
# Interoperability Specification

# Sylantro Application Server

# IMS ISC Interface

## (Pre-Release – v1.4)

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***Record Page***

| Author(s) | Issue | Date | Comments |
|---|---|---|---|
| Mohsen Soroush | 1.0 | 09/08/2005 | Initial Draft. This document contains information on pre-release implementation of ISC interface on Sylantro Application Server and is provided soley for the purpose of interoperability discussions and IOT testing only. |
| Mohsen Soroush | 1.1 | 11/08/2005 | Updated. |
| Mohsen Soroush | 1.2 | 11/14/2005 | Editorial updates. |
| Mohsen Soroush | 1.3 | 12/21/2005 | Added additional call flow examples and filter criteria information. |
| Mohsen Soroush | 1.4 | 01/16/2006 | Errata corrections |

# Table of Contents

# 1. Introduction

## 1.1 Document Purpose and Scope

This specification describes SIP interoperability requirements and sample call flows supported by the Sylantro SIP Application Server (AS) ISC interface in an IMS compliant architecture. The information is presented in a format that can be distributed to Sylantro partners and customers as a basis for interoperability discussion and testing. The document does not contain any design, detailed architecture, or configuration information of the Sylantro Application Feature Server.

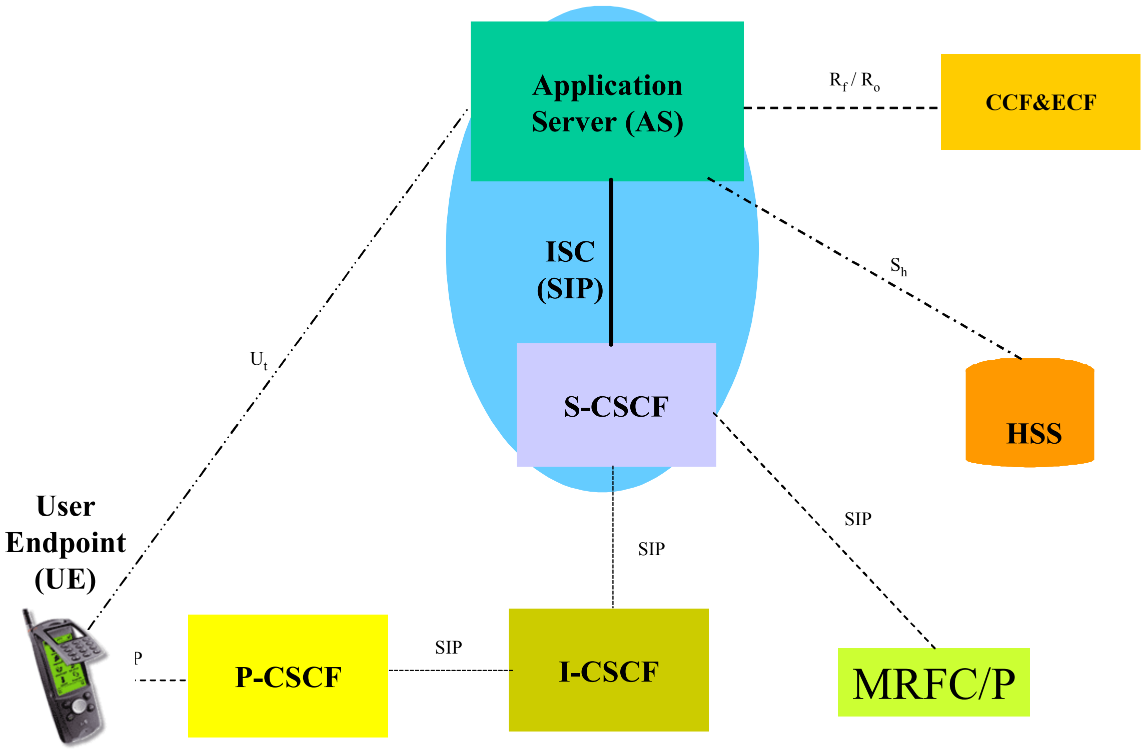
The scope of this specification is directed towards interoperability between the Sylantro SIP Application Server and IMS Serving Call Session Control Function (S-CSCF).

## 1.2 Applicability and Target Audience

This document represents a method by which a 3rd party may understand the general requirements for SIP interoperability with Sylantro SIP Application Server. This document does not represent a detailed design of the Sylantro AS. Sylantro Systems reserves the right to change and/or modify the contents of this document without notice. In particular, the call flows in this document may not be applicable to any prior or later releases of the Sylantro AS.

This document is intended for Sylantro and 3rd party partner and customer personnel involved with the interoperability project. The primary audience is network architects, system architects, systems engineers, and test engineers, but may be useful for business development, project planning, and sales engineering. ***The call flow traces in this document are a snapshot from an interoperability test bed. As such, no definite inference should be made with regard to timing and exact sequence of message exchanges as they may be different in other network deployment configurations.***

# 2. Reference IMS Architecture




## 2.1 General Information

In the above simplified IMS reference architecture, the ISC SIP interface provides the common signaling component between the IMS network and the Sylantro SIP Application Server. Sylantro SIP Application Server acts as a Back-to-Back User Agent (B2BUA) supporting SIP originations, SIP terminations, and media renegotiation through re-INVITE processing.

RTP streams flow between endpoints and network elements that process media. Sylantro AS does not originate or terminate RTP streams.

**The focus of this document is on the ISC interface and does not address the interactions between the AS and other functions such as HSS or CCF/ECF.**

## 2.2 Application Server Role

Many application servers may exist in the reference architecture, each performing a specific function in the delivery of features and applications using the common IMS signaling network. For purposes of this interoperability specification, the Sylantro Application Server hosts Business and/or Consumer subscribers that are configured for advanced features and services. The S-CSCF routes all calls originating from or terminating to Sylantro-hosted subscribers to the Sylantro AS based on initial Filter Criteria (iFC) configurations and triggers at S-CSCF.

## 2.3 Other Network Elements

There are functions that may be performed by other network elements in the reference architecture. These components are not described in the scope of this document.

# 3. IMS ISC Interface

The ISC interface is based on RFC 3261 (SIP) and specified further in 3GPP technical specifications:

1. 3GPP TS 23.228: "IP Multimedia Subsystem, Stage 2."
2. 3GPP TS 24.228: "Signaling flows for the IP multimedia call control based on SIP and SDP, Stage 3."
3. 3GPP TS 24.229: "IP Multimedia call control protocol based on SIP and SDP, Stage 3."
4. 3GPP TS 23.218 v6.1.0 "IP Multimedia (IM) session handling, IM call model, Stage 2."

Support for SIP loose routing is the cornerstone of the request routing among SIP elements within the IMS network. The pre-loaded route header values in the requests towards the AS indicate the session case (originating, terminating, terminating unregistered) and may include opaque token values that are used by the S-CSCF to correlate different call legs within a session.

The Sylantro AS acts as a B2BUA and 3$^{rd}$ party call controller in providing the advanced services and features. Depending on the features invoked for susbcribers, the Sylantro AS may act in one of two roles:

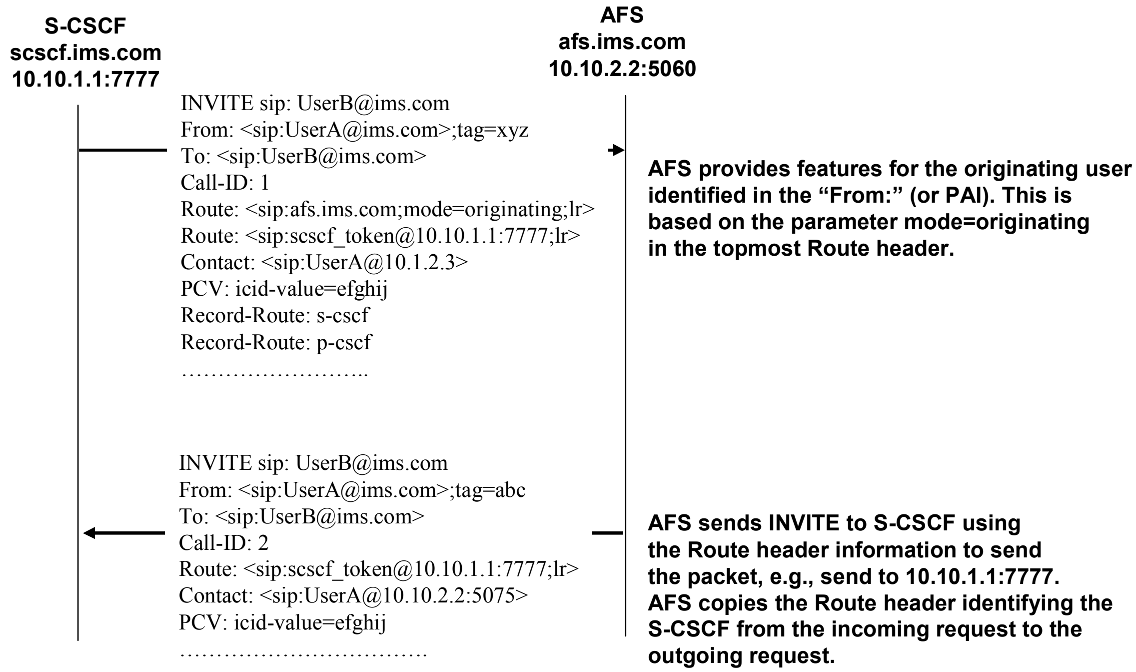
1. Routing B2BUA: AS receives a request from S-CSCF, terminates it and generates a new request, which is based on the received request.
2. Initiating B2BUA: AS initiates request(s) on behalf of hosted subscriber(s), which may be logically connected together at the AS.

The following sections describe the Sylantro Application Server interactions with S-CSCF based on the role it plays (i.e., Routing or Initiating B2BUA) and the pre-loaded route header values and parameters that identify the session case (i.e., whether the SIP request is originated by the served subscriber, terminated to the registered subscriber, or addressed to the served unregistered subscriber).

## 3.1 Originating (MO) Trigger Processing

When the originating user, identified with the SIP URI in the P-Asserted-Identity (PAI) or From header, is a subscriber hosted on the Sylantro AFS, the S-CSCF routes the request to AFS based on iFC provisioned for the subscriber at S-CSCF (Mobile_Originating trigger). The topmost Route header includes the SIP URI that identifies the AFS and a parameter or token that specifies the session case (i.e., originating). AFS processes the request and provides any features that apply to the originating user and initiates a related outgoing request. The outgoing request is routed to the S-CSCF based on the SIP URI in Route header that is copied into the outgoing message from the incoming request.

**Route header processing Call Flow**
**(url param in the AS Route header to identify originating trigger)**



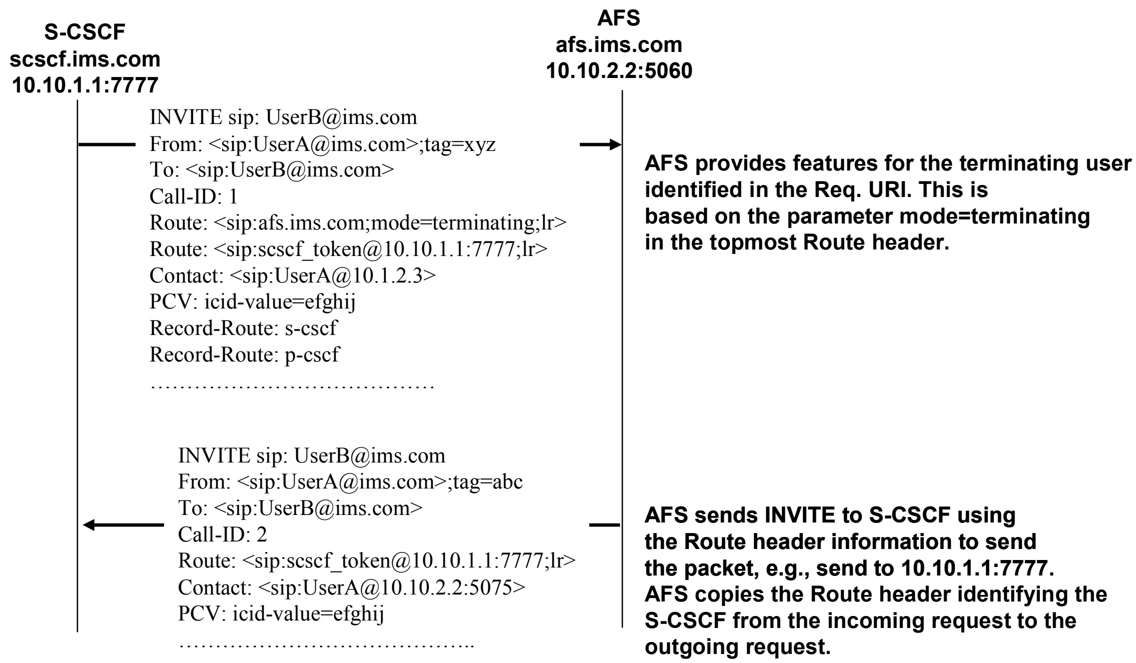

The Sylantro AS can identify the originating trigger based on one of the following optional parameters in the topmost Route header in the incoming SIP request:

- "orig" string as user part in the SIP URI (Route:<sip:***orig***@afs.ims.com;lr>): This is the preferred option and can be configured in the Originating Trigger configuration by setting the value of the application server SIP URI to <sip:orig@FQDN or IP address of AS;lr>
- mode=originating (Route:<sip:afs.ims.com;***mode=originating***;lr>)
- call=orig (Route:<sip:afs.ims.com;***call=orig***;lr>)

## 3.2 Terminating (MT) Trigger Processing

When the terminating user, identified with the SIP URI in the Request URI, is a subscriber hosted on the Sylantro AFS, the S-CSCF routes the request to AFS based on iFC provisioned for the subscriber at S-CSCF (Mobile_Terminating trigger). The topmost Route header includes the SIP URI that identifies the AFS and a parameter or token that specifies the session case (i.e., terminating). AFS processes the request and provides any features that apply to the terminating user and initiates a related outgoing request. The outgoing request is routed to the S-CSCF based on the SIP URI in Route header that is copied into the outgoing message from the incoming request.

**Route header processing Call Flow**
**(url param in the AS Route header to identify terminating trigger)**




The Sylantro AS can identify the terminating trigger based on one of the following optional parameters in the topmost Route header in the incoming SIP request:

– “term” string as user part in the SIP URI (Route:<sip:***term***@afs.ims.com;lr>): This is the preferred option and can be configured in the Terminating Trigger configuration by setting the value of the application server SIP URI to <sip:term@FQDN or IP address of AS;lr>

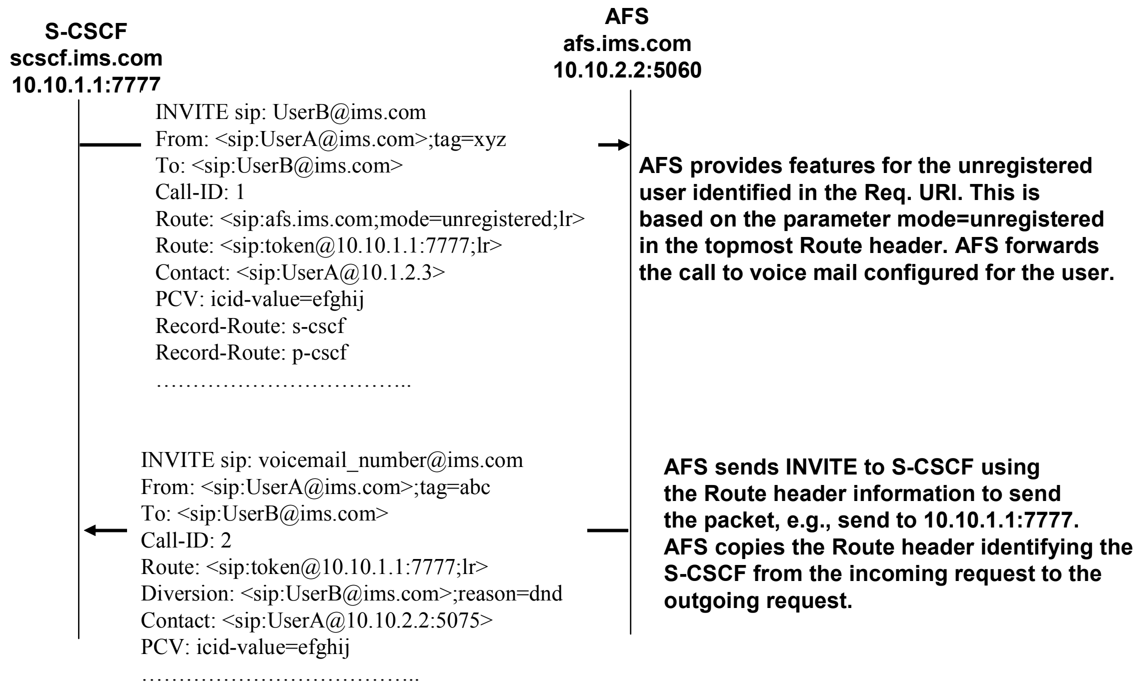
– mode=terminating (Route:<sip:afs.ims.com;***mode=terminating***;lr>)

– call=term_registered (Route:<sip:afs.ims.com;***call=term_registered***;lr>)

## 3.3 Terminating Unregistered Trigger Processing

When the terminating user, identified with the SIP URI in the Request URI, is a subscriber hosted on the Sylantro AFS and is not registered, the S-CSCF routes the request to AFS based on iFC provisioned for the subscriber at S-CSCF. The topmost Route header includes the SIP URI that identifies the AFS and a parameter or token that specifies the session case (i.e., terminating unregistered). AFS processes the request and provides any features that apply to the terminating user (forwarding to voice mail) and initiates a related outgoing request. The outgoing request is routed to the S-CSCF based on the SIP URI in Route header that is copied into the outgoing message from the incoming request.

**Route header processing Call Flow**
**(url param in the AS Route header to identify terminating unregistered trigger)**




The Sylantro AS can identify the terminating unregistered trigger based on one the following optional parameters in the topmost Route header in the incoming SIP request:

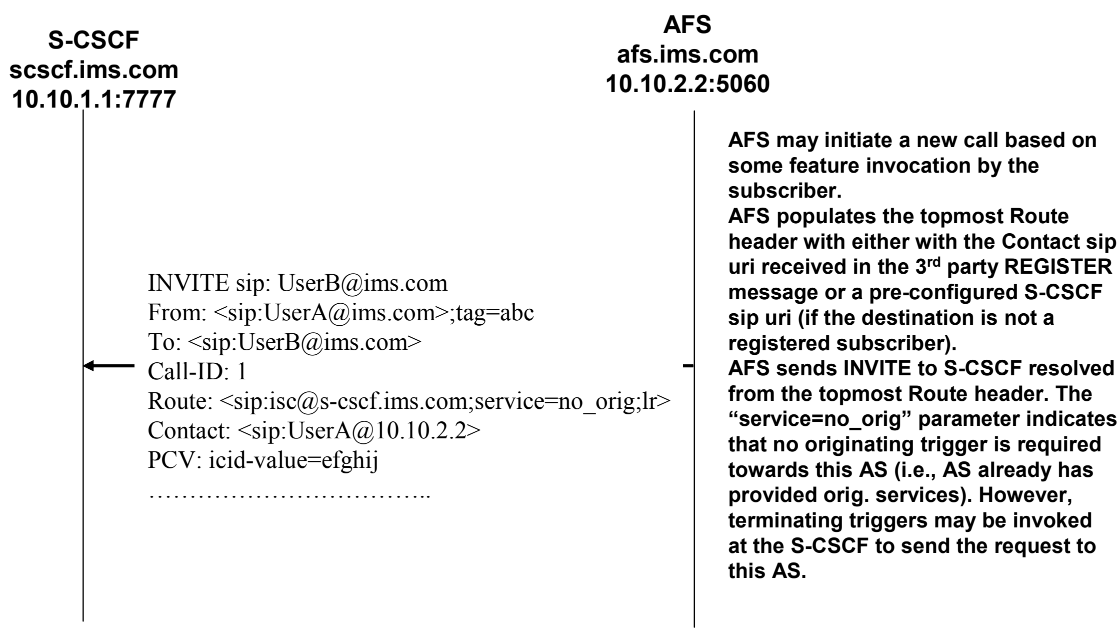
– "unregistered" as user part in the SIP URI (Route:<sip:***unregistered***@afs.ims.com;lr>): This is the preferred option and can be configured in the Terminating Unregistered Trigger configuration by setting the value of the application server SIP URI to <sip:unregistered@FQDN or IP address of AS;lr>
– mode=unregistered (Route:<sip:afs.ims.com;***mode=unregistered***;lr>)
– call=term_unregistered (Route:<sip:afs.ims.com;***call=term_unregistered***;lr>)

## 3.4 Requests initiated by the Application Server (Initiating B2BUA)

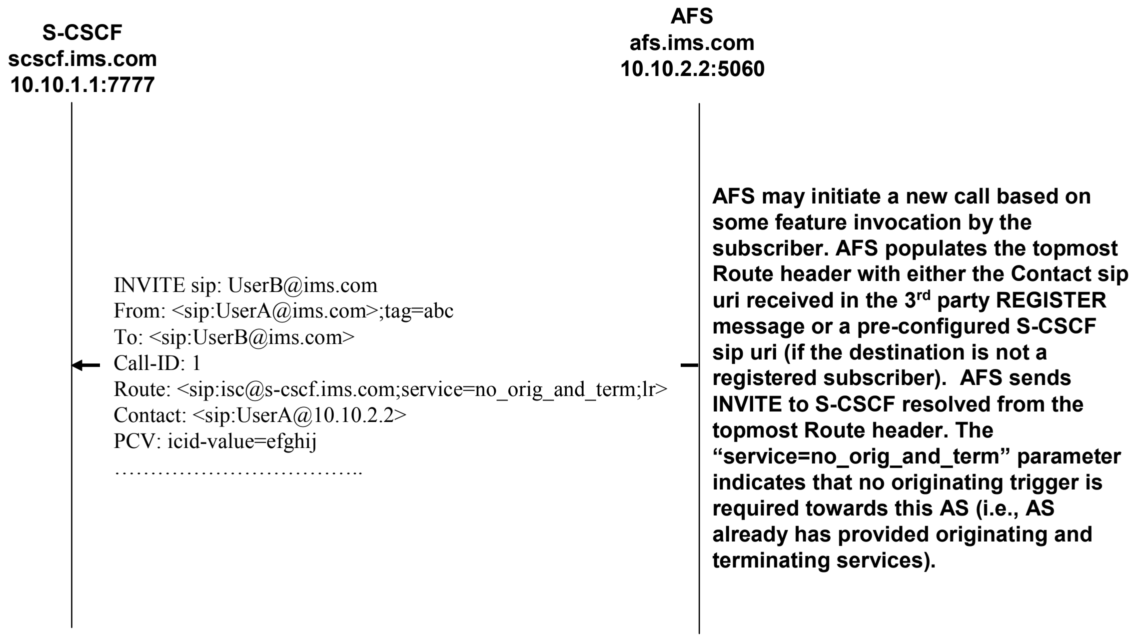
Sylantro AFS supports a suite of advanced features which makes use of new call requests initiated by the AFS on behalf of the subscribers hosted on the AFS. Examples of features that use this capability include (but not limited to) Click-to-Call, Blind Transfer, and Find Me/ Follow Me.

Since the Application Server initiates the call requests for these features on behalf of subscribers, it already has applied originating and/or terminating features for the given subscribers. In these circumstances, the S-CSCF must not apply the originating or terminating triggers that would normally invoke the Sylantro AFS for the given subscriber(s). The Sylantro AFS inserts an opaque "service" parameter in the topmost Route header that includes the S-CSCF SIP URI. The service parameter has values of "***no_orig***" and "***no_orig_and_term***" to indicate that "no originating trigger" or "no originating and terminating trigger" towards the AFS must be applied to the given request. The service parameter values can be used in the configuration of the iFC in the S-CSCF/HSS for given subscribers. ***S-CSCF is not required to intrepret the service parameter. It is only used in the evaluation of the initial filter criteria.***

**Call initiated by the Application Server:**
**No Originating trigger is required towards the AS**



**Call initiated by the Application Server:**
**No Originating and Terminating triggers are required towards the AS**




## 3.5 Third-Party Registration by Application Server

Sylantro AFS supports third-party registration of subscribers by the S-CSCF. The SIP URI in the Contact header of the 3rd party REGISTER message identifies the current Serving CSCF that is assigned to the subscriber. For new calls initiated by the AFS towards a registered subscriber (AS acting as an initiating B2BUA), the topmost Route header will be populated with the SIP URI received in the Contact header of the REGISTER message. This ensures that the request is routed to the correct Serving CSCF for terminating processing.

## 3.6 Initial Filter Criteria (iFC) Requirements

For IMS subscribers that are hosted on Sylantro Application Server, the follwing filter criteria must be configured:

1. Each user must have a Trigger Point for SIP REGISTER (results in 3rd party registration by S-CSCF) :

   **Trigger Point = {Method = REGISTER}**

   This trigger results in the S-CSCF to initiate a 3rd party REGISTER towards the Sylantro Application Server when it receives a REGISTER request from the Subscriber's UE.

2. Each user must have Originating Trigger Point:

   **Trigger Point = {Session Case = Originating}**

   **AND**

   **{(Method = INVITE) OR (Method = SUBSCRIBE) OR (Method = NOTIFY)}**

   **AND**

   **{NOT [ (Route header that includes parameter "service = no_orig") OR**

   **(Route header that includes parameter "service = no_orig_and_term") ] }**

   This trigger results in INVITE, SUBSCRIBE, or NOTIFY requests initiated by the subscriber; as identified by the address of record in the P-Asserted-ID header (or From header if no PAI header in the request); and do not include a Route header that has a "servie=no_orig" or "service=no_orig_and_term" parameters to be routed to the Application Server. When the INVITE/SUBSCRIBE/NOTIFY requests include a Route header with "service=no_orig" or "servie=no_orig_and_term" parameter, the S-CSCF will not invoke the originating trigger towards the Sylantro Application Server.

3. Each user must have Terminating Trigger Point:

   **Trigger Point = {Session Case = Terminating}**

   **AND**

   **{(Method = INVITE) OR (Method = SUBSCRIBE) OR (Method = NOTIFY)}**

   **AND**

   **{NOT (Route header that includes parameter "service = no_orig_and_term")}**

   This trigger results in INVITE, SUBSCRIBE, or NOTIFY requests destined towards the subscriber; as identified by the address of record in the Request URI; and do not include a Route header that has a "service=no_orig_and_term" parameters to be routed to the Application Server. When the INVITE/SUBSCRIBE/NOTIFY requests include a Route header with "servie=no_orig_and_term" parameter, the S-CSCF will not invoke the terminating trigger towards the Sylantro Application Server.

# 4. Sample Call Flows

In the following sections several sample call flows are described based on the features supported by the Sylantro Application Feature Server. The intent of these examples are to display the use cases for typical SIP and call processing steps that must be supported between the AFS and the IMS Core Network elements via the ISC interface. **For a more complete set of Sylantro AFS feature call flows, please refer to "*Sylantro Application Feature Server SIP Implementation and Call Flows*" document.**

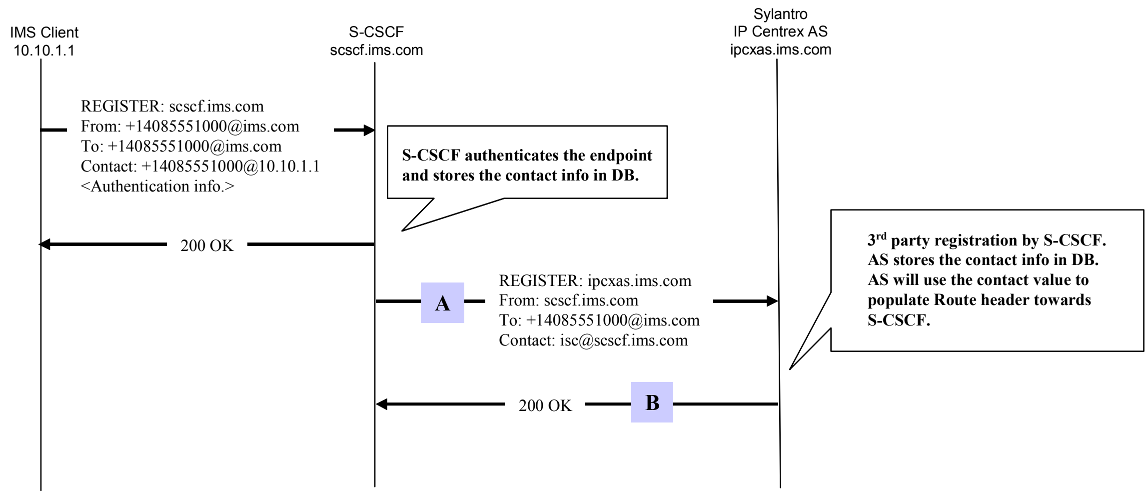
## 4.1 Assumptions

1. IMS SIP clients register with S-CSCF.
2. S-CSCF supports 3rd party SIP registration towards the Sylantro AS (In case, the S-CSCF does not support the 3rd party registration mechanism, Sylantro AFS supports a provisioning interface to bind subscribers to the SIP URI of the S-CSCF).
3. The S-CSCF manages the SIP routing to MRFC/P for features that require interaction with media servers. The Sylantro AFS initiates SIP requests for establishment of media sessions between subscribers and media servers (routes to S-CSCF as in Initiating B2BUA). It also supports direct http/vxml interface with the media server for fetching of media clips and control of media interactions.
4. The S-CSCF manages routing to SIP UM AS via ISC interface
5. Sylantro AFS is configured for IMS only (mixed operation is not yet supported)
6. Example flows use the "mode" parameter for the session case determination. However the same flows apply when the "call" parameter or strings as "user" are used to identify the session case.

**Note:** *In the following call flows, some intermediary signaling messages are not shown to reduce the clutter.* Detailed traces of specific messages between the AFS and S-CSCF are provided to highlight the interactions.

## 4.2 SIP Third-Party Registration

The following call flow diagram illustrates the IMS client SIP registration. S-CSCF performs the endpoint registration. The S-CSCF performs 3rd party registration towards the AS (based on iFC). The AS accepts third-party registration from S-CSCF based on the domain name identified in the From header SIP URI in the REGISTER message.




AS uses the SIP URI in the Contact header received in the REGISTER message to populate a Route header for insertion in the new requests (e.g., Click-to-Call, Blind Transfer, Find Me/Follow Me, etc.) that are initiated by the AS (i.e., initiating B2BUA mode).

### A. S-CSCF → Sylantro AFS

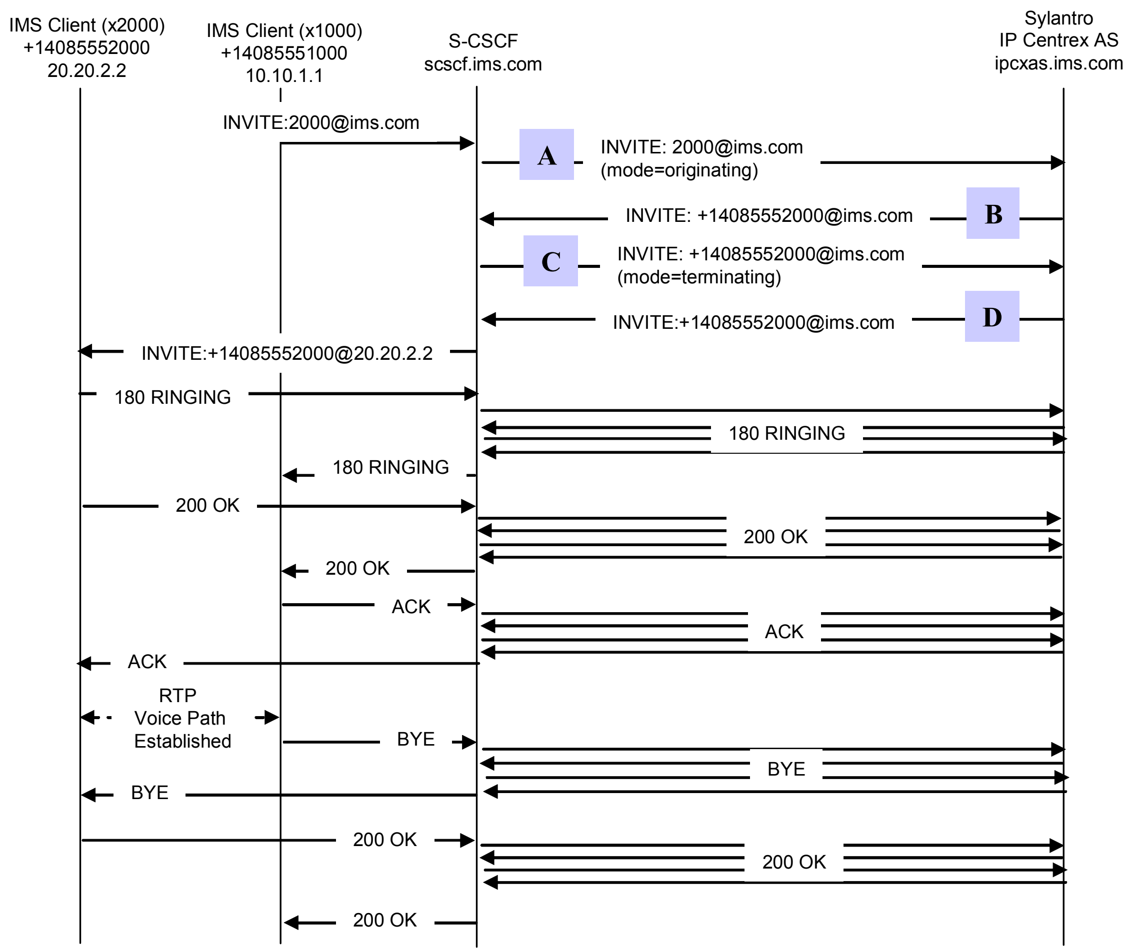
```
REGISTER sip:ipcxas.ims.com SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
From: <sip:s-cscf.ims.com>;tag=1234
To: <sip:+15105551001@ims.com;user=phone>
Call-ID: 1-3964@scscf.ims.com
Cseq: 1 REGISTER
Route:<sip:ipcxas.ims.com;lr>
Contact: <sip:isc@s-cscf.ims.com:5077;transport=tcp>
Max-Forwards: 70
Expires: 7200
```

### B. Sylantro AFS → S-CSCF

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP scscf.ims.com
CSeq: 1 REGISTER
Call-ID: 1-3964@scscf.ims.com
From: <sip:s-cscf.ims.com>;tag=1234
To: <sip:+15105551001@ims.com;user=phone>;tag=1168179722325771
Contact: <sip:isc@s-cscf.ims.com:5077;transport=tcp>
Content-Length: 0
```

## 4.3 IP Centrex-to-IP Centrex Call (Extension-to-Extension call)

The following call flow diagram illustrates the case when an enterprise subscriber places a call to a user within the same tenant (i.e., extension dialing). This call flow illustrates the case where the originating and terminating subscribers are hosted on the same Application Server. AS invokes originating user features on the "originating" trigger and terminating user features on "terminating" trigger.




Sample SIP messages between the S-CSCF and Sylantro AFS are as follows:

### A. S-CSCF → Sylantro AFS (Originating trigger):

```
INVITE sip:2000@ims.com;user=phone SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP pcscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: UserA <sip:+14085551000@ims.com;user=phone>;tag=1234
To: UserB <sip:2000@ims.com;user=phone>
Call-ID: 1-1520@10.10.1.1
Cseq: 1 INVITE
```

```
Route:<sip:ipcxas.ims.com;mode=originating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
Record-Route: <sip:pcscf.ims.com;lr>
Contact: <sip:+14085551000@10.10.1.1;user=phone>
P-Charging-Vector: icid-value=003400300a141e15
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: …..
```

### B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Originating trigger):

```
INVITE sip:+14085552000@ims.com SIP/2.0
From: "John, Doe" <sip:+14085551000@ims.com>;tag=c8a75184+cadb573e
To: +14085552000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Asserted-Identity: "John, Doe" <tel:+14085551000>
P-Asserted-Identity: "John, Doe" <sip:+14085551000@ims.com>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+14085551000@100.10.1.1:5065;transport=tcp>
Content-Length: ……
```

### C. S-CSCF → Sylantro AFS (Terminating trigger):

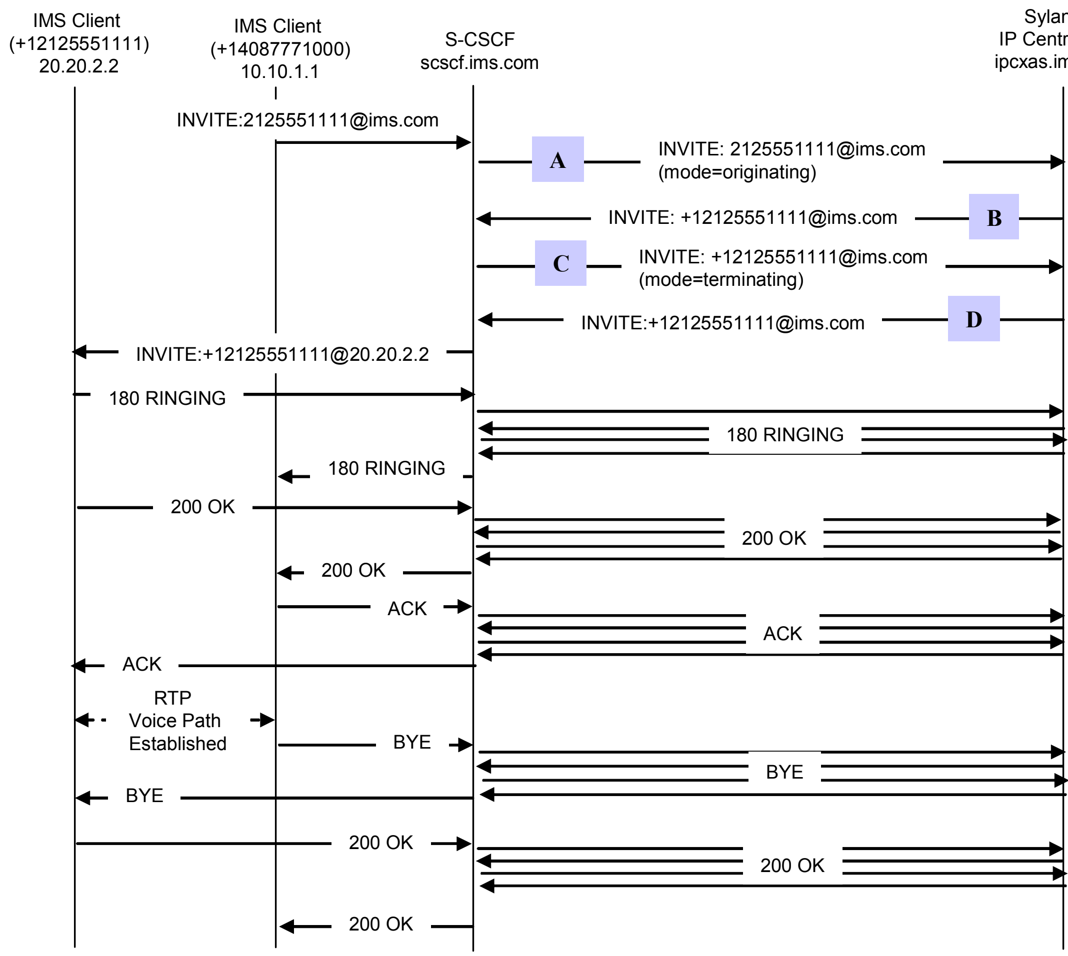
```
INVITE sip:+14085552000@ims.com SIP/2.0

From: "John, Doe" <sip:+14085551000@ims.com>;tag=c8a75184+cadb573e

To: +14085552000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Asserted-Identity: "John, Doe" <tel:+14085551000>
P-Asserted-Identity: "John, Doe" <sip:+14085551000@ims.com>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+14085551000@100.10.1.1:5065;transport=tcp>
Content-Length: …….
```

## D. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085552000@ims.com SIP/2.0
From: "JD, Co-Worker" <sip:+14085551000@ims.com>;tag=c8a75184+cadb573e
To: +14085552000 <sip:+14085552000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
P-Asserted-Identity: " JD, Co-Worker" <tel:+14085551000>
P-Asserted-Identity: " JD, Co-Worker" <sip:+14085551000@ims.com>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+14085551000@100.10.1.1:5065;transport=tcp>
Content-Length: …….
```

## 4.4 IP Centrex-to-IP Centrex Call (between two different tenants)

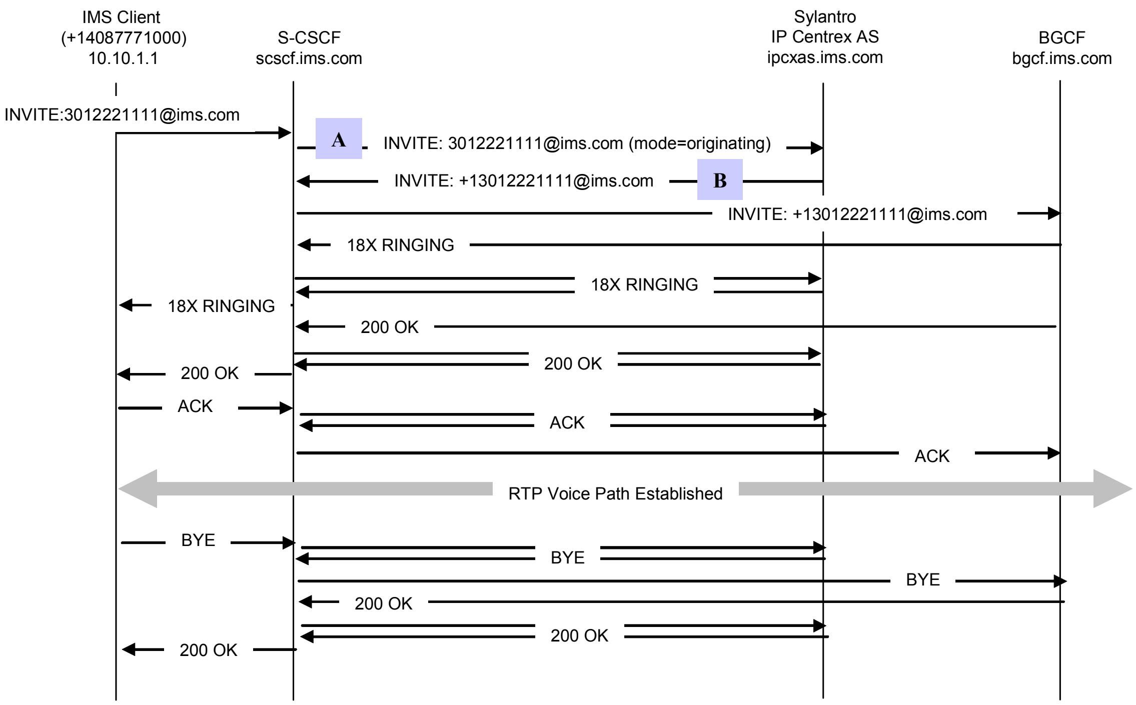
The following call flow diagram illustrates the case when an enterprise user places a call to a user in a different enterprise. This call flow also illustrates the case where the originating and terminating subscribers are hosted on the same Application Server. AS invokes originating user features on the "originating" trigger and terminating user features on "terminating" trigger.




The sample messages A-D in this scenario are similar to ones in section 4.3.

## 4.5 IP Centrex to Non-IP Centrex Call Types

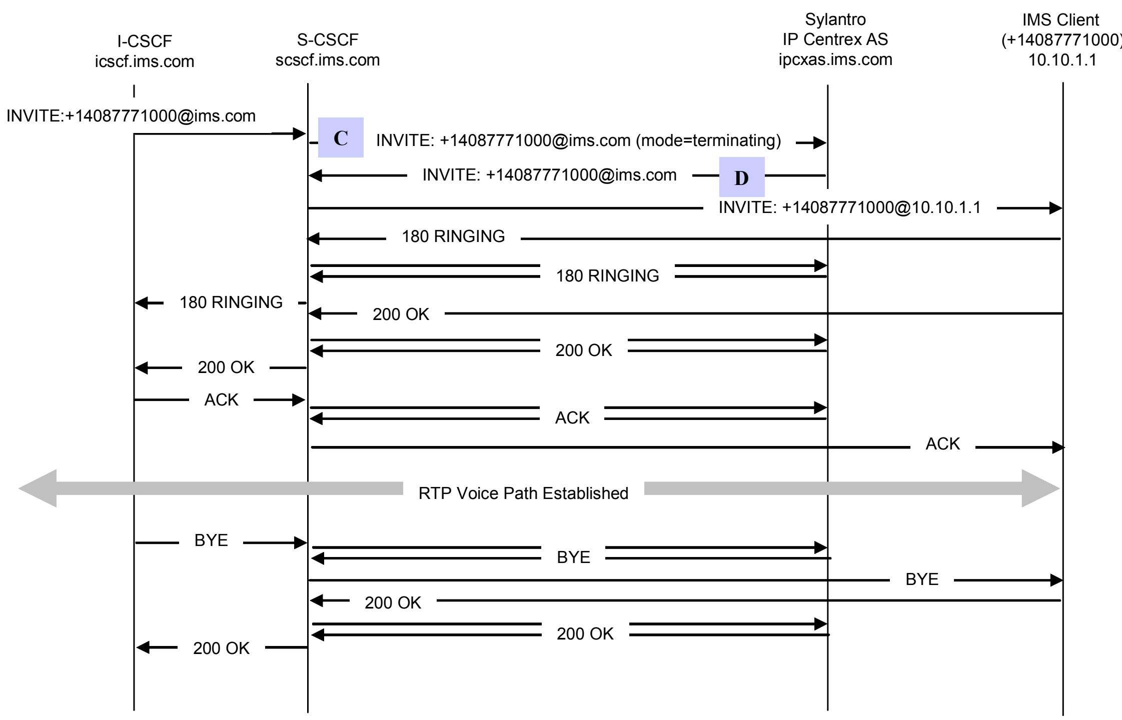
The following call flow illustrates the case when an enterprise user places calls to a non-IP Centrex user. For example, the destination can be another IMS user or a PSTN user. This call flow illustrates the case where only the originating subscriber is hosted on the Application Server. AS invokes originating user features on the "originating" trigger and routes the call to S-CSCF for subsequent processing and routing.




The sample messages A-B in this scenario are similar to ones in section 4.3.

## 4.6 Calls to IP Centrex user from Non-IP Centrex Originations

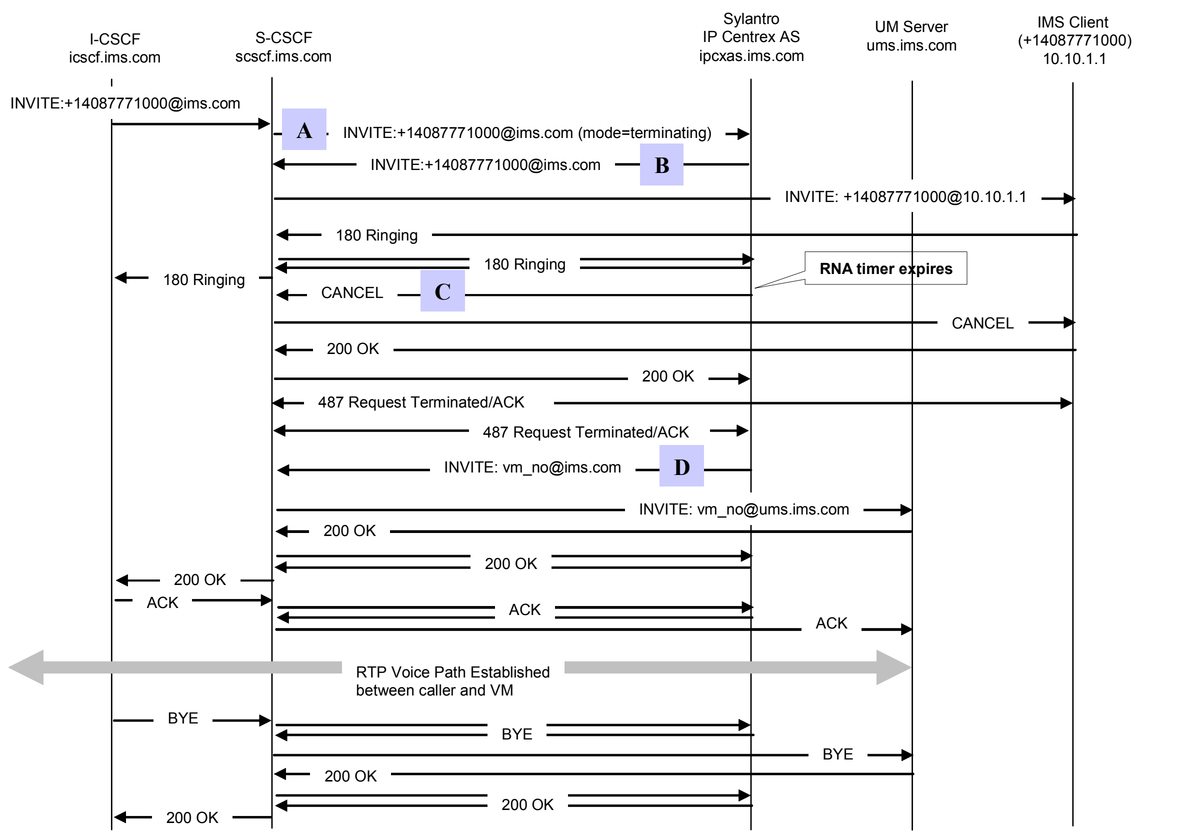
The following call flow illustrates the case where the called party is an IP Centrex user while the caller is not an IP Centrex user. For example, the caller may be an IMS user or a PSTN user. This call flow illustrates the case where only the terminating subscriber is hosted on the Application Server. AS invokes terminating user features on the "terminating" trigger and routes the call to S-CSCF for subsequent processing and routing.




The sample messages C-D in this scenario are similar to ones in section 4.3.

## 4.7 Non-IP Centrex-to-IP Centrex: Forwarding on Ring/No Answer to Voice Mail

In the following call flow, the terminating party, which is an IP Centrex user, does not answer the call. The call will therefore be forwarded to voice mail by the AS. This call flow example illustrates the AS processing for basic call forwarding scenarios.




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
```

```
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP icscf.ims.com
Contact: <sip:+12122341111@10.20.2.3:5069;transport=tcp>
Content-Length: ………
```

### B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Content-Length: ………
```

### C. Sylantro AFS → S-CSCF (Cancelling previous INVITE based on RNA timer expiry):

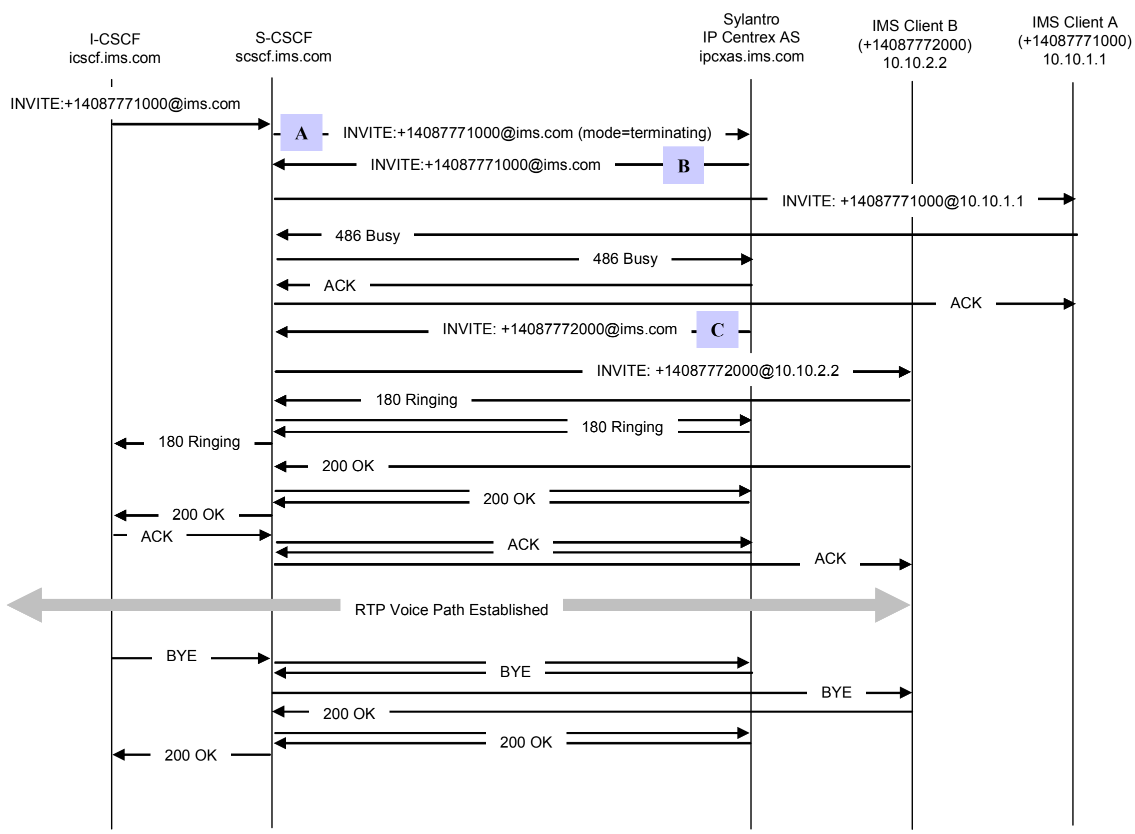
```
CANCEL sip:+14085551000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 CANCEL
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
P-Charging-Vector: icid-value=003400300a141e15
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Max-Forwards: 70
Supported: timer
Content-Length: 0
```

### D. Sylantro AFS → S-CSCF (Outgoing INVITE towards the Voice Mail Pilot number based on RNA timer expiry):

```
INVITE sip:+14085559000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-888112333@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Diversion: <sip:+14085551000@ims.com>;reason=no-answer;counter=1
Content-Length: .......
```

## 4.8 Non-IP Centrex-to-IP Centrex: Forwarding on Busy

In the following call flow, the terminating party (User A), is busy. Based on Call Forwarding setting of User A, call is forwarded to User B upon “Busy” response from User A client.




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
```

```
Via: SIP/2.0/TCP icscf.ims.com
Contact: <sip:+12122341111@10.20.2.3:5069;transport=tcp>
Content-Length: ………
```

### B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

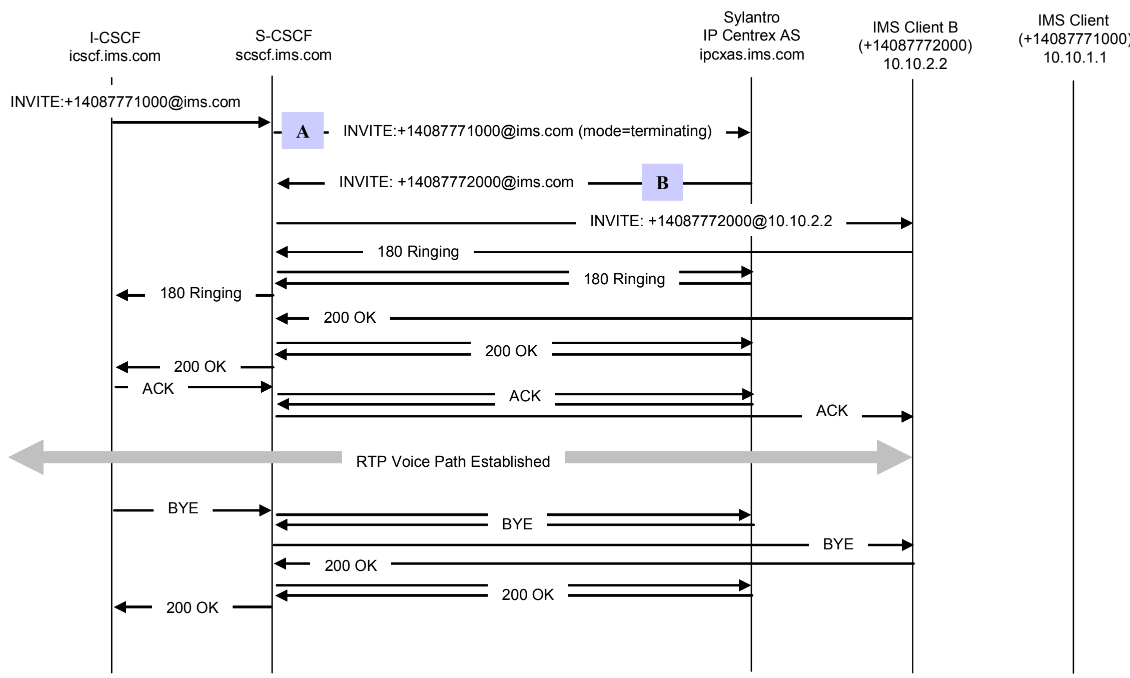
```
INVITE sip:+14085551000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Content-Length: ………
```

### C. Sylantro AFS → S-CSCF (Outgoing INVITE to Forwarded Destination based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085552000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Diversion: <sip:+14085551000@ims.com>;reason=user-busy;counter=1
Content-Length: ………
```

## 4.9 Non-IP Centrex-to-IP Centrex: Forwarding All (Unconditional)

In the following call flow, based on Call Forwarding setting of User A, all incoming calls to User A are forwarded to User B.




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0

From: <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e

To: +14085551000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP icscf.ims.com
Contact: <sip:+12122341111@10.20.2.3:5069;transport=tcp>
Content-Length: .........
```

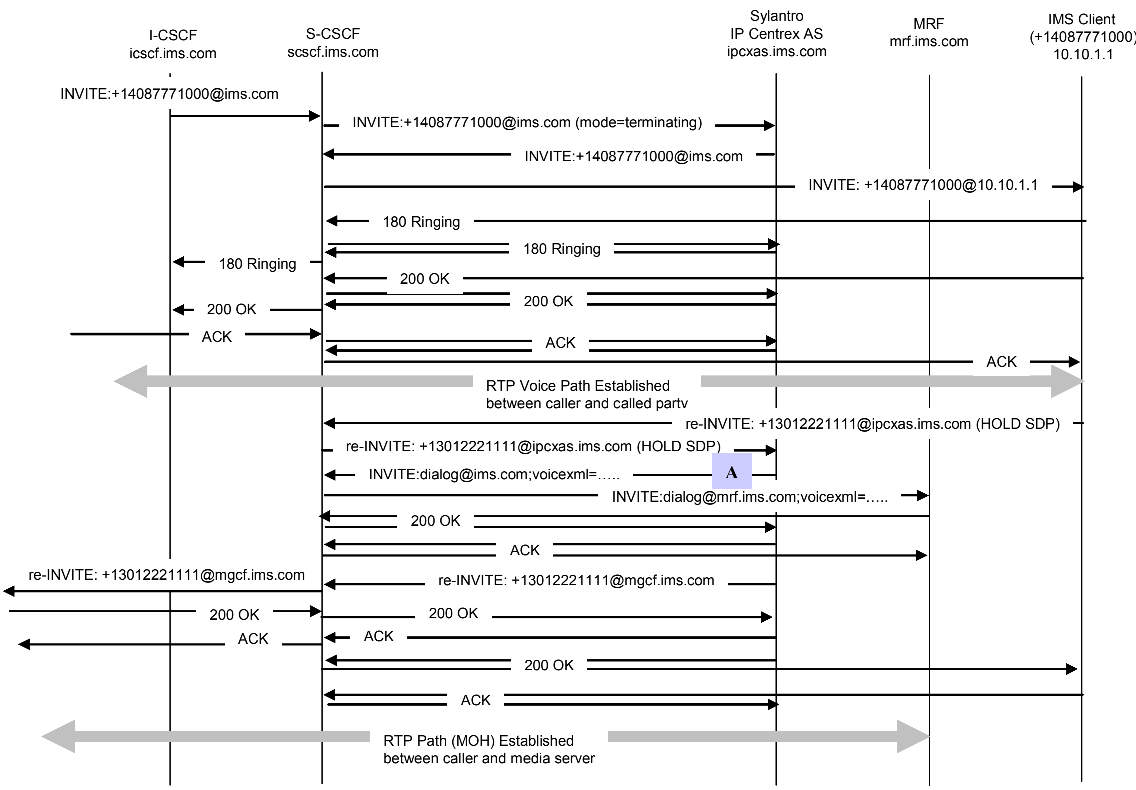
### B. Sylantro AFS → S-CSCF (Outgoing INVITE to Forwarded Destination based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085552000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
```

```
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Diversion: <sip:+14085551000@ims.com>;reason=do-not-disturb;counter=1
Content-Length: ………
```

## 4.10 Music on Hold

In the following call flow, the IP Centrex subscriber places an established call on hold. The held party will be connected to MRF for MOH. In this example, the AS acts as an Initiating B2BUA when sends a request to invoke the services of a media server. The Request URI parameters in this request identify the http/vxml parameters for AS/media server interactions.

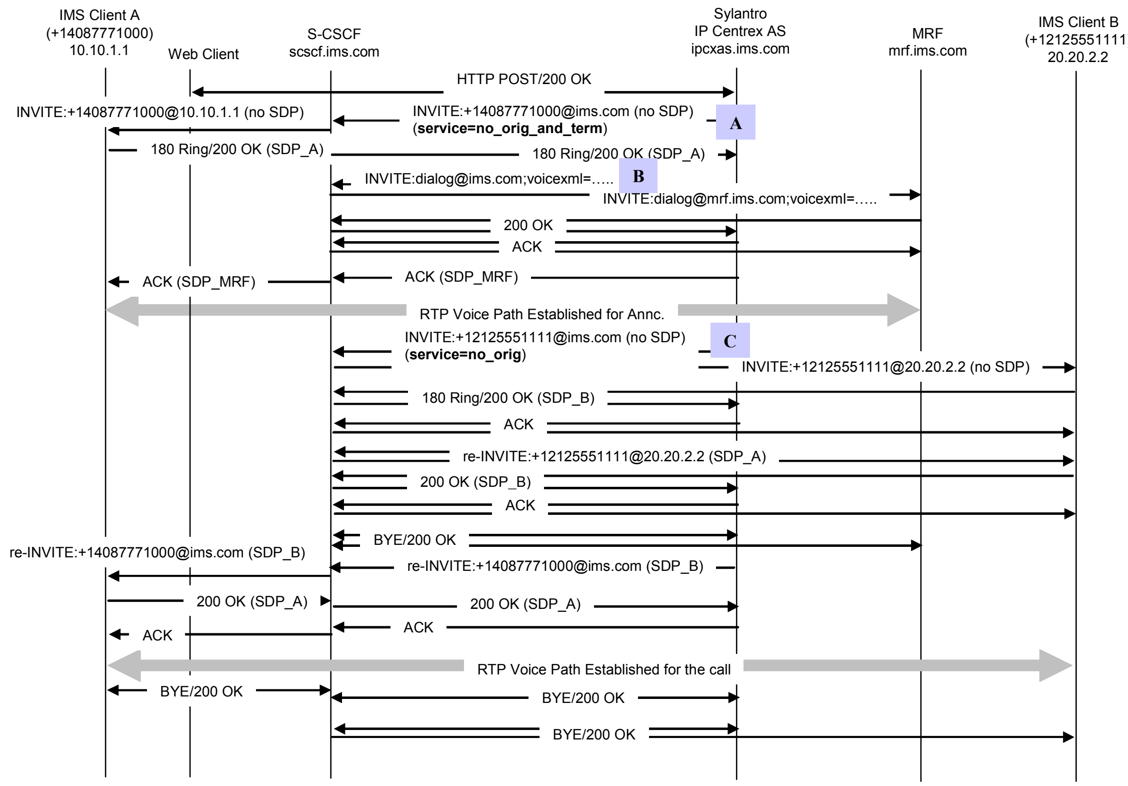



### A. Sylantro AFS → S-CSCF (Outgoing INVITE for MOH invocation):

```
INVITE sip:dialog@ims.com;
transport=tcp;voicexml=http://ipcxas.ims.com/proxy/servlet%3Fsubject%3Dmoh
&action%3Dplay&type%3Dsystem&applicationid%3Dmoh&language%3Den-
us&file%3Dmoh.pcm&mediaformat%3D1 SIP/2.0
From: <sip:controlserver@ims.com>;tag=731686176579224
To: <sip:dialog@ims.com>
Call-ID: 22-1462786503@100.10.1.1
CSeq: 1 INVITE
Max-Forwards: 70
Content-Type: application/sdp
Route: <sip: s-cscf.ims.com;transport=tcp;service=no_orig;lr>
Via: SIP/2.0/TCP 100.10.1.1:5075;branch=z9hG4bK999212340584877
Contact: <sip:dialog@100.10.1.1:5075;transport=tcp>
Content-Length: …….
```

## 4.11 Click To Call

The following call flow illustrates the case where the IP Centrex user invokes the click-to-call capability from the web portal and originates a call between her IMS client and a Non-IP Centrex IMS user. In this example, the AS acts as an Initiating B2BUA sending requests toward the originating subscriber, destination, and media server.




### A. Sylantro AFS → S-CSCF (Outgoing INVITE towards subscriber that initiated C2C):

```
INVITE sip:+14087771000@ims.com SIP/2.0
From: "John, Jones" <sip:+14087771000@ims.com>;tag=ea772646+ead127ae
To: +14087771000 <sip:+14087771000@ims.com>
Call-ID: 5-810954715@100.10.1.1
CSeq: 1 INVITE
Route: <sip:isc@scscf.ims.com:5067;transport=tcp;
service=no_orig_and_term;lr>
P-Asserted-Identity: "John, Jones" <tel:+14087771000>
P-Asserted-Identity: < sip:+14087771000@ims.com>
P-Charging-Vector: icid-value=eab7527a@100.10.1.1;icid-generated-
at=100.10.1.1;orig-ioi=ims.com
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK1275654407647276
Contact: <sip:+14087771000@100.10.1.1:5065;transport=tcp>
Content-Length: 0
```

(No SDP)

### B. Sylantro AFS → S-CSCF (Outgoing INVITE towards the media server):

```
INVITE sip:dialog@ims.com;
transport=tcp;voicexml=http://ipcxas.ims.com/proxy/servlet%3Fsubject%3DCOM
RIO&session%3D-1&to%3D4087771000 SIP/2.0
From: “John, Jones” <sip:+14087771000@ims.com>;tag=28ed005e+29deb296
To: <sip:dialog@ims.com>
Call-ID: 2-1504770107@100.10.1.1
CSeq: 1 INVITE
Max-Forwards: 70
Content-Type: application/sdp
Route: <sip:scscf.ims.com;transport=tcp;service=no_orig;lr>
P-Charging-Vector: icid-value=29bd8b20@100.10.1.1;icid-generated-
at=100.10.1.1;orig-ioi=ims.com
Via: SIP/2.0/TCP 100.10.1.1:5075;branch=z9hG4bK1510858799379376
Contact: <sip:dialog@100.10.1.1:5075;transport=tcp>
Content-Length: …..
```

### C. Sylantro AFS → S-CSCF (Outgoing INVITE towards the destination subscriber):

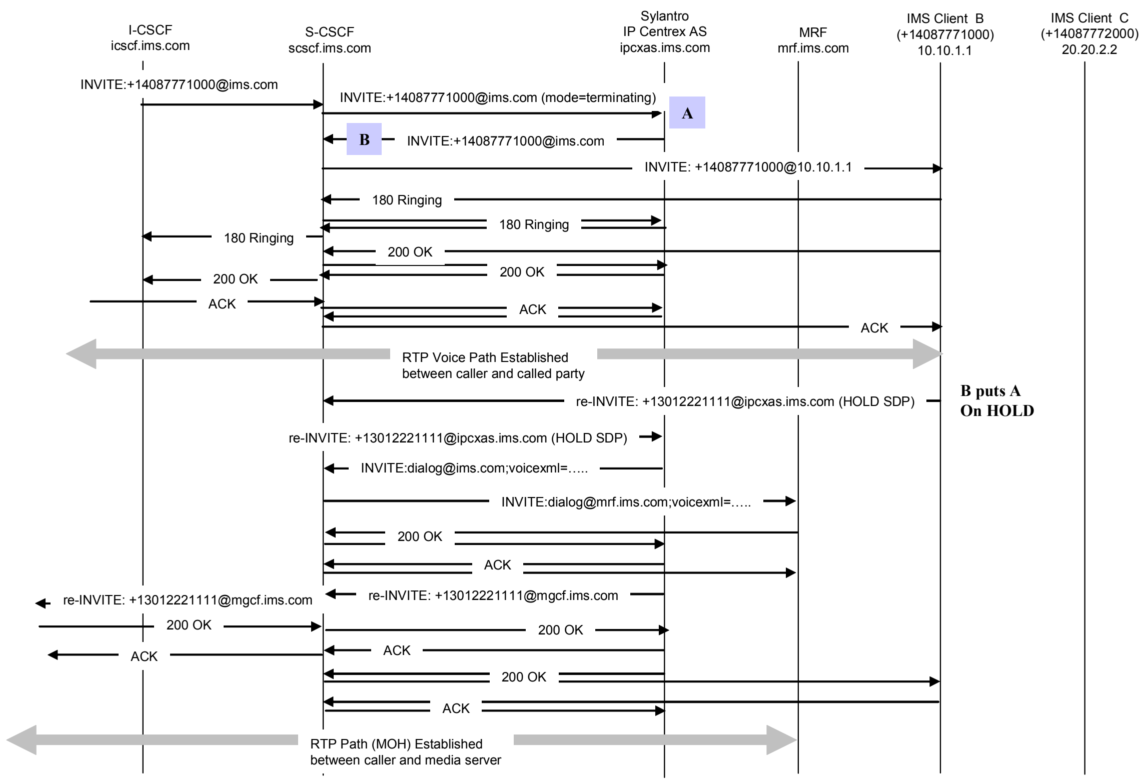
```
INVITE sip:+12125551111@ims.com SIP/2.0
From: "John, Jones" <sip:+14087771000@ims.com>;tag=f60d8d1c+f8463dd6
To: <sip:+12125551111@ims.com>
Call-ID: 4-388804331@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route: <sip:isc@scscf.ims.com:5077;transport=tcp;service=no_orig;lr>
P-Asserted-Identity: <tel:+14087771000>
P-Asserted-Identity: <sip:+14087771000@ims.com>
P-Charging-Vector: icid-value=f80a34b2@100.10.1.1;icid-generated-
at=100.10.1.1;orig-ioi=ims.com
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK1043365641853888
Contact: <sip:+14087771000@100.10.1.1:5065;transport=tcp>
Content-Length: ……….
```

## 4.12 Call Transfer

Sylantro AFS supports call transfer based on the REFER method outlined in RFC 3515. Two variations of call transfer feature are illustrated here:

1. Blind Transfer: The user invokes the transfer and is out of the scenario while the caller is on music on hold until the transfer target answers.
2. Consultative Transfer: The user invokes transfer, consults with transfer target after answer, and completes the transfer after consultation. The caller is on music on hold while transfer is taking place.

The initial call set up and placing the call on hold is common to both Blind and Consultative scenarios:




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122345555@ims.com>;tag=1234abcd
To: <sip:+14085551000@ims.com>
Call-ID: 8-111905773@200.20.2.2
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
```

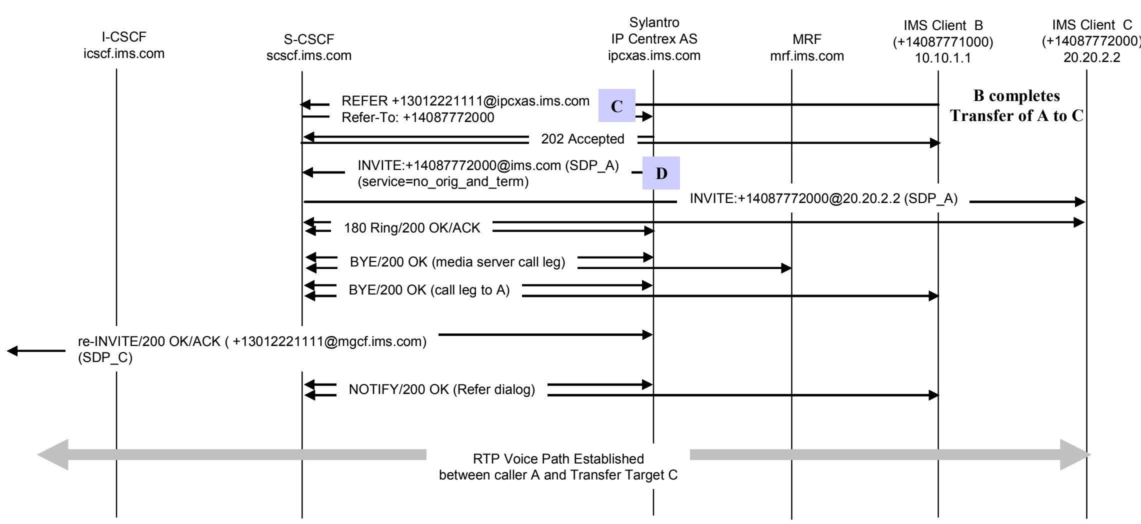
```
P-Asserted-Identity: "John, Doe" <tel:+12122345555>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 200.20.2.2:7777;branch=z9hG4bK821533665715080
Contact: <sip: +12122345555@200.20.2.2:5060;transport=tcp>
Content-Length: …….
```

### B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122345555@ims.com>;tag=5678efjh
To: <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
P-Asserted-Identity: <tel:+14085551000>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122345555@100.10.1.1:5065;transport=tcp>
Content-Length: …….
```

## 4.12.1 Blind Transfer

For Blind Transfer to User C, User B client sends an "in-dialog" REFER directed at calling party. AS terminates the REFER and sends a new INVITE towards User C. After User C answers the call, AS manages the media negotiation through a sequence of re-INVITEs towards calling party and Transfer target (User C):



### C. S-CSCF → Sylantro AFS (REFER to Blind Transfer A to C):

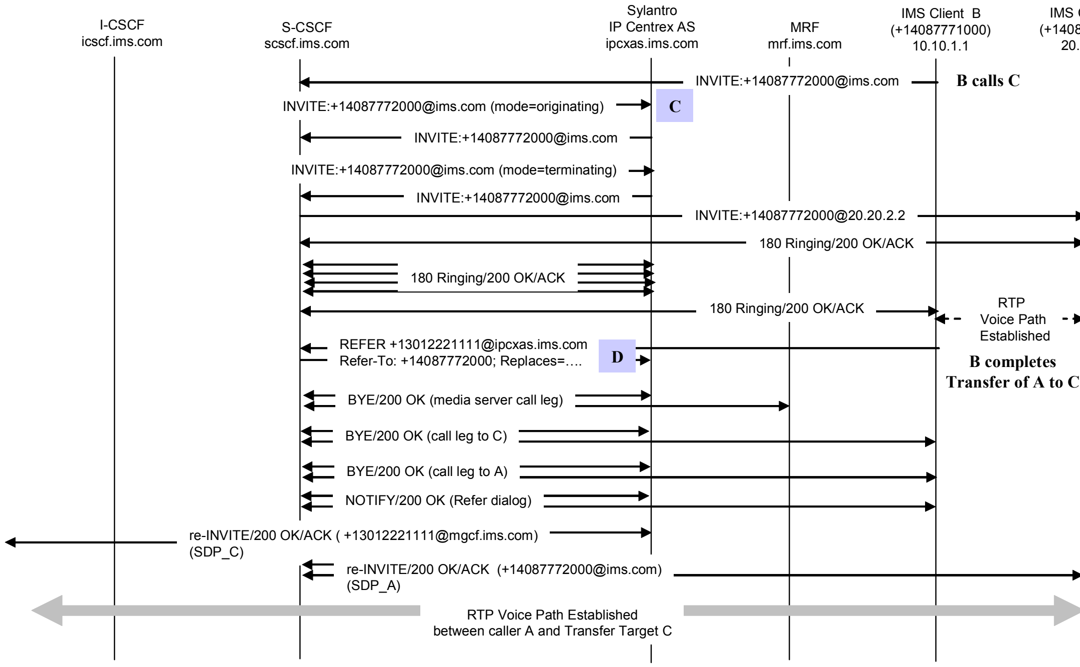
```
REFER sip:+12122345555@100.10.1.1:5065;transport=tcp SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP pcscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: <sip:+14085551000@ims.com;user=phone>;tag=91011ijkl
To: <sip:+15105551001@38.187.114.246:5060>;tag=5678efjh
Call-ID: 9-222905885@100.10.1.1
Cseq: 3 REFER
Contact: <sip:+14085551000@10.10.1.1:5060;transport=tcp>
Refer-To: sip: 4085552000@ims.com
Referred-By: sip:+14085551000@ims.com
Max-Forwards: 70
Content-Length: 0
```

### D. Sylantro AFS → S-CSCF (Outgoing INVITE towards User C):

```
INVITE sip:+14085552000@ims.com SIP/2.0
From: <sip:+14085551000@ims.com>;tag=121314mnop
To: <sip:+14085552000@ims.com>
Call-ID: 10-333906776@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route: <sip:isc@scscf.ims.com:5067;transport=tcp;
service=no_orig_and_term;lr>
P-Asserted-Identity: <tel:+14085551000>
P-Charging-Vector: icid-value=00078964300b156e5
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+14085551000@100.10.1.1:5065;transport=tcp>
Content-Length: …….
```

## 4.12.2 Consultative Transfer

For Consultative Transfer to User C, User B client calls User C. After consultation with User C, User B client sends an “in-dialog” REFER with the “Replaces” header directed at calling party. AS terminates the REFER and manages the media negotiation through a sequence of re-INVITEs towards calling party and Transfer target (User C):




## C. S-CSCF → Sylantro AFS (Originating trigger):

```
INVITE sip:2000@ims.com;user=phone SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP pcscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: UserB <sip:+14085551000@ims.com;user=phone>;tag=151617qpst
To: UserC <sip:2000@ims.com;user=phone>
Call-ID: 1-1520@10.10.1.1
Cseq: 1 INVITE
Route:<sip:ipcxas.ims.com;mode=originating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
Record-Route: <sip:pcscf.ims.com;lr>
Contact: <sip:+14085551000@10.10.1.1;user=phone>
P-Charging-Vector: icid-value=003400300a141e15
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: …..
```

## D. S-CSCF → Sylantro AFS (REFER with Replaces header to Transfer A to C):

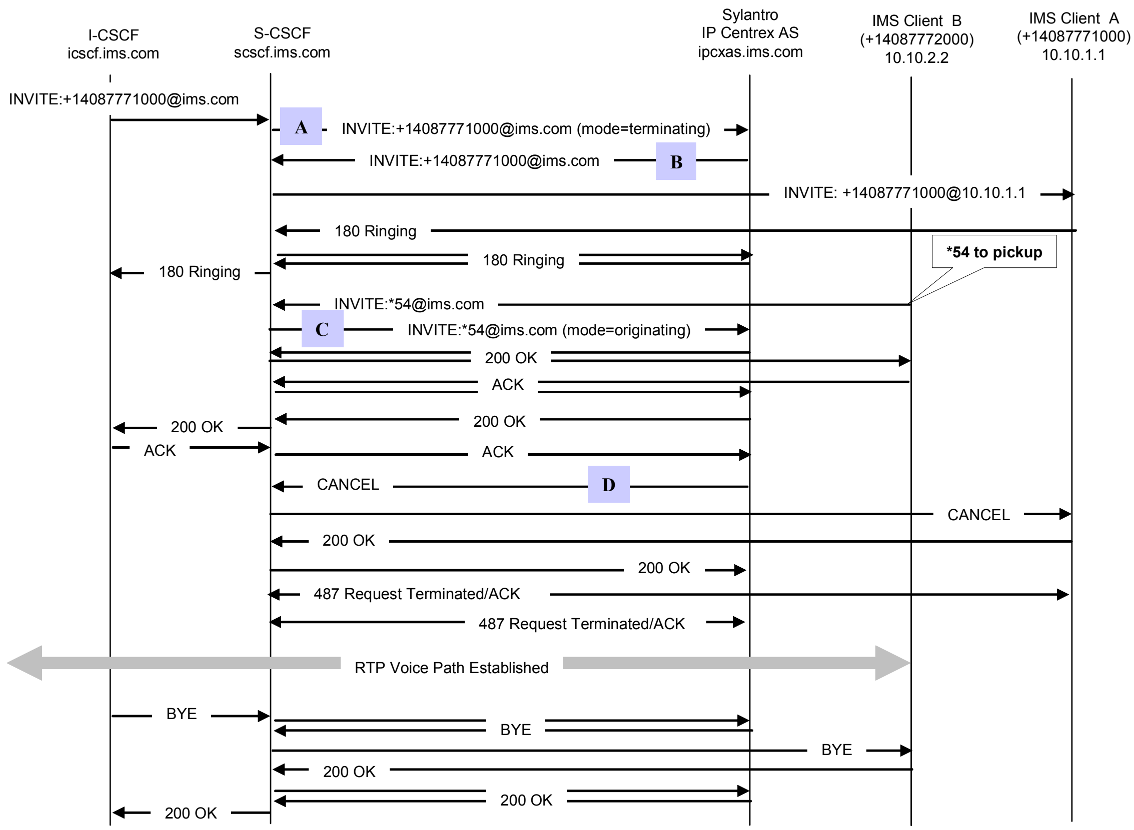
```
REFER sip:+12122345555@100.10.1.1:5065;transport=tcp SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP pcscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: <sip:+14085551000@ims.com;user=phone>;tag=91011ijkl
To: <sip:+15105551001@38.187.114.246:5060>;tag=5678efjh
Call-ID: 9-222905885@100.10.1.1
Cseq: 3 REFER
Contact: <sip:+14085551000@10.10.1.1:5060;transport=tcp>
```

```
Refer-To: <sip:4085552000@ims.com;user=phone?Replaces=1-
1520%4010.10.1.1%3Bto-tag%3D181920uvxy%3Bfrom-tag%3D151617qpst>
Referred-By: sip:+14085551000@ims.com
Max-Forwards: 70
Content-Length: 0
```

The Replaces header identifies the SIP dialog to be replaced (in this example the dialog established between User B and AFS with INVITE C in the above figure).

## 4.13 Group Call Pickup

This feature allows users within a call group (as defined by a Tenant or Group administrator) to pick up a ringing call at a group member destination. In this example, Users A and B are in the same call group. User B dials GCP feature code (*54) to pick up a ringing call at User A client.




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085552000@ims.com>
Call-ID: 8-111905773@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
```

```
Via: SIP/2.0/TCP icscf.ims.com
Contact: <sip:+12122341111@10.20.2.3:5069;transport=tcp>
Content-Length: ………
```

### B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122341111@100.10.1.1:5065;transport=tcp>
Content-Length: ………
```

### C. S-CSCF → Sylantro AFS (Originating trigger for INVITE to pickup a group call):

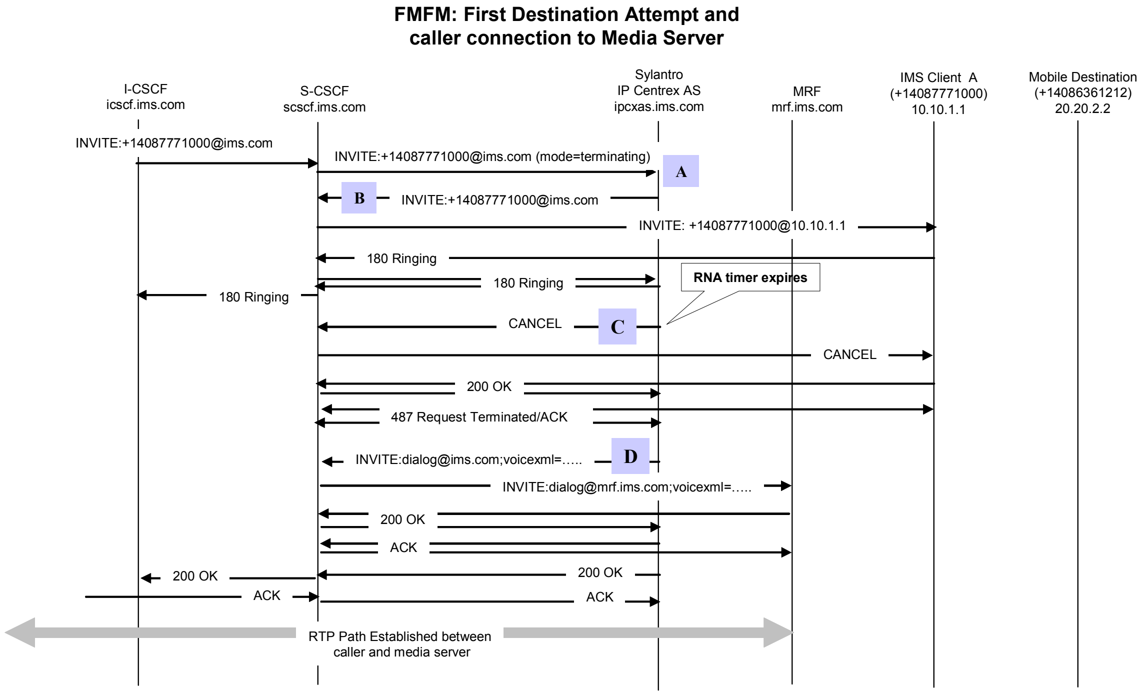
```
INVITE sip:*54@ims.com SIP/2.0

From: <sip:+14085552000@ims.com>;tag=c8a75184+cadb573e

To: <sip:*54@ims.com>
Call-ID: 5-12345678@10.10.2.2
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=originating;lr>
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Charging-Vector: icid-value=005600300a141e29
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP icscf.ims.com
Contact: <sip:+14085552000@10.10.2.2:5060;transport=tcp>
Content-Length: ………
```

### D. Sylantro AFS → S-CSCF (Cancelling INVITE in Step A):

```
CANCEL sip:+14085551000@ims.com SIP/2.0
From: "PSTN, Friend" <sip:+12122341111@ims.com>;tag=c8a75184+cadb573e
To: +14085551000 <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 CANCEL
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
P-Charging-Vector: icid-value=003400300a141e15
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Max-Forwards: 70
Supported: timer
Content-Length: 0
```

## 4.14 Find Me/Follow Me: Sequential Ringing

With this feature, the AFS will attempt to route an incoming call to a subscriber to multiple destinations, based on the call treatments configured by the subscriber. In this example, an incoming call is first routed to subscriber's public number. After ring/no answer timeout, the caller is connected to media server for announcement, while the AS attempts the 2nd destination (e.g., a mobile public #) set up by the subscriber. Once a destination answers the call, the AS connects the caller to the answered destination by initiating re-INVITEs towards the calling and called parties.




### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0

From: <sip:+12122347777@ims.com>;tag=1234abcd

To: <sip:+14085551000@ims.com>
Call-ID: 8-111905773@200.20.2.2
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Asserted-Identity: "John, Doe" <tel:+12122347777>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 200.20.2.2:7777;branch=z9hG4bK821533665715080
Contact: <sip: +12122347777@200.20.2.2:5060;transport=tcp>
Content-Length: …….
```

## B. Sylantro AFS → S-CSCF (Outgoing INVITE based on Incoming INVITE with Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=5678efjh
To: <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122347777@100.10.1.1:5065;transport=tcp>
Diversion: <sip:+14085551000@ims.com;user=phone>;reason=unknown;counter=1
Content-Length: …….
```

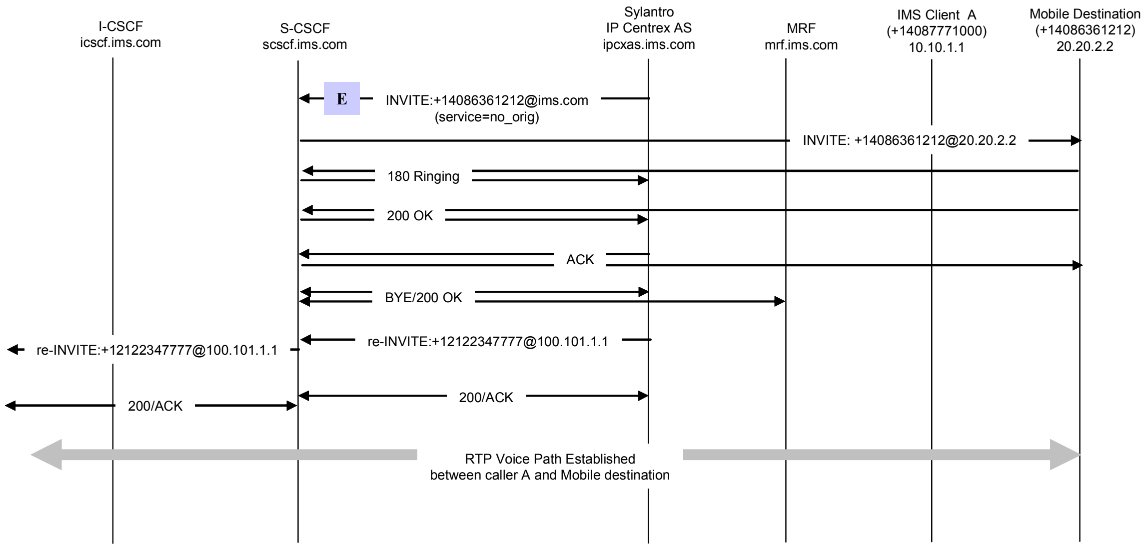
## C. Sylantro AFS → S-CSCF (Cancelling previous INVITE based on RNA timer expiry):

```
CANCEL sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=5678efjh
To: <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 CANCEL
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
P-Charging-Vector: icid-value=003400300a141e15
Route:<sip:ISC_TOKEN@10.20.30.40:5067;transport=tcp;lr>
Max-Forwards: 70
Supported: timer
Content-Length: 0
```

## D. Sylantro AFS → S-CSCF (INVITE towards Media Server):

```
INVITE sip:dialog@ims.com;
voicexml=http://ipcxas.ims.com/proxy/servlet%3Fsubject%3DFMFM&session%3D-
1&to%3D5105551002&callid%3D9f7ecc38-1dd1-11b2-82b3-
b03162323164:ae393a42&tenantID%3Dimsrocks&subscriberID%3D1000 SIP/2.0
From: <sip:+12122347777@ims.com>;tag=8d87f8fe
To: <sip:5105551000@38.187.114.243>
Call-ID: 21-1952312128@100.10.1.1
CSeq: 1 INVITE
Max-Forwards: 70
Content-Type: application/sdp
Route: <sip:scscf.ims.com;transport=tcp;service=no_orig;lr>
P-Charging-Vector: icid-value=003400300a141e15
Via: SIP/2.0/TCP 100.10.1.1:5075;branch=z9hG4bK1943499534989522
Contact: <sip:+12122347777@100.10.1.1:5075;transport=tcp>
Content-Length: ......
```

**FMFM: Second Destination Attempt and caller re-connection to Answered Destination**




## E. Sylantro AFS → S-CSCF (INVITE towards 2nd Destination in FMFM treatment):

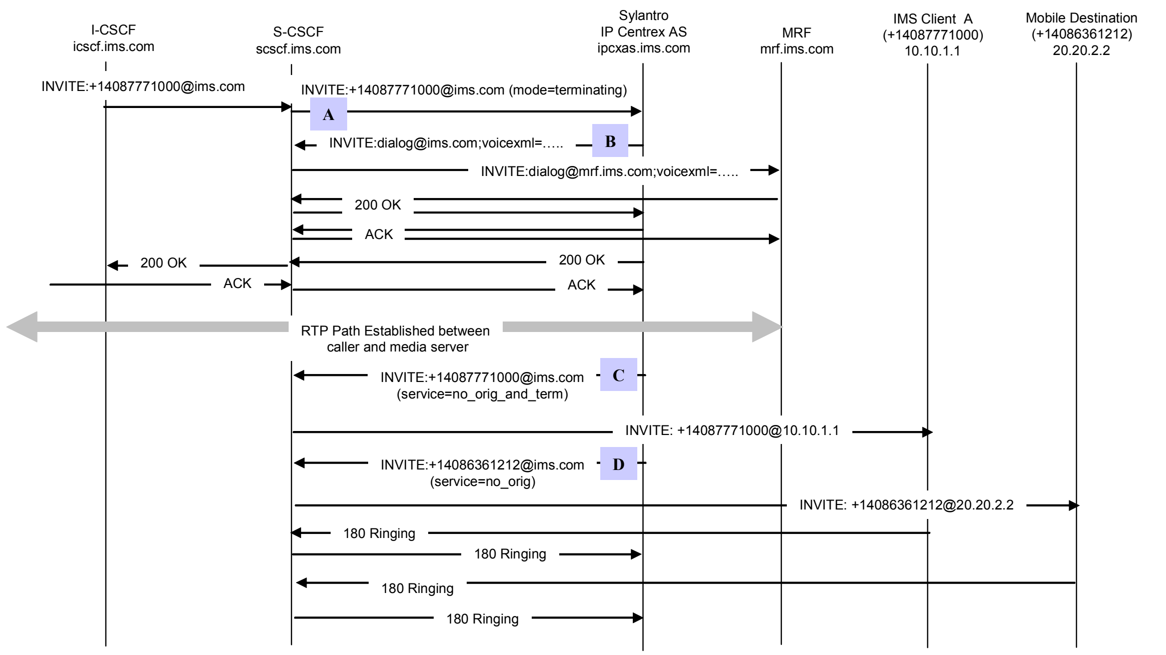
```
INVITE sip:+14086361212@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=97ec7414
To: <sip:+14086361212@ims.sylantro.com>
Call-ID: 39-410825972@100.10.1.1
CSeq: 1 INVITE
Route: <sip:scscf.ims.com;transport=tcp;service=no_orig;lr>
P-Asserted-Identity: "+12122347777" <tel:+12122347777>
P-Asserted-Identity: "+12122347777" <sip:+14082347777@ims.com>
Diversion: <sip:5105551000@ims.com;user=phone>;reason=no-answer;counter=3
P-Charging-Vector: icid-value=97c4c00e@100.10.1.1;icid-generated-
at=100.10.1.1;orig-ioi=ims.com
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK567641482721730
Contact: <sip:+12122347777@100.10.1.1:5065;transport=tcp>
Content-Length: 0
```

(NO SDP)

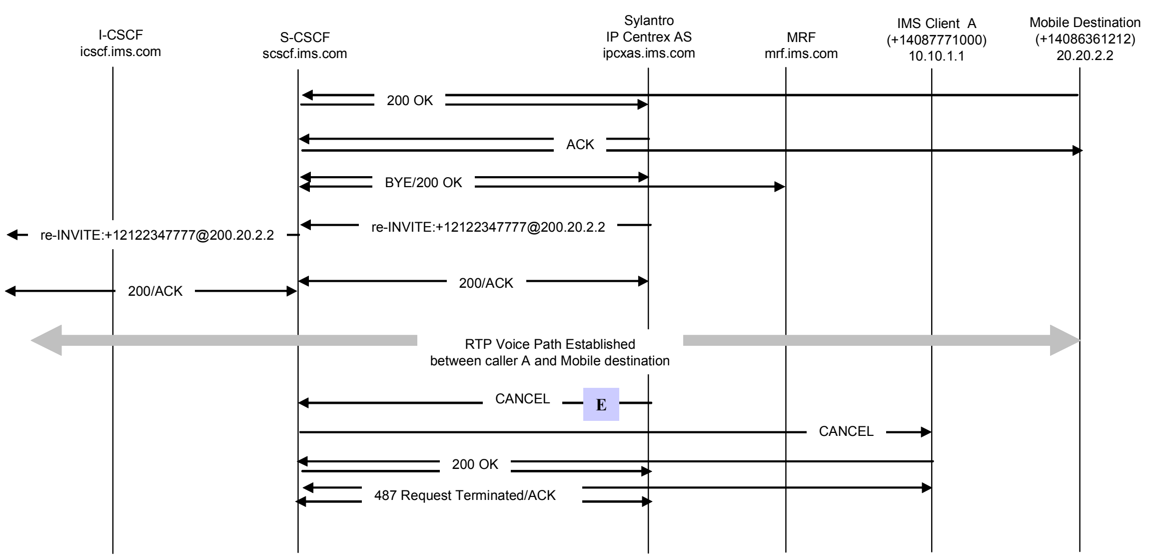
## 4.15 Find Me/Follow Me: Simultanous Ringing

This feature is a variation of the FMFM feature where the caller is connected to an announcement first, while the AFS will attempt to route an incoming call to a subscriber to multiple destinations simultanously, based on the call treatments configured by the subscriber. Once a destination answers the call, the AS connects the caller to the answered destination by initiating re-INVITEs towards the calling and called parties and cancelling the unanswered call attempts.

**Find Me/Follow Me: SimRing – routing to two destinations simultanously**




**Find Me/Follow Me: SimRing – Mobile destination answers**



### A. S-CSCF → Sylantro AFS (Terminating trigger):

```
INVITE sip:+14085551000@ims.com SIP/2.0

From: <sip:+12122347777@ims.com>;tag=1234abcd

To: <sip:+14085551000@ims.com>
Call-ID: 8-111905773@200.20.2.2
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip:ipcxas.ims.com;mode=terminating;lr>
Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>
Record-Route: <sip:scscf.ims.com;lr>
P-Asserted-Identity: "John, Doe" <tel:+12122347777>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 200.20.2.2:7777;branch=z9hG4bK821533665715080
Contact: <sip: +12122347777@200.20.2.2:5060;transport=tcp>
Content-Length: …….
```

### B. Sylantro AFS → S-CSCF (INVITE towards Media Server):

```
INVITE sip:dialog@ims.com;
voicexml=http://ipcxas.ims.com/proxy/servlet%3Fsubject%3DFMFM&session%3D-
1&to%3D5105551002&callid%3D9f7ecc38-1dd1-11b2-82b3-
b03162323164:ae393a42&tenantID%3Dimsrocks&subscriberID%3D1000 SIP/2.0
From: <sip:+12122347777@ims.com>;tag=8d87f8fe
To: <sip:5105551000@38.187.114.243>
Call-ID: 21-1952312128@100.10.1.1
CSeq: 1 INVITE
Max-Forwards: 70
Content-Type: application/sdp
Route: <sip:scscf.ims.com;transport=tcp;service=no_orig;lr>
P-Charging-Vector: icid-value=003400300a141e15
Via: SIP/2.0/TCP 100.10.1.1:5075;branch=z9hG4bK1943499534989522
Contact: <sip:+12122347777@100.10.1.1:5075;transport=tcp>
Content-Length: ......
```

### C. Sylantro AFS → S-CSCF (INVITE towards first Destination in FMFM treatment):

```
INVITE sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=5678efjh
To: <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 INVITE
Content-Type: application/sdp
Route:<sip: sip:scscf.ims.com;service=no_orig_and_term;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Supported: timer
Min-SE: 1800
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Contact: <sip:+12122347777@100.10.1.1:5065;transport=tcp>
Diversion: <sip:+14085551000@ims.com;user=phone>;reason=no-
answer;counter=1
```

```
Content-Length: 0
```

(NO SDP)

### D. Sylantro AFS → S-CSCF (INVITE towards 2nd Destination in FMFM treatment):

```
INVITE sip:+14086361212@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=97ec7414
To: <sip:+14086361212@ims.sylantro.com>
Call-ID: 39-410825972@100.10.1.1
CSeq: 1 INVITE
Route: <sip:scscf.ims.com;transport=tcp;service=no_orig;lr>
P-Asserted-Identity: "+12122347777" <tel:+12122347777>
P-Asserted-Identity: "+12122347777" <sip:+14082347777@ims.com>
Diversion: <sip:5105551000@ims.com;user=phone>;reason=no-answer;counter=3
P-Charging-Vector: icid-value=97c4c00e@100.10.1.1;icid-generated-
at=100.10.1.1;orig-ioi=ims.com
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK567641482721730
Contact: <sip:+12122347777@100.10.1.1:5065;transport=tcp>
Content-Length: 0
```

(NO SDP)

### E. Sylantro AFS → S-CSCF (Cancelling INVITE to first destination):

```
CANCEL sip:+14085551000@ims.com SIP/2.0
From: <sip:+12122347777@ims.com>;tag=5678efjh
To: <sip:+14085551000@ims.com>
Call-ID: 9-222905885@100.10.1.1
CSeq: 1 CANCEL
Route:<sip: sip:scscf.ims.com;service=no_orig_and_term;transport=tcp;lr>
P-Charging-Vector: icid-value=003400300a141e15
Via: SIP/2.0/TCP 100.10.1.1:5065;branch=z9hG4bK821533665715080
Max-Forwards: 70
Supported: timer
Content-Length: 0
```

## 4.16 SUBSCRIBE/NOTIFY Flows

***Note: This section is provided for informational purposes only and does not imply the below features are currently functional in an IMS testing environment.***

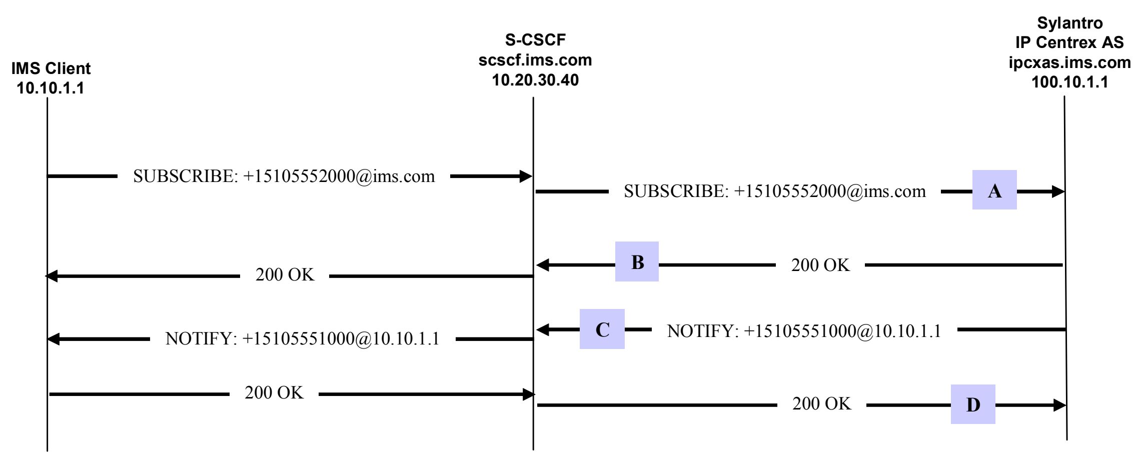
Sylantro AFS supports several advanced features based on the SIP Event Notification framework (RFC 3265). At present, the specific features that make use of this framework include:

- Message Waiting Indication ("message summary" event package)
- Group Call Pickup ("dialog" event package)
- Directed Call Pickup ("dialog" event package)
- ACD Agent Checkin/Checkout ("presence" event package)
- SIP Bridged Line Appearance/Shared Call Appearance ("dialog" event package)
- Call Forwarding Indication Notification ("missed-call-summary" event package)

All the above features require the SIP user agent client to support the feature call flows and related RFCs and internet drafts. Support for these features in IMS architecture require the IMS Core elements (CSCFs) to support the same call flows. Additional discussion of feature call flows and verification of the same with IMS Core partners is required before testing the above features.

The following sections illustrate the generic call flows for the subscription by the user agent (AFS acts as Notifier) and subscription by the AFS (user agent acts as Notifier).

### 4.16.1 Subscription by User Agent




### A. S-CSCF → Sylantro AFS (Originating trigger):

```
SUBSCRIBE sip:+15105552000@ims.com SIP/2.0
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: <sip:+15105551000@ims.com>;tag=ABCD
To: <sip:+15105552000@ims.com>
```

```
CSeq: 1 SUBSCRIBE
Call-ID: fdc49228-7721a52-a2d0bbaf@10.10.1.1
```

***Route:<sip:ipcxas.ims.com;mode=originating;lr>***
***Route:<sip:ISC_TOKEN2@10.20.30.40:5067;transport=tcp;lr>***

```
Record-Route: <sip:scscf.ims.com;lr>
Contact: <sip:+15105551000@10.10.1.1;transport=tcp>
Event: [package_name: e.g.,dialog]
User-Agent: XYZ-IPPhone
Accept: application/dialog-info+xml
Max-Forwards: 70
Expires: 3600
Content-Length: 0
```

### B. Sylantro AFS → S-CSCF:

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
Record-Route: <sip:scscf.ims.com;lr>
CSeq: 1 SUBSCRIBE
Call-ID: fdc49228-7721a52-a2d0bbaf@10.10.1.1

From: <sip:+15105551000@ims.com>;tag=ABCD
To: <sip:+15105552000@ims.com>;tag=1234
Allow-Events:  message-summary, missed-call-summary, dialog, presence
Expires: 3700
Contact: <sip:AS_TOKEN@100.10.1.1:5097;transport=tcp>
Content-Length: 0
```

### C. Sylantro AFS → S-CSCF (in-dialog NOTIFY transaction):

```
NOTIFY sip: +15105551000@10.10.1.1;transport=tcp SIP/2.0
From: <sip:+15105551000@ims.com>;tag=1234
To: <sip:+15105552000@ims.com>;tag=ABCD
Call-ID: fdc49228-7721a52-a2d0bbaf@10.10.1.1

CSeq: 2 NOTIFY
```

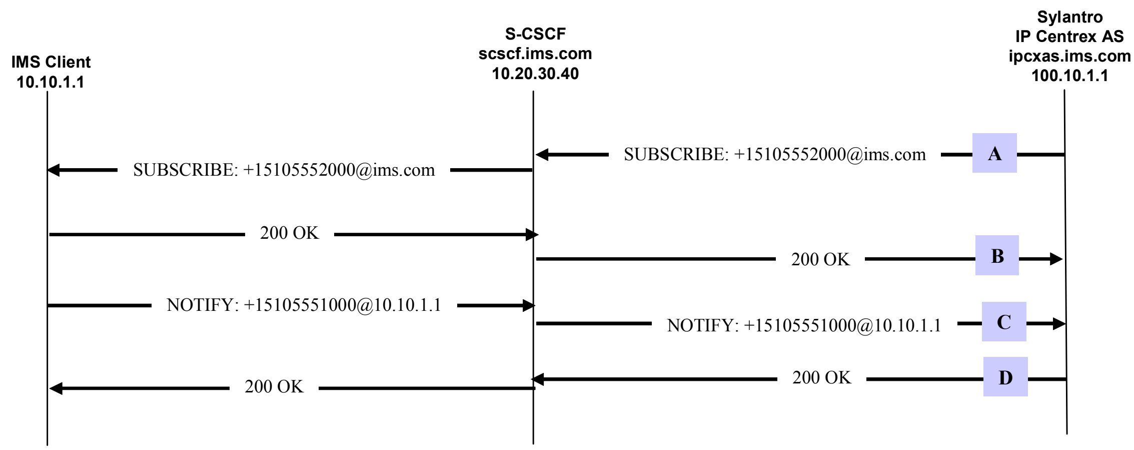
***Route: <sip:scscf.ims.com;lr>***

```
Via: SIP/2.0/TCP 100.10.1.1:5097;branch=z9hG4bK2073488092167246
Max-Forwards: 70
Content-Type: application/dialog-info+xml
Event: [package_name: e.g.,dialog]
Subscription-State: active
Contact: <sip:AS_TOKEN@100.10.1.1:5097;transport=tcp>
Content-Length: .....

(XML payload)
```

### D. S-CSCF → Sylantro AFS:

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 100.10.1.1:5097;branch=z9hG4bK2073488092167246
From: <sip:+15105551000@ims.com>;tag=1234
To: <sip:+15105552000@ims.com>;tag=ABCD
CSeq: 2 NOTIFY
Call-ID: fdc49228-7721a52-a2d0bbaf@10.10.1.1

Contact: <sip:+15105551000@10.10.1.1;transport=tcp>
Event: [package_name: e.g.,dialog]
User-Agent: XYZ-IPPhone
Content-Length: 0
```

## 4.16.2 Subscription by Application Server




### A. Sylantro AFS → S-CSCF: *(the Route header value is populated with the default S-CSCF SIP URI or the SIP URI value received in the Contact header of 3rd party REGISTER)*

```
SUBSCRIBE sip:+15105552000@ims.com SIP/2.0
Via: SIP/2.0/TCP ipcxas.ims.com
From: <sip:+15105551000@ims.com>;tag=ABCD
To: <sip:+15105552000@ims.com>
CSeq: 1 SUBSCRIBE
Call-ID: fdc49228-7721a52-a2d0bbaf@100.10.1.1

Route: <sip:scscf.ims.com;transport=tcp;service=no_orig_and_term;lr>
Contact: <sip:AS_TOKEN@100.10.1.1;transport=tcp>
Event: [package_name: e.g.,dialog]
Accept: application/dialog-info+xml
Max-Forwards: 70
Expires: 3600
Content-Length: 0
```

### B. S-CSCF → Sylantro AFS:

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ipcxas.ims.com
Record-Route: <sip:scscf.ims.com;lr>
```

```
CSeq: 1 SUBSCRIBE
Call-ID: fdc49228-7721a52-a2d0bbaf@100.10.1.1

From: <sip:+15105551000@ims.com>;tag=ABCD
To: <sip:+15105552000@ims.com>;tag=1234
Expires: 3600
Contact: <sip:+15105552000@10.10.1.1:5060;transport=tcp>
Content-Length: 0
```

### C. S-CSCF → Sylantro AFS (in-dialog NOTIFY transaction):

```
NOTIFY sip: AS_TOKEN@100.10.1.1;transport=tcp SIP/2.0

Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: <sip:+15105552000@ims.com>;tag=1234
To: <sip:+15105551000@ims.com>;tag=ABCD
Call-ID: fdc49228-7721a52-a2d0bbaf@100.10.1.1

CSeq: 2 NOTIFY

Max-Forwards: 70
Content-Type: application/dialog-info+xml
Event: [package_name: e.g.,dialog]
Subscription-State: active
Contact: <sip:AS_TOKEN@10.10.1.1:5060;transport=tcp>
Content-Length: .....

(XML payload)
```

### D. Sylantro AFS → S-CSCF:

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP scscf.ims.com
Via: SIP/2.0/TCP 10.10.1.1
From: <sip:+15105552000@ims.com>;tag=1234
To: <sip:+15105551000@ims.com>;tag=ABCD
Call-ID: fdc49228-7721a52-a2d0bbaf@100.10.1.1

CSeq: 2 NOTIFY

Contact: <sip:+15105551000@10.10.1.1;transport=tcp>
Event: [package_name: e.g.,dialog]
Content-Length: 0
```

# 5. SIP Protocol Compliance

Sylantro is strongly committed to the implementation of features and services based on IETF SIP standards and best current practices. Sylantro SIP Application Server has implemented fully or partially, the following RFCs and Internet Drafts in support of SIP Interworking and specific application features:

| SIP RFC and Internet Drafts Compliance | Comments |
|---|---|
| RFC 2327 “SDP: Session Description Protocol“ | Compliant. The Sylantro AFS does not initiate or terminate media RTP packets. As a B2BUA, it facilitates the exchange of SDP information between different endpoints in a given session. |
| RFC 3087 "Control of Service Context using SIP Request-URI" | Compliant. |
| RFC 3261 "SIP" | Compliant. *See Sylantro-SIP-Interop-R3dot2 document for more details..* |
| RFC 3262 "Reliable provisional responses" | Sylantro AFS sends a PRACK response to the SIP UA that requires it. SDP in PRACK is not supported. |
| RFC 3263 “Locating SIP Servers” | Compliant |
| RFC 3264 "Offer/Answer Model" | Partial compliance. Support only SDP requirements on modifying the session and accepting Hold SDP options. Other requirements in RFC 3264 are not supported. |
| RFC 3265 “SIP-Specific Event Notification” | Compliant. |
| RFC 3325 "Private Extensions to the SIP for Asserted Identity within Trusted Networks" | P-Asserted Identity and P-Preferred Identity headers |
| RFC 3326 “The Reason Header Field for SIP” | Compliant. |
| RFC 3515 "The SIP Refer Method" | Compliant. |
| RFC 3842 “A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)” | Compliant. |
| RFC 3891 “The Session Initiation Protocol (SIP) "Replaces" Header” | Compliant. |
| <draft-anil-sipping-bla-02.txt> | The SIP BLA IETF draft. |
| <draft-burger-sipping-netann-07> | For Music On Hold and Conferencing only |
| <draft-ietf-sipping-cc-conferencing-01> | The framework is used in the implementation of centralized conferencing feature. |
| <draft-ietf-sipping-dialog-package-06> | Used in SIP BLA, DCP and GCP implementations. |
| <draft-ietf-sipping-service-examples-06> - Call park and transfer examples | Partial. Park and Transfer features. |

| | |
|---|---|
| <draft-ietf-sip-privacy-04.txt> | Remote-Party ID header and Anonymity headers only. Available only on SIP gateway interfaces. |
| <draft-ietf-sip-serverfeatures-05.txt> | Supported and Require header supported. 421 Extension required response supported. |
| <draft-ietf-sip-session-timer-15.txt> "Session Timers in SIP" | Compliant (on network interface) |
| <draft-levy-diversion-05.txt> "Diversion Indication in SIP" | Partial. Used only to identify the diverting party and the reason for diversion at Sylantro AFS. The Sylantro AFS do not forward receive diversion headers. Diversion headers are sent only to configured SIP gateways on the server. Sylantro AFS does not forward diversion headers towards subscriber and client devices. |

## 5.1 SIP Methods

| SIP METHODS | AFS Support |
|---|---|
| ACK | Yes (Send/Receive) |
| BYE | Yes (Send/Receive) |
| CANCEL | Yes (Send/Receive) |
| INVITE | Yes (Send/Receive) |
| MESSAGE | No |
| NOTIFY | Yes (Send/Receive) |
| OPTIONS | Yes (Send/Receive) |
| PRACK | Yes (Send/Receive) |
| PUBLISH | No |
| REFER | Yes (Receive only) |
| REGISTER | Yes (Receive only) |
| SUBSCRIBE | Yes (Send/Receive) |
| UPDATE | No |
| INFO | No |

## 5.2 SIP Event Packages

| Eevent Packages | AFS Support |
|---|---|
| dialog | Yes |
| message summary | Yes |
| missed-call-summary | Yes |
| presence | Yes |
| reg-event | Yes (internally) |

Other event packages not yet supported.

## 5.3 SIP P-Headers

The following Private extension headers are supported:

| P-Headers | AFS Support |
|---|---|
| P-Charging Vector | Yes<br>– Send the received value in Routing mode<br>– Generate a value in Initiating mode |
| P-Asserted-Identity | Yes (send/receive) |
| P-Access-Network-Info | Yes<br>Send the received value in Routing mode |
| P-Visited-Network-ID | Yes<br>Send the received value in Routing mode |

# 6. Interoperability Requirements

## 6.1 General Requirements

The following are basic requirements of any SIP device or network element that interoperates with the Sylantro SIP Application Server:

1. Most Sylantro advanced features rely on the ability of a SIP UA to support re-INVITE requests (as defined in RFC 3261) to notify the UA of changes in SDP media session information. User agents that interoperate with Sylantro AFS MUST support re-INVITE mechanism as specified in RFC 3261.
2. User agents MUST honor the changes in the SDP when negotiated successfully and modify media sessions accordingly.
3. User agents MUST NOT changes the RTP port with every receipt of re-INVITE request in order for the SDP negotiation, facilitated via the Sylantro AFS (B2BUA), to converge on a final set of parameters.
4. User agents MUST be able to process INVITE requests without SDP payload. Furthermore, they MUST be able to process SDP payload in ACK requests.
5. Unified Messaging systems MUST be able to receive and process Diversion header information.
6. SIP/TCP should be supported for IMS since the size of the SIP messages frequently exceed the 1300 byte MTU limit.
7. S-CSCF/HSS configuration MUST be able to support configuration of filter criteria for users based on different SIP methods (in particular INVITE, REFER, and SUBSCRIBE/NOTIFY) and values and parameters associated with the SIP headers.
8. S-CSCF MUST be able to support maintaining the Request URI parameters sent by the AFS for requests towards media servers (see call flow examples in section 4).
9. The following SIP URIs are reserved for announcement and conferencing features:
    a. sip: dialog@<ims_domain_name>;[optional parameters]
    b. sip: conf@<ims_domain_name>;[optional parameters]
    c. sip: conf=<AS_generated_token>@<ims_domain_name>;[optional parameters]