of the Via header stack.
3. *Max-Forwards* – The S-CSCF decrements the value by one.
4. *Record-Route* – The S-CSCF inserts itself into the Record-Route header.

There may be separate IBCFs in each (provider-a and provider-b) network.

4. INVITE Intercepted Number IBCF -> Intercepted Number S-CSCF

INVITE sip:+1-NPA-NXX-DIAL@provider-b.com; user=phone SIP/2.0

Via: SIP/2.0/UDP ibcf-1.provider-b.com:5060;branch=z9hG4bK74bi2

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch= z9hG4bK74bg0

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

Max-Forwards: 67

Record-Route: < sip:ibcf-1.provider-b.com>

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

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

From: < sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>;tag=1234567

To: < sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060>

P-Asserted-Identity: "+1NPA-NXX-ORIG" < sip:+1NPA-NXX-ORIG@provider-a.com;user=phone>

P-Charging-Vector: icid-value=2345bc9876e;

 icid-generated-at=192.1.6.8;

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orig-voi=provider-b.com

Content-Type: application/sdp

Content-Length: ...

The ingress node routes the call to the S-CSCF. It 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 has changed:

- a. *Via* – The IBCF adds a *Via* header identifying itself to the top of the *Via* header stack.
- b. *Max-Forwards* – The IBCF decrements the value.
- c. *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.
- d. *P-Charging-Vector* – A new *P-Charging-Vector* is generated for use in this domain.

There may be separate IBCFs in each (provider-a and provider-b) network. Each IBCF makes similar changes to the INVITE.

5. INVITE Intercepted Number S-CSCF -> Intercepted Number AS

INVITE sip:+1-NPA-NXX-DIAL@provider-b.com; user=phone SIP/2.0

Via: SIP/2.0/UDP s-cscf-2.provider-b.com:5060;branch=z9hG4bK74bj3

Via: SIP/2.0/UDP ibcf-1.provider-b.com:5060;branch=z9hG4bK74bi2

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bg0

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

Max-Forwards: 66

Record-Route: <sip:s-cscf-2.provider-b.com>

Record-Route: <sip:ibcf-1.provider-b.com>

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

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

From: <sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>; tag=1234567

To: <sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060>

P-Asserted-Identity: "+1NPA-NXX-ORIG" <sip:+1NPA-NXX-ORIG@provider-a.com;user=phone>

P-Charging-Vector: icid-value=2345bc9876e;

icid-generated-at=192.1.6.8;

orig-voi=provider-b.com

Content-Type: application/sdp

Content-Length: ...

Even though there is no longer a user associated with this number, the provider associated with the intercepted number maintains records about this number, including the address of the AS to which incoming requests for this number are to be forwarded. This is done in order to implement the Intercept service. The S-CSCF forwards the request to this AS the same as it would for any other incoming request that gets routed to an AS. It formulates the INVITE appropriately and sends it to on the AS.

Therefore, this INVITE has changed:

- a. *Via* – The S-CSCF adds a *Via* header identifying itself to the top of the *Via* header stack.
- b. *Max-Forwards* – The S-CSCF decrements the value by one.
- c. *Record-Route* – The S-CSCF inserts itself into the *Record-Route* header.

5a. AS-MS Signaling Intercepted Number AS <--> MS

The AS identifies the appropriate intercept announcement to be played, identifies an appropriate, available MS to play this message, and performs the appropriate signaling to cause the MS to play this message via RTP to the caller. The signaling associated with this step is the same as for any other connection of an MS to play announcement, and is only specific to the intercept service insofar that the specified announcement is intercept specific. The MS plays the announcement to the caller, and in a typical scenario, the user would hang up, potentially noting any new number that might be provided as part of this service.

6. Intercepted Number AS -> Intercepted Number S-CSCF

SIP/2.0 183 Session Progress

Via: SIP/2.0/UDP s-cscf-2.provider-b.com:5060;branch=z9hG4bK74bj3

Via: SIP/2.0/UDP ibcf-1.provider-b.com:5060;branch=z9hG4bK74bi2

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bg0

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

Record-Route: <sip:s-cscf-2.provider-b.com>

Record-Route: <sip:ibcf-1.provider-b.com>

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

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

From: <sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone>;tag=1234567

To: <sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone>; tag=2345678

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060>

Content-Type: application/sdp

Content-Length: ...

The 183 is passed backward to the UE, removing the appropriate Via headers along the way.

A.2. Redirect to NGN Operator Intercept Service

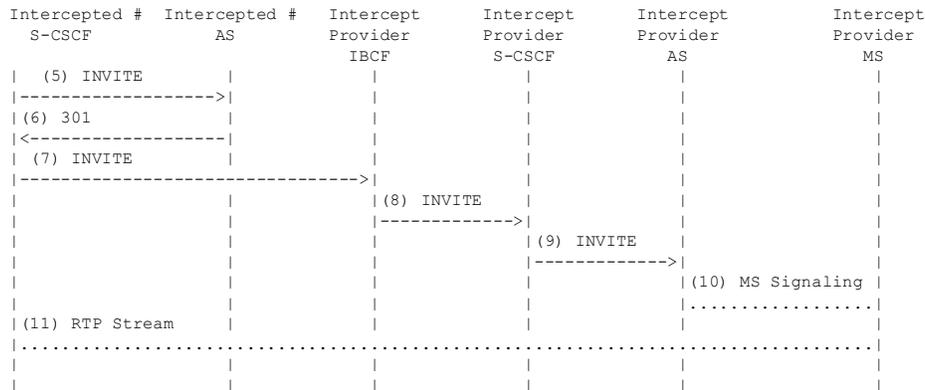


Figure A.2: Call Setup Redirected to an NGN Operator Intercept Service AS

Steps (1) through (5) in Figure A.2 are the same as the corresponding steps in Figure A.1.

```

6. 301 Redirect AS->Intercepted # S-CSCF
SIP/2.0 301 Moved Permanently
From: <sip:+1-NPA-ORIG@provider-a.com; user=phone>;tag=1234567
To: sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone
CSeq: 101 INVITE
Call-ID: f81d4fae-7dec-11d0-a765-00a0c91e6bf6@192.168.1.2
Contact: <sip:intercept-regular@provider-c.com>
    
```

The AS analyses the dialed digits along with user profile information, and determines that this is a call requiring Intercept treatment, and identifies the provider where this is to be sent. It thus redirects the request from “sip: +1-NPA-NXX-DIAL@provider-b.com” to “sip:intercept-regular@provider-c.com”. It accomplishes this by creating a 301 “Moved Permanently” response, populating the Contact header field with the URI to which the request should be redirected, and returning this message in response to the received INVITE message. [Note that if the “intercept-regular” is in the provider-b network, then steps (8) and (9) below are replaced by a single step routing this INVITE to intercept-regular@provide-b.com.]

The 301 includes (but is not limited to):

- d. Response code of "301".

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- e. Contact header specifying the new destination to which the request is to be forwarded.
- f. All other header fields are populated as for any other non final SIP response. No SDP is included and no media session is established.

7. INVITE Intercepted Number S-CSCF -> Intercept Provider IBCF

INVITE sip:intercept-regular@provider-c.com SIP/2.0

Via: SIP/2.0/UDP s-cscf-2.provider-b.com:5060;branch=z9hG4bK74bj3

Via: SIP/2.0/UDP ibcf-1.provider-b.com:5060;branch=z9hG4bK74bi2

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bg0

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

Max-Forwards: 65

Record-Route: < sip:s-cscf-2.provider-a.com >

Record-Route: < sip:ibcf-1.provider-b.com >

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

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

From: < sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone >; tag=1234567

To: < sip:+1-NPA-NXX-DIAL@ provider-a.com; user=phone >

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060 >

P-Asserted-Identity: "+1NPA-NXX-ORIG" < sip:+1NPA-NXX-ORIG@provider-a.com; user=phone >

P-Charging-Vector: icid-value=3456bc9876e;

 icid-generated-at=192.2.6.8;

 orig-voi=provider-c.com

History-Info: < sip:+1-NPA-NXX-DIAL@provider-a.com; user=phone?Reason=SIP;Cause=301;text="Moved Permanently" >; index=1,

 < sip: intercept-regular@provider-c.com; user=phone >; index=1.1

Diversion: < sip:+1-NPA-NXX-DIAL;phone-context=provider-b.com; reason=unconditional >

Content-Type: application/sdp

Content-Length: ...

When the S-CSCF receives the 301 response from the AS, it retries the request using the value contained in the Contact header field of the 301 response as the new Request-URI in the INVITE. Additionally, it adds a History-Info header field identifying the original called number (i.e., the intercepted number).

Therefore, this INVITE has changed (relative to the INVITE in step 5):

- a. *Request-URI* – This is based on the value contained in the Contact header field of the 301 response. In this instance, it contains the called number as the user part, but the domain is that of the intercept provider.

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- b. *History-Info* – The History-Info with an index=1 identifies the previous value of the Request-URI, which is the original called number. The Regular Intercept service may use this value as the intercepted number when determining the referral number. The History-Info with an index=1.1 confirms that the signaling has visited an AS that is redirecting the signaling to the Regular Intercept service.
- 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. *P-Charging-Vector* – A new P-Charging-Vector is generated for use in this domain.

The remaining headers and message bodies correspond to those in step 5 in Annex A.1.

8. INVITE Intercept Provider IBCF -> I-CSCF

INVITE sip:intercept-regular@provider-c.com SIP/2.0

Via: SIP/2.0/UDP ibcf-2.provider-c.com:5060; branch=z9hG4bK74bk3

Via: SIP/2.0/UDP s-cscf-2.provider-b.com:5060;branch=z9hG4bK74bj3

Via: SIP/2.0/UDP ibcf-1.provider-b.com:5060;branch=z9hG4bK74bi2

Via: SIP/2.0/UDP s-cscf-1.provider-a.com:5060;branch=z9hG4bK74bh1

Via: SIP/2.0/UDP p-cscf-1.provider-a.com:5060;branch=z9hG4bK74bg0

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

Max-Forwards: 64

Record-Route: < sip:s-cscf-2.provider-a.com >

Record-Route: < sip:ibcf-1.provider-b.com >

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

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

From: < sip:+1-NPA-NXX-ORIG@provider-a.com; user=phone >;tag=1234567

To: < sip:+1-NPA-NXX-DIAL@ provider-a.com; user=phone >

CSeq: 101 INVITE

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

Contact: < sip:+1-NPA-NXX-ORIG@192.0.4.1:5060 >

P-Asserted-Identity: "+1NPA-NXX-ORIG" < sip:+1NPA-NXX-ORIG@provider-a.com;user=phone >

P-Charging-Vector: icid-value=3456bc9876e;

 icid-generated-at=192.2.6.8;

 orig-ioi=provider-c.com

History-Info: < sip:+1-NPA-NXX-DIAL@provider-a.com?Reason=SIP;Cause=301;text="Moved Permanently"; user=phone >; index=1,

 < sip:AS-b@provider-b.com; user=phone >; index=1.1

Diversion: < sip:+1-NPA-NXX-DIAL;reason=unconditional >

Content-Type: application/sdp

Content-Length: ...

Since this INVITE is destined to a new network, it is routed to the IBCF in the provider-c.com network, where the address of regular-intercept@provider-c.com can be resolved.

This INVITE has changed:

- a. *Via* – The IBCF adds a Via header identifying itself to the top of the Via header stack.
- b. *Max-Forwards* – The IBCF decrements the value.

The IBCF does not add to the Record Route, but does add to the Via header and changes the Max-Forwards.

Steps 9, 10, 10a, and 11, are equivalent to steps 4, 5, 5a, and 6 for delivering the call to the appropriate AS and managing the interaction with the MS in the illustrative call flow in section A.1.

A.3. Call Forwarding to NGN Operator Intercept Service

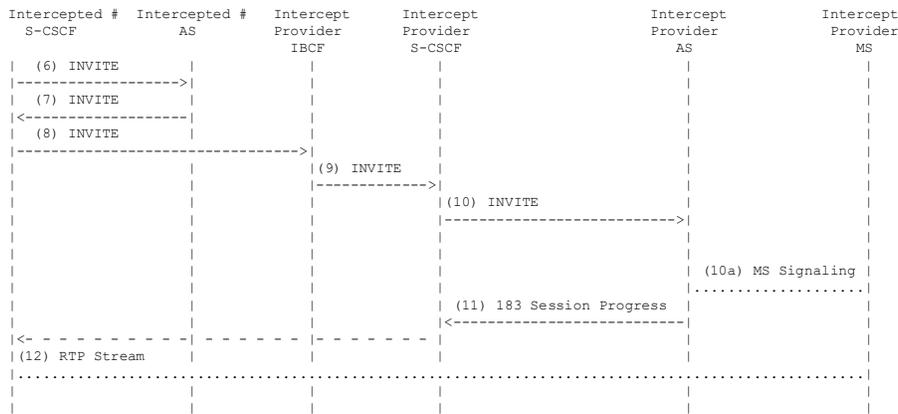


Figure A.3: Call Setup Forwarded to an NGN Operator Intercept Service AS

Steps (1) through (6) in Figure A.3 are the same as the corresponding steps in Figure A.1.

7. INVITE Intercepted Number AS -> Intercepted Number S-CSCF

INVITE sip: intercept-regular@provider-c.com; SIP/2.0

Via: SIP/2.0/UDP AS.provider.b:5060;branch=z9hG4bK74bx1

Max-Forwards: 64

Record-Route: <sip:AS.provider-b.com>

From: < AS-b@provider-b.com >;tag=3456789

To: < intercept-regular@provider-c.com >

CSeq: 101 INVITE

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

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Contact: <sip:AS-b.provider-b.com>

P-Asserted-Identity: "+1NPA-NXX-ORIG" <sip:+1NPA-NXX-ORIG; user=phone>

P-Charge-Info: <sip:+1NPA-NXX-ORIG@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: < sip:+1-NPA-NXX-DIAL@provider-a.com?Reason=SIP;Cause=301;text="Moved Permanently"; user=phone >; index=1,

 < sip:AS-b@provider-b.com; user=phone >; index=1.1

Diversion: < sip:+1-NPA-NXX-DIAL;reason=unconditional>

Content-Type: application/sdp

Content-Length: ...

The AS analyses the dialed digits along with user profile information, and determines that this is a call requiring Intercept treatment, and identifies the provider where this is to be sent. It thus retargets the request from "sip: NPA-NXX-DIAL" to "sip:regular-intercept@provider-c.com". [Note that "sip:regular-intercept..." could refer to an NGN Operator Intercept service-specific AS or it could refer to a more general operator services AS that provides NGN Operator Intercept service. Also note that this use case uses "provider-c.com" to refer to a different network than the network of the AS. If the networks are the same, then steps (5) and (6) below are replaced by a single step routing this INVITE to the NGN Operator Intercept Service AS.]

The AS, acting as a B2BUA, initiates a new call leg, sending an INVITE to the S-CSCF. Therefore, this 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: intercept-regular@provider-c.com".
- b. *History-Info* – The History-Info with an index=1 identifies the previous value of the Request-URI, which is the original called number. The Regular Intercept service may use this value as the intercepted number when determining the referral number. The History-Info with an index=1.1 confirms that the signaling has visited an AS that is redirecting the signaling to the Regular Intercept service.
- 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. The population of these is not unique to operator services.
- e. *P-Asserted-Identity* – The AS passes the received value unchanged.
- f. *P-Charge-Info* – The AS passes the received value unchanged.

Steps 8, 9, 10, and 10a are equivalent to steps 4, 5, 6, and 6a for delivering the call to the appropriate AS and managing the interaction with the MS in the illustrative call flow in section A.1.

At step 11, the 183 Session Progress follows the list of VIA headers to the B2BUA at the Intercepted Number AS, where it is regenerated and passed back to the UE, again following the list of VIA headers.