Implementing End-to-End SIP Vol 2: SIP Telephone Signaling and Dial Plan Options

Avaya Aura Feature Package 4:
Communication Manager 6.3 Service Pack 6.0
System Manager 6.3.8

Document Number: 157210
September, 2014

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Introduction

"Implementing End-to-End SIP Vol 2: SIP Telephone Signaling and Dial Plan Options" is a companion document to the "Implementing End-to-End SIP Vol 1: Endpoint Deployment, Issue 2" White Paper. Volume 2 addresses Communication Manager 6.3 Service Pack 6.0 and System /Session Manager 6.3.8 known collectively as Avaya Aura® Feature Package 4.

Information provided in this document can be used to provision SIP Users in System/Session Manager and Communication Manager Evolution Server or Communication Manager Feature Server with support for sequenced applications.

1 IP Multimedia Subsystem (IMS) Signaling Flows

The Avaya Aura® SIP solution is based on IMS signaling architecture to support sequenced applications for calls to and from SIP and non-SIP endpoints. It is important to understand signaling flows between SIP endpoints using this architecture for proper administration of Session Manager (SM) utilizing System Manager (SysMgr) and Avaya Communication Manager Evolution Server (CM-ES) or Feature Server (CM-FS).

CM-FS supports SIP trunk connections to SM for access to:

- subscribed SIP users registered to SM
- access SIP Public Switched Telephone Network
- other on-net CM systems, CS1000
- Named Applications such as Avaya Aura Messaging

CM-FS does <u>not</u> support H.323 TDM or analog endpoints. CM-FS is connected to SM via a SIP signaling groups, which are IMS enabled. An IMS enabled signaling group is used to support the <u>half call model</u> for the sequencing of calls and features of the subscribed SIP users.

Starting in CM 6.0 CM can be configured as an Evolution Server. CM-ES supports

- all traditional H.323, TCM or analog endpoints and trunks
- all SIP trunk connections to SM specified above for FS

The biggest change in CM 6.0 is that SM replaces Session Enablement Server (SES) as a registrar for SIP telephones. CM-ES is connected to SM via a SIP signaling groups, which are <u>not</u> IMS enabled. A non-IMS signaling group is used to support the <u>full call model</u> for the limited sequencing of calls and features of the subscribed SIP users.

Describing specific sequenced applications that require the half call model implemented in CM-FS, and sequenced applications that can be supported by both CM-ES and CM-FS call models, is beyond the scope of this document. In this document the only sequenced application described in detail is one that involves a single CM-ES or CM-FS. Although both CM-ES and CM-FS alternatives are described in this paper CM-ES should be the configuration used to address a majority of customer requirements.

1.1 Application Sequencing

CM-ES, CM-FS and SM use IMS phase tags in SIP messages to support application sequencing. It is important to understand phase tags in CM-ES and CM-FS

A SIP station to SIP station call on a single Feature or Evolution Server uses four signaling legs between CM and SM (not including the initial off-hook imsorig):

1. imsorig
2. origdone
3. imsterm
4. termdone

A SIP station to SIP trunk uses imsorig and origdone signaling legs to the SIP Phone. A SIP trunk to SIP station uses imsterm and termdone signaling legs to the SIP Phone.

Following is a diagram that outlines call flows between SIP phones on the same CM-ES or CM-FS with multiple sequenced applications:

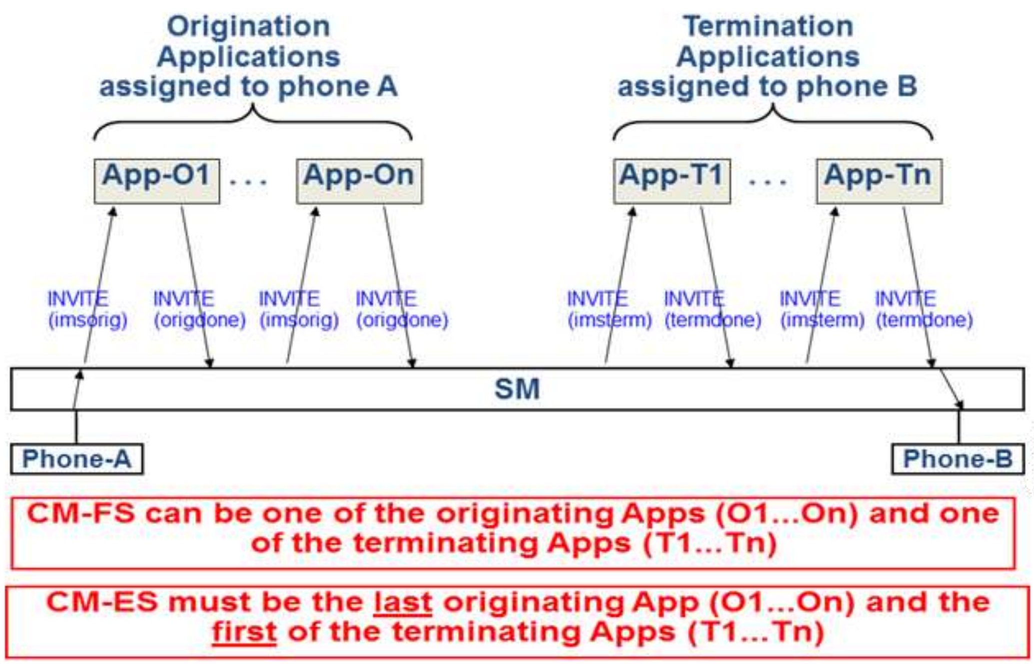


Figure 1: Application Sequencing in CM-ES and CM-FS

In this diagram, SIP Phone-A originates a call to SIP Phone-B. System Manager is used to define, originating and terminating sequenced applications in User Management for each SIP user. At least one of these sequenced applications on the origination and termination sides of the call must include CM-ES or CM-FS (Note: For each SIP user the same CM system must be specified for origination and termination sequencing). There can be multiple applications sequenced for a SIP user that initiates (originating sequence) a call as well as multiple applications sequenced for a SIP user that answers (terminating sequence) a call. Each sequenced application on the origination side of the call utilizes imsorig and origdone IMS tags. Each sequenced application on the terminations side of the call utilizes imsterm and termdone IMS tags

Sequenced applications are provisioned for users based on whether the user is on the originating or on the terminating side of a call. When designing sequences for applications that act on endpoints controlled by a CM-ES, the CM must be the last application defined on the origination side of a call and it must be the first application defined on the termination side of the call. A CM-ES in the previous diagram would need to be the last application in the origination sequence (App-On) and the first application in the termination sequence (App-T1). This is because the Evolution Server has implemented the full call model for call processing and all origination and termination feature processing occurs on the origination side of the call.

Implementing application sequencing for SIP users receiving features from a CM-FS is more flexible. Sequencing of CM-FS can occur before or after other applications on both the origination and termination sides of the call. This is because the CM-FS has implemented the half-call model separating origination processing from termination processing of SIP calls on the same Feature Server. It is beyond the scope of this paper to describe the ramifications of these rules on application development and deployment of sequenced applications with CM-ES or CM-FS.

1.2 CM-ES Origination and Termination Call Processing

On station to station calls, origination and termination feature processing occurs between imsorig and origdone legs of the call in the Evolution Server full call model. CM-ES does not do feature processing between imsterm and termdone call legs. Origination feature processing includes line and bridge appearance lamp updates that occur when the user goes off-hook. Termination features include: line and bridge appearance updates, call forwarding, send all calls, or coverage treatment that occur when phones ring and are answered.

In Figure 2 SIP Phone-A and Phone-B are on the same CM system (CM-A)

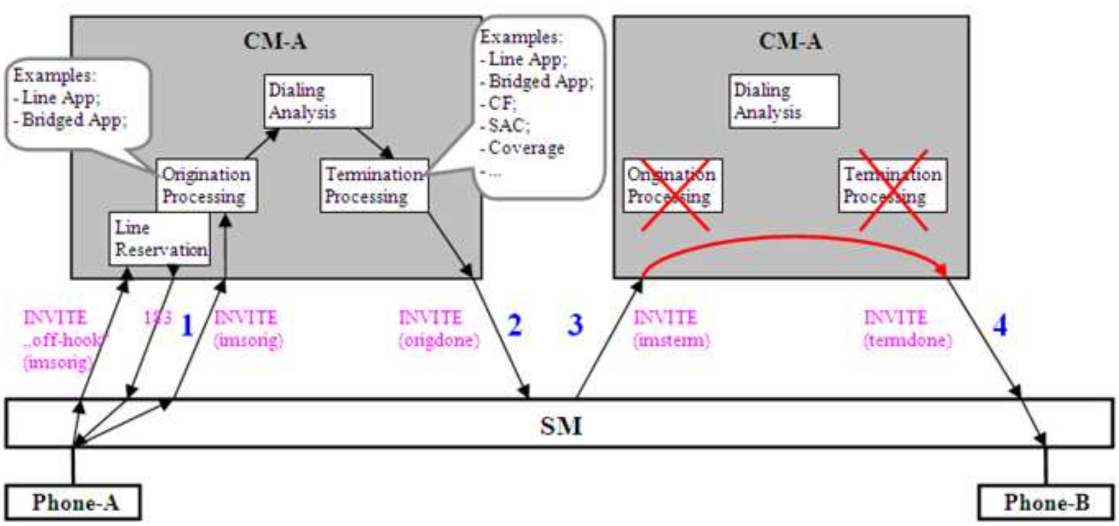


Figure 2: CM-ES Origination and Termination Call Processing

1.3 CM-FS Origination and Termination Call Processing

Origination feature processing occurs between the imsorig and origdone legs of the call in CM-FS half call model. Termination feature processing occurs between the imsterm and termdone legs of the call.

Figure 3 shows call flow for SIP Phone-A and Phone-B on the same CM system (CM-A)

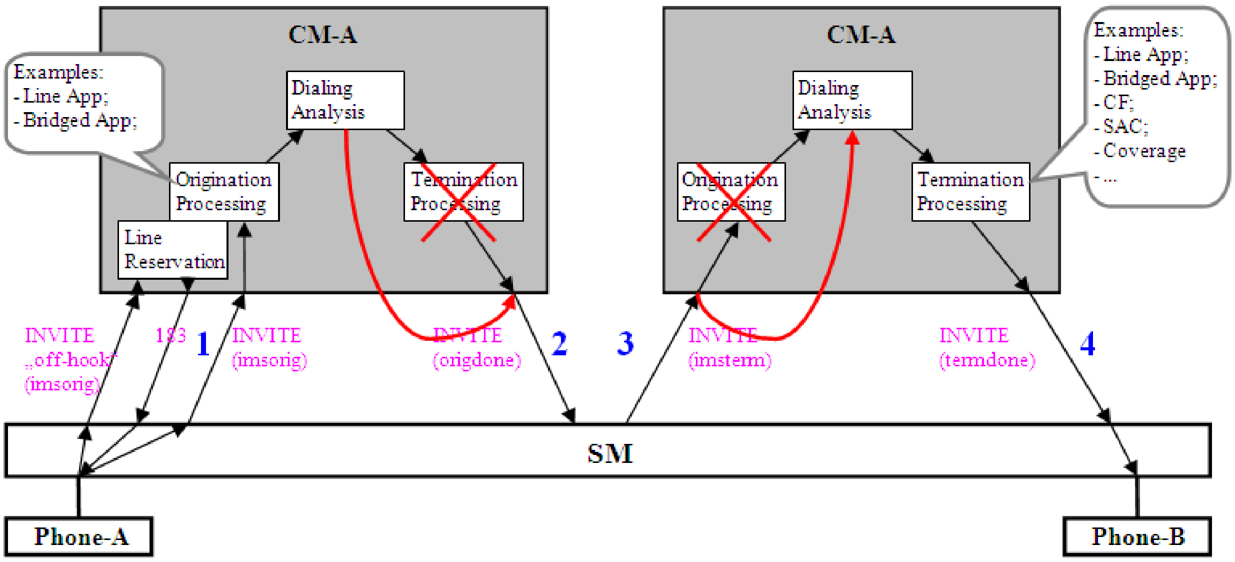


Figure 3: CM-FS Origination and Termination Call Processing

On station to station calls, separation of origination and termination processing in CM-FS provides full sequence application flexibility.

1.4 Call Legs and Call Records

A steady state SIP station to SIP station call on CM-FS uses four call legs (four IMS trunks). Two call records are used; a call record for the originating user (imsorig and origdone) and a call record for the terminating user (imsterm and termdone).

A steady state SIP station to SIP station call on CM-ES uses two call legs (two IMS trunks). A single call record is used for the originating user (imsorig and origdone). For a CM-ES, the SIP signaling groups are configured as “non-IMS” and only the imsorig and origdone legs remain active (one call record) on a steady state call. This is because the CM-ES on the originating half of the call has performed all the call processing functions for the terminating user as well as the originating user. CM-ES adds *atrp=shortcut* to the Accept-Contact header in the origdone SIP message. When CM-ES receives the imsterm phase tag with *atrp=shortcut*, it acts as a proxy and directly sends the message back to SM in the termdone leg without any further call processing or use of any additional trunk resources.

The description assumes that both users A and B are assigned to the same CM-ES or CM-FS. If both users are on different CMs, CM-A will do origination call processing for the originating SIP phone and CM-B will do termination call processing for the terminating SIP phone.

2 Session Manager Roles

When implementing SIP telephony, it is important to understand the roles of Session Manager.

SM performs two major functions:

1. Centralized router for calls to non-SIP users using Network Routing Policies (NRP) for
    a. All outbound PSTN calls (off-net)
    b. All inbound PSTN calls to
        i. non-SIP users on access elements and CM-ES,
        ii. VDNs and vectors on access elements, CM-ES and CM-FS
    c. Named applications
        i. Meeting Exchange,
        ii. Avaya Experience Portal,
        iii. Avaya Aura Messaging
    d. All private network calls (on-net calls) between non-SIP telephone users on access elements and CM-ES
2. Registrar for SIP users and associated application sequencing to CM-ES or CM-FS based on SIP user profiles.

CM-ES can support both SIP and non-SIP users on the same system. SM routes on-net and off-net calls to non-SIP phones on CM-ES based on matching SM Dial Patterns and Network Routing Policies (NRP). Calls to SIP phone users are placed by SM based on matching on SIP user handles and subsequent application sequencing to the proper CM.

2.1 SM as a Centralized Router

SM uses a numbering plan to centralize routing of calls to non-SIP users by administering SM Network Routing Policy (NRP) "routing applications". Applications supported include: Domains, Locations, Adaptations, SIP Entities, Entity Links, Routing Policies, and Dial Patterns.

The Dial Patterns application, in particular defines the enterprise numbering plan (Note: "Number Patterns" is a more precise description of this application than "Dial Patterns" application since the numbers administered in many cases are not dialed digits). Dial plans are local and contained within CM Access Elements, Evolution, or Feature Servers.

SM adapts the dial plan used by CM or other PBX systems to the numbering plan in SM for analysis and centralized routing. SM uses ingress adaptation modules to modify addresses from non-SIP users. After analysis is done and routing determined, SM uses egress adaptation modules to modify the analyzed numbering plan digits to the local dial plan requirements of the non-SIP user.

Best Practice:

CM adapts SIP Station OPTIM call legs within the same CM-ES or CM-FS system and not ingress or egress adaptations in SM. Supporting a mix of SIP and non-SIP endpoints on CM-ES requires understanding of the underlying SIP call flows described in this document in order to avoid adapting SIP user, sequenced call flows. Examples that follow in this paper rely on CM to adapt numbers to public or private format and not SM when handling SIP telephone user call flows to insure integrity of SIP call processing of Subscribe, Publish and Notify messages between CM and SM.

2.2 SM as a Registrar

As a registrar, SM is responsible for processing SIP Registration messages from SIP users logging in, and SIP Invite messages from a SIP user originating a call. In both cases SM looks for a match in the Contact header with a SIP handle administered in SM. If there is a match, SM sequences the subsequent Subscribe and Invite messages to the associated CM.

When Session Manager receives a SIP Invite message from a non-SIP user that is originating a call, it first looks for a match of R-URI header (called number) with a SIP user handle for termination sequencing. If there is no match SM then looks for matches in NRP dial patterns.

SIP handles used for application sequencing to SIP users and dial patterns used by NRP to route calls to non-SIP users must be coordinated for proper routing of calls between these two elements.

Sections 3-5 address design decisions that must be made before implementing end-to-end SIP telephony designs:

- The Numbering Plan used by SM to analyze and route calls
- SIP handles administered in SM to support registrar function for SIP telephones
- Extension numbers implemented in CM-ES or CM-FS based on dial plan

3 Numbering Plans

Numbering plans address the numbering scheme used to address telephony users globally and within an enterprise.

Following are two recommended numbering plans that SM can use for centralized routing and SIP handle sequencing:

- E.164 Public Numbering Plan
- Enterprise Canonical (Private) Numbering Plan

Attributes that are part of any numbering plan include:

- Geographic
- Scalable
- Global
- Unique numbers
- Explicit numbers (minimize occurrences of short inter-digit timeouts)

3.1 E.164 Public Numbering Plans

Public numbering plans are numbering schemes used in the telecommunications industry to ensure access to the public telephony PSTN infrastructure. The public telephony numbering plan that is currently recognized globally is the ITU-T E.164 Numbering Plan Recommendation. Adoption of the E.164 numbering plan in the enterprise guarantees unique numbers for routing; E.164 numbers are universally identified by appending a "+" at the beginning of the numbers.

The E.164 format is: + Country/Region Code (Area Code) Subscriber Number

- "+" - indicates that the number is in E.164 format
- Country/Region Code - the standard country/region code that identifies the country or region for a phone number.
- (Area Code) - the area or city code for the phone number
- Subscriber Number - the number for a phone subscriber

Best Practice:

All public entries in the SM Dial Patterns table are in E.164 format with the + sign followed by the country code and national number. There are many advantages to this approach:

- International calls route with minimal number of entries using the + sign rather than routing based on country specific international prefixes.
- All calls beginning with + route to a default SIP entity. Exceptions to this default entry can be based on + followed by country codes or other more specific entries.
- E.164 format aligns with LDAP, OCS, service provider, and wireless carrier conventions.
- On ingress adaptations to SM from CM-ES or CM-FS, the international prefix ("00", "011" etc.) is deleted and replaced with the + sign before analysis and routing.
- On egress adaptations from SM the + is deleted along with other modifications in order to complete calls to non-SIP users (and VDNs) on access elements, CM-ES and CM-FS.

Since E.164 format is recommended for routing of public numbers, national and international prefixes are not administered in the Dial Patterns table. SIP users that require access to the public network should have at least one handle that is in E.164 format. SIP users can have additional handles that are enterprise canonical (unique).

Best Practice:

Avoid using a national numbering plan format (without the + and the country code) as an alternative to the E.164 numbering plan format. The only exception to this is national Service Codes such as "911" in North America.

3.2 Private (Enterprise Canonical) Numbering Plans

Enterprise and governmental agencies use private numbering plans for dialing within an organization. This reduces the total number of digits analyzed and dialed to reach other subscribers in the private network. Private numbering plans can also be implemented by Service Providers using Software Defined Networks (SDN). The goal of a private numbering plan, like public numbering plans, is to insure unique numbers in the enterprise for analysis and routing of calls.

Private numbering plans usually contain two parts:

- Routing Prefix
- Extension

With 7-digit private numbering plans the routing prefix is usually the first 3-digits and the extension is the last 4 digits.

Enterprise canonical private numbering plans can coexist in SM with the E.164 Public Numbering Plan. SM uses the Dial Pattern table to analyze and route private network numbers. These private format dial patterns do not contain a + sign. This presents a strategy that clearly demarks public from private number routing and analysis. Use of both public and private numbers for analysis and routing in SM would certainly be used in the case where the private numbering plan has no relationship with the public numbering plan. Examples of this include retail stores that have routing prefixes based on store numbers and extensions based on departments within the store that are common within all store locations.

The extension number part of a private numbering plan can be associated with a public numbering plan Direct Inward Dial (DID, IDD) number. Numbers can be adapted from a private number to a public number by deleting the routing prefix and inserting the lead digits of the public number in front of the extension number. When there is a relationship between the enterprise private and public numbers, decisions must be made as to how to route all of these numbers.

SM can maintain public and private numbering plans in SM for analysis and routing. Another strategy is to use adaptations in CM or ingress adaptations in SM to convert both private and public number formats received from CM to E.164 numbering format for analysis and routing by SM. In this case dial plans in Communication Manager can be based on public and private numbering plans, but standardize on E.164 numbering for analysis and routing in SM. SM can then use egress adaptations to convert the public numbers back to the private numbering plan for analysis by Communication Manager.

Best Practice:
Following are several deployment strategies to adapt public to private and private to public numbers for calls that overflow from private to public facilities (on-net to off-net calls).

1. Adapt private numbers to public numbers in CM before sending to SM using Automatic Alternate Routing (AAR) Digit Conversion. All analysis and routing in SM is done based on E.164 numbers. Egress adaptations are used to convert E.164 to dial plan used by CM.
2. CM sends both public and private format numbers to SM. SM uses both E.164 and private number plans for analysis and routing. SM egress adaptations are used to convert E.164 numbers to the dial plan used by CM.
3. Adapt private to public numbers using ingress adaptations in SM. SM analyses and routes calls based on E.164 numbers. Egress adaptations are used to convert E.164 to dial plan used by CM. Any additions to the private to public table must then be applied to every ingress adaptation associated with SIP Entities.

Best Practice: The following **strategy** addresses calling party number (CPN) contained in the SIP P-Asserted Identity (PAI) and History-Info headers on calls that overflow from private to public facilities. CPN information is displayed and logged at the destination user. Log files can then be used to return a call. Decisions regarding format of CPN should be made in CM:

- Privately dialed numbers should use private numbering table and send private formatted CPN. Private format includes CPN in national number format.
- Publically dialed numbers should use the public-unknown-numbering table and send E.164 formatted CPN to SM. CM inserts + to all calls to SM when using this table.

4 Dial Plans

A dial plan is functionally different than a numbering plan. A dial plan specifies the digits dialed within the constraints of a numbering plan. A dial plan describes how the numbering plan can be used by subscribers. A dial plan is geographically significant whereas a numbering plan is global in nature. There are multiple national dial plans associated with the E.164 Numbering Plan. Within a national or geographic boundary there can also be numerous dial plans. A typical dialed phone number is comprised of digits that need not always be dialed and digits that must always be dialed including national and international prefixes.

SM uses numbering plans to analyze and route calls. SM does NOT manage dial plans. Dial Plans are defined by the communication systems connected to SM. This includes CM configured as an access element, Evolution Server, or Feature Server. CM dial plans are used by analog, digital, IP, and SIP telephones.

CM and SM use the following procedure to communicate dial plan information to SIP phones with Advanced SIP Telephony (AST) capabilities:

1. CM synchronizes Dial Plan Analysis, AAR, and ARS information with SM.
2. The dial plan information is stored in the Personal Profile Manager (PPM) database.
3. SM passes dial plan formats to SIP phones using PPM when the phones initially register to SM and subscribe to CM, or when this information is manually pushed to the phones using System Manager.
4. When a SIP user dials a call from a SIP phone keypad, this information is used to
    a. recognize AAR and ARS feature codes and subsequent dial tone,
    b. determine when to send digits to SM for feature processing by CM-ES or CM-FS

When the phone has a dial string that matches the dial plan in the phone, it sends the digits to SM in a SIP INVITE message. SM looks up the user profile of the originating SIP telephone and forwards the call to the associated CM-ES or CM-FS specified in the origination application sequence. On the termination side SM looks up the user profile of the terminating SIP telephone and forwards the call to the associated CM-ES or CM-FS specified in the termination application sequence.

4.1 Public Dial Plans

The structure of a dial plan based on the E.164 numbering plan includes:

- An international access code (international direct dialing prefix) for dialing international calls.
- Country Code for dialing calls between countries.
- National access code (national direct dialing prefix) for dialing national (non local) calls; it is never dialed for calls between countries.
- Area Code/City Code dialed from inside or outside the code area.
- Local number (subscriber number) dialed within an area/city code when permitted.

If a user can dial the + sign then there is no need to dial the international prefix if the service provider supports this format. This is usually seen in mobile phone service provider networks. Users on a cell phone can dial the + sign and the country code, city/area code/ subscriber number and not need to know the international prefix of the country where the call originates. International travelers usually have cell phone directories in E.164 format to reduce complexity of understanding the international prefix rules of the country that they are visiting. The latest versions of Avaya SIP 96x1 telephones also support dialing of the + sign.

4.2 Private Dial Plans

Dial plans developed with enterprise canonical (private) numbering plan formats are usually a uniform length. Enterprises or governmental agencies use these plans for dialing within the organization.

Private dial plans do use routing codes. Following is a typical Private Dial Plan format RNX-XXXX where:

- RNX = [2-7][0-9][0-9]
- Station = [0-9][0-9][0-9][0-9]

Private dial plans do not necessarily need to be 7-digits; this is an example. Often, customers try to match the RNX of a 7-digit extension with the last 7 digits of the public number. This strategy can end up with conflicts between locations that may have the same RNX in this case or a conflict at 7-digits between the public number and either the ARS or AAR access code within a PBX dial plan.

4.3 CM Dial Plan Considerations:

Private Branch Exchange (PBX) systems have historically developed Public and Private Dial Plans.

A typical PBX dial plan includes the following attributes:

- feature access code used to access public routing tables and provide least cost routing
  - the ARS feature access code can begin with a * or # sign
  - allows the digit "9" to be used as the leading digit of an extension
- feature access code used to access private routing tables and reflects private numbering plan
  - In CM, the AAR feature access code can begin with a * or # sign
  - allows the digit "8" to be used as the leading digit of an extension
- leading digits defined as extensions and operator with some defined length
  - In CM the extension length can be up to 13 digits
  - Short codes for dialing within a location
- additional feature access codes used to access features and for direct access to trunks

Best Practice:

PBX dial plan extension numbers must be unique and non-ambiguous within CM to minimize short inter-digit timeouts. Short inter-digit timeouts occur when there are two (or more) potential matches to digits being dialed. Short inter-digit timeouts can result in misdialed numbers or introduce delays in dialing when the shorter dial string is dialed.

5 SIP Handles and CM Extension Numbers

5.1 SIP Handles Overview

SIP handle(s) must be unique in the enterprise based on the enterprise numbering plan. Decisions about choice of SIP handle format are closely aligned with the numbering plan. If E.164 numbering format is chosen for enterprise routing, then the SIP handle should be E.164 or based on E.164. If enterprise canonical format is chosen, then the SIP handle should be enterprise canonical as well. There are cases where both number formats are used for routing in the enterprise.

SIP handles associated with SIP users are used for: login to SIP telephones by SIP users, registration to SM, subscription to CM and subsequent sequencing of SIP user calls. The handles discussed in this document are numeric followed by a domain. The SIP user logs into the SIP telephone by dialing the numeric part of a user handle. Domains are discussed further in Section 11.

When using E.164 numbering plan for routing, two SIP handles per user is required:

- E.164 handle in E.164 format with the leading + sign to interface with SM NRP
- SIP handle without the + sign for SIP users
    - to log in and register to SM and subscribe
    - to log in and subscribe to CM-ES or CM-FS

5.2 CM Extensions Overview

CM extensions must match or be a subset of the SIP handle. In CM, there is a direct mapping between the SIP handle used to register the user and the extension number administered on the "off-pbx-telephone station-mapping".

CM extensions and short codes are administered as part of the dial plan. Depending on the size of the CM system and the size and number of DID numbers used by this CM system, the length of the extension number can be short (3-digits for example) or can be relatively long based on E.164 number. All extensions must be unique within the CM system. All short codes need to be unique within a location defined in the CM system (short codes are beyond the scope of this document). Extension numbers do not need to be unique in the enterprise.

Consolidated CM systems that cover large geographic areas with large number of phones usually have longer extension lengths than smaller standalone systems because of the need for uniqueness of extensions within the CM system. These larger systems will usually have extension numbers that are also enterprise canonical and can be the SIP handle as well. Extension numbers in smaller standalone systems are usually not enterprise canonical and will be a subset of the number used as the SIP handle.

5.3 SIP Handle and Extension Options

Following are four different SIP handle and extension options used in conjunction with an existing or proposed enterprise wide numbering and dial plan. These options are not exhaustive, but do represent a sample of combinations handles and extensions that can be utilized. In many cases more than one of these options can be deployed in the same enterprise; every one of these options has an E.164 handle to coordinate with Session Management Network Routing Policies (NRP) based, at least in part, on E.164 Numbering Plan.

5.3.1 Option One: Extension Based on E.164 Numbering Plan

This configuration can be used with consolidated CM system with many SIP endpoints. The handles used are the Avaya E.164 handle with the + sign and an Avaya SIP handle without the + sign. In this case the Avaya SIP handle used to login to the SIP phone is the same as the extension number. The Avaya SIP handle is also the preferred handle used by SM on imsorig and imsterm call legs. Following is an example based on North America:

- SIP Handles
    - Avaya E.164 +19952250022
    - Avaya SIP 19952250022 (Public Long)
- Extension Number 19952250022 (Public Long)

5.3.2 Option Two: Extension Based on Private Long Number

This is another typical configuration of a consolidated CM system with many SIP endpoints. The handles used are the Avaya E.164 with the + sign and an Avaya SIP handle that reflects enterprise canonical numbering plan. In this case, again, the Avaya SIP handle used to login to the SIP phone is the same as the extension number. The Avaya SIP handle is also the preferred handle used by SM on imsorig and imsterm call legs. In the following example based on North America, only the last four digits match the Private Long number:

- SIP Handles
    - Avaya E.164 +19952252222
    - Avaya SIP 3212222 (Private Long)
- Extension Number 3212222 (Private Long)

5.3.3 Option Three: Extension Based on Subset of E.164 Numbering Plan

This configuration is appropriate for CM system in which shorter length extension numbers are desirable. These extension numbers are unique within the CM system, but not unique in the enterprise. The extension length in this case is usually 4, 5, or 7-digits and reflects a subset of the E.164 based handle. The key here is that the extension number is a subset of the Avaya SIP handle used to login into the SIP phone. The Avaya SIP handle is also the preferred handle used by SM on imsorig and imsterm call legs. Following is an example based on North America:

- SIP Handles
    - Avaya E.164 +19952250222 (E.164)
    - Avaya SIP 19952250222 (Public Long)
- Extension Number 50222 (Public Short)

5.3.4 Option Four: Extension Based on Subset of Private Numbering Plan

Smaller CM systems with shorter length extension numbers can use configuration. These extension numbers are unique within the CM system, but not unique in the enterprise. The extension length in this case is usually 4 or 5-digits and reflects a subset of the enterprise canonical based handle which is usually 6 or 7 digits The extension number is a subset of the Avaya SIP handle used to login into the SIP phone. The Avaya SIP handle is also the preferred handle used by SM on imsorig and imsterm call legs. In the following example based on North America, only the last four digits match the Private Long number:

- SIP Handles
  - Avaya E.164 +19952252002
  - Avaya SIP 3212002 (Private Long)
- Extension Number 2002 (Private Short)

5.4 Summary of SIP Handle and CM Extension Definitions

Following is a summary of definitions used to describe SIP handles and CM extension numbers.

- "**Enterprise Canonical Number**" (ECN): Number unique in enterprise. The number is in private long number format as defined by the enterprise private numbering plan. SIP handles to register to SM and subscribe to CM can use ECN format. In CM ENC numbers can be defined in the dial plan as an extension.
- **E.164 Number** is a number with a leading "+" followed by country code and national number. These numbers are unique in the enterprise. SIP handles to register to SM and subscribe to CM can use E.164 number format. In CM, E.164 numbers cannot be defined in the dial plan as an extension.
- **Public Long Number** is a public number that is unique in the enterprise. SIP handles to register to SM and subscribe to CM can use the public long number format.
  - E.164 number without the leading "+" can be defined in CM dial plan as an extension.
  - National number in single country enterprise implementations of SM can be defined in CM dial plan as an extension.
- **Private Long Number** is a number that is part of a private numbering plan used to route in the enterprise network and is usually 6 or 7 digits in length. The private long number can be used to register to SM and subscribe to CM. In CM this can be defined in dial plan as an extension.
- **Public Short Number** is a number that is a subset of the Public Long Number and is not enterprise canonical. The public short number cannot be used to register to SM or subscribe to CM. In CM it can be defined in the dial plan as an extension or short code.
- **Private Short Number** is a number that is a subset of the Private Long Number and is not enterprise canonical. The private short number cannot be used to register to SM or subscribe to CM. In CM it can be defined in the dial plan as an extension or short code.
- "**avext parameter**": An extension (short) number can be converted to a long number using the public-unknown-numbering or private-numbering table (before origdone in CM-ES and termdone in CM-FS). In this case, the extension number is added as avext parameter to the PAI and Contact messages. The avext parameter is used to display extension number on the answering SIP Phone rather than PAI. See ICHT Section 6.1.5.

6 SM and CM SIP Telephone Call Processing

This section describes Session Manager and Communication Manager Call Processing to support SIP telephone call flows; it is based on the four options described in section 5.3. All four cases that follow support: registration of SIP phones to SM, subscription of SIP phones to Communication Manager, and subsequent INVITE, PUBLISH, NOTIFY and REFER messages sent during normal call processing.

The user handle length can be E.164 as well as Public or Private Long numbers. The extension number can be the same format as the Public or Private Long numbers, or they can be shorter in length. In cases where the handle and the associated login of the user is longer than the extension number, the handle/login is the long form of the number and the extension is the short form of the number.

Best Practice:
It is important to coordinate translations between SM and CM because SM only knows about the long number (canonical) forms and CM only knows about the short forms of the number.

6.1 CM tables for calling and called party

The following tables are used to convert from long number forms to short number forms and vice versa.

6.1.1 "off-pbx-telephone station-mapping" table

The off-pbx-telephone station-mapping table is used to support multiple applications. CM uses this table to provide features and media for SIP stations over Off-PBX Integration and Mobility (OPTIM) trunks using the Outboard Proxy SIP (OPS) application.

This table is used when a SIP phone makes a call (origination mapping) or when a SIP phone receives a call (termination mapping). Option 4 provides a good example of the relationship between short and long number manipulation because E.164 handle, SIP handle, and extension number are all different.

```
display off-pbx-telephone station-mapping 3212002              Page   1 of   3
                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number     Trunk        Config  Dual
 Extension                  Prefix                      Selection    Set     Mode
 2002              OPS         -       3212002           aar         1
 2003              OPS         -       3212003           aar         1
```

In Option 4 there are two SIP handles for each user: +19952252002/+19952252003 (E.164) and 3212002/3212003 (private long) which is also the <u>preferred handle</u>. The extension numbers in this example are 2002/2003. When provisioning this phone in System Manager the "preferred handle" 3212002/3212003 and "extension" 2002/2003 are pushed to this form in CM as the "Phone Number" and the "Station Extension". The "Phone Number" is the long form of the number associated with the SIP handle used to login as a user.

When 2002 calls 2003, origination mapping is used to convert the long form of the number 3212002 to the extension number 2002 upon receipt of imsorig message from SM when 3212002 initiates the call. Termination mapping is used to convert the extension number 2003 to 3212003 before sending termdone to SM to terminate a call to phone 3212003 (see Section 10).

6.1.2 "public-unknown-numbering" table

This table converts short number (extension) calling party/PAI to E.164 number when the associated signaling group/trunk group is connected to SM. In CM-ES and CM-FS this conversion occurs prior to origdone/terminating call leg when a SIP phone initiates a call to a SIP trunk.

SIP station to station calls are a different matter and highlight one of the main differences between use of this table in CM-ES and CM-FS. In CM-ES Server this short to long conversion of calling party/PAI occurs before origdone/terminating in the same manner as calls to SIP trunks.

In CM-FS, this table is used to convert short number to long number for both called party/R-URI and calling party/PAI before origdone processing on the origination side SIP station to station calls. On the termination side of the call CM-FS uses this table in a traditional manner and converts short number (extension) calling party/PAI to E.164 number before termdone call leg.

6.1.3 "private-numbering" table

This table converts short number (extension) calling party/PAI to private long number. In CM-ES and CM-FS this conversion occurs before origdone/terminating call leg when a SIP phone initiates a call to a SIP trunk.

SIP station to station calls are a different matter and highlight one of the main differences between use of this table in CM-ES and CM-FS. In CM-ES Server this short to long conversion of calling party/PAI occurs before origdone/terminating in the same manner as calls to SIP trunks.

In CM-FS, this table is used to convert short number to long number for both called party/R-URI and calling party/PAI before origdone processing on the origination side SIP station to station calls. On the termination side of the call CM-FS uses this table in a traditional manner and converts short number (extension) calling party/PAI to private long number before termdone call leg.

6.1.4 CM algorithm used to gain access to public and private tables

Within CM, the public-unknown-numbering or private-numbering table maps calling party information (CM-ES and CM-FS) as well as called party information (CM-FS prior to origdone leg) from short to long forms. Calling party information is contained in SIP "From", and "PAI" headers. Called party information is contained in the SIP "To" and "Request URI" headers. This information is used by CM and SM for origination and termination call processing.

CM uses the following algorithm to determine which table to use:

1. For calls initiated by SIP phones, CM uses the public and private tables based on:
    a. call type of the dialed number string match in Automatic Alternate Route (AAR)/Automatic Route Selection (ARS) and/or
    b. numbering format administered on the route pattern preference chosen, based on numbering format of the associated trunk group
2. The SIP trunk group format can be set to either public or private.
3. For all trunk groups including SIP
    a. If the trunk group numbering format is set to private and
        i. call type set to npvt or lpvt in ARS or
        ii. call type set to unku or lev0, lev1 in AAR or set to or
        iii. unk-unk or lev0 on the route pattern (use of numbering format on the route pattern overrides the table to use specified by call type set in AAR/ARS)
    b. Then the call will use the private numbering table.
4. For all trunk groups including SIP
    a. If the trunk group numbering format is set to private and
        i. call type set to fnpa, hnpa, intl, natl, pubu or
        ii. call type set to aar or intl in AAR and
        iii. numbering format is blank on the route pattern (use of numbering format on the route pattern overrides the table to use specified by call type set in AAR/ARS)
    b. Then the call will use the public numbering table.
5. For all trunk groups including SIP
    a. If the trunk group numbering format is set to public,
    b. Then **all** calls use public table regardless of call type or route pattern administration.
6. Calls initiated by non-SIP phones use the same rules as above except that lpvt and npvt call types in ARS use the public table even if trunk group numbering format is set to private.

NOTE: In CM, Notify, Publish and Subscribe messages and the History-Info and Diversion Headers

1. use the public table if trunk group numbering format is public and
2. use the private table if trunk group numbering format is private

These messages do not use the algorithm specified above. In order to support proper population of SIP messages, if the trunk group uses private numbering format, both the public and private numbering tables must be filled out properly.

Use of the private numbering format on the trunk group is an important capability. It allows the administrator of the system the flexibility to use either the public or private numbering table on a route pattern by route pattern basis for access to on-net and off net users using a single set of trunks.

1. If a user dials a public number the administrator can route that call to a route pattern that populates the calling party number with a public number.
2. If a user dials a private number, the administrator can route the call to a route pattern that populates the calling party number with a private number by specifying unk-unk on the numbering format of the route pattern.

6.1.5 "inc-call-handling-trmt" table

The incoming call handling treatment (ICHT) table converts public/private long numbers to short numbers:

- On SIP station to station and SIP station to SIP trunk calls ICHT converts called party/Request-URI in CM-ES and CM-FS upon receipt of imsorig message.
- ICHT converts <u>both</u> called party/Request-URI and calling party/PAI) in CM-FS upon receipt of imsterm message on the termination side of SIP station to station calls.
- For Subscribe Messages to CM, the ICHT does long to short digit manipulation rather than "off-pbx-telephone station-mapping" originating mapping. This is important to note when the extension number is not the same as the handle used to log into the phone (see Option Three and Option Four below).

CM uses the ICHT table for two additional purposes in addition to manipulation of R-URI in CM-ES and R-URI and PAI in CM-FS from long to short forms.

CM uses ICHT to construct "avext parameter" on SIP station to station calls in CM-ES and CM-FS. In CM-ES the avext parameter is constructed using ICHT prior to origdone. In CM-FS, the avext parameter is constructed using ICHT prior to termdone processing. This results in ability of the terminating SIP phone to display the extension number of the calling party SIP phone (short form) rather than PAI (long form) generated by public-unknown-numbering or private numbering tables.

CM also uses ICHT to determine whether the response to imsorig call leg from SM on behalf of SIP station originating a call to another SIP station or SIP trunk should be origdone or terminating call leg. Following is a discussion of this call flow process for SIP station to station calls and SIP station to SIP trunk calls.

On SIP station to station calls, when CM receives an imsorig call leg from SM it processes the call and applies origination processing in CM-FS case and both the origination and termination processing in the CM-ES case. One that processing is complete CM now needs to send the call back to SM for further call processing. Recall, that before responding back to SM the extension number (short number) of the originating SIP station is converted to the long number as specified above in either public-unknown-numbering or private-numbering table.

CM uses ICHT of the outbound SIP OPTIM trunk leg to "audit" the PAI generated by public-unknown-numbering or private-numbering table of the station. Audit of PAI means that the conversion administered in ICHT is applied to see if the originating station is a SIP station, but does NOT actually change PAI in the call flow.

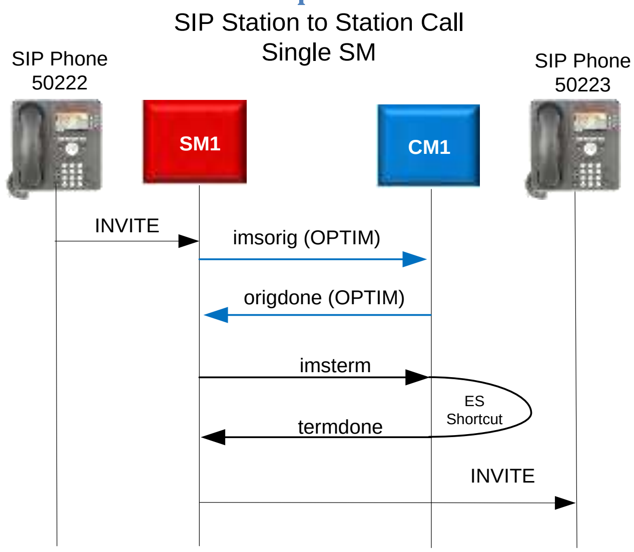
- For station to station calls, ICHT should have an entry on the associated outbound trunk that converts PAI constructed by Private/Public table to extension number.
- Audit of the outbound OPTIM trunk using the associated ICHT will result in short form match of the originating extension. The origdone call leg is returned to <u>same</u> SM that initiated the imsorig. , The avext parameter is added in PAI and Contact headers in CM-ES in origdone messages. In CM-FS, the avext parameter is added in termdone messages and populates PAI with the long form of the number originally generated by Private/Public tables.

Following is a diagram of a SIP station to station call on a single SM using an Option Three call flow example. In this case the public-unknown-numbering table has built PAI of +19952250222 for extension 50222. The ICHT table for the outbound OPTIM trunk has the following entry:

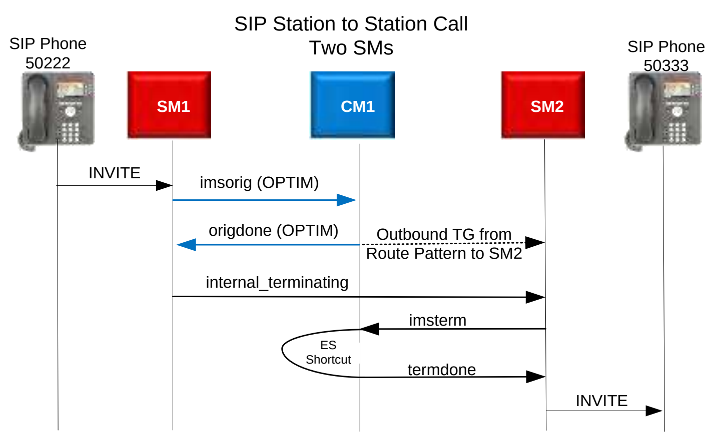
```
change inc-call-handling-trmt trunk-group 910                   Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
tie             12 +1995225         7
```

In the audit, there is a match with the extension number 50222 after converting +19952250222. CM proceeds with the call using origdone call leg with +19952250222 in PAI.

6.1.6 ICHT Call Flow Examples



If SIP stations are controlled by different SMs the call flow looks like this:



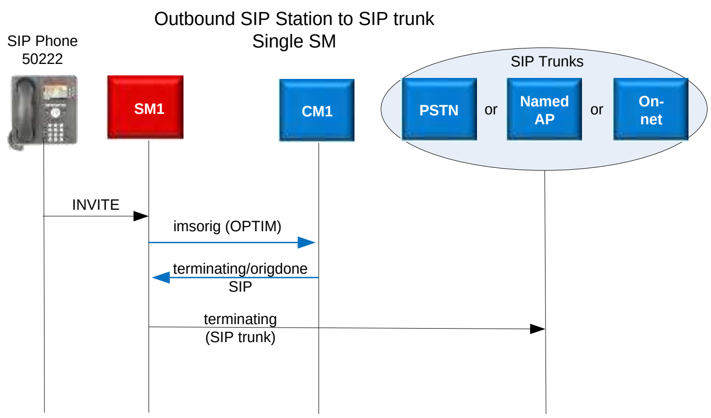
With multiple SMs in the call, the outbound call flow is dependent on how the incoming call handling treatment (ICHT) table in CM is administered for outbound SIP trunk group(s). CM audits the PAI of the originating station (long form of +19952250222) built from the public numbering table prior to sending the call back to SM with ICHT to determine if the short form is the SIP extension of the originating station. <u>If the audit of the outbound trunk ICHT results in short form matching the extension, then the signaling for the call is returned to the originating SM as origdone call leg:</u> signaling for the origdone call leg goes back to SM1 while the trunk group chosen in CM to SIP phone 50333 uses trunk group to SM2.

If ARS/AAR routing for origdone processing is to a different SM than the SM that initiated imsorig, CM will still show trunk specified in routing as in use. Signaling for origdone and trunk group used can end up on different SMs. If explicit sequencing of origination applications after CM is required, origdone call processing must be utilized. In any case this is the recommended administration for SIP station to station calls.

For SIP stations originating calls to SIP PSTN, CM also uses ICHT of the outbound SIP trunk leg to <u>audit</u> the PAI generated by public-unknown-numbering or private-numbering table of the station.

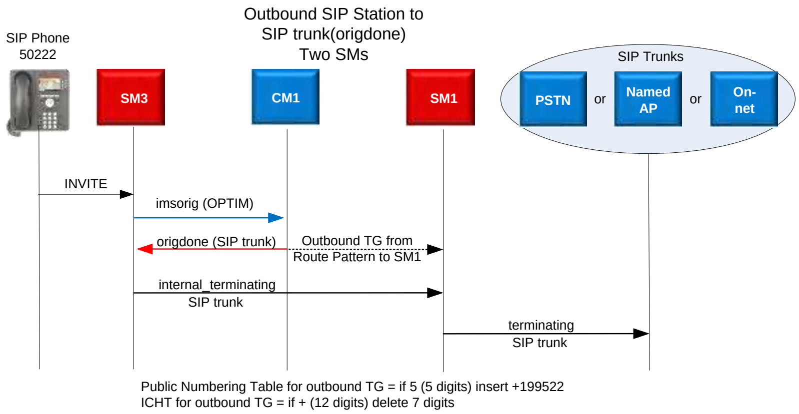
- If the audit of the outbound SIP trunk using associated ICHT results in short form match of the originating extension, then the call is returned to same SM that initiated imsorig SM as origdone instead of terminating.
- If there is no match to the extension after the audit, CM returns the call to the SM specified in AAR/ARS routing with associated trunk group member being used with a terminating phase tag.
- Implicit call sequencing including Collaboration Environment can use terminating call legs. In this case the signaling and trunk group used are to the same SM.

On outbound SIP station to SIP trunk calls, the imsorig (SIP station) call leg between SM and CM uses an OPTIM trunk group. A non-OPTIM trunk group(s) is used to access: PSTN trunks; on-net trunks to CM and other PBXs; or named applications such as Aura and Modular Messaging, Meeting Exchange and Avaya Aura Conferencing. The SIP trunk call leg is either terminating or origdone depending on administration of ICHT of the outbound SIP SIP trunk group.

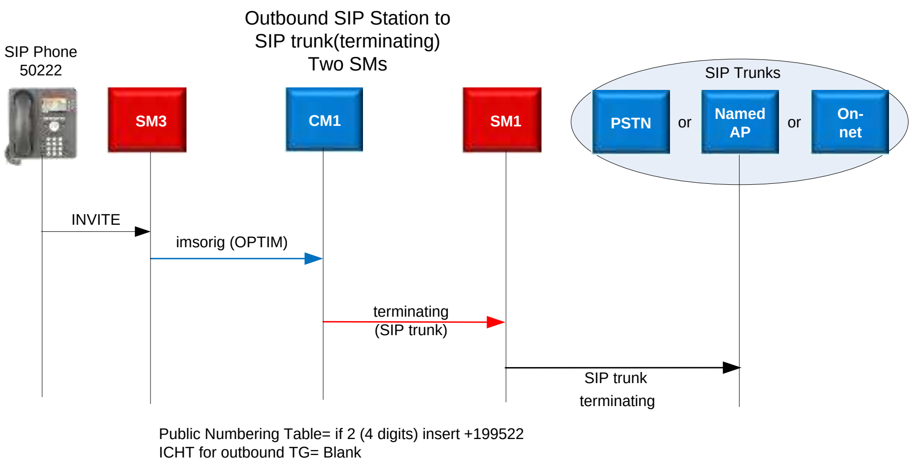


With multiple SMs in the call, the outbound call flow is dependent on how the incoming call handling treatment (ICHT) table in CM is administered for outbound SIP trunk group(s). CM audits the PAI of the station (long form of +19952250222) built from the public numbering table prior to sending the call back to SM with ICHT to determine if the short form is the SIP extension of the originating station. <u>If audit of outbound trunk ICHT results in short form matching the extension the call is returned to the originating SM as origdone call leg</u>.

In the following example extension 50222 is making an outbound SIP trunk call. The public numbering table is used to build PAI to +19952250222. The ICHT audit results short number 50222 which matches the SIP extension number. This call is sent back to the same SM as an origdone call leg, but CM internally creates a call record for the outbound trunk group to SM1 from the route pattern chosen by ARS analysis. PAI sent is the original number built in public numbering table: +19952250222.



If the ICHT on the outbound SIP trunk is blank the following call flow is the result:



In this call flow the trunk chosen in ARS/AAR analysis reflects the signaling path of the call. Unless there is a need for explicit sequencing of the outbound call leg, this is the recommended call flow for SIP station to SIP trunk calls; using terminating call leg instead of origdone call leg reflects the actual call path of the call. In either case, the OPTIM trunk call leg from CM is separate from the outbound SIP trunk in this call flow. In examples that follow for the four options, terminating call legs are preferable for outbound calls unless explicit call sequencing of applications after CM is required.

6.2 CM Evolution Server Call Processing

The following diagram summarizes the use of calling and called party number tables in Evolution Server for a SIP station to SIP station call on the same Evolution Server.

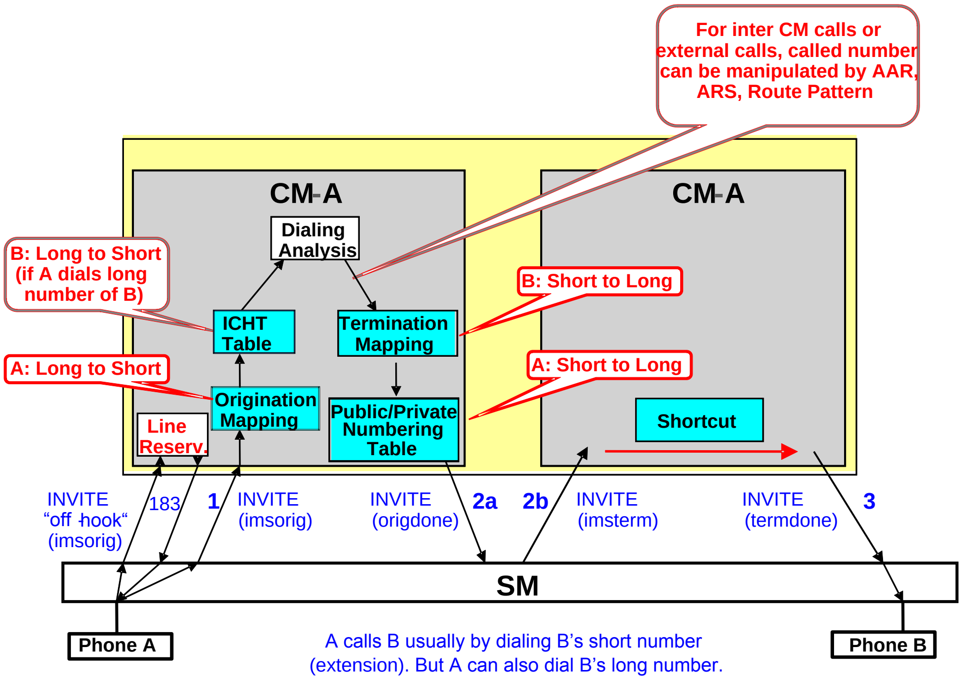


Figure 4: Evolution Server Call Processing

Note that in an ES-CM, all of the call processing, both origination and termination, is done between imsorig and origdone call legs. The ICHT table is used to convert long to short for called party number, if necessary, and the public/private tables are used to convert short to long for calling party number. This is traditional use of these tables.

6.3 CM Feature Server Call Processing

For a SIP station to station call on the same CM Feature Server (CM-FS), origination processing is done between the IMS imsorig and origdone call legs. Termination processing is done between the IMS imsterm and termdone call legs. To support termination processing, the "off-pbx-telephone station-mapping" form used for termination mapping and public/private numbering table processing is moved from the originating side of the call before origdone leg to the terminating side of the call to support the term done leg of the call.

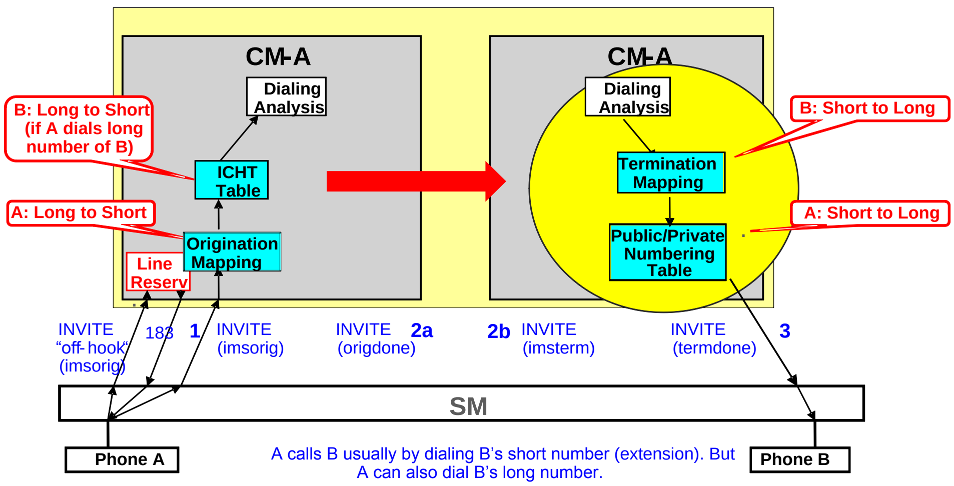


Figure 5: CM-FS Move of Term Mapping and Private/Public Tables to termdone

There now needs to be CM tables that can be used to process the flow of the call to origdone leg and from imsterm leg in the CM-FS. In the following table the public/private numbering table (to support origdone) and the ICHT table (to support imsterm) are used for these functions.

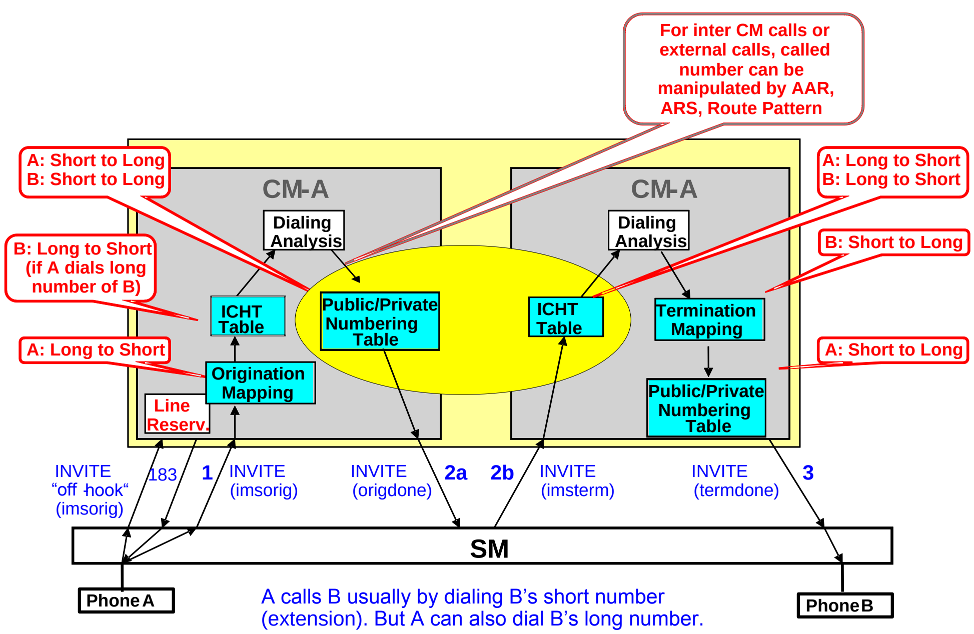


Figure 6: CM-FS use of Public/Private and ICHT

The public/private tables in this call flow execute short to long digit manipulation for BOTH calling (From and PAI) and called party (Request-URI) information and not just calling party information prior to sending the call to SM on origdone. The ICHT table executes long to short digit manipulation for BOTH calling and called party information and not just called party information on call from SM on the imsterm leg.

The public/private table executes calling party **only** short to long digit manipulation prior to the call being sent to SM on the termdone leg of the call. The ICHT table executes called party information **only** long to short digit manipulation on the imsorig leg of the call from SM.

Following is a complete diagram of digit manipulation in CM-FS for long to short and short to long digit manipulation.

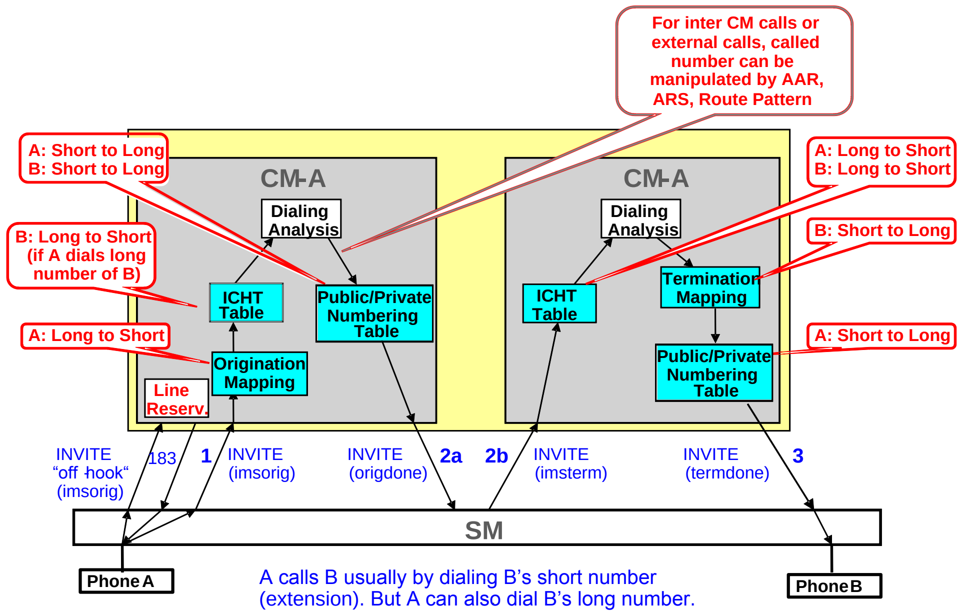


Figure 7: Summary of CM-FS Call Processing

6.4 SM SIP Entity and CM Signaling Groups for Evolution and Feature Server

6.4.1 SM SIP Entity

For all of the options described below there is a single SIP Entity from each SM to CM using FQDN to support SIP station to station and SIP station to PSTN calls.   The default TLS port 5061 interfaces to all of the signaling groups in CM:

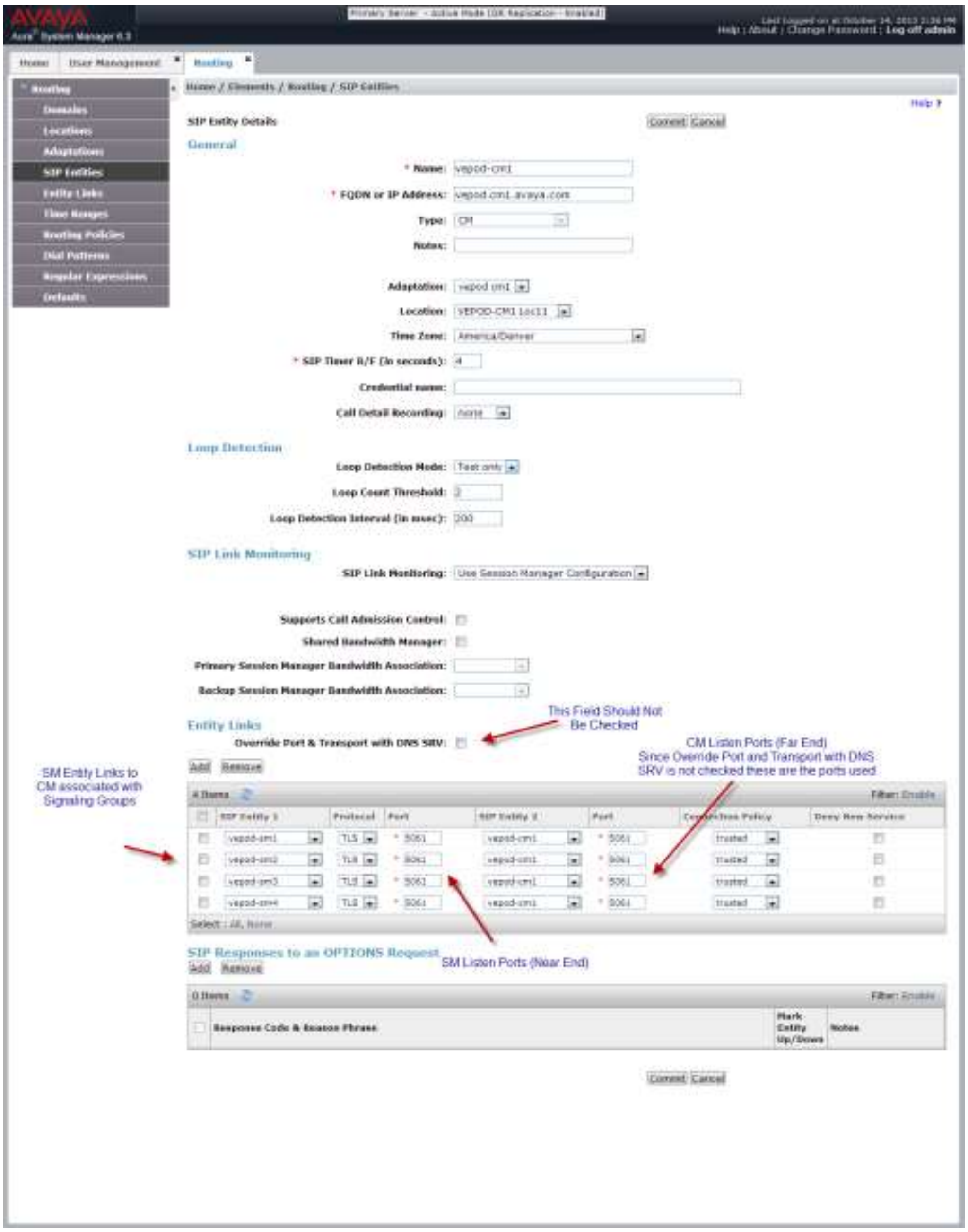


Figure 8: SM SIP Entity to CM Main and Survivable Server

The FQDN “vepod.cm1.avaya.com” specified on the CM SIP Entity is resolved to SIP Entity Links using Local Host Name Resolution (LHNR).

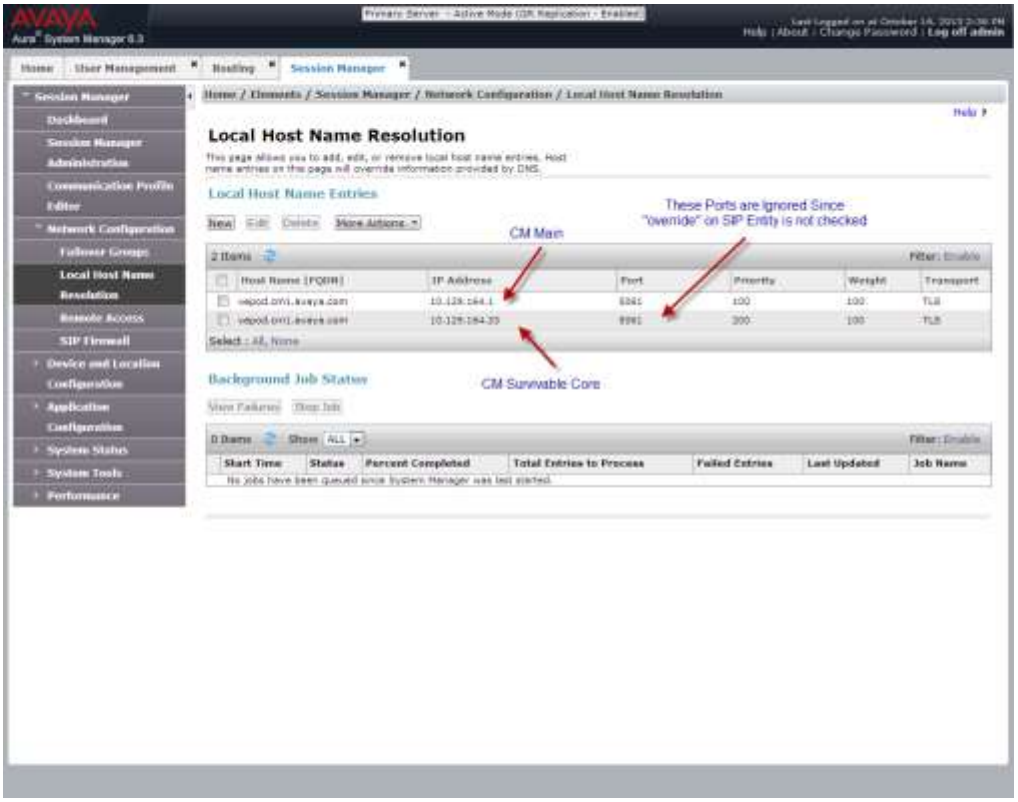


Figure 9: LHNR to Define Entity Links to CM Main and Survivable Core

6.4.2 CM Evolution Server or CM Feature Server Signaling Groups

The full call model in CM-ES and half call model in CM-FS is based on administration of the "IMS Enabled Field" on the signaling group form. In CM-ES this field is set to "n". In CM-FS, this field is set to yes. All signaling groups administered in CM to any of the SMs use Far-End Listen Port 5061 to match the single CM SIP Entity.

```
display signaling-group xxx                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910            Group Type: sip
  IMS Enabled? n or y      Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                    Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region:
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

Setting of the IMS field is what determines whether or not calls follow the full call model on CM-ES or half call model on CM-FS. In either case:

Set "Peer Detection Enabled" to y.

1. When initially adding a signaling group:
   a. "Peer Detection Enabled" is set to yes as the default
   b. "Peer Server" is set to Others and is not administrable (When connection is established to SM, then "Peer Server" will change to SM automatically)
   c. "Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?" is set to no and is administrable
   d. "Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers?" is set to yes and is administrable
   e. For SIP phones, the defaults when adding a signaling group for these fields should not be changed
2. Once the signaling group is added, the two new fields change dynamically to match the Peer Server field, as follows:
   a. If it changes to "Others" through peer detection or changed manually by changing "Peer Detection Enabled" to no, "Prepend" becomes "n" and "Remove" becomes "y" and the fields are administrable.
   b. If it changes to "SM" through peer detection or changed manually, "Prepend" changes to "y" and "Remove" changes to "n" and both fields are read only. This reflects the same behavior seen before introduction of the new fields

6.5 SIP Design Requirement Best Practices

Design requirements to address SIP and OPTIM signaling/trunk group separation under normal CM call processing loads include:

Best Practice:
Dedicate OPTIM signaling/trunk groups for SIP station call legs. These OPTIM signaling/trunk groups must be separate from SIP signaling/ trunk groups used for: inbound and outbound PSTN trunks, on-net SIP trunks, or named applications:

Best Practice:
SIP station call routing from CM (origdone and termdone) to SM must use OPTIM signaling/trunk groups that reflect the primary and secondary SMs administered for that SIP user. Extensions that use SM-x as the primary SM should route to a pattern with preferences for OPTIM trunk groups to SM-x first and OPTIM trunk groups with preferences to SM-y second. Extensions that use SM-y as the primary SM should route to a pattern with preferences for trunk groups to SM-y first and trunk groups with preferences to SM-x second.

Best Practice:
Assign users to primary and secondary SMs based primarily on blocks of extension numbers rather than communities of interest, locations or other categories. This results in optimal use of finite AAR (preferred) or ARS tables to properly route calls in CM. This requirement impacts the strategy used to coordinate primary and secondary SM controllers specified in the 46xxsetting.txt file with the Session Manager Profile specified in SysMgr administration for each user. Following are two strategies that can be used for this purpose:

1. In 46xxsettings.txt file, administer the parameter "SET SIP_CONTROLLER_LIST" to specify the same two SM controllers for all SIP users
    - Uses two SMs specified in 46xxsettings .txt file to respond to all initial registration requests from phones using "301 Permanently Moved" to communicate SM controllers specified in SIP user profile.
    - Requires the ability of the SIP phone to respond properly to "301 Permanently Moved" responses to SIP phone registration requests.
2. As an alternative, coordinate 46xxsettings.txt file parameter "SET SIP_CONTROLLER_LIST" with controllers specified in System Manager. This requires use of "groups" assigned manually on the SIP phone with groups administered in the 46xxsettings.txt file. Each group specified on the SIP phones has a group in 46xxsetting.txt file with "SET SIP_CONTROLLER_LIST" administered to match controllers specified in SM user profile. This has several challenges:
    - Requires manually setting of group on each SIP phone coordinated with group in 46xxsetting.txt file that reflects that users primary and secondary SM
    - Requires implementation of groups in 46xxsetting.txt file to reflect SMs administered in SM user profiles
    - Does not address third party SIP devices such as E129 SIP phone that does not use 46xxsettings.txt file and groups

Best Practice:
Segregation of OPTIM and SIP trunks based on SIP trunk type (‘public-ntwrk’ or ‘tie’) is necessary for CM to execute “Call Processing Overload Mitigation” software administered in “system-parameters features”.

A proactive overload mitigation strategy that coordinates CM, SM and SBC is critical to meeting customer concerns about events that can lead to a flood of SIP call center calls and potential CM call processing overload. CM call processing overload mitigation software is designed to isolate stations or trunks first in response to processor overload which is defined as processor load greater than 92.5% for 20 consecutive seconds. In the event of processor overload, CM sheds traffic by responding to SIP INVITE and OPTION messages over “tie” or “public-ntwrk’ trunks with “**503 Service Unavailable”** with a “retry after 30 seconds”.

Prior to FP4, CM activated “Call Processing Overload Mitigation” based on all trunks and did not mitigate based on ‘public-ntwrk’. “Call Processing Overload Mitigation” in FP4 now mitigates based on ‘public-trunks-first’ or ‘all-trunks-first’ by administering “Call Processing Overload Mitigation” to “public-trunks-first”:

```
change system-parameters features                               Page   3 of  20
                    FEATURE-RELATED SYSTEM PARAMETERS
TTI/PSA PARAMETERS

  WARNING!  SEE USER DOCUMENTATION BEFORE CHANGING TTI STATE

        Terminal Translation Initialization (TTI) Enabled? n


               Customer Telephone Activation(CTA) Enabled? n

                     Hot Desking Enhancement Station Lock? n

EMU PARAMETERS
           EMU Inactivity Interval for Deactivation(hours):

CALL PROCESSING OVERLOAD MITIGATION
           Restrict Calls: public-trunks-first
```

The example below uses the signaling group/trunk group numbering convention to separate SIP from OPTIM trunks:

1. Signaling/trunk group 1xy where:
    a. “1” represents inbound SIP signaling/trunk group
    b. “x” represents the SM number
    c. “y” represents the signaling/trunk group
    d. Signaling/trunk group 110 would be an inbound trunk group to SM1, and this is the first trunk of this group. If multiple signaling/trunk groups to SM1 are required because of traffic requirements, additional signaling/trunk groups could be added (111, 112, 113, etc.)
    e. Service Type: public-ntwrk

2. Signaling/trunk group 2xy
    a. “9” represents outbound SIP signaling/trunk group
    b. “x” represents the SM number

c. "y" represents the signaling group
d. Signaling/trunk group 910 would be an OPTIM trunk group to SM1, and this is the first trunk of this group. If multiple signaling/trunk groups to SM1 are required because of traffic requirements, additional signaling/trunk groups could be added (911, 912, 913, etc.)
e. Service Type: public-ntwrk

3. Signaling/trunk group 9xy
   a. "9" represents OPTIM SIP signaling/trunk group
   b. "x" represents the SM number
   c. "y" represents the signaling group
   d. Signaling/trunk group 910 would be an OPTIM trunk group to SM1, and this is the first trunk of this group. If multiple signaling/trunk groups to SM1 are required because of traffic requirements, additional signaling/trunk groups could be added (911, 912, 913, etc.)
   e. Service Type: tie

6.6 System Manager and CM Sequence Application Administration

Following screen shows CM-ES or CM-FS as an application that can be sequenced

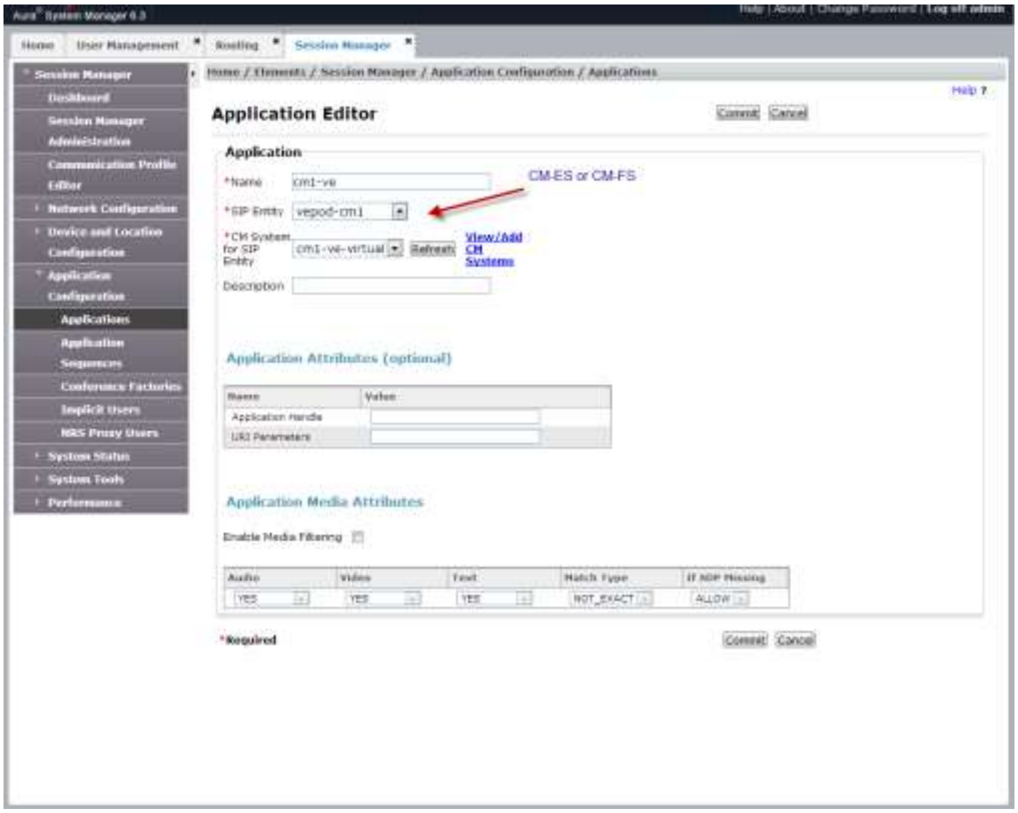


Figure 10: CM-ES or CM-FS Defined as an Application

The following screen defines an application sequence that includes only CM-ES or CM-FS. Application sequencing is specified in System Manger for each SIP user in Home/Users/UserManagement/Manage Users path in Web interface. This application sequence is used for origination and termination sequencing to SIP phone users. If CM is the only application in the sequence, CM-ES should be used. Multiple applications included in the origination or termination sequence may require CM-FS to support the call flows (see Section 1).

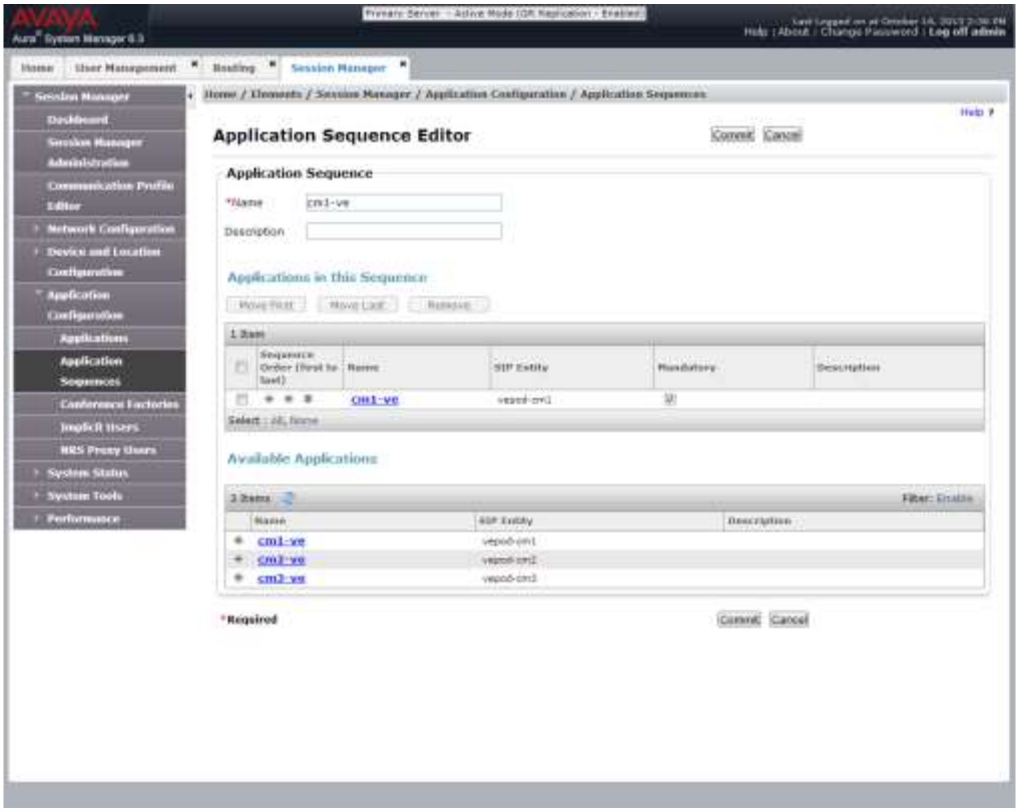


Figure 11: Application Sequence for CM-ES or CM-FS

6.7 System Manager and CM Administration for Call Options

All four call options specified in Sections 7-10 have a System Manager component that describes handles and domains used by the SIP telephones for login as well as authentication in Communication Manager. Recall that a proper SIP handle must be chosen that is unique across the enterprise served by SM. An E.164 handle as well as one other handle that is used by the SIP user attempting to register to SM and subsequently used to Subscribe to CM must be administered. Proper administration of domain and login password is necessary as well as the format of the associated extension number.

Administration for CM-ES and CM-FS is the same with the exception of the IMS Enabled field on the signaling group form outlined in Section 6.4. Any additional or different administration between CM-ES and CM-FS is contained within each of the respective section for each of the options. It is how a CM-ES or CM-FS uses this administration is what is different and documented in this paper.

In the following discussion trunks used for SIP phone call control will be referred to as OPTIM trunks and all other trunks will be referred to as SIP trunks.

7 Option One: Extensions Based on E.164 Numbering Plan

Option one CM extensions are based on E.164 numbering plan without the + sign. This is a typical configuration of a consolidated CM system with many SIP endpoints. The handles used are E.164 with the + sign and another handle without the + sign. In this case the handle used to login to the SIP phone is the same as the extension number. Following is an example based on North America:

- Handles
  - <u>Avaya E.164</u> +19952250022
  - <u>Avaya SIP</u>   19952250022 (Public Long /Preferred Handle)
- Extension Number
  - 19952250022 (Public Long)

7.1 SIP Station to SIP Station Call Flow-Option One

7.1.1 Evolution Server

Following is an Option One diagram of a CM-ES SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example).

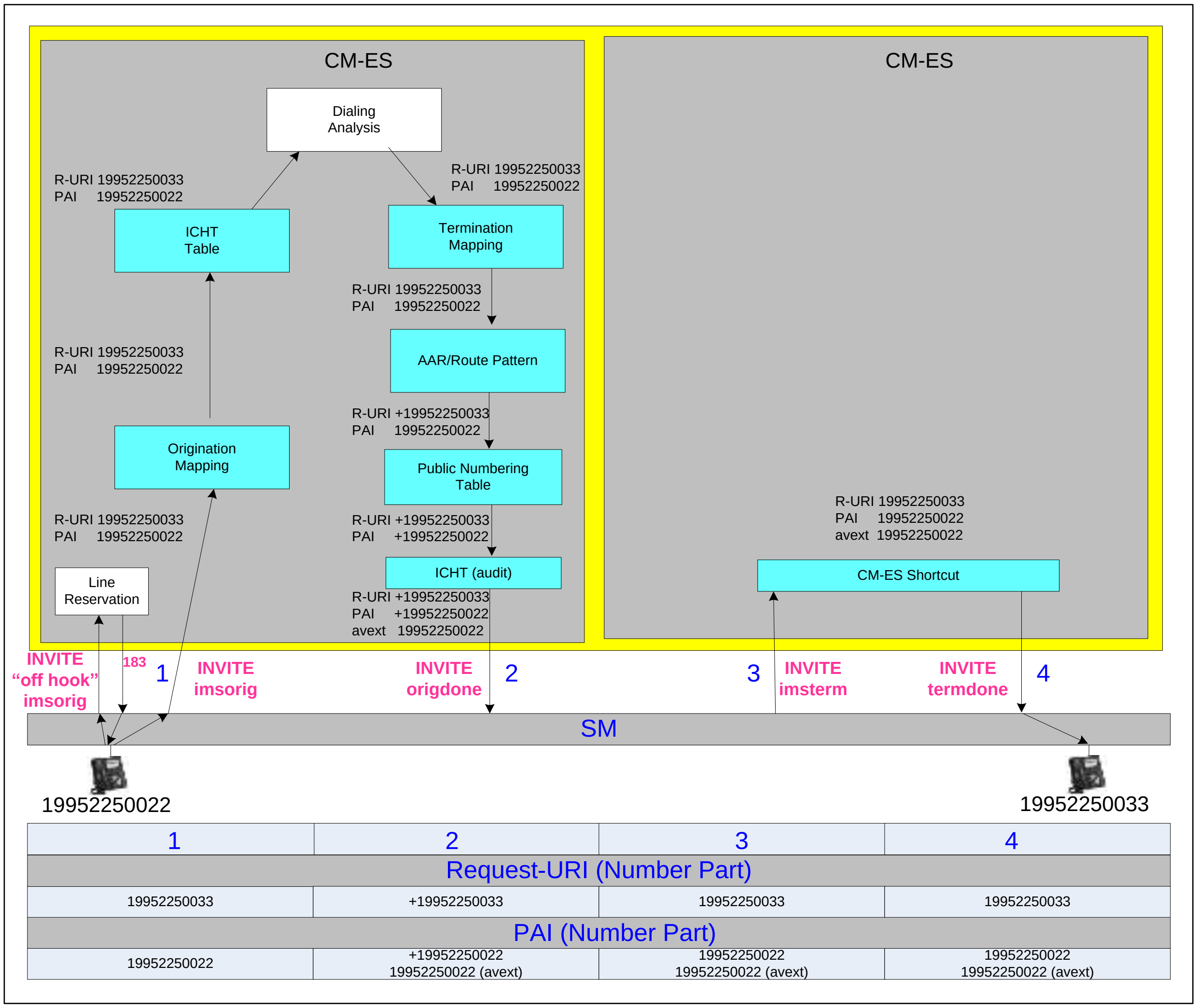


Figure 12: Evolution Server Call Flow-Option One

In this example, 19952250022 is the calling SIP station and 19952250033 is the called SIP station. SIP station 19952250022 logs in as 19952250022 which matches the Avaya SIP handle in SM (same is true for 19952250033 login). In the SIP INVITE message from the originating SIP phone, the SIP handle 19952250022 appears in the Contact header and the dialed digits 19952250033 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-ES

    a. SM does a lookup of 19952250022 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 19952250022 since it is the preferred handle specified in SysMgr.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES prior to origdone call leg to SM
    a. The call flows through origination mapping for station 19952250022 and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
    b. The call next flows through the ICHT table on SIP telephone trunk group (TG910/930) to change R-URI from long to short form, but since there is no match there is no change to the R-URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    d. CM converts extension number of terminating SIP extension 19952250033 to phone number 19952250033 using term mapping. In this case both numbers are using the public long form.
    e. CM AAR routes the call to the proper route pattern based on <u>terminating</u> phone number 19952250033.
    f. The public-unknown numbering table adapts the calling party information (PAI) from SIP station extension number 19952250022 (public long) to E.164 format by adding the "+".
    g. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 19952250022 (public long).
        i. ICHT must have an entry that deletes the +.
        ii. There is now a match with the originating SIP station 19952250022 after "+" is deleted, CM appends avext parameter with extension (public long number) to the E.164 PAI header and sends both forms back to SM as origdone rather than terminating.
        iii. If there is no match CM sends the call back to SM as "terminating" with E.164 PAI format with no avext parameter.
    h. A "+" is also added to R-URI since there is a "p" in the route pattern insert column.

3. Processing by SM prior to imsterm call leg to CM-ES
    a. SM now looks at Request-URI of +19952250033
    b. SM recognizes this as a handle associated with extension 19952250033 (note: SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
    c. SM forwards this call back to CM based on termination sequence administration in System Manager.
    d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 19952250022 (PAI) and 19952250033 (R-URI) and neither contain a "+".

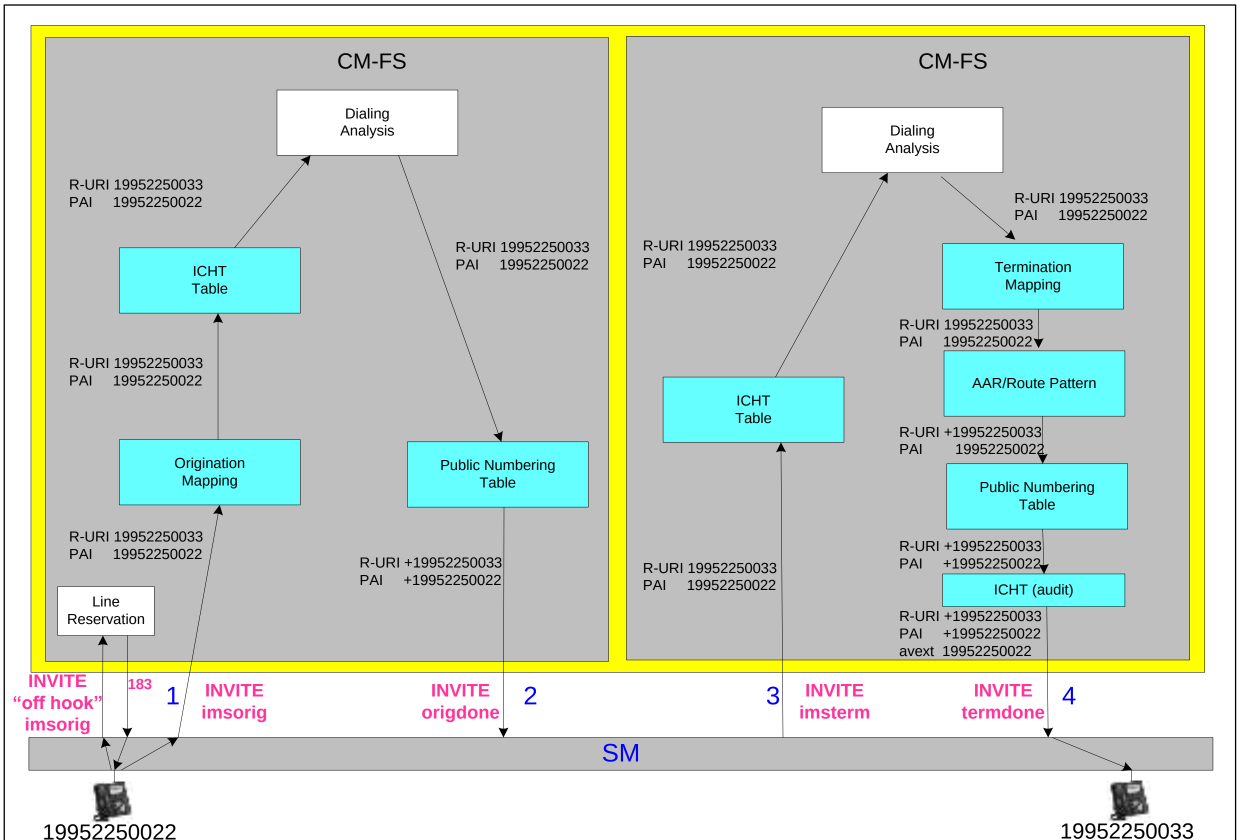
4. Processing by CM-ES prior to termdone back to SM
    a. CM-ES does "shortcut" of imsterm and sends the SIP invite back to SM in termdone with no further processing of the call.
    b. Both PAI and R-URI are 11-digits in length without the "+".

5. SM now matches the R-URI with the called user profile and completes the call to the phone

6. SIP Phone displays avext (extension number) if available, otherwise PAI is displayed.

7.1.2 Feature Server

Following is an Option One diagram of a CM-FS SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):



1	2	3	4
Request-URI (Number Part)			
19952250033	+19952250033	19952250033	+19952250033
PAI (Number Part)			
19952250022	+19952250022	19952250022	+19952250022 19952250022 (avext)

Figure 13: CM-FS Call Flow-Option 1

In this example, 19952250022 is the calling SIP station and 19952250033 is the called SIP station. SIP station 19952250022 logs in as 19952250022 which matches the SIP handle in SM (same is true for 19952250033 login). In the SIP INVITE message from the originating SIP phone, the SIP handle 19952250022 appears in the Contact header and the dialed digits 19952250033 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
   a. SM does a lookup of 19952250022 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.

   b. The PAI header in imsorig contains 19952250022 since it is the preferred handle specified.
   c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-FS prior to origdone call leg to SM
   a. The call flows through origination mapping for station 19952250022 and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
   b. The call next flows through the ICHT table on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
   c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
   d. CM AAR routes the call to the proper route based on originating phone number 19952250022.
   e. The public-unknown numbering table adapts the calling party information (PAI) AND called party information (R-URI) from SIP station extension public long to E.164 format by adding the "+".
   f. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 19952250022 (public long).
      i. ICHT must have an entry that deletes the +.
      ii. There is now a match with the originating SIP station 19952250022 after "+" is deleted, CM sends the call back to SM as origdone call leg rather than terminating.
      iii. If there is no match CM sends the call back to SM as "terminating" with E.164 PAI format.
   g. NOTE: the "p" in the route pattern insert column is ignored in CM-FS prior to origdone

3. Processing by SM prior to imsterm call leg to CM-FS is the same as for CM-ES
   a. SM now looks at Request-URI of +19952250033
   b. SM recognizes this as a handle associated with extension 19952250033 (note: SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
   c. SM forwards this call back to CM based on termination sequence administration in System Manager.
   d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 19952250022 (PAI) and 19952250033 (R-URI) and neither contain a "+".

4. Processing by CM-FS prior to termdone back to SM
   a. CM-FS looks for match on R-URI and PAI in ICHT.
      i. This is a special case where CM-FS attempts to do long to short processing on both R-URI and PAI.
      ii. Both PAI and R-URI are 11-digits in length (long form of the number) without the "+" and do not match anything in the ICHT table
   b. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
   c. CM converts extension number of terminating SIP station 19952250033 from extension number to phone number using term mapping. In this case both extension number and telephone number are the same public long format.

d. CM AAR routes the call to the proper route pattern based on <u>terminating</u> phone number 19952250033.
e. The public-unknown numbering table adapts the calling party information (PAI) from SIP station extension number 19952250022 (public long) to E.164 format by adding the "+".
f. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 19952250022 (public long).
    i. ICHT must have an entry that deletes the +.
    ii. There is now a match with the originating SIP station 19952250022 after "+" is deleted, CM appends avext parameter with extension (public long number) to the E.164 PAI header and sends both forms back to SM as termdone call leg.
    iii. If there is no match CM sends the call back to SM with E.164 PAI format as termdone call leg with no avext parameter.
g. A "+" is added to the R-URI since there is a "p" in the route pattern insert column.
h. Now both PAI and R-URI are E.164 with the "+" on both.

5. SM now matches the R-URI with the called user profile and sends the call to the phone

6. SIP Phone displays avext (extension number) if available, otherwise PAI is displayed.

7.1.3 System Manager, CM-ES and CM-FS Administration-Option One

For SIP users in System Manager User Profile for both Evolution and Feature Server:

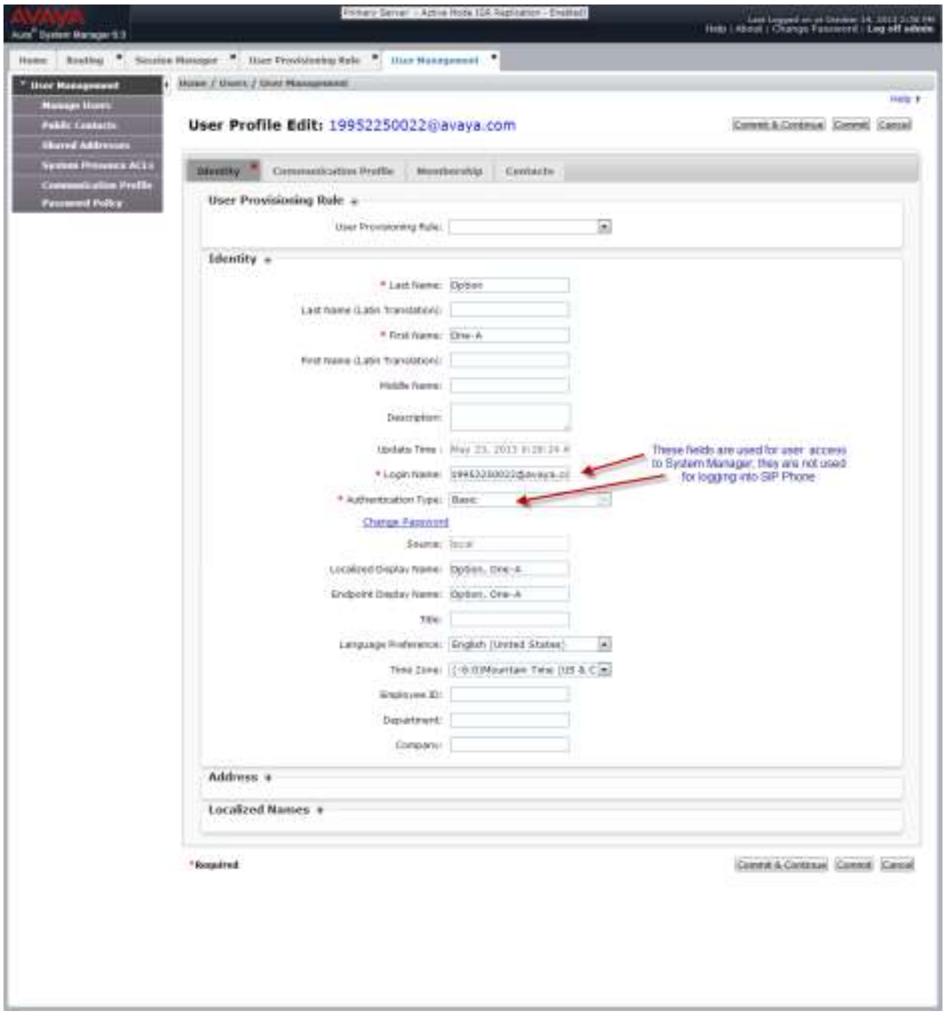


Figure 14: System Manager User Profile Identity-Option One

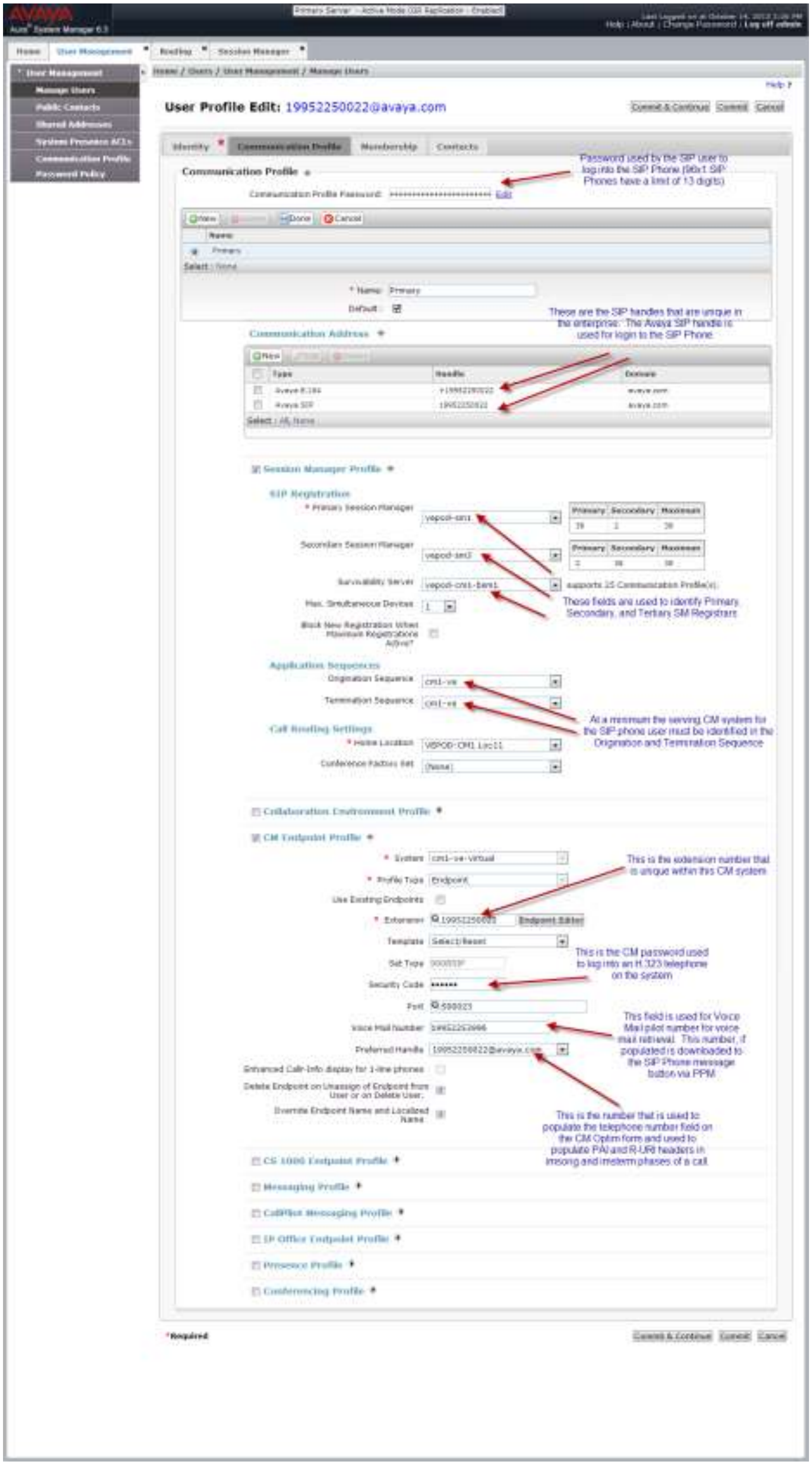


Figure 15: System Manager User Communication Profile-Option One

System Manager automatically populates CM station as well as off-pbx-telephone station-mapping forms with the following translations using the selected template for both Evolution and Feature Server:

```
display station 19952250022                                  Page   1 of   6
                                     STATION

Extension: 19952250022                 Lock Messages? n              BCC: 0
     Type: 9608SIP                     Security Code: 123456          TN: 1
     Port: S00023                    Coverage Path 1:                COR: 1
     Name: Option, One-A             Coverage Path 2:                COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                Location:                Time of Day Lock Table:
              Loss Group: 19
                                               Message Lamp Ext: 19952250022

        Display Language: english                Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                        IP SoftPhone? n

                                                       IP Video? n
```

```
display station 19952250022                                  Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                             Headset? n
       Jack:                                             Speaker? n
      Cable:                                            Mounting: d
      Floor:                                         Cord Length: 0
   Building:                                           Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

```
display station 19952250022                                  Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS

         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

Note: the default routing for this station is aar and that the phone will get three call appearances.

System Manager also populates the off-pbx telephone station mapping form with the application type OPS. System Manager uses the CM Endpoint Profile “Preferred Handle” in the User Communication Profile to populate the “phone number” field in CM.

```
display off-pbx-telephone station-mapping 19952250022          Page   1 of    3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
 Station        Application Dial    CC   Phone Number    Trunk        Config  Dual
 Extension                  Prefix                       Selection    Set     Mode
 1995-225-0022     OPS          -      19952250022      aar         1
 1995-225-0033     OPS          -      19952250033      aar         1
```

The off-pbx-telephone station-mapping form is used for long to short digit manipulation of the calling station on the origination side of the call (Phone Number to Station Extension). The off-pbx-telephone station-mapping form is used for short to long digit manipulation of the called station on the termination side of the call (Station Extension to Phone Number).

Prior to administering the System Manager User Profile, the following **minimum** administration must be done in Communication Manager (Note that administration that follows is the same for Evolution and Feature Server):

The dialplan analysis form has the following administration:

- The dialed string 1, 11-digits in length to support 11-digit extensions (1995-225-0022 & 1995-225-0033).
- The dialed string *8 for Automatic Alternate Routing (AAR) and *9 for Automatic Route Selection (ARS). The AAR and ARS feature access codes must be defined.
- The dial string *, 4-digits in length to accommodate SIP trunk dial access codes.

```
display dialplan analysis                                      Page   1 of   12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all         Percent Full: 5

    Dialed    Total   Call      Dialed    Total   Call      Dialed    Total   Call
    String    Length Type       String    Length Type       String    Length Type
   1            11   ext
   *8           2    fac
   *9           2    fac
   *            4    dac
```

Following is the minimum translations for system features:

```
display feature-access-codes                                   Page   1 of   11
                               FEATURE ACCESS CODE (FAC)
      Auto Alternate Routing (AAR) Access Code: *8
    Auto Route Selection (ARS) - Access Code 1: *9    Access Code 2:
```

The AAR/ARS codes do not have to be what is shown here but they do need to be administered.

A dedicated signaling group(s) needs to be set up for use by the SIP telephones to the primary SM and secondary SM. In the example there are actually four SMs: vepod-sm1 and vepod-sm2 in Data Center One and vepod-sm3 and vepod-sm4 in Data Center Two. Assume that for SIP telephones in this example that vepod-sm1 is the primary SM and vepod-sm3 is the secondary SM. The Signaling group to vepod-sm1 is 901, and the signaling group to vepod-sm3 is 930. Administration for Signaling group 910 to vepod-sm1 is shown here:

```
display signaling-group 910                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910             Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                   Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region:
                                     Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                               Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                       IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

In this example IMS Enabled field is set to n. This is the proper setting for a CM-ES. Setting of the IMS field is what determines whether or not calls follow the full call model on CM-ES or half call model on CM-FS. It is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

For CM-FS the signaling group needs to be set up for use by the SIP telephones to each SM with IMS Enabled set to y (in the example there are two SMs), all other administration is the same as CM-ES.

```
display signaling-group 910                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910             Group Type: sip
  IMS Enabled? y         Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                   Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region:
                                     Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                               Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                       IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-FS will select this signaling group.

Following is administration for SIP trunk group associated with the SIP signaling group 910.

```
display trunk-group 910                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 910                  Group Type: sip           CDR Reports: y
  Group Name: OPTIM SM1                   COR: 1        TN: 1        TAC: *910
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? N
                                             Member Assignment Method: auto
                                                      Signaling Group: 910
                                                    Number of Members: 15
```

```
display trunk-group 910                                         Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? Y

                     Numbering Format: public
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n                     SIP ANAT Supported? n
```

As discussed in Section 6.1.4, the “Numbering Format” field on page 3 of the form is used to determine whether to use the public or private numbering table for various call flows in Evolution and Feature Server. In this example the numbering format used is public since all calls should display E.164 format for calling (PAI) and called number (R-URI) information.

In the Evolution Server, station to station calls use the routing associated with the terminating OPS phone number 19952250033 for origdone call leg and shortcuts the term side of the call. In the Feature Server, station to station calls use routing associated with the originating OPS phone number (19952250022) for the origdone call leg and terminating OPS station phone number (19952250033) on the call for termdone call leg.

Routing for 19952250022 and 19952250033 is based on AAR and uses route pattern 910.

```
display aar analysis 0                                                    Page   1

                           AAR DIGIT ANALYSIS REPORT

                            Location:  all

               Dialed            Total         Route    Call      Node
               String          Min    Max     Pattern   Type      Number

         1995225                11     11      910       aar
```

NOTE: insure that the AAR Digit Conversion Table is <u>not</u> converting the Matching Pattern “1” and routing the call to ARS (default entry). Need to delete the entry in red if it is in the system as default.

```
display aar digit-conversion 0                                    Page   1 of   2
                       AAR DIGIT CONVERSION TABLE
                                 Location: all                 Percent Full: 0
  Matching Pattern      Min   Max   Del   Replacement String   Net   Conv ANI Req
  0                     1     28    0                          ars    y        n
  x11                   3     3     0                          ars    y        n
  1                     4     28    0                          ars             n
```

```
display route-pattern 910                                         Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP SM1 and SM3
                             SCCAN? n      Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 910  0                        p                                      n   user
 2: 930  0                        p                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                           next
 2: y y y y y n  n            rest                                           none
```

In this example Look Ahead Routing (LAR) has been implemented in the event that CM gets failure messages on vepod-sm1 it can retry on vepod-sm3 using trunk group 930. The preferences in this route pattern match the primary and secondary SM specified for the SIP users in System Manager based on the best practice cited in Section 6.5. Since public numbering format is being used on the trunks, the numbering format is not applicable and is left blank. In this example the public-unknown-numbering table is based on the CM algorithm described in section 6.1.4.

In these examples, SM dial patterns are based on E.164 format, therefore a + is inserted on R-URI by administering a “p” on the route pattern. The sending of a “+” for R-URI is not mandatory since there is a handle for SIP stations with and without the “+”. With this translation a + will appear on R-URI in origdone, but it will be deleted on imsterm by SM since SM uses the Preferred Handle of 19952250333. The + is included in this case so that all route patterns are consistent.

Following is the administration for the public-unknown-numbering table.

```
display public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext            Trk        CPN         CPN
Len Code           Grp(s)     Prefix      Len
                                                Total Administered: 1
 11 1                                     11       Maximum Entries: 9999

                                                Note: If an entry applies to
                                                a SIP connection to Avaya
                                                Aura(R) SM,
                                                the resulting number must
                                                be a complete E.164 number.

                                                Communication Manager
                                                automatically inserts
                                                a '+' digit in this case.
```

CM-ES and CM-FS both use this translation for short to long form administration based on call flows described in Section 7.1.1 and 7.1.2

Following is administration for the incoming call handling treatment for trunk group 910 (and 930)

```
display inc-call-handling-trmt trunk-group 910                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
  tie             12 +               1[1]
```

CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 7.1.1 and 7.1.2.

[1] This entry is also used to convert from long to short form for Subscribe messages rather than using off-pbx-telephone station-mapping (see 6.1.5).

7.2 SIP Station to Outbound SIP PSTN Call Flow-Option One

7.2.1 CM-ES and CM-FS

Following is the call flow for number dialed to the PSTN from extension 19952250022 to 1720-356-4567:

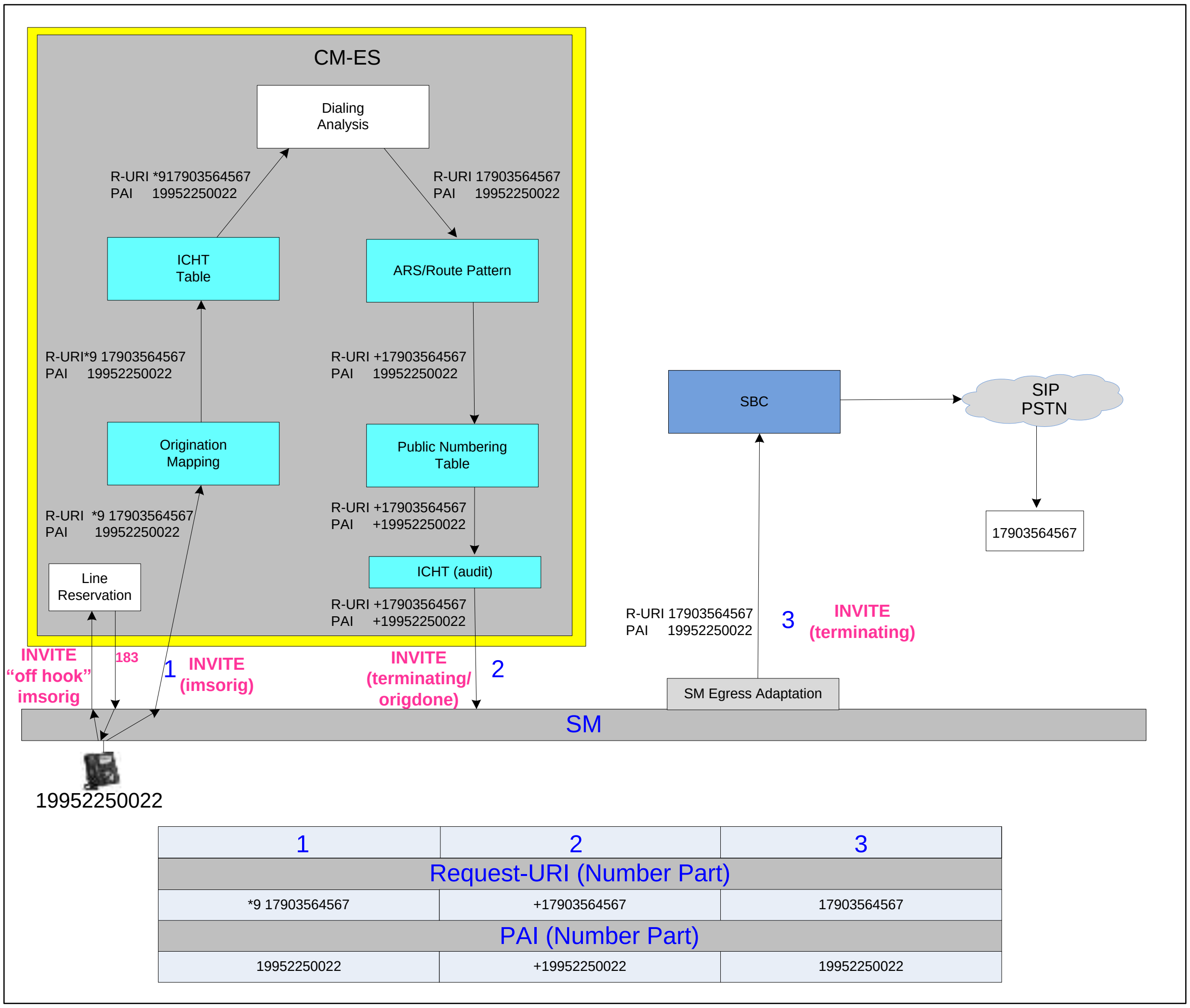


Figure 16: Outbound PSTN Call Flow in Feature or CM-ES Server-Option One

End user dials the ARS access code *9 and the PSTN number

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
    a. SM does a lookup of 19952250222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 19952250022 since it is the preferred handle specified.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES and CM-FS prior to origdone call leg to SM
    a. The call flows through origination mapping for station 19952250022 and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
    b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. Call is routed to proper route pattern
    e. The public-unknown numbering table is now used to adapt the calling party information from public long number to E.164 with the "+" prior to sending the call to SM in the origdone leg of the call.
    f. CM uses ICHT on SIP PSTN trunk group (TG110/130) to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 19952250022 (public long).
        i. If ICHT has an entry that deletes the + there is now a match with the originating SIP station 19952250022 after "+" is deleted
            1. CM uses origdone call leg with PAI in E.164 format.
            2. Explicit sequencing of origination applications after CM requires origdone call processing.
            3. CM <u>always</u> sends signaling for origdone call leg back to the same SM that initiated imsorig call processing regardless of what is specified in AAR/ARS routing;
            4. If AAR/ARS routing for origdone is different than the SM used for imsorig, CM call processing still shows use of signaling group/trunk group specified in ARS/AAR.
            5. Since this is not a station to station call, avext is not appended to PAI header.
        ii. If ICHT does NOT have an entry that deletes the + there is no match with the originating station 19952250022
            1. CM uses terminating call leg with PAI in E.164 format
            2. Implicit sequencing, including Collaboration Environment is supported since CE does not require origdone call processing.
            3. <u>Terminating call legs unlike origdone call legs do not need to return to the same SM that initiated imsorig call processing.</u>
            4. In this case, CM sends the terminating call leg to SM specified in ARS routing as "terminating" even if it is different than the SM used for imsorig.
            5. CM call processing shows trunk group usage to SM chosen by ARS.
    g. The "p" in the route pattern inserts the "+" on the R-URI
    h. CM sends PAI and R-URI numbers to SM in E.164 format[2]
3. Processing by SM prior to terminating to SBC
    a. SM determines routing policy
    b. Applies egress adaptation to SBC based on SBC and PSTN requirements

---

[2] Additional considerations need to be applied to international calls. In North America, the international prefix dialed is "011" (many other parts of the world it is "00"). These digits can be deleted on an international route pattern and the "+" inserted or the call can be sent to SM with the international prefix. In this case, SM would have an adaptation to delete the international prefix and insert "+" for analysis and routing. Here is a case where an SM ingress adaptation is being used in SM that does not impact SIP telephone call processing

7.2.2 CM-ES and CM-FS Administration

Separate signaling group(s) need to be set up for access to the PSTN SIP trunks from CM to each SM (in the example assume two SMs). Administration for signaling group 110 to vepod-sm1 is shown here (also need signaling group 130 to vepod-sm3):

```
display signaling-group 110                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 110             Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 241
                                    Far-end Secondary Node Name:
Far-end Domain: sbccore.avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from PSTN trunks (imsterm) have a domain of sbccore.avaya.com and CM will select this signaling group.

Following is a sample trunk used for PSTN access via SM:

```
display trunk-group 110                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 110                  Group Type: sip            CDR Reports: y
  Group Name: SIP PSTN SM1                COR: 1        TN: 1        TAC: *110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 110
                                                    Number of Members: 15
```

```
display trunk-group 110                                        Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y

 DSN Term? n                     SIP ANAT Supported? n
```

Since “public” is specified in the “Numbering Format” field all calls to this trunk group will use the “public-unknown numbering” table

Following is ARS Analysis table for access to SIP PSTN trunks:

```
display ars analysis 0                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    011                   10   18    110       intl         n
    1                     11   11    110       natl         n
    911                   3    3     911       emer         n
```

Assume that 11 digit North America numbers are being dial and are then routed using route pattern 110 as specified in ARS Analysis:

```
display route-pattern 110                                      Page   1 of   3
                    Pattern Number: 110    Pattern Name: SBC DC1
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 110  0                        p                                   n   user
 2: 130  0                        p                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                     next
 2: y y y y y n  n            rest                                     none
```

Route pattern 110 inserts the + to the Request-URI based on the “p” entered in the inserted digits field so that SM can route on E.164 number. In North America 1+10 digits is in E.164 format if the + is appended to the dial string. In route pattern 110 the numbering format field is not applicable since these are public trunks. Based on the algorithm used for public trunks, this call will use the public-unknown-numbering table.

The public-unknown-numbering table administration for SIP station to station calls can be used for PSTN calls as well.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 and 130 the ICHT for the PSTN trunk group 110/130 is to insure that the call is sent as origdone rather than “terminating” on PSTN calls:

```
display inc-call-handling-trmt trunk-group 910                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
  tie             12 +               1
```

CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 7.2.1.

7.3 Inbound SIP PSTN to SIP Station Call Flow-Option One

7.3.1 CM-ES and CM-FS

Following is the call flow for number dialed from the PSTN to extension 19952250022:

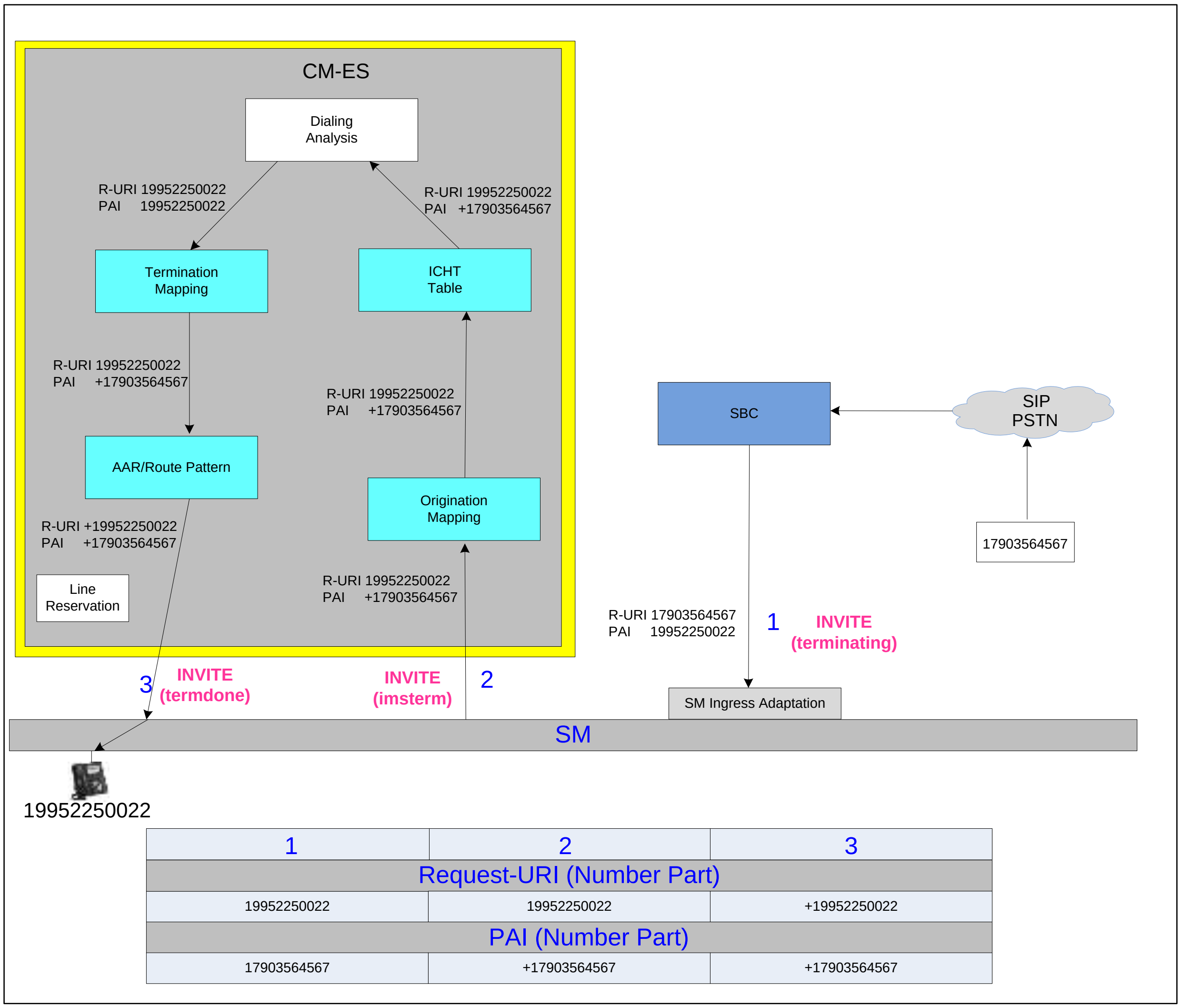


Figure 17: Inbound PSTN Call Flow in Feature or CM-ES Server-Option One

PSTN user 17203564567 dials PSTN number 19952250022

1. Processing by SM on terminating call leg from SBC
    a. SBC sends call to SM using terminating phase.
    b. SM adapts R-URI to E.164 format +19952250022
    c. SM adapts PAI header to E.164 format +17203564567
2. Processing by SM prior to imsterm call leg to CM
    a. SM does a lookup of R-URI from SBC of +19952250222 and sees that it is a registered user and forwards the call to CM based on termination sequence administration in System Manger using preferred handle 19952250222.

    b. SM sends PAI to CM as E.164

3. Processing by CM prior to termdone call leg to SM
    a. The call flows through origination mapping and there is no match with PAI +17203564567 and sees no match.
    b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG110/130) to convert R-URI from long to short form and since there is no short form there is no change.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. The call flows through termination mapping for station 19952250022 for short to long form processing and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the R-URI.
    e. CM AAR routes the call to the proper route pattern based on terminating phone number 19952250022
    f. The "p" in the route pattern inserts the "+" on the R-URI
    g. CM sends PAI and R-URI numbers to SM in E.164 format
4. SM now matches the R-URI with the called user profile and sends the call to the phone
5. SIP Phone displays E.164 number +17203564567

8 Option Two: Extensions Based on the Private Long Number

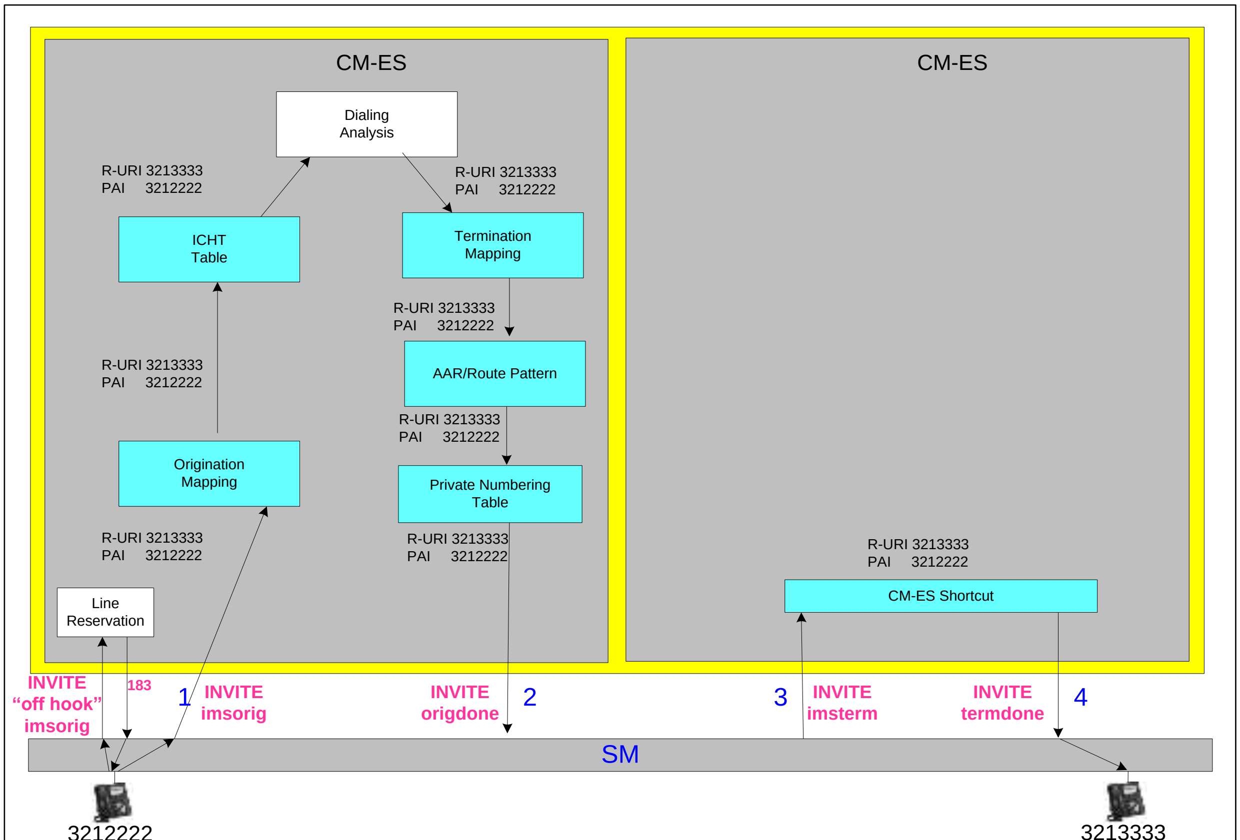
Option two extensions are based on the private long number. This is another typical configuration of a consolidated CM system with many SIP endpoints. The handles used are E.164 with the + sign and another handle that reflects enterprise canonical numbering plan. In this case the handle used to login to the SIP phone is the same as the extension number. Following is an example based on North America (Note, in this example, only the last four digits match the E.164 number):

- Handles
  - <u>Avaya E.164</u> +19952252222
  - <u>Avaya SIP</u>   3212222 (Private Long/Preferred Handle)
- Extension Number
  - 3212222 (Private Long)

8.1 SIP Station to Station Call Flow-Option Two

8.1.1 Evolution Server

Following is an Option Two diagram of a CM-ES SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):



1	2	3	4
Request-URI (Number Part)			
3213333	3213333	3213333	3213333
PAI (Number Part)			
3212222	3212222	3212222	3212222

Figure 18: Evolution Server Call Flow-Option Two

In this example, 3212222 is the calling SIP station and 3213333 is the called SIP station. SIP station 3212222 logs in as 3212222 which matches the SIP handle in SM (same is true for 3213333 login). In the SIP INVITE message from the originating SIP phone the SIP handle 3212222 appears in the Contact header and the dialed digits 3213333 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-ES

    a. SM does a lookup of 3212222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 3212222 since it is the preferred handle specified in SysMgr.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES prior to origdone call leg to SM
    a. The call flows through origination mapping for station 3212222 and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
    b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) to change R-URI from long to short form, but since there is no match there is no change to the R-URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    d. CM-ES converts extension number of terminating SIP extension 3213333 to associated phone number using term mapping. In this case both numbers are using the private long form.
    e. CM-ES routes the origdone call leg using AAR routing based on <u>terminating</u> phone number 3213333.
    f. The private numbering table adapts the calling party information (PAI) from short to long formats. SIP station extension number 321222 is already in private long format so PAI does not change.
    g. Since PAI private long form is used for extension and PAI, CM-ES
        i. does not need to use ICHT to determine the station that matches private long form of PAI
        ii. does not append avext to PAI
        iii. sends call back to SM as origdone rather than terminating

3. Processing by SM prior to imsterm call leg to CM-ES
    a. SM now looks at Request-URI of 3213333
    b. SM recognizes this as a handle associated with extension 3213333 (note: SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
    c. SM forwards this call back to CM based on termination sequence administration in System Manager.
    d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 3212222 (PAI) and 3213333 (R-URI).

4. Processing by CM-ES prior to termdone back to SM
    a. CM-ES does "shortcut" of imsterm and sends the SIP invite back to SM in termdone with no further processing of the call.
    b. Both PAI and R-URI are 7-digits in length.

5. SM now matches the R-URI with the called user profile and completes the call to the phone

6. SIP phone displays PAI

8.1.2 Feature Server

Following is an Option Two diagram of a CM-FS SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):

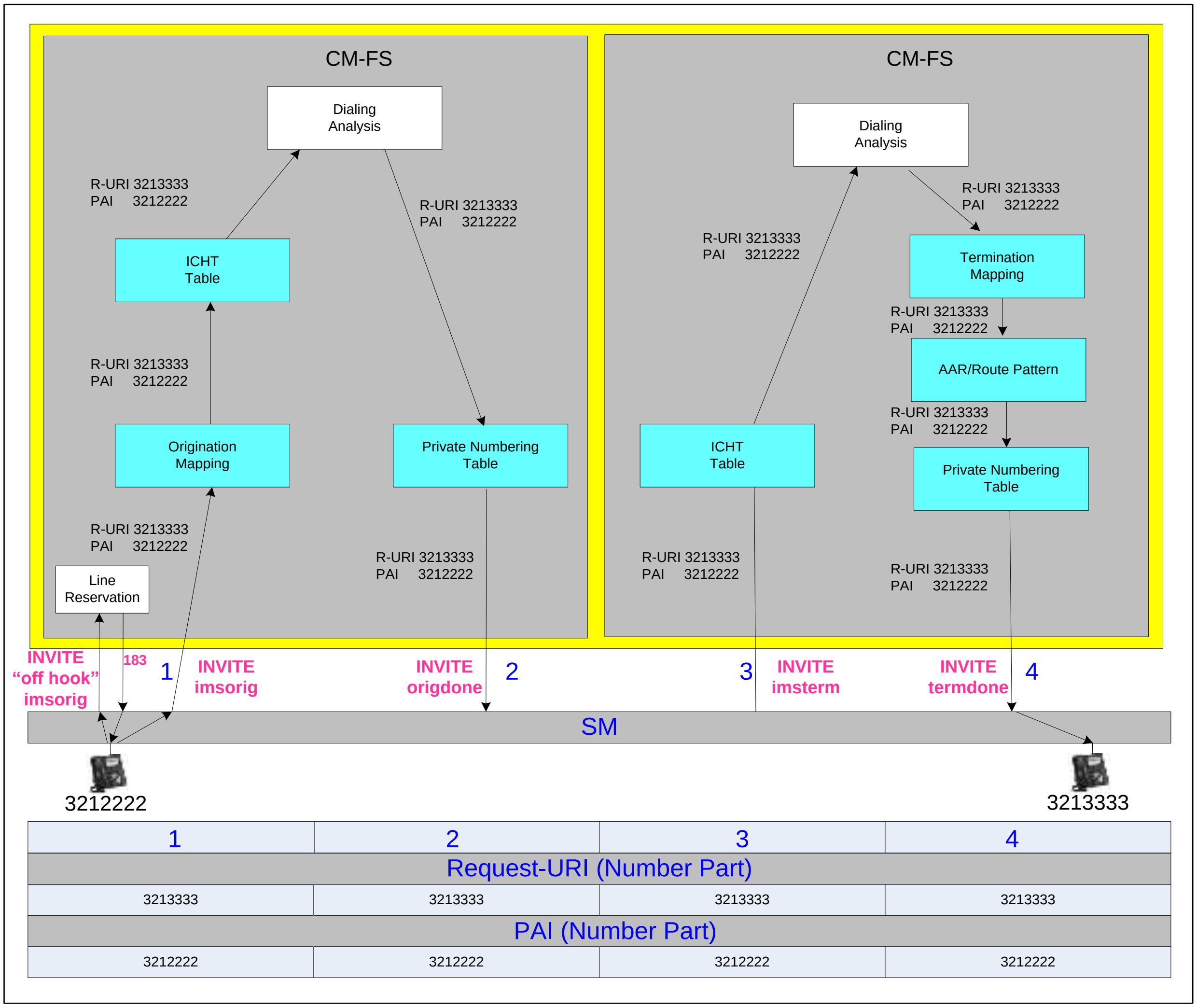


Figure 19: CM-FS Call Flow-Option Two

In this example, 3212222 is the calling SIP station 3213333 is the called SIP station. SIP station 3212222 logs in as 3212222 and matches on that handle in SM (same is true for 3213333 login). In the SIP INVITE message from the originating SIP phone the SIP handle 3212222 appears in the Contact header and the dialed digits 3213333 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
   a. SM does a lookup of 3212222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
   b. The PAI header in imsorig contains 3212222 since it is the preferred handle specified.
   c. The R-URI contains the digits dialed by the end user and is not looked at by SM.
2. Processing by CM-FS prior to origdone call leg to SM

a. The call flows through origination mapping for station 3212222 and since the phone number matches the station extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) to change R-URI from long to short form, but since there is no match there is no change to the R-URI.
c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
d. CM routes the origdone call leg using AAR routing based on <u>originating</u> phone number 3212222.
e. The private numbering table adapts the calling party information (PAI) <u>AND</u> called party information (R-URI) from short to longs. In this case the extensions are already in private long format and do not need to be adapted.
f. Since PAI private long form is used for extension and PAI, CM-FS
    i. does not need to use ICHT to determine the station that matches private long form of PAI
    ii. does not append avext to PAI in any case
    iii. sends call back to SM as origdone rather than terminating

3. Processing by SM prior to imsterm call leg to CM-FS is the same as for CM-ES
    a. SM now looks at Request-URI of 3213333
    b. SM recognizes this as a handle associated with station extension 3213333 (note: SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
    c. SM forwards this call back to CM based on termination sequence administration in SM.
    d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 3212222 (PAI) and 3213333 (R-URI).
4. Processing by CM-FS prior to termdone back to SM
    a. CM-FS looks for match on R-URI and PAI in ICHT.
        i. This is a special case where CM-FS attempts to do long to short processing on both R-URI and PAI.
        ii. Both PAI and R-URI are 7-digits in length (long form of the number) and do not match anything in the ICHT table
    b. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    c. CM converts extension number of terminating SIP station 3213333 from extension number to telephone number using term mapping. In this case both extension number and telephone number are the same private long format.
    d. CM AAR routes the call to the proper route pattern based on terminating phone number 3213333
    e. The private numbering table adapts the calling party information (PAI) from SIP station extension number 3213333 from private short to private long. In this case both are private long and no adaptation is necessary.
    f. Since PAI private long form is used for extension and PAI, CM-FS
        i. does not need to use ICHT to determine the station that matches private long form of PAI
        ii. does not append avext to PAI
        iii. sends call back to SM as origdone rather than terminating

5. SM now matches the R-URI with the called user profile and sends the call to the phone.
6. SIP phone displays PAI

8.1.3 System Manager, CM-ES and CM-FS Administration-Option Two

For SIP users in System Manager User Profile for both Evolution and Feature Server:

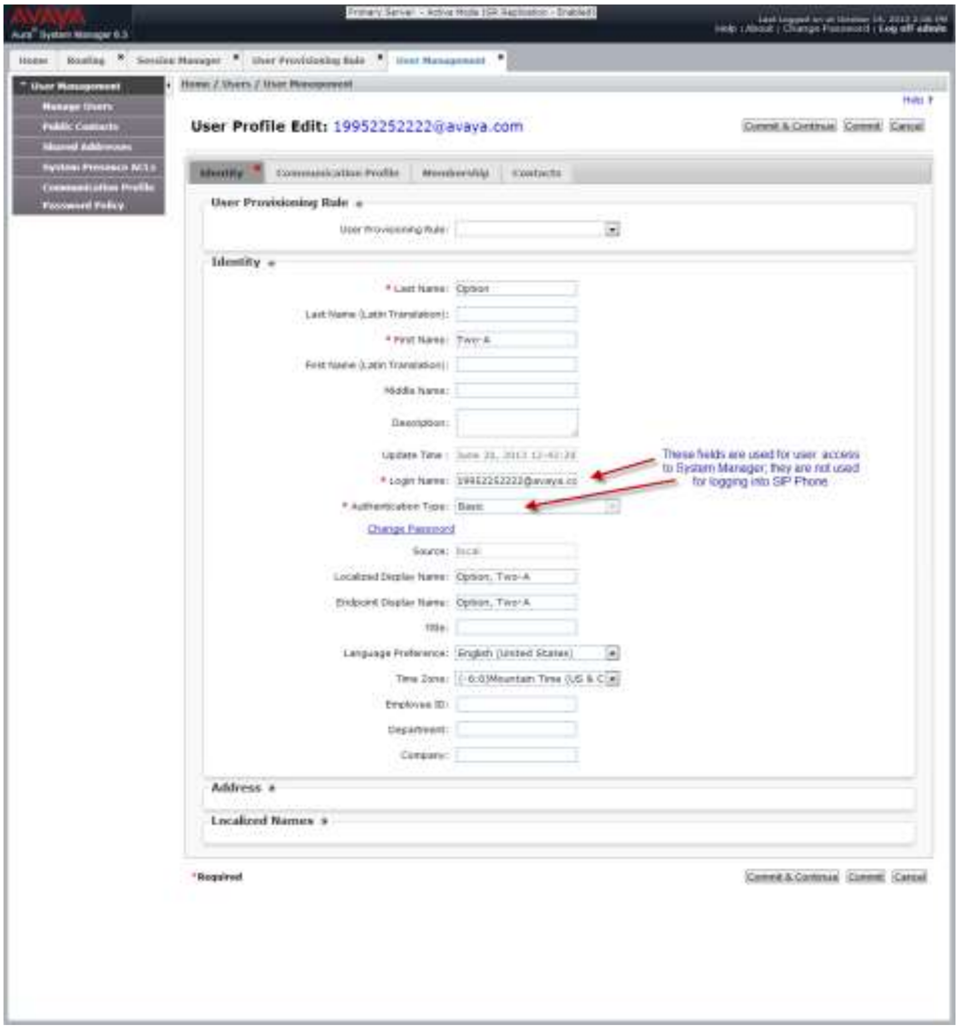


Figure 20: System Manager User Profile Identity-Option Two

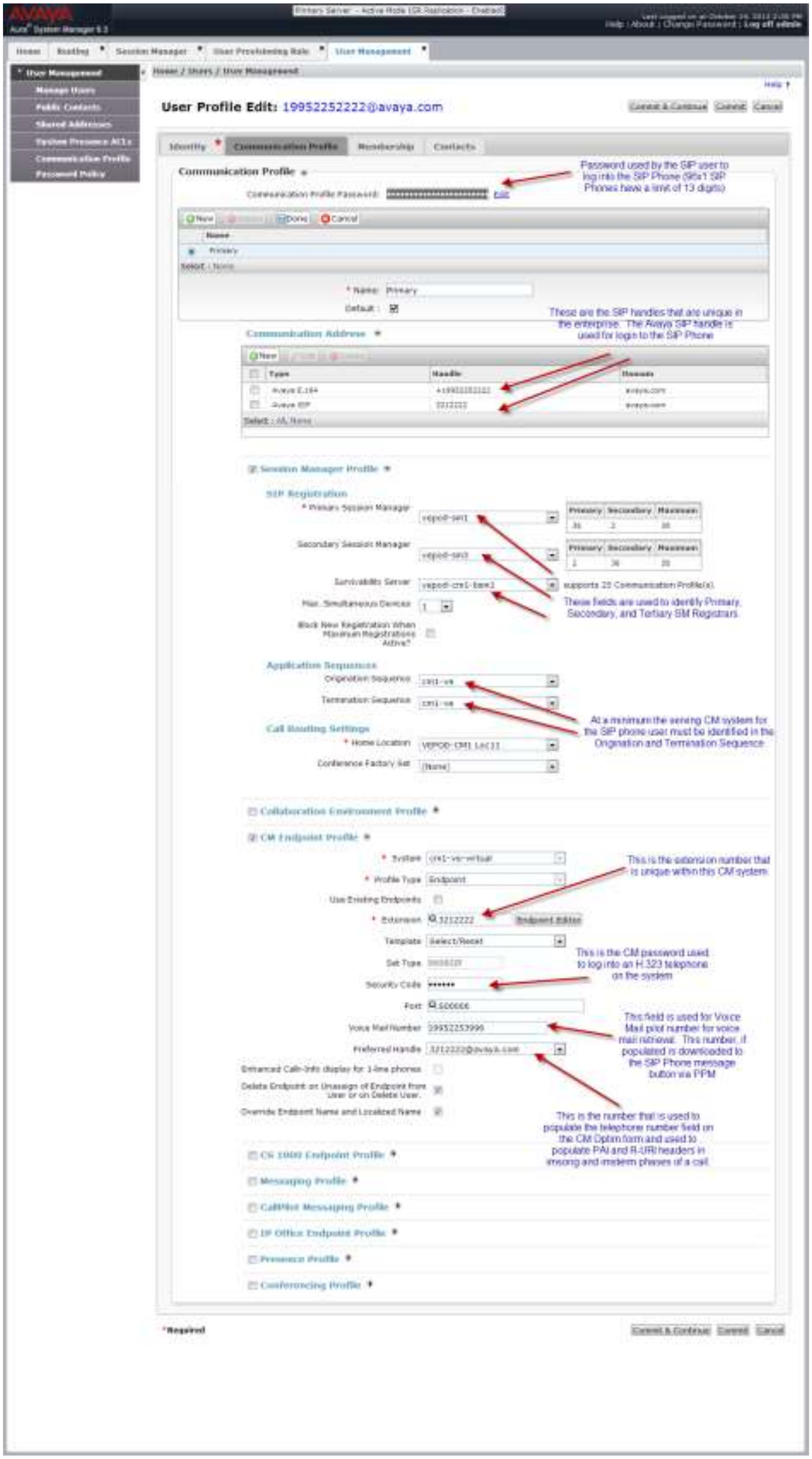


Figure 21: System Manager User Communication Profile-Option Two

System Manager automatically populates CM station as well as off-pbx-telephone station-mapping forms with the following translations using the selected template for both Evolution and Feature Server:

```
display station 3212222                                         Page   1 of   6
                                     STATION

Extension: 321-2222                      Lock Messages? n             BCC: 0
     Type: 9608SIP                       Security Code: 123456         TN: 1
     Port: S00006                      Coverage Path 1:               COR: 1
     Name: Option, Two-A               Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                Location:                  Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 321-2222

        Display Language: english                  Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
```

```
display station 3212222                                         Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                          Headset? n
       Jack:                                          Speaker? n
      Cable:                                         Mounting: d
      Floor:                                     Cord Length: 0
   Building:                                        Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                            5:
 2: call-appr                            6:
 3: call-appr                            7:
 4:                                      8:
```

```
display station 3212222                                         Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

Note: the default routing for this station is aar and that the phone will get three call appearances.

System Manager also populates the off-pbx telephone station mapping form with the application type OPS. System Manager uses the CM Endpoint Profile “Preferred Handle” in the User Communication Profile to populate the “phone number” field in CM.

```
display off-pbx-telephone station-mapping 3212222              Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 321-2222         OPS          -      3212222          aar        1
 321-3333         OPS                 3213333          aar        1
```

The off-pbx-telephone station-mapping form is used for long to short digit manipulation of the calling station on the origination side of the call (Phone Number to Station Extension). The off-pbx-telephone station-mapping form is used for short to long digit manipulation of the called station on the termination side of the call (Station Extension to Phone Number).

Prior to administering the System Manager User Profile the following **minimum** administration must be done in Communication Manager (Note: administration that follows is the same for Evolution and Feature Server).

The dialplan analysis form has the following administration:

- The dialed string 3, 7-digits in length to support 7-digit extensions (3212222 & 3213333).
- The dialed string 8 for Automatic Alternate Routing (AAR) and dialed string 9 for Automatic Route Selection (ARS). The AAR and ARS feature access codes must be defined.
- The dialed string *, 4-digits in length to accommodate SIP trunk dial access codes.

```
display dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 5

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   3            7   ext
   8            1   fac
   9            1   fac
   *            4   dac
```

Following is the minimum translations for system features:

```
display feature-access-codes                                   Page   1 of  11
                               FEATURE ACCESS CODE (FAC)
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9    Access Code 2:
```

The AAR/ARS codes do not have to be what is shown here, but they do need to be administered.

A dedicated signaling group(s) needs to be set up for use by the SIP telephones to the primary SM and secondary SM. In the example there are actually four SMs: vepod-sm1 and vepod-sm2 in data center one and vepod-sm3 and vepod-sm4 in data center two. Assume that for SIP telephones in this example that vepod-sm1 is the primary SM and vepod-sm3 is the secondary SM. The Signaling group to vepod-sm1 is 910, and the signaling group to vepod-sm3 is 930. Administration for Signaling group 910 to vepod-sm1 is shown here:

```
display signaling-group 910                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910            Group Type: sip
  IMS Enabled? n          Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                   Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region:
                                     Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

In this example IMS Enabled field is set to n. This is the proper setting for a CM-ES. Setting of the IMS field is what determines whether or not calls follow the full call model on CM-ES or half call model on CM-FS. It is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

For CM-FS the signaling group needs to be set up for use by the SIP telephones to each SM with IMS enabled set to y (in the example there are two SMs), all other administration is the same as CM-ES.

```
display signaling-group 910                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910            Group Type: sip
  IMS Enabled? y          Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                   Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region:
                                     Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

Following is administration for SIP trunk group associated with the SIP signaling group 910:

```
display trunk-group 910                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 910                  Group Type: sip           CDR Reports: y
  Group Name: OPTIM SM1                   COR: 1        TN: 1        TAC: *910
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

```
display trunk-group 910                                         Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y

 DSN Term? N
```

As discussed in Section 6.1.4, the "Numbering Format" field on page 3 of the form is used to determine whether to use the public or private numbering table for various call flows in Evolution and Feature Server. In this example the numbering format used is private; on-net calls and SIP station to station calls use the private table.

In the Evolution Server, station to station calls use the routing associated with the terminating OPS phone number 3212222 for origdone call leg (no routing associated on termdone since CM-ES shortcuts the term side of the call. In the Feature Server, station to station calls use routing associated with the originating OPS phone number 3212222 for the origdone call leg and terminating OPS station 3213333 on the call for termdone call leg.

Routing for 3212222 and 3213333 is based on AAR and uses route pattern 910.

```
display aar analysis 0                                                Page   1

                           AAR DIGIT ANALYSIS REPORT

                            Location:  all

               Dialed            Total        Route    Call      Node
               String          Min    Max    Pattern   Type     Number

               3                 7      7      910      aar
```

```
display route-pattern 910                                       Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP SM1 and SM3
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 910  0                                                             n   user
 2: 930  0                                                             n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                unk-unk  next
 2: y y y y y n  n            rest                                unk-unk  none
```

In this example look ahead routing has been implemented in the event that CM gets failure messages on vepod-sm1 it can retry on vepod-sm3 using trunk group 930. The preferences in this route pattern match the primary and secondary SM specified for the SIP users in System Manager based on the best practice cited in Section 6.5. The numbering format on this route pattern is set to unk-unk. Since trunk group 910 and 930 are private trunks the private-numbering table is used based on the CM algorithm specified in section 6.1.4.

Following is administration in the private-numbering table.

```
display private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 7  3                                          7     Total Administered: 5
                                                      Maximum Entries: 540
```

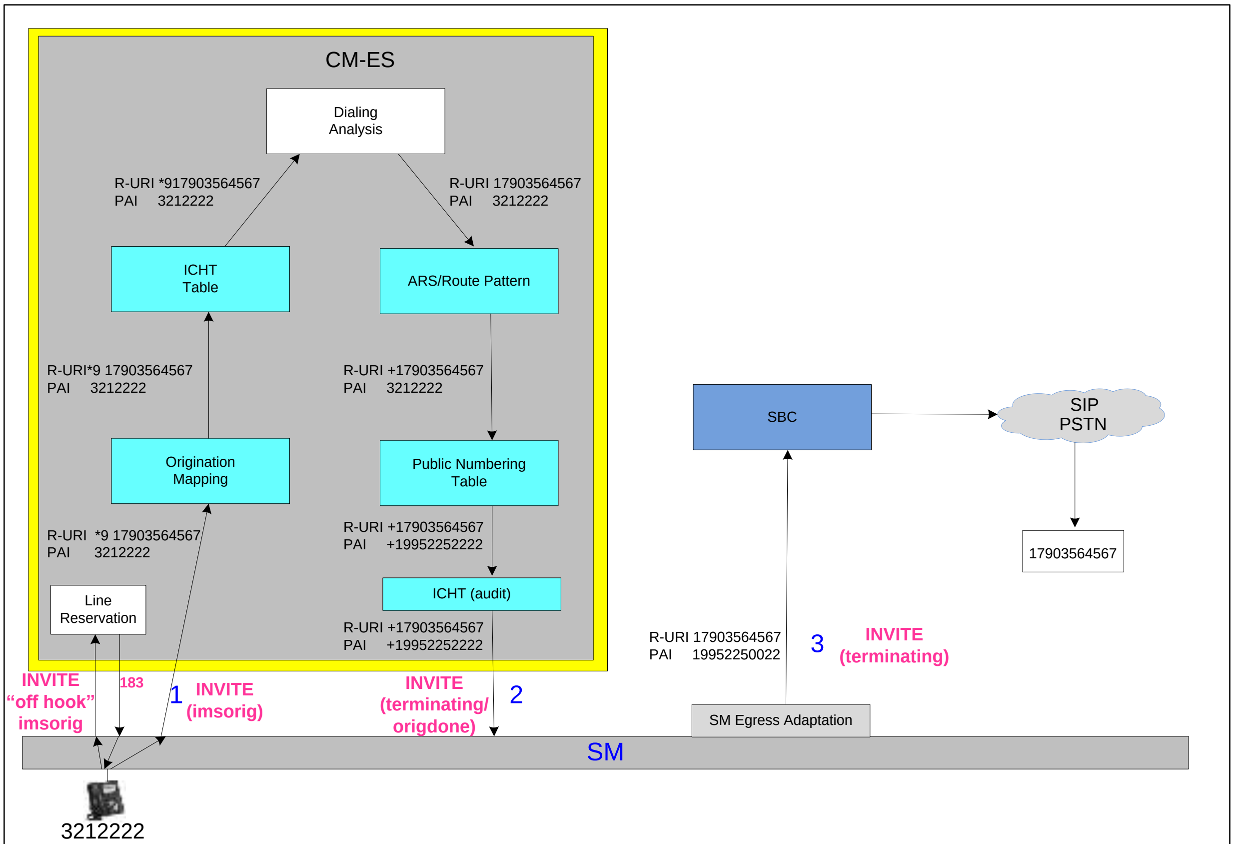
CM-ES and CM-FS both use this private-numbering translation for short to long form administration based on call flows described in Section 8.1.1 and 8.1.2. Matches in this table will <u>not</u> result in "+" being appended to the number.

ICHT does not have to be filled out to support private long to private short form for subscriptions or SIP user calls since extension and SIP handle in this example both use private long form.

8.2 SIP Station to Outbound SIP PSTN Call Flow-Option Two

8.2.1 CM-ES and CM-FS

Following is call flow for 11-digit North American Number to PSTN from station extension 3212222 to 1720-356-4567:



1	2	3
Request-URI (Number Part)		
*9 17903564567	+17903564567	17903564567
PAI (Number Part)		
3212222	+19952252222	19952252222

Figure 22: Outbound PSTN Call Flow in Feature or Evolution Server-Option Two

End user dials the ARS access code *9 and the PSTN number

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
    a. SM does a lookup of 3212222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 3212222 since it is the preferred handle specified.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.
2. Processing by CM-ES and CM-FS prior to origdone call leg to SM

a. The call flows through origination mapping for station 3212222 and since the phone number matches the station extension number on the off-pbx-telephone station-mapping form, no change is made to the PAI.
b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
d. Call is routed to proper route pattern
e. The public-unknown-numbering table is now used to adapt the calling party information from private long number to E.164 with the "+" prior to sending the call to SM in the origdone leg of the call.
f. CM uses ICHT on SIP PSTN trunk group (TG110/130) to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 3212222 (public long).
    i. If ICHT has an entry that deletes +1995225 and inserts 321. There is now a match with the originating SIP station 3212222
        1. CM sends the call back to SM as origdone call leg with PAI in E.164 format.
        2. Explicit sequencing of origination applications after CM requires origdone call processing.
        3. CM <u>always</u> sends signaling for origdone call leg back to the same SM that initiated imsorig call processing regardless of what is specified in AAR/ARS routing;
        4. If AAR/ARS routing for origdone is different than the SM used for imsorig, CM call processing still shows use of signaling group/trunk group specified in ARS/AAR.
        5. Since this is not a station to station call, avext is not appended to PAI header.
    ii. If ICHT does NOT have an entry that deletes +1995225 and inserts 321
        1. CM sends the call back to SM as "terminating" with E.164 PAI format.
        2. Implicit sequencing, including Collaboration Environment is supported since CE does not require origdone call processing.
        3. <u>Terminating call legs unlike origdone call legs do not need to return to the same SM that initiated imsorig call processing.</u>
        4. In this case, CM sends the terminating call leg to SM specified in ARS routing as "terminating" even if it is different than the SM used for imsorig.
        5. CM call processing shows trunk group usage to SM chosen by ARS.
g. The "p" in the route pattern inserts the "+" on the R-URI
h. CM sends PAI and R-URI numbers to SM in E.164 format[3]

3. Processing by SM prior to terminating to SBC
    a. SM determines routing policy
    b. Applies egress adaptation to SBC based on SBC and PSTN requirements

[3] Additional considerations need to be applied to international calls. In North America, the international prefix dialed is "011" (many other parts of the world it is "00"). These digits can be deleted on an international route pattern and the "+" inserted or the call can be sent to SM with the international prefix. In this case, SM would have an adaptation to delete the international prefix and insert "+" for analysis and routing. Here is a case where an SM ingress adaptation is being used in SM that does not impact SIP telephone call processing

8.2.2 System Manager, CM-ES and CM-FS Administration-Option Two

Separate signaling group(s) need to be set up for access to the PSTN SIP trunks to each SM (in the example assume two SMs). Administration for signaling group to vepod-sm1 is shown here also need signaling group 130 to vepod-SM3):

```
display signaling-group 110                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 110             Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: SM1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 241
                                    Far-end Secondary Node Name:
Far-end Domain: sbccore.avaya.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from PSTN trunks (imsterm) have a domain of sbccore.avaya.com and CM will select this signaling group.

Following is a sample trunk used for PSTN access via SM:

```
display trunk-group 110                                        Page   1 of  21
                                TRUNK GROUP

Group Number: 110                  Group Type: sip           CDR Reports: y
  Group Name: SIP PSTN SM1                COR: 1        TN: 1        TAC: *110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 110
                                                     Number of Members: 15
```

```
display trunk-group 110                                        Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n                     SIP ANAT Supported? n
```

Following is a simple ARS Analysis table for access to SIP PSTN trunks:

```
display ars analysis 0                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    011                   10   18    110      intl         n
    1                     11   11    110      natl         n
    911                   3    3     911      emer         n
```

Assume that 11 digit North America numbers are being dial and are then routed using route pattern 110 as specified in ARS Analysis:

```
display route-pattern 110                                      Page   1 of   3
                    Pattern Number: 110    Pattern Name: SBC DC1
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 110  0                        p                                    n   user
 2: 130  0                        p                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                          next
 2: y y y y y n  n            rest                                          next
```

Route pattern 110 inserts the + to the Request-URI based on the “p” entered in the inserted digits field so that SM can route on E.164 number. In North America 1+10 digits is in E.164 format if the + is appended to the dial string. In route pattern 101 the numbering format field is not applicable since these are public trunks. Based on the algorithm used for public trunks, this call will use the public-unknown-numbering table.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 and 130 the ICHT for the PSTN trunk group 110/130 is to insure that the call is sent as origdone rather than "terminating" on PSTN calls:

```
display public-unknown-numbering 0                            Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN           CPN
Len Code            Grp(s)    Prefix        Len
                                                  Total Administered: 8
 7  321                        1995225      11       Maximum Entries: 9999
                                                  Note: If an entry applies to
                                                  a SIP connection to Avaya
                                                  Aura(R) SM,
                                                  the resulting number must
                                                  be a complete E.164 number.
```

CM-ES and CM-FS both use this translation for short to long form administration based on call flows described in Section 8.2.1.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 and 130 the ICHT for the PSTN trunk group 110/130 is to insure that the call is sent as origdone rather than "terminating" on PSTN calls:

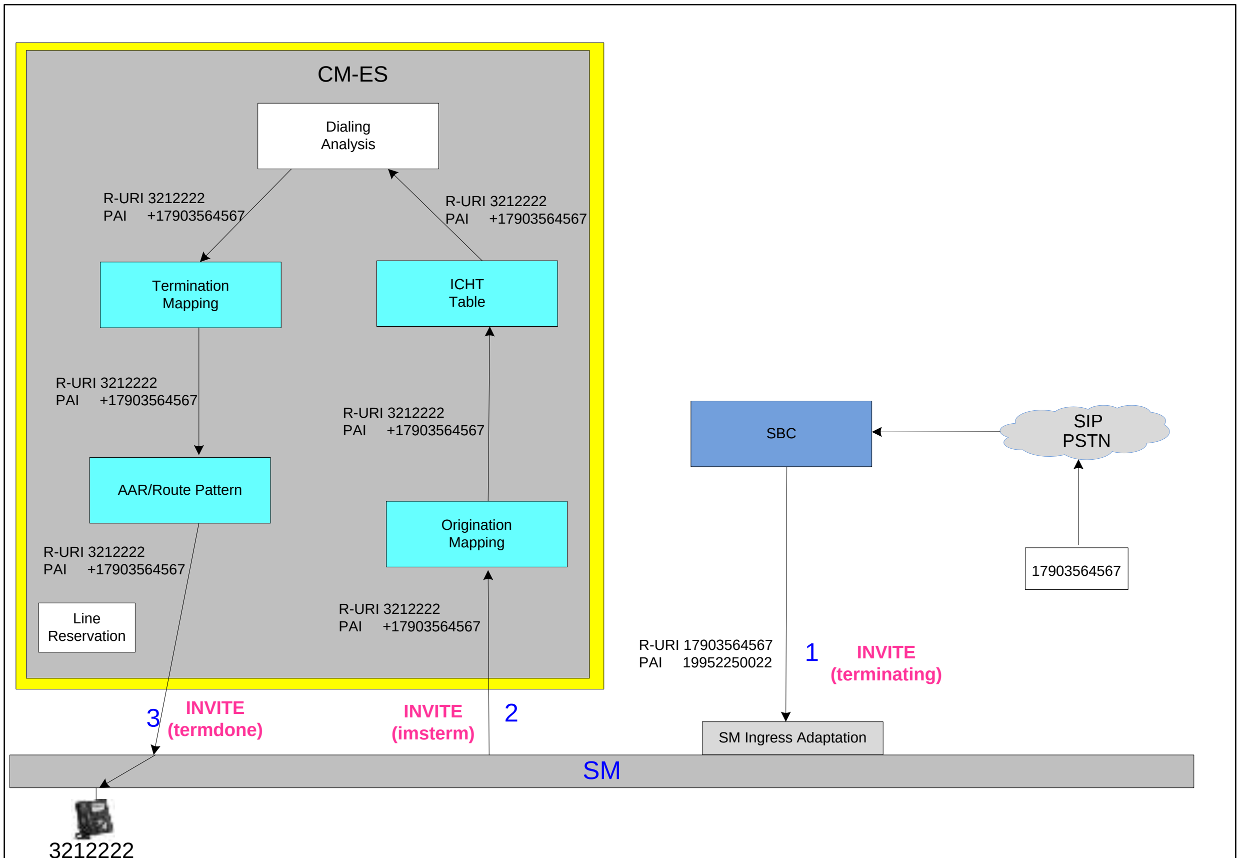
```
change inc-call-handling-trmt trunk-group 110                 Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             12 +1995225         8   321
```

CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 8.2.1.

8.3 Inbound SIP PSTN to SIP Station Call Flow-Option Two

8.3.1 CM-ES and CM-FS

Following is the call flow for a number dialed from the PSTN to extension 3212222



1	2	3
Request-URI (Number Part)		
19952252222	3212222	3212222
PAI (Number Part)		
17903564567	+17903564567	+17903564567

Figure 23: Inbound PSTN Call Flow in Feature or CM-ES Server-Option Two

PSTN user 17203564567 dials PSTN number 19952252222

1. Processing by SM on terminating call leg from SBC
    a. SBC sends call to SM using terminating phase.
    b. SM adapts R-URI to E.164 format +19952252222
    c. SM adapts PAI header to E.164 format +17203564567
2. Processing by SM prior to imsterm call leg to CM
    a. SM does a lookup of R-URI from SBC of +19952252222 and sees that it is a registered user and forwards the call to CM based on termination sequence administration in System Manger using preferred handle 321222.
    b. SM sends PAI to CM as E.164

3. Processing by CM prior to termdone call leg to SM
    a. The call flows through origination mapping and there is no match with PAI +17203564567 and sees no match.
    b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG110/130) to convert R-URI from long to short form and since there is no short form there is no change.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. The call flows through termination mapping for station 321222 for short to long form processing and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the R-URI.
    e. CM AAR routes the call to the proper route pattern based on terminating phone number 3212222
    f. CM sends PAI in E.164 format and R-URI in private long format to SM
4. SM now matches the R-URI with the called user profile and sends the call to the phone
5. SIP Phone displays E.164 number +17203564567

9 Option Three: Extensions Based on a Subset of the E.164 Numbering Plan

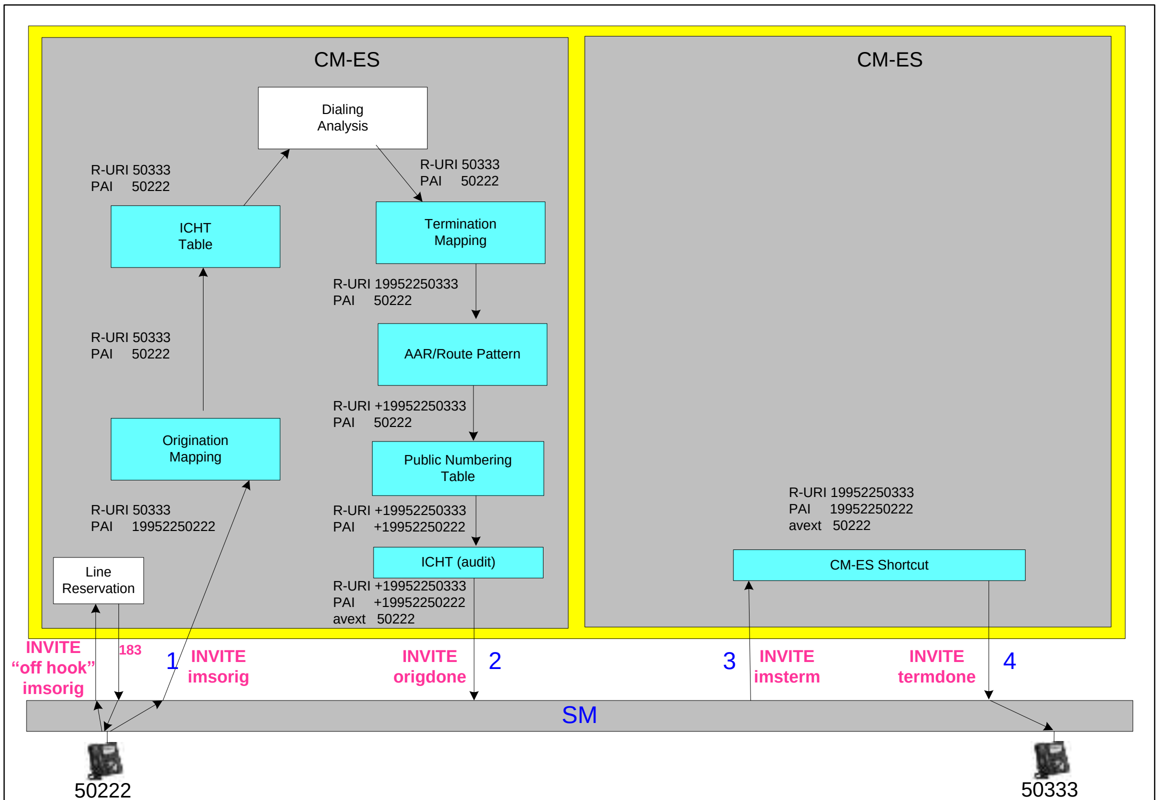
Option three extensions are based on a subset of the E.164 numbering plan. This is a configuration that can be used by a smaller CM system in which shorter length extension numbers are desirable. These extension numbers are unique within the CM system, but not unique in the enterprise, therefore the user needs to log into the SIP Phone using the Public Long form of the number. The extension number is a subset of the E.164 handle used to login into the SIP phone. Following is an example based on North America:

- Handles
  - Avaya E.164 +19952250222
  - Avaya SIP 19952250222 (Public Long/Preferred Handle)
- Extension Number
  - 50222 (Public Short)

9.1 SIP Station to SIP Station Call Flow-Option Three

9.1.1 Evolution Server

Following is an Option Three diagram of CM-ES SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):



1	2	3	4
Request-URI (Number Part)			
50333	+19952250333	19952250333	19952250333
PAI (Number Part)			
19952250222	+19952250222 50222 (avext)	19952250222 50222 (avext)	19952250222 50222 (avext)

Figure 24: Evolution Server Call Flow-Option Three

In this example, 50222 is the calling SIP station and 50333 is the called SIP station. SIP station 50222 logs in as 19952250222 which matches the SIP handle in SM (same is true for 50333 login). In the SIP INVITE message from the originating SIP phone the SIP handle 19952250222 appears in the Contact header and the dialed digits 50333 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-ES
   a. SM does a lookup of 19952250222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.

    b. The PAI header in imsorig contains 19952250222 since it is the preferred handle specified in SysMgr.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.
2. Processing by CM-ES prior to origdone call leg to SM
    a. The call flows through origination mapping and a match for phone number 19952250222 is found and is replaced by the station extension number 50222 shown on the off-pbx-telephone station-mapping form; a public long to public short conversion has been done.
    b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) to change R-URI from long to short form, but since there is no match there is no change to the R-URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    d. CM converts extension number of terminating SIP extension 50333 (public short) to the associated phone number 19952250333 (public long) using term mapping.
    e. CM AAR routes the call to the proper route pattern based on terminating phone number 19952250333.
    f. The public-unknown numbering table adapts the calling party information (PAI) from SIP station extension number 50222 (public short) to +19952250222 E.164 format.
    g. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 50222 (public short).
        i. ICHT has an entry on the trunk group used to route the call that deletes 7 digits +199522.
        ii. There is now a match with the originating SIP station 50222 after "+" is deleted, CM appends avext parameter with extension (public short number) to the E.164 PAI header and sends both forms back to SM as origdone rather than terminating.
        iii. If there is no match CM sends the call back to SM as "terminating" with E.164 PAI format with no avext parameter.
    h. A "+" is also added to R-URI since there is a "p" in the route pattern insert column.

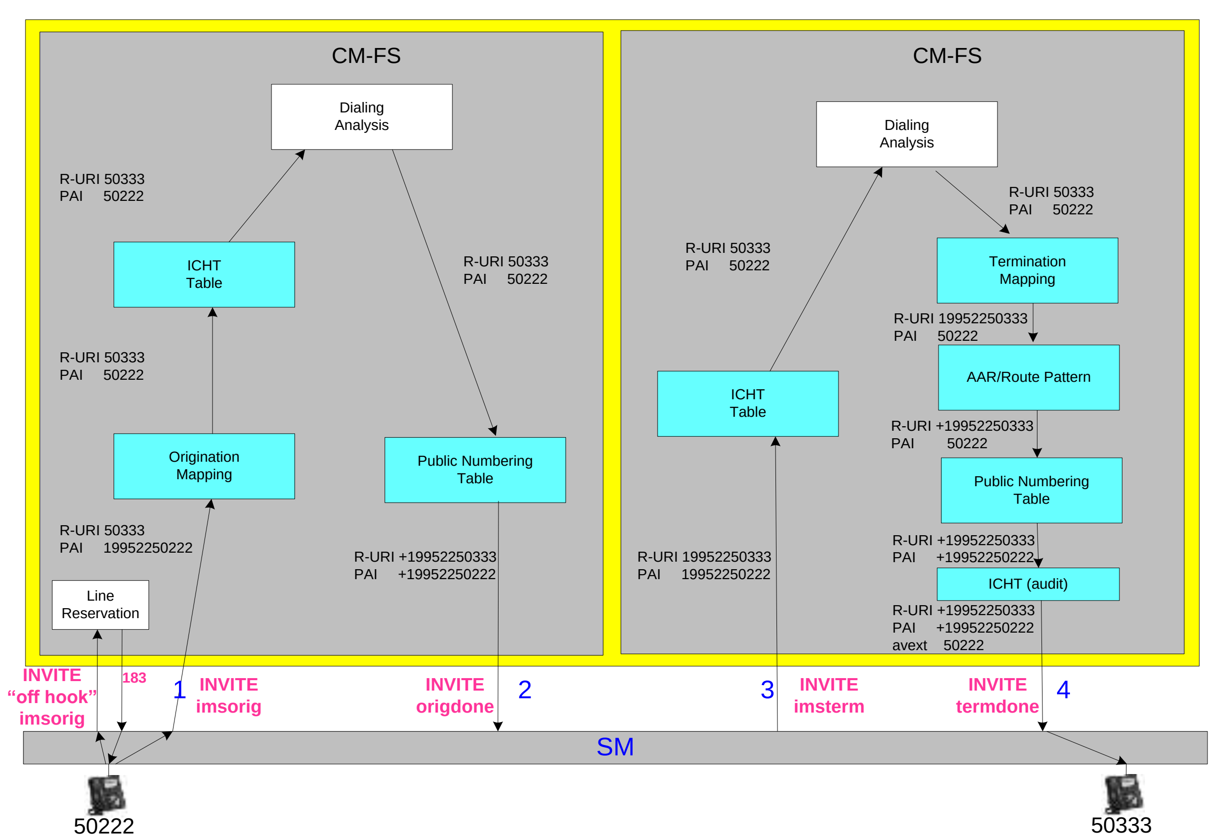
3. Processing by SM prior to imsterm call leg to CM-ES
    a. SM now looks at Request-URI of +19952250333
    b. SM recognizes this as a handle associated with station extension SM forwards this call back to CM based on termination sequence administration in System Manager.
    c. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 19952250222 (PAI) and 19952250333 (R-URI) and neither contain a "+".

4. Processing by CM-ES prior to termdone back to SM
    a. CM-ES does "shortcut" of imsterm and sends the SIP invite back to SM in termdone with no further processing of the call.
    b. Both PAI and R-URI are 11-digits in length without the "+" and avext appended to PAI header is 50222.

5. SM now matches the R-URI with the called user profile and completes the call to the phone
6. SIP phone displays avext (extension number 50222) since it is available, otherwise PAI is displayed.

9.1.2 Feature Server

Following is an Option Three diagram of a CM-FS SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):



1	2	3	4
Request-URI (Number Part)			
50333	+19952250333	19952250333	+19952250333
PAI (Number Part)			
19952250222	+19952250222	19952250222	+19952250222 50222 (avext)

Figure 25: CM-FS Call Flow-Option Three

In this example, SIP station 50222 is the calling SIP station and 50333 is the called SIP station. SIP station 50222 logs in as 19952250222 which matches the SIP handle in SM (same is true for 50333 login). The SIP INVITE message from the originating SIP phone 19952250222 appears in the Contact header and 50333 from the terminating SIP phone appears in the Request-URI.

1. Processing by SM prior to imsorig call leg to Feature Server
    a. SM does a lookup of 19952250222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 19952250222 since it is the preferred handle specified in SysMgr.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-FS prior to origdone call leg to SM
    a. The call flows through origination mapping (public long to public short conversion) and a match for phone number 19952250222 is found and is replaced by the station extension number 50222 shown on the off-pbx-telephone station-mapping form.
    b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    d. CM AAR routes the call to the proper route based on originating phone number 19952250222.
    e. The public-unknown-numbering table adapts calling party information (PAI) <u>AND</u> called party information (R-URI) from SIP station extension (public short) to E.164 format by adding +199522 to 50222 and 50333.
    f. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 50222 (public long).
        i. ICHT must have an entry that deletes the +199522.
        ii. There is now a match with the originating SIP station 50222 after the digits are deleted; CM sends the call back to SM as origdone call leg rather than terminating.
        iii. If there is no match CM sends the call back to SM as “terminating” with E.164 PAI format.
    g. NOTE: the “p” in the route pattern insert column is ignored in CM-FS prior to origdone

3. Processing by SM prior to imsterm call leg to CM-FS
    a. SM now looks at Request-URI of +19952250333
    b. SM recognizes this as a handle associated with extension 50333 (note SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
    c. SM forwards this call back to CM based on termination sequence administration in System Manager.
    d. Both R-URI and PAI sent back to CM on imsterm leg based on the preferred handles administered in SM: 19952250222 (PAI) and 19952250333 (R-URI).

4. Processing by CM-FS prior to termdone leg back to SM
    a. CM-FS looks for match on R-URI and PAI in ICHT.
        i. This is a special case where CM-FS attempts to do long to short processing on both R-URI and PAI.
        ii. Both PAI and R-URI are 11-digits in length (public long form)
        iii. There needs to be an entry in ICHT to delete 6 digits (199522) to convert to extensions 50222 and 50333 (public short form)
    b. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    c. CM Term mapping converts R-URI from the short form (station extension 5033) to the long form (phone number 19952250333).
    d. CM routes the termdone call leg using AAR routing based on <u>terminating</u> phone number 19952250333
    e. The public-unknown-numbering table adapts the calling party information (PAI) from SIP station extension number 50222 (public short) to E.164 by adding +199522.

f. CM uses ICHT to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 50222 (public short).
    i. ICHT must have an entry that deletes the +199522.
    ii. There is now a match with the originating SIP station 50222 after "+" is deleted, CM appends avext parameter with extension 50222 (public short number) to the E.164 PAI header and sends both forms back to SM as termdone call leg.
    iii. If there is no match CM sends the call back to SM with E.164 PAI format as termdone call leg with no avext parameter.

g. A "+" is added to the R-URI since there is a "p" in the route pattern insert column.

h. Now both PAI and R-URI are E.164 with the "+" on both.

5. SM now matches the R-URI with the called user profile and sends the call to the phone
6. SIP Phone displays avext (extension number) if available, otherwise PAI

9.1.3 System Manager, CM-ES and CM-FS Administration-Option Three

For SIP users in System Manager User Profile for both Evolution and Feature Server:

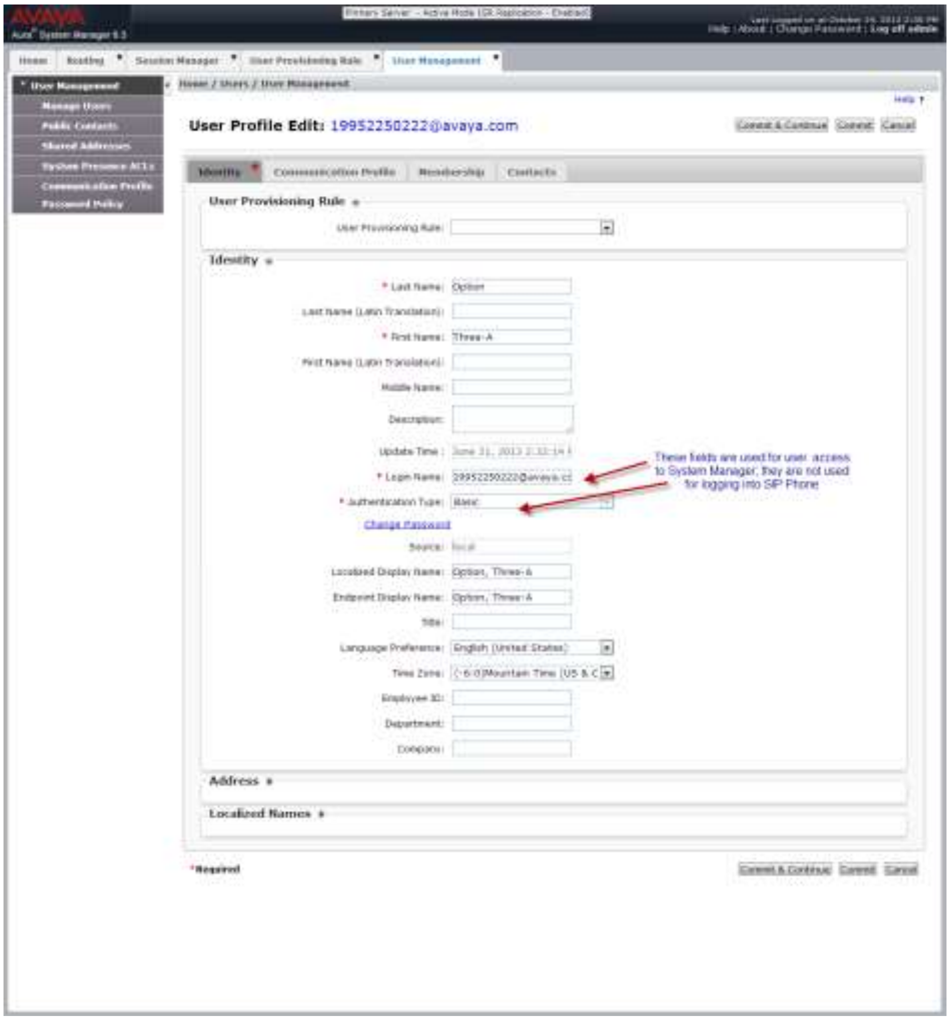


Figure 26: System Manager User Profile Identity-Option Three

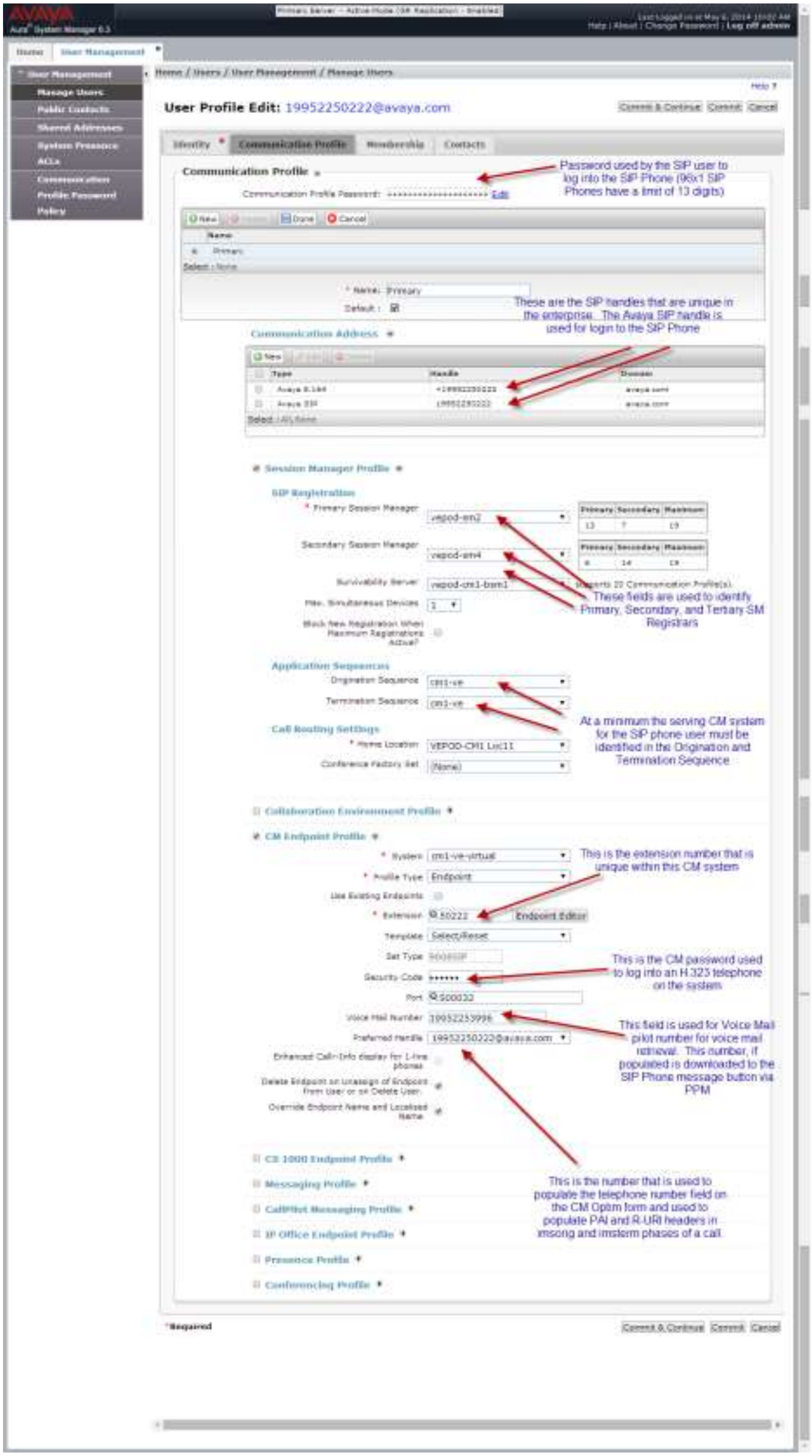


Figure 27: System Manager User Communication Profile-Option Three

System Manager automatically populates CM station as well as off-pbx-telephone station-mapping forms with the following translations using the selected template for both Evolution and Feature Server:

```
display station 50222                                           Page   1 of   6
                                     STATION

Extension: 50222                        Lock Messages? n             BCC: 0
     Type: 9608SIP                      Security Code: 123456         TN: 1
     Port: S00245                     Coverage Path 1:               COR: 1
     Name: Option, Three-A            Coverage Path 2:               COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                Location:                 Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 82222

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n
```

```
display station 50222                                           Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                           Headset? n
       Jack:                                           Speaker? n
      Cable:                                          Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:
```

```
display station 50222                                           Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

Note that the default routing for this station is aar and that the phone will get three call appearances.

System Manager also populates the off-pbx telephone station mapping form with the application type OPS. System Manager 6.2 uses the CM Endpoint Profile “Preferred Handle” in the User Communication Profile to populate the “phone number” field in CM

```
display off-pbx-telephone station-mapping 50222                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                Prefix                      Selection   Set     Mode
 50222            OPS          -     19952250222       aar        1
 50333            OPS          -     19952250333       aar        1
```

The off-pbx-telephone station-mapping form is used for long to short digit manipulation of the calling station on the origination side of the call (Phone Number to Station Extension). The off-pbx-telephone station-mapping form is used for short to long digit manipulation of the called station on the termination side of the call (Station Extension to Phone Number).

Prior to administering the System Manager User Profile the following **minimum** administration must be done in Communication Manager (Note that administration that follows is the same for Evolution and Feature Server):

The dialplan analysis form has the following administration:

- The dialed string 5, 5-digits in length to support 5-digit extensions (50222 & 50333).
- The dialed string *8 for Automatic Alternate Routing (AAR) and dialed string *9 for Automatic Route Selection (ARS). The AAR and ARS feature access codes must be defined.
- The dialed string *, 4-digits in length to accommodate SIP trunk dial access codes.

```
display dialplan analysis                                     Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 5

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   5           5   ext
   *8          2   fac
   *9          2   fac
   *           4   dac
```

Following is the minimum translations for system features:

```
display feature-access-codes                                  Page   1 of  11
                               FEATURE ACCESS CODE (FAC)
      Auto Alternate Routing (AAR) Access Code: *8
    Auto Route Selection (ARS) - Access Code 1: *9    Access Code 2:
```

The AAR/ARS codes do not have to be what is shown here, but they do need to be administered.

A dedicated signaling group(s) needs to be set up for use by the SIP telephones to the primary SM and secondary SM. In the example there are actually four SMs: vepod-sm1 and vepod-sm2 in data center one and vepod-sm3 and vepod-sm4 in data center two. Assume that for SIP telephones in this example that vepod-sm1 is the primary SM and vepod-sm3 is the secondary SM. The Signaling group to vepod-sm1 is 910, and the signaling group to vepod-sm3 is 930. Administration for Signaling group 910 to vepod-sm1 is shown here:

```
Display signaling-group 910                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910            Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region:
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

In this example IMS Enabled field is set to n. This is the proper setting for a CM-ES. Setting of the IMS field is what determines whether or not calls follow the full call model on CM-ES or half call model on CM-FS. It is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

For CM-FS the signaling group needs to be set up for use by the SIP telephones to each SM with IMS Enabled set to y (in the example there are two SMs), all other administration is the same as CM-ES.

```
display signaling-group 910                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910            Group Type: sip
  IMS Enabled? y        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                 Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region:
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

Following is administration for SIP trunk group associated with the SIP signaling group.

```
display trunk-group 910                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 910                  Group Type: sip          CDR Reports: y
  Group Name: OPTIM SM1                   COR: 1        TN: 1        TAC: *910
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

```
display trunk-group 910                                        Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n
```

As discussed in Section 6.1.4, the “Numbering Format” field on page 3 of the form is used to determine whether to use the public-unknown-numbering or private-numbering table for various call flows in Evolution and Feature Server. In this example the numbering format used is public since all calls should display E.164 format for calling (PAI) and called number (R-URI) information.

In the Evolution Server, station to station calls use the routing associated with the terminating OPS phone number 19952250333 for origdone call leg (no routing associated on termdone since CM-ES shortcuts the term side of the call. In the CM-FS, station to station calls use routing associated with the originating OPS station 19952250322 for the origdone call leg and terminating OPS station 19952250333 on the call for termdone call leg.

Routing for 19952250222 and 19952250333 is based on AAR and uses route pattern 910.

```
change aar analysis                                                    Page   1

                           AAR DIGIT ANALYSIS REPORT

                            Location:  all

               Dialed             Total         Route    Call      Node
               String           Min    Max     Pattern   Type      Number

         1995225                 11     11       910      aar
```

NOTE: insure that the AAR Digit Conversion Table is not converting the Matching Pattern "1" and routing the call to ARS (default entry). Need to delete the entry in red if it is in the system as default.

```
display aar digit-conversion 0                                        Page   1 of   2
                       AAR DIGIT CONVERSION TABLE
                                 Location: all                  Percent Full: 0
  Matching Pattern        Min    Max   Del   Replacement String   Net   Conv ANI Req
  0                       1      28    0                          ars    y        n
  x11                     3      3     0                          ars    y        n
  1                       4      28    0                          ars             n
```

```
display route-pattern 910                                             Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP SM1 and SM3
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                 QSIG
                             Dgts                                        Intw
 1: 910  0                         p                                      n   user
 2: 930  0                         p                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                            next
 2: y y y y y n  n            rest                                            none
```

In this example look ahead routing has been implemented in the event that CM gets failure messages on vepod-sm1 it can retry on vepod-sm3 using trunk group 930. The preferences in this route pattern match the primary and secondary SM specified for the SIP users in System Manager based on the best practice cited in Section 6.5. Since public numbering format is being used on the trunks, the numbering format is not applicable and is left blank. In this example the public-unknown-numbering table is based on the CM algorithm described in section 6.1.4.

In these examples, SM dial patterns are based on E.164 format, therefore a + is inserted on R-URI by administering a "p" on the route pattern. The sending of a "+" for R-URI is not mandatory since there is a handle for SIP stations with and without the "+". With this translation a plus will appear on R-URI in origdone, but it will be deleted on imsterm by SM since SM uses the Preferred Handle of 19952250333. The + is included in this case so that all route patterns are consistent.

Following is the administration in the public-unknown-numbering table.

```
display public-unknown-numbering 0                                  Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk        CPN         CPN
Len Code            Grp(s)     Prefix      Len
                                                 Total Administered: 8
 5   5                         199522      11       Maximum Entries: 9999

                                                 Note: If an entry applies to
                                                 a SIP connection to Avaya
                                                 Aura(R) SM,
                                                 the resulting number must
                                                 be a complete E.164 number.
```

CM-ES and CM-FS both use this translation for short to long form administration based on call flows described in Section 9.1.1 and 9.1.2

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 910 (and 930)

```
change inc-call-handling-trmt trunk-group 910                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
  tie             11 1995225          6[4]
  tie             12 +1995225         7
```

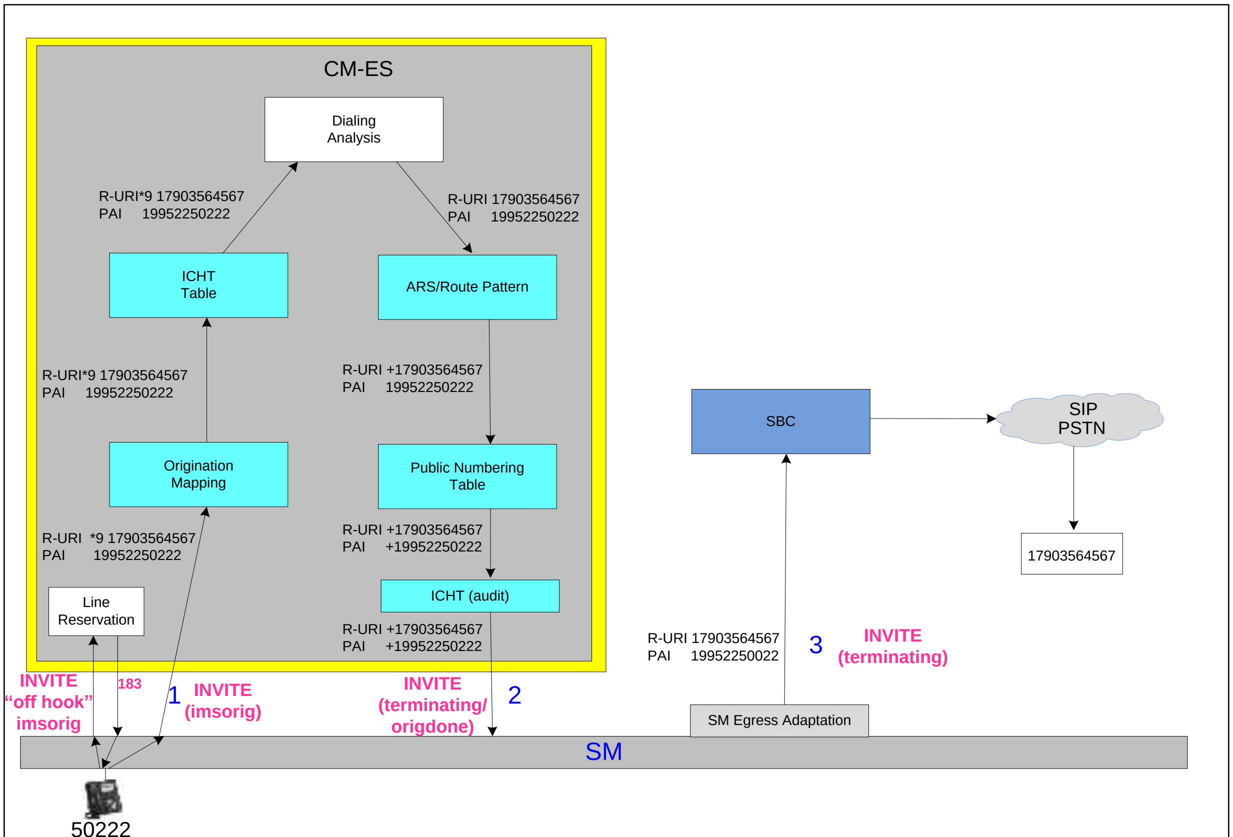
CM-ES and CM-FS both uses the +1995225 delete 7 digits translation for long to short form administration based on call flows described in Section 9.1.1 and 9.1.2

[4] This entry is used to convert from long to short form for Subscribe messages rather than using off-pbx-telephone station-mapping (see section 6.1.5).

9.2 SIP Station to Outbound SIP PSTN Call Flow-Option Three

9.2.1 CM-ES and CM-FS

Following is a call flow for number 17903564567 dialed from SIP station extension 50222:



1	2	3
Request-URI (Number Part)		
*9 17903564567	+17903564567	17903564567
PAI (Number Part)		
19952250222	+19952250222	19952250222

Figure 28: PSTN Call Flow in Feature or Evolution Server-Option Three

End user dials the ARS access code *9 and the PSTN number

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
   a. SM does a lookup of 19952250222 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
   b. The PAI header in imsorig contains 19952250222 since it is the preferred handle specified.
   c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES and CM-FS prior to origdone call leg to SM
    a. The call flows through origination mapping for station 50222 and since the phone number 19952250222 does not match the extension number 50222 on the off-pbx-telephone station-mapping form it converts to the extension number 50222 (public short form).
    b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. Call is routed to proper route pattern.
    e. The public-unknown numbering table is now used to adapt the calling party information from public short number to E.164 with the "+" prior to sending the call to SM in the origdone leg of the call.
    f. CM uses ICHT on SIP PSTN trunk group (TG110/130) to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 19952250022 (public long).
        i. If ICHT has an entry that deletes the +199522 there is now a match with the originating SIP station 50222
            1. CM sends the call back to SM as origdone call leg with PAI in E.164 format.
            2. Explicit sequencing of origination applications after CM requires origdone call processing.
            3. CM <u>always</u> sends signaling for origdone call leg back to the same SM that initiated imsorig call processing regardless of what is specified in AAR/ARS routing;
            4. If AAR/ARS routing for origdone is different than the SM used for imsorig, CM call processing still shows use of signaling group/trunk group specified in ARS/AAR.
            5. Since this is not a station to station call, avext is not appended to PAI header.
        ii. If ICHT does NOT have an entry that deletes the +199522 there is no match with the originating station 50222
            1. CM sends call back to SM as terminating call leg with PAI in E.164 format
            2. Implicit sequencing, including Collaboration Environment is supported since CE does not require origdone call processing.
            3. <u>Terminating call legs unlike origdone call legs do not need to return to the same SM that initiated imsorig call processing.</u>
            4. In this case, CM sends the terminating call leg to SM specified in ARS routing as "terminating" even if it is different than the SM used for imsorig.
            5. CM call processing shows trunk group usage to SM chosen by ARS.
    g. The "p" in the route pattern inserts the "+" on the R-URI
    h. CM sends PAI and R-URI numbers to SM in E.164 format[5]

[5] Additional considerations need to be applied to international calls. In North America, the international prefix dialed is "011" (many other parts of the world it is "00"). These digits can be deleted on an international route pattern and the "+" inserted or the call can be sent to SM with the international prefix. In this case, SM would have an adaptation to delete the international prefix and insert "+" for analysis and routing. Here is a case where an SM ingress adaptation is being used in SM that does not impact SIP telephone call processing

3. Processing by SM prior to terminating to SBC
    a. SM determines routing policy
    b. Applies egress adaptation to SBC based on SBC and PSTN requirements

9.2.2 CM-ES and CM-FS Administration

Separate signaling group(s) need to be set up for access to the PSTN SIP trunks from CM to each SM (in the example assume two SMs). Administration for signaling group 110 to vepod-sm1 is shown here (also need signaling group 130 to vepod-sm3):

```
display signaling-group 110                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 110            Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 241
                                    Far-end Secondary Node Name:
Far-end Domain: sbccore.avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from PSTN trunks (imsterm) have a domain of sbccore.avaya.com and CM will select this signaling group.

Following is a sample trunk used for PSTN access via SM:

```
display trunk-group 110                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 110                  Group Type: sip           CDR Reports: y
  Group Name: SIP PSTN SM1                COR: 1        TN: 1        TAC: *110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 110
                                                    Number of Members: 15
```

```
display trunk-group 110                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y

                     Numbering Format: public
                                                 UUI Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n
                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n                      SIP ANAT Supported? n
```

Since “public” is specified in the “Numbering Format” field all calls to this trunk group will use the “public unknown-numbering table”

Following is a simple ARS Analysis table for access to SIP PSTN trunks:

```
display ars analysis 0                                           Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    011                   10   18    110       intl          n
    1                     11   11    110       natl          n
    911                   3    3     911       emer          n
```

Assume that 11 digit North America numbers are being dial and are then routed using route pattern 110 as specified in ARS Analysis:

```
display route-pattern 110                                        Page   1 of   3
                    Pattern Number: 110    Pattern Name: SBC DC1
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 110  0                        p                                   n   user
 2: 130  0                        p                                   n   user


     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                        next
 2: y y y y y n  n            rest                                        next
```

Route pattern 110 inserts the + to the Request-URI based on the “p” entered in the inserted digits field so that SM can route on E.164 number. In North America 1+10 digits is in E.164 format if the + is appended to the dial string. In addition, route pattern 110 the numbering format field is not applicable since these are public trunks. Based on the algorithm for non-ims signaling groups, this call will use the public-unknown-numbering table.

The public-unknown-numbering table already administered for SIP station to station calls can be used for PSTN calls as well.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 (and 130) so that origdone processing is done to support explicit sequencing of calls:

```
change inc-call-handling-trmt trunk-group 110                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
  tie             12 +1995225         7
```

If explicit sequencing is <u>not</u> required this administration should be eliminated so that signaling and trunk group reflect the same SM in configurations with multiple SMs using terminating call legs.   CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 9.2.1.

9.3 Inbound SIP PSTN to SIP Station Call Flow-Option Three

9.3.1 CM-ES and CM-FS

Following is the call flow for number dialed from the PSTN to extension 50222

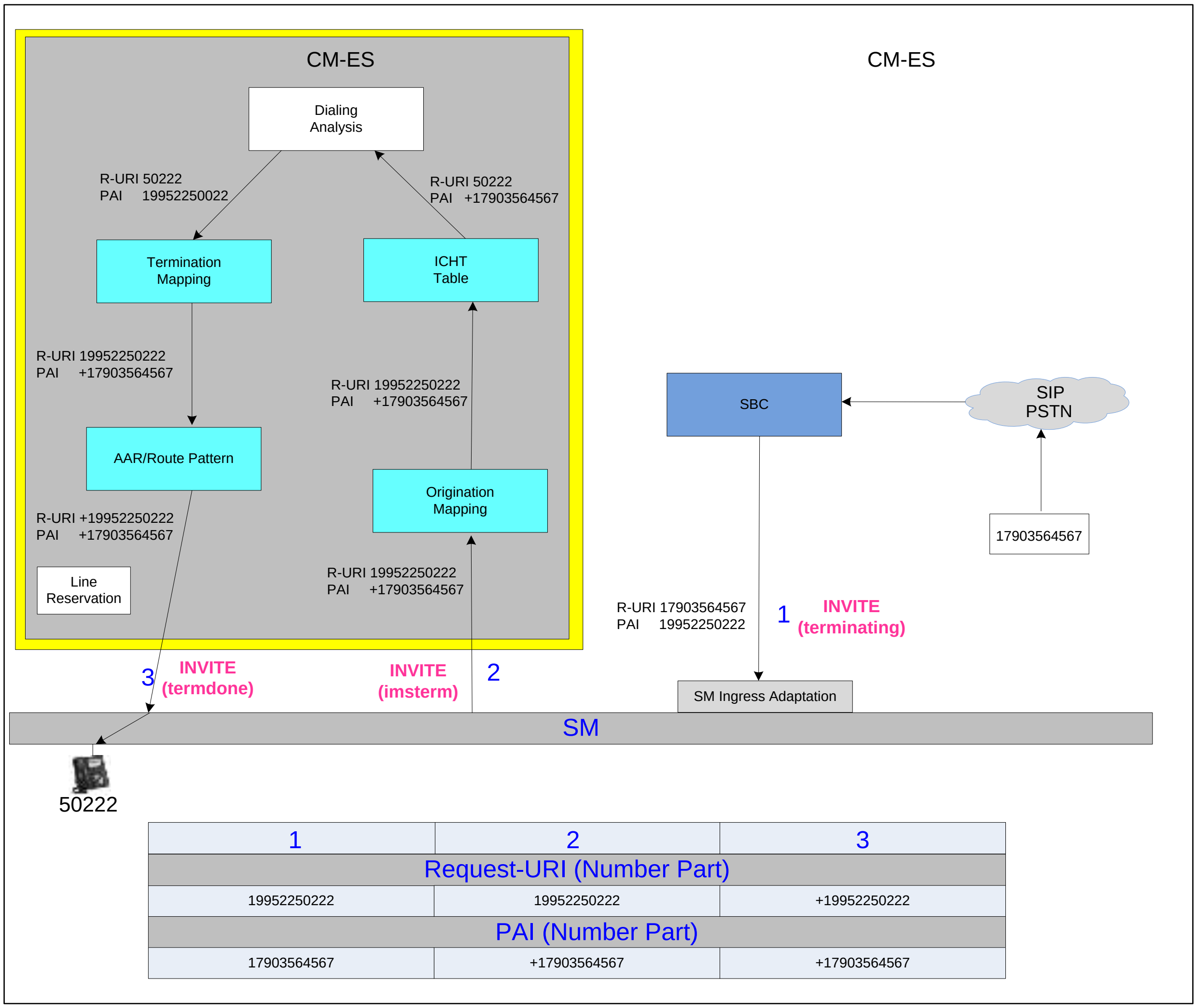


Figure 29: Inbound PSTN Call Flow in Feature or CM-ES Server-Option Three

PSTN user 17903564567 dials PSTN number 19952250222

1. Processing by SM on terminating call leg from SBC
   a. SBC sends call to SM using terminating phase.
   b. SM adapts R-URI to E.164 format +19952250222
   c. SM adapts PAI header to E.164 format +17203564567
2. Processing by SM prior to imsterm call leg to CM
   a. SM does a lookup of R-URI from SBC of +19952250222 and sees that it is a registered user and forwards the call to CM based on termination sequence administration in System Manger using preferred handle 19952250222.
   b. SM sends PAI to CM as E.164

3. Processing by CM prior to termdone call leg to SM
    a. The call flows through origination mapping and there is no match with PAI +17203564567 and sees no match.
    b. The call next flows through ICHT on SIP telephone trunk group (TG110/130) to convert R-URI from public long form 19952250222 to public short form 50222.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. The call flows through termination mapping for station 50222 for short to long form processing and since the phone number matches the extension number on the off-pbx-telephone station-mapping form, no change is made to the R-URI.
    e. CM AAR routes the call to the proper route pattern based on terminating phone number 19952250222
    f. CM sends PAI in E.164 format and R-URI in private long format to SM
4. SM now matches the R-URI with the called user profile and sends the call to the phone
5. SIP Phone displays E.164 number +17203564567

9.3.2 CM-ES and CM-FS Administration

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 (and 130)

```
change inc-call-handling-trmt trunk-group 110                      Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len       Digits
  tie              11  1995225            6
  tie              12  +1995225           7
```

CM-ES and CM-FS both use the +1995225 delete 6 digits translation for long to short form administration added to TG 110/130 based on call flows described in Section 9.3.1.

10 Option Four: Extensions Based on a Subset of Enterprise Numbering Plan

Option four extensions are based on a subset of the enterprise canonical numbering plan. This is a configuration that can be used by a smaller CM system in which shorter length extension numbers are desirable. These extension numbers are unique within the CM system, but not unique in the enterprise. The extension length in this case is usually 4 or 5-digits and reflects a subset of the enterprise canonical based handle which is usually 6 or 7 digits. The key here is that the extension number is a subset of the enterprise canonical handle used to login into the SIP phone. Following is an example based on North America (**NOTE**: only the last four digits match the E.164 number):

In this configuration there are two SIP Handles:

- Handles
  - Avaya E.164 +19952252002
  - Avaya SIP 3212002 (Private Long/Preferred Handle)
- Extension Number
  - 2002 (Private Short)

10.1 SIP Station to Station Call Flow-Option Four

10.1.1 Evolution Server

Following is an Option Four diagram of a CM-ES SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example):

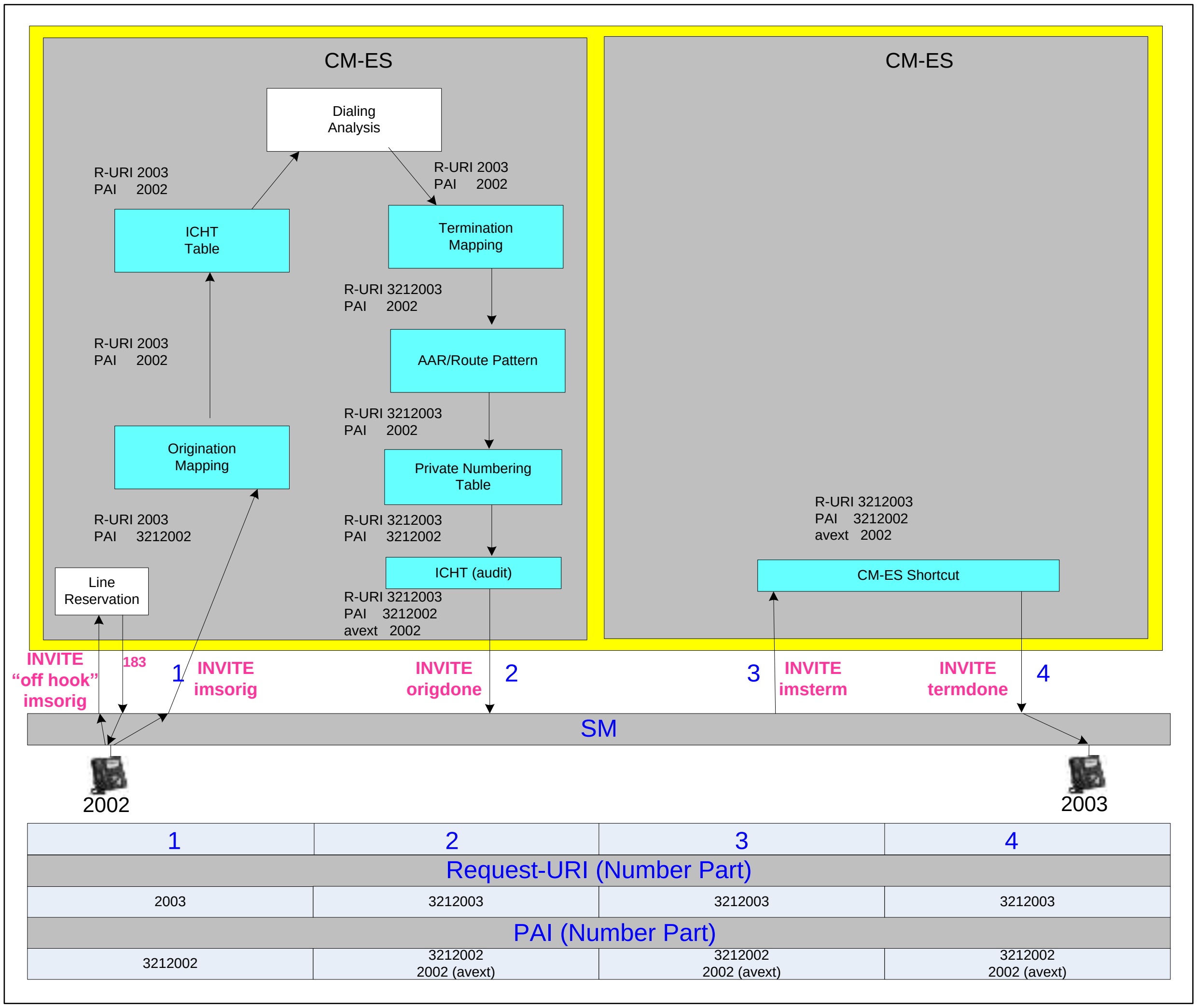


Figure 30: Evolution Server Call Flow: Option Four

In this example, 2002 is the calling SIP station and 2003 is the called SIP station. SIP station 2002 logs in as 3212002 which matches the SIP handle in SM (same is true for 2003 login). In the SIP INVITE message from the originating SIP phone the SIP handle 3212002 appears in the Contact header and the dialed digits 2003 appear in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-ES
    a. SM does a lookup of 3212002 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.

b. The PAI header in imsorig contains 3212002 since it is the preferred handle specified in SysMgr.
c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES prior to origdone call leg to SM
   a. The call flows through origination mapping and a match for phone number 3212002 is found and is replaced by the station extension number 2002 shown on the off-pbx-telephone station-mapping form; a private long to private short conversion has been done.
   b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) to change R-URI from long to short form, but since there is no match there is no change to the R-URI.
   c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
   d. CM converts extension number of terminating SIP extension 2003 (private short) to the associated phone number 3212003 (private long) using term mapping.
   e. CM AAR routes the call to the proper route pattern based on <u>terminating</u> phone number 3212003.
   f. The private-numbering table adapts the calling party information (PAI) from SIP station extension number 2002 (private short) to 3212002 (private long)
   g. CM uses ICHT to determine if the private long form of PAI (3212002) generated by the private-numbering table is based on the originating SIP station extension 2002 (private short).
      i. ICHT must have an entry that deletes the 321.
      ii. There is now a match with the originating SIP station 2002 after 321 is deleted, CM appends avext parameter with extension (private short number) to the E.164 PAI header and sends both forms back to SM as origdone rather than terminating.
      iii. If there is no match CM sends the call back to SM as “terminating” with E.164 PAI format with no avext parameter

3. Processing by SM prior to imsterm call leg to CM-ES
   a. SM now looks at Request-URI of 3202003.
   b. SM recognizes this as a handle associated with station extension 2003 (note: SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
   c. SM forwards this call back to CM based on termination sequence administration in System Manager.
   d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 3212002 (PAI) and 3212003 (R-URI).

4. Processing by CM-ES prior to termdone back to SM
   d. CM-ES does “shortcut” of imsterm and sends the SIP invite back to SM in termdone with no further processing of the call.
   e. Both PAI and R-URI are 7-digits in length.

5. SM now matches the R-URI with the called user profile and completes the call to the phone
6. SIP Phone displays avext (extension number) if available, otherwise PAI.

10.1.2 Feature Server

Following is an Option Four diagram of a CM-FS SIP station to SIP station call flow (Subscribe, Notify, and Publish messages are not shown in this example)

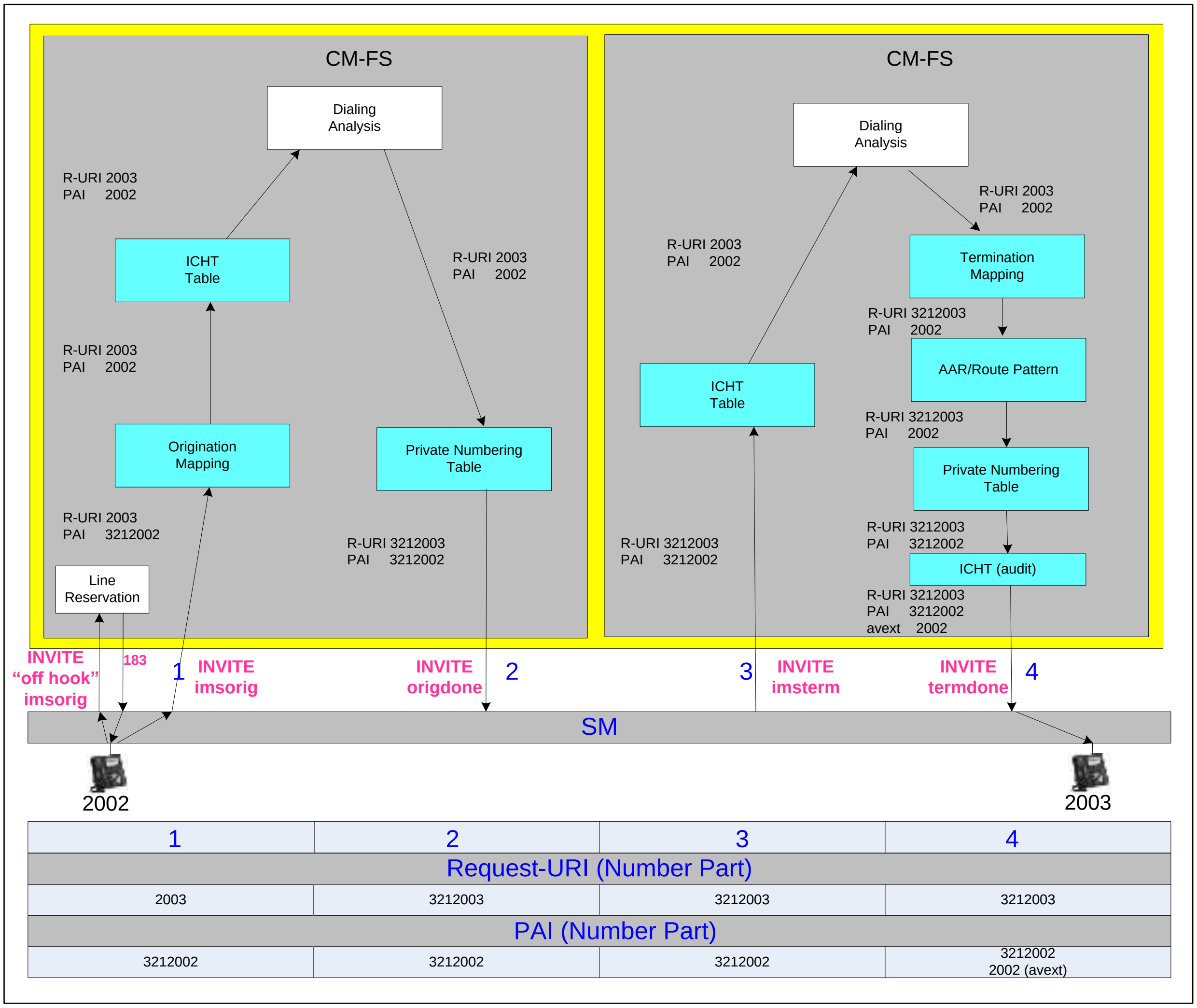


Figure 31: CM-FS Call Flow-Option Four

In this example, SIP station 2002 is the calling SIP station and 2003 is the called SIP station. SIP station 2002 logs in as 3212002 which matches the SIP handle in SM (same is true for 2003 login). The SIP INVITE message from the originating SIP phone 3212002 appears in the Contact header and 2003 from the terminating SIP phone appears in the Request-URI.

1. Processing by SM prior to imsorig call leg to CM-FS
   a. SM does a lookup of 3212002 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
   b. The PAI header in imsorig contains 3212002 since it is the preferred handle specified in SysMgr.
   c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-FS prior to origdone call leg to SM
    a. The call flows through origination mapping (private long to private short conversion) and a match for phone number 3212002 is found and is replaced by the station extension number 2002 shown on the off-pbx-telephone station-mapping form.
    b. The call next flows through the ICHT table on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    d. CM AAR routes the call to the proper route based on originating phone number 2002.
    e. The private-numbering table adapts the calling party information (PAI) AND called party information (R-URI) from SIP station extension private short form to private long form by appending "321".
    f. CM uses ICHT to determine if the private long form of PAI generated by the private numbering table is based on the originating SIP station extension 2002 (private short).
        iv. ICHT must have an entry that deletes the "321".
        v. There is now a match with the originating SIP station 2002 after "321" is deleted, CM sends the call back to SM as origdone call leg rather than terminating.
        vi. If there is no match CM sends the call back to SM as "terminating" with private long format.

3. Processing by SM prior to imsterm call leg to Feature Server
    a. SM now looks at Request-URI of 3212003
    b. SM recognizes this as a handle associated with station extension 2003 (note SM knows nothing about CM station extensions, it just knows the handles administered in System Manager).
    c. SM forwards this call back to CM based on termination sequence administration in System Manager.
    d. Both R-URI and PAI sent back to CM are based on the preferred handles administered in SM: 3212002 (PAI) and 3212003 (R-URI).

4. Processing by CM-FS prior to termdone leg back to SM
    a. CM-FS looks for match on R-URI and PAI in ICHT.
        i. This is a special case where CM-FS attempts to do long to short processing on both R-URI and PAI.
        ii. Both PAI and R-URI are 7-digits in length (private long form)
        iii. There needs to be an entry in ICHT to delete 3 digits (321) to convert both to extensions 2002 and 2003 (private short form
    b. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis.
    c. CM term mapping converts R-URI from the short form (station extension 2003) to the long form (phone number 3212003).
    d. CM AAR routes the call to the proper route pattern based on terminating phone number 3212003.
    e. The private-numbering table adapts the calling party information from SIP station extension number 2002 (private short) to 3212002 (private long)

f. CM uses ICHT to determine if the private long form of PAI (3212002) generated by the private-numbering table is based on the originating SIP station extension 2002 (private short).
    i. ICHT must have an entry that deletes the 321.
    ii. There is now a match with the originating SIP station 2002 after 321 is deleted, CM appends avext parameter with extension 2002 (private short number) to the E.164 PAI header and sends both forms back to SM as origdone rather than terminating.
    iii. If there is no match CM sends the call back to SM as “terminating” with E.164 PAI format with no avext parameter

5. SM now matches the R-URI with the called user profile and sends the call to the phone.
6. SIP Phone displays avext (extension number) if available, otherwise PAI.

10.1.3 System Manager, CM-ES and CM-FS Administration-Option Four

Following are translations for the SIP user in System Manager User Profile:

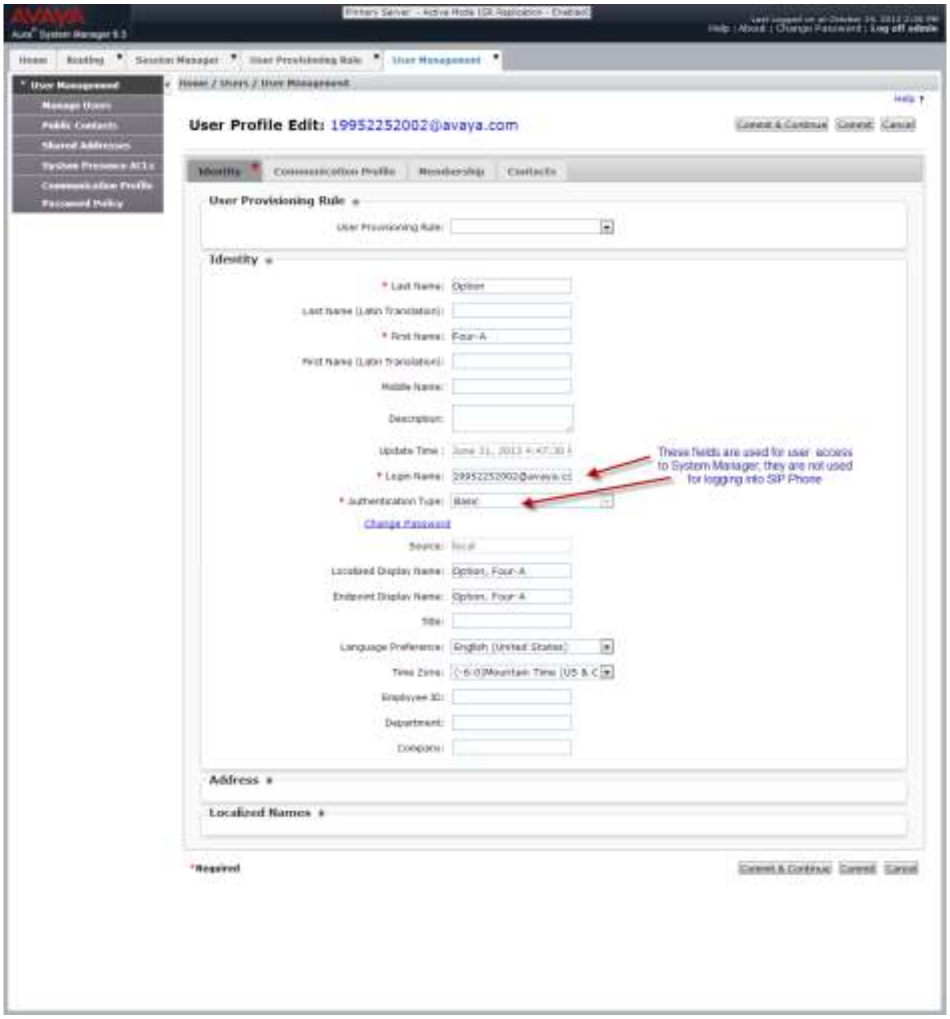


Figure 32: System Manager User Profile Identity

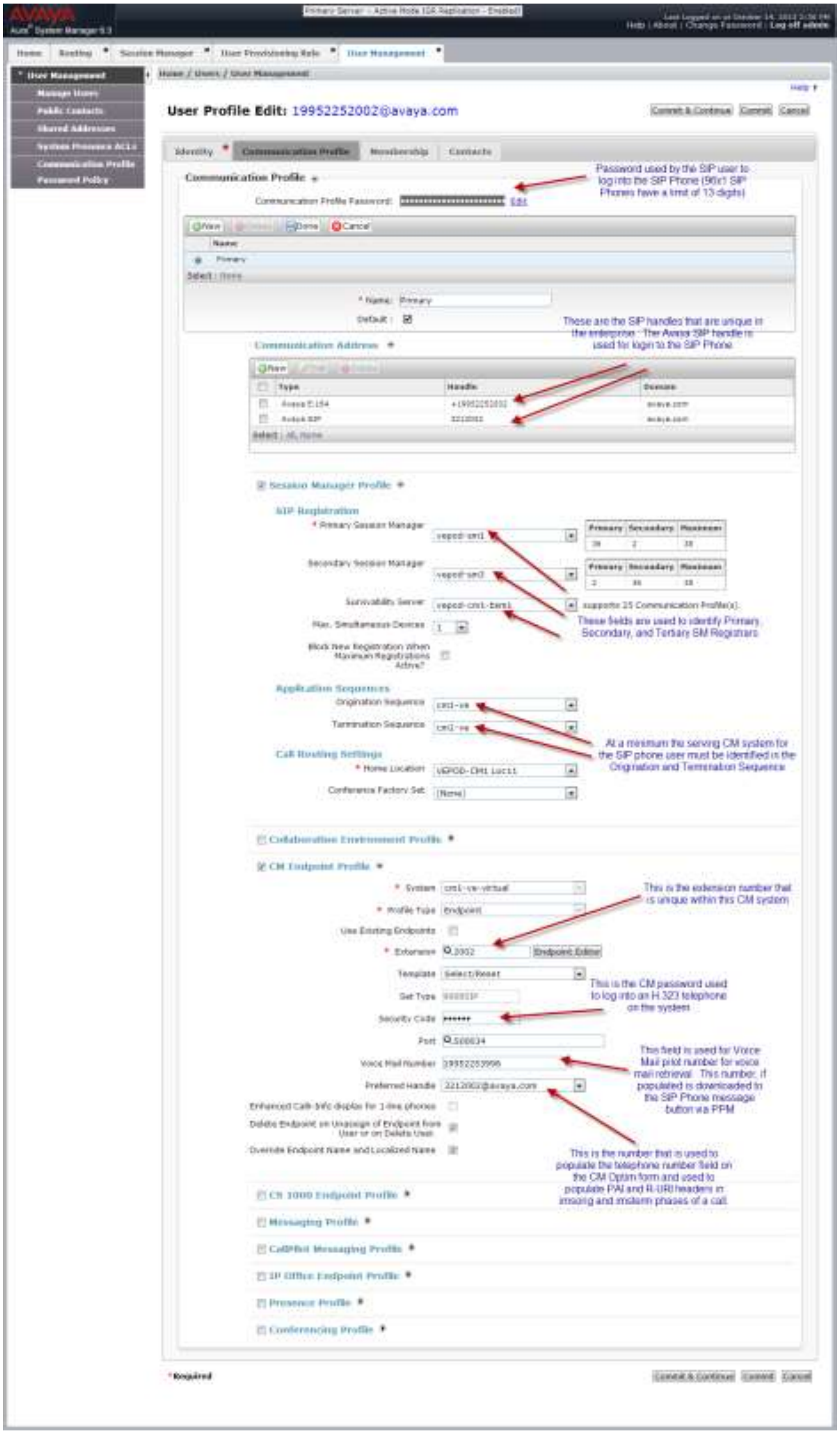


Figure 33: System Manager User Communication Profile

System Manager automatically populates CM station as well as off-pbx-telephone station-mapping forms with the following translations using the selected template for both Evolution and Feature Server:

```
display station 2002                                            Page   1 of   6
                                     STATION

Extension: 2002                         Lock Messages? n              BCC: 0
     Type: 9608SIP                      Security Code: 123456          TN: 1
     Port: S00245                     Coverage Path 1:                COR: 1
     Name: Option, Four-A             Coverage Path 2:                COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                Location:                 Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 2002

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n
```

```
display station 2002                                             Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                          Headset? n
       Jack:                                          Speaker? n
      Cable:                                         Mounting: d
      Floor:                                      Cord Length: 0
   Building:                                        Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:
```

```
display station 2002                                             Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

Note that the default routing for this station is aar and that the phone will get three call appearances.

System Manager also populates the off-pbx telephone station mapping form with the application type OPS. System Manager 6.2 uses the CM Endpoint Profile “Preferred Handle” in the User Communication Profile to populate the “phone number” field in CM.

```
Display   off-pbx-telephone station-mapping 2002                Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2002    OPS          -              3212002          aar         1
 2003    OPS          -              3212003          aar         1
```

The off-pbx-telephone station-mapping form is used for long to short digit manipulation of the calling station on the origination side of the call (Phone Number to Station Extension). The off-pbx-telephone station-mapping form is used for short to long digit manipulation of the called station on the termination side of the call (Station Extension to Phone Number).

Prior to administering the System Manager User Profile the following **minimum** administration must be done in Communication Manager (Note that administration that follows is the same for Evolution and Feature Server):

The dialplan analysis form has the following administration:

- The dialed string 2, 4-digits in length to support 4-digit extensions (2002 & 2003).
- The dialed string 8 is for Automatic Alternate Routing (AAR) and dialed string 9 for Automatic Route Selection (ARS). The AAR and ARS feature access codes must be defined.
- The dialed string *, 4-digits in length to accommodate SIP trunk dial access codes.

```
display dialplan analysis                                    Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 5

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   2           4   ext
   8           1   fac
   9           1   fac
   *           4   dac
```

Following is the minimum translations for system features:

```
display feature-access-codes                                 Page   1 of  11
                               FEATURE ACCESS CODE (FAC)
      Auto Alternate Routing (AAR) Access Code:  8
    Auto Route Selection (ARS) - Access Code 1:  9    Access Code 2:
```

The AAR/ARS codes do not have to be what is shown here, but they do need to be administered.

A dedicated signaling group(s) needs to be set up for use by the SIP telephones to the primary SM and secondary SM. In the example there are actually four SMs: vepod-sm1 and vepod-sm2 in data center one and vepod-sm3 and vepod-sm4 in data center two. Assume that for SIP telephones in this example that vepod-sm1 is the primary SM and vepod-sm3 is the secondary SM. The Signaling group to vepod-sm1 is 910, and the signaling group to vepod-sm3 is 930. Administration for Signaling group 910 to vepod-sm1 is shown here:

```
Display signaling-group 910                                    Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910             Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: VEPOD-SM1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:
                                    Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

In this example IMS Enabled field is set to n. This is the proper setting for a CM-ES. Setting of the IMS field is what determines whether or not calls follow the full call model on CM-ES or half call model on CM-FS. It is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

For CM-FS the signaling group needs to be set up for use by the SIP telephones to each SM with IMS Enabled set to y (in the example there are two SMs), all other administration is the same as CM-ES.

```
display signaling-group 910                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 910             Group Type: sip
  IMS Enabled? y         Transport Method: tls
        Q-SIP? n
     IP Video? y           Priority Video? n        Enforce SIPS URI for SRTP? Y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                  Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:
                                    Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from SIP stations (imsorig) have a domain of avaya.com and CM-ES will select this signaling group.

Following is administration for SIP trunk group associated with the SIP signaling group 910.

```
display trunk-group 910                                             Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y

                     Numbering Format: private
                                                 UUI Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? N
```

As discussed in Section 6.1.4, the “Numbering Format” field on page 3 of the form is used to determine whether to use the public or private numbering table for various call flows in Evolution and Feature Server. In this example the numbering format used is private; on-net calls and SIP station to station calls use the private table.

In CM-ES, station to station calls use the routing associated with the terminating OPS phone number 3212003 for origdone call leg (no routing associated on termdone since CM-ES shortcuts the term side of the call. In CM-FS, station to station calls use routing associated with the originating OPS phone number 3212002 for the origdone call leg and terminating OPS phone number 3213003 on the call for termdone call leg.

Routing for 3212002 and 3213003 is based on AAR and uses route pattern 910.

```
display aar analysis 0                                                       Page   1

                           AAR DIGIT ANALYSIS REPORT

                            Location:  all
               Dialed           Total        Route    Call      Node
               String         Min    Max    Pattern   Type     Number
         321                   7      7      910      aar
```

The call type in this example is “aar” which means that CM will use the public-unknown-numbering table even if numbering format on the chosen trunk group is public or private unless overridden on the route pattern form.

```
display route-pattern 910                                       Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP SM1 and SM3
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0                                                             n   user
 2: 2    0                                                             n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                unk-unk  next
 2: y y y y y n  n            rest                                unk-unk  next
```

In this example look ahead routing has been implemented in the event that CM gets failure messages on vepod-sm1 it can retry on vepod-sm3 using trunk group 930. The preferences in this route pattern match the primary and secondary SM specified for the SIP users in System Manager based on the best practice cited in Section 6.5. The numbering format on this route pattern is set to unk-unk. Since trunk group 910 and 930 are private trunks the private-numbering table is used based on the CM algorithm specified in section 6.1.4.

Following is administration in the private-numbering table.

```
display private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 4  2                         321              7     Total Administered: 5
                                                      Maximum Entries: 540
```

CM-ES and CM-FS both use this translation for short to long form administration based on call flows described in Section 10.1.1 and 10.1.2.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 910 (and 930)

```
display inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
tie             7  321              3[6]
```

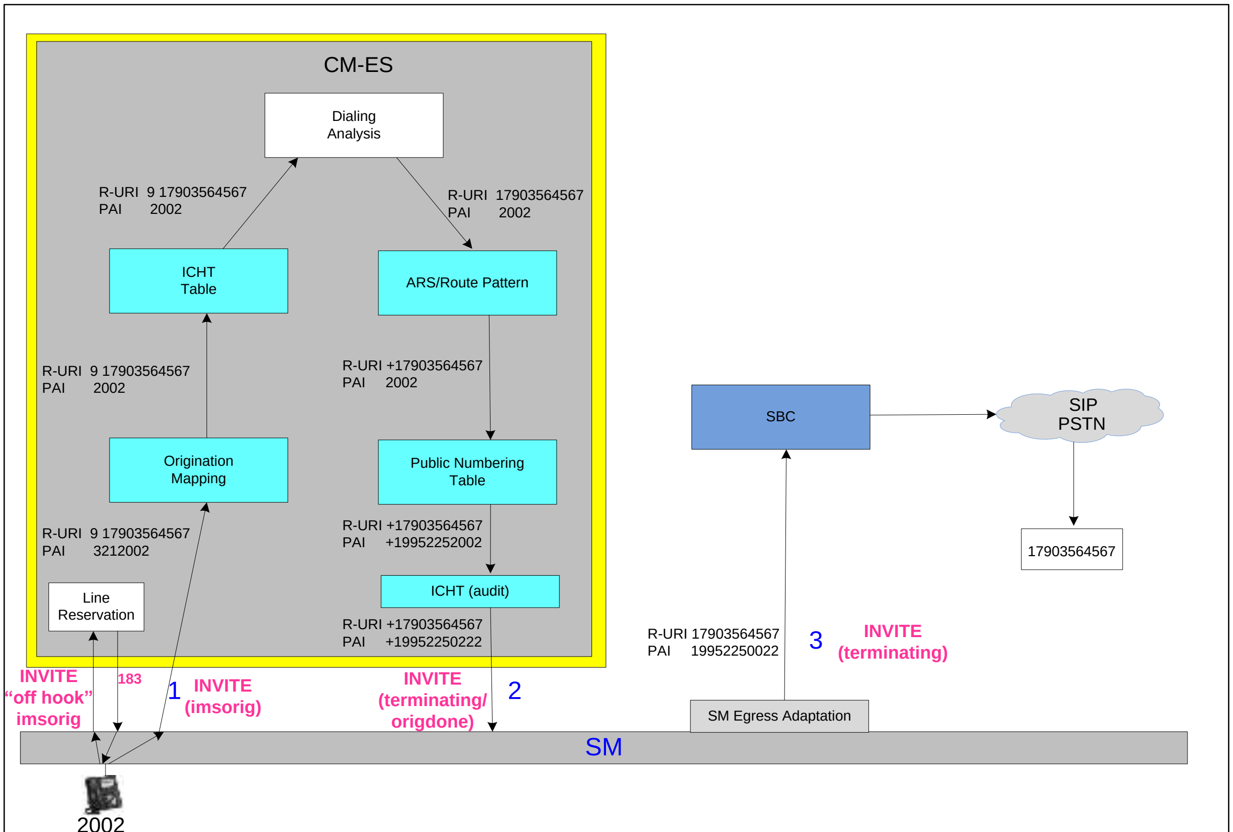
CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 10.1.1 and 10.1.2.

[6] This entry is also used to convert from long to short form for Subscribe messages rather than using off-pbx-telephone station-mapping (see section 6.1.5).

10.2 SIP Station to Outbound SIP PSTN Call Flow-Option Four

10.2.1 CM-ES and CM-FS

Following is call flow for 11-digit North American Number to PSTN from station extension 2222 to 1720-356-4567:



1	2	3
Request-URI (Number Part)		
9 17903564567	+17903564567	17903564567
PAI (Number Part)		
3212002	+19952252002	19952252002

Figure 34: PSTN Call Flow in Feature or Evolution Server-Option Four

End user dials the ARS access code 9 and the PSTN number

1. Processing by SM prior to imsorig call leg to CM-FS is the same as for CM-ES
    a. SM does a lookup of 3212002 and sees that it is a registered user and forwards the call to CM based on origination sequence administration in System Manger.
    b. The PAI header in imsorig contains 3212002 since it is the preferred handle specified.
    c. The R-URI contains the digits dialed by the end user and is not looked at by SM.

2. Processing by CM-ES and CM-FS prior to origdone call leg to SM
    a. The call flows through origination mapping for station 2002 and since the phone number 3212002 does not match the extension number 2002 on the off-pbx-telephone station-mapping form it converts to the extension number 2002 (private short form).
    b. The call next flows through ICHT on SIP telephone trunk group on SIP telephone trunk group (TG910/930) and since there is no match there is no change to the Request URI.
    c. Call processing now proceeds through Dialing analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. Call is routed to proper route pattern
    e. The public-unknown numbering table is now used to adapt the calling party information from private short number to E.164 +1995225 prior to sending the call to SM in the origdone leg of the call.
    f. CM uses ICHT on SIP PSTN trunk group (TG110/130) to determine if the E.164 form of PAI generated by the public-unknown-numbering table is based on the originating SIP station extension 2002 (private short).
        i. If ICHT has an entry that deletes "+1995225" there is now a match with the originating SIP station 2002.
            1. CM sends the call back to SM as origdone call leg with PAI in E.164 format.
            2. Explicit sequencing of origination applications after CM requires origdone call processing.
            3. CM <u>always</u> sends signaling for origdone call leg back to the same SM that initiated imsorig call processing regardless of what is specified in AAR/ARS routing;
            4. If AAR/ARS routing for origdone is different than the SM used for imsorig, CM call processing still shows use of signaling group/trunk group specified in ARS/AAR.
            5. Since this is not a station to station call, avext is not appended to PAI header.
        ii. If ICHT does NOT have an entry that deletes the +1995225 there is no match with the originating station 2002.
            1. CM sends call back to SM as terminating call leg with PAI in E.164 format
            2. Implicit sequencing, including Collaboration Environment is supported since CE does not require origdone call processing.
            3. <u>Terminating call legs unlike origdone call legs do not need to return to the same SM that initiated imsorig call processing.</u>
            4. In this case, CM sends the terminating call leg to SM specified in ARS routing as "terminating" even if it is different than the SM used for imsorig.
            5. CM call processing shows trunk group usage to SM chosen by ARS.
    g. The "p" in the route pattern inserts the "+" on the R-URI
    h. CM sends PAI and R-URI numbers to SM in E.164 format[7]

[7] Additional considerations need to be applied to international calls. In North America, the international prefix dialed is "011" (many other parts of the world it is "00"). These digits can be deleted on an international route pattern and the "+" inserted or the call can be sent to SM with the international prefix. In this case, SM would have an adaptation to delete the international prefix and insert "+" for analysis and routing. Here is a case where an SM ingress adaptation is being used in SM that does not impact SIP telephone call processing

3. Processing by SM prior to terminating to SBC
    a. SM determines routing policy
    b. Applies egress adaptation to SBC based on SBC and PSTN requirements

10.2.2 CM-ES and CM-FS Administration

Separate signaling group(s) need to be set up for access to the PSTN SIP trunks to each SM (in the example assume two SMs). Administration for signaling group to vepod-sm1 is shown here also need signaling group 130 to vepod-SM3):

```
display signaling-group 110                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 110             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

   Near-end Node Name: procr                    Far-end Node Name: vepod-sm1
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region: 241
                                      Far-end Secondary Node Name:
Far-end Domain: sbccore.avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

In this example it is assumed that the PAI of any inbound calls from PSTN trunks (imsterm) have a domain of sbccore.avaya.com and CM will select this signaling group.

Following is a sample trunk used for PSTN access via SM:

```
display trunk-group 110                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 110                  Group Type: sip          CDR Reports: y
  Group Name: SIP PSTN SM1                COR: 1        TN: 1        TAC: *110
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 110
                                                   Number of Members: 15
```

```
display trunk-group 110                                         Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y

                     Numbering Format: public
```

```
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y

 DSN Term? n                     SIP ANAT Supported? n
```

Since “public” is specified in the “Numbering Format” field all calls to this trunk group will use the “public-unknown numbering table”

Following is a simple ARS Analysis table for access to SIP PSTN trunks:

```
display ars analysis 0                                         Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                  Location: all          Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    011                   10   18    110      intl         n
    1                     11   11    110      natl         n
    911                   3    3     911      emer         n
```

Assume that 11 digit North America numbers are being dial and are then routed using route pattern 110 as specified in ARS Analysis:

```
display route-pattern 110                                      Page   1 of   3
                    Pattern Number: 110    Pattern Name: SBC DC1
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 110  0                        p                                    n   user
 2: 130  0                        p                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                       next
 2: y y y y y n  n            rest                                       next
```

Route pattern 110 inserts the + to the Request-URI based on the “p” entered in the inserted digits field so that SM can route on E.164 number. In North America 1+10 digits is in E.164 format if the + is appended to the dial string. In addition, route pattern 110 the numbering format field is not applicable since these are public trunks.

Based on the algorithm for non-ims signaling groups, this call will use the public-unknown-numbering table. This table is filled out as follows:

```
display public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN            CPN
Len Code            Grp(s)    Prefix         Len
                                                 Total Administered: 8
 4  2                          1995225        11      Maximum Entries: 9999
                                                 Note: If an entry applies to
                                                 a SIP connection to Avaya
                                                 Aura(R) SM,
                                                 the resulting number must
                                                 be a complete E.164 number.
```

CM-ES and CM-FS both use this translation for short to long form administration based on call flows described in Section 10.2.1.

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 (and 130):

```
change inc-call-handling-trmt trunk-group 110                   Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             7  +1995225         8   321
```

CM-ES and CM-FS both use this translation for long to short form administration based on call flows described in Section 10.2.1.

10.3 Inbound SIP PSTN to SIP Station Call Flow-Option Four

10.3.1 CM-ES and CM-FS

Following is the call flow for number dialed from the PSTN to extension 2002

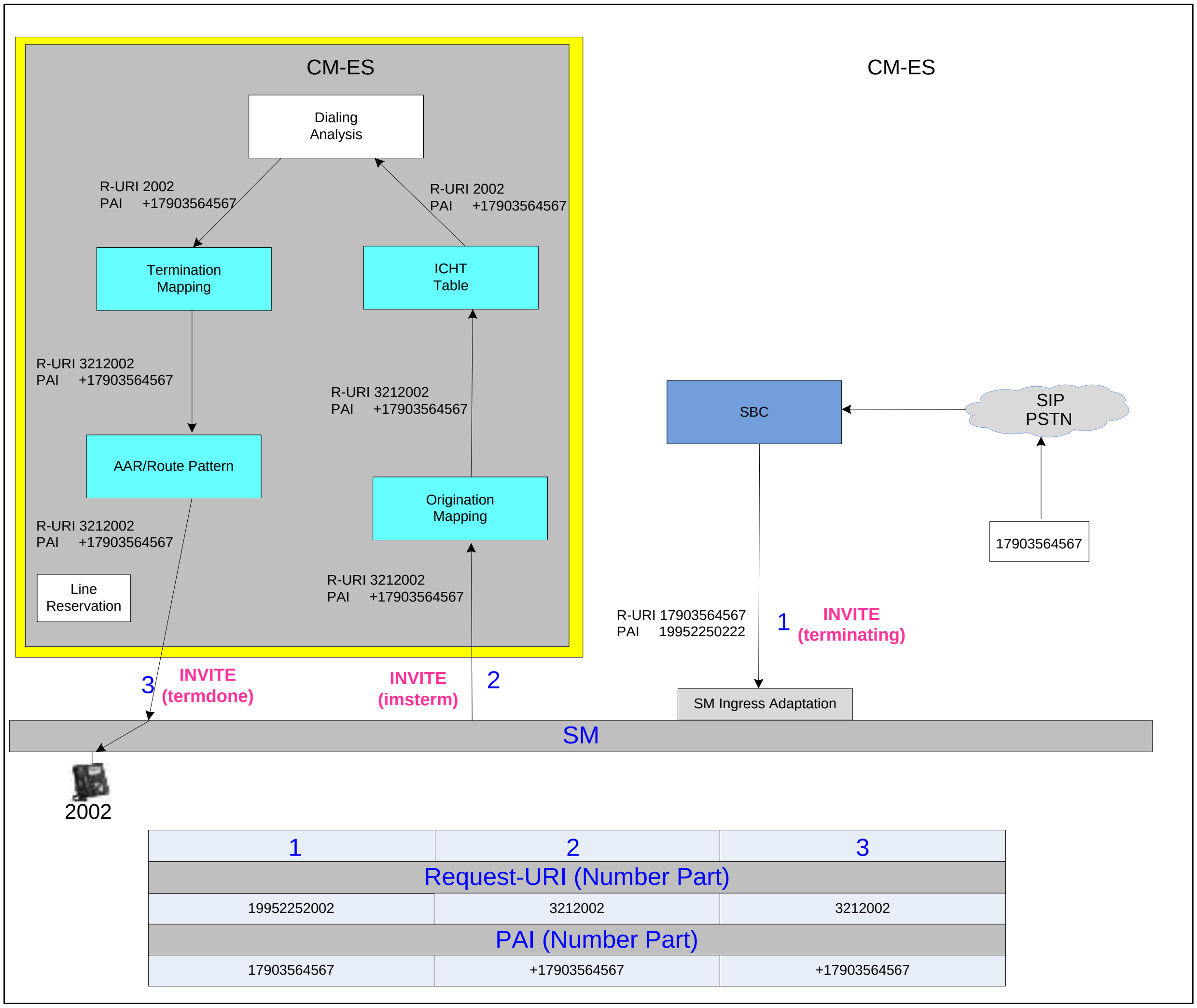


Figure 35: Inbound PSTN Call Flow in Feature or CM-ES Server-Option One

PSTN user 17203564567 dials PSTN number 19952250222

1. Processing by SM on terminating call leg from SBC

    a. SBC sends call to SM using terminating phase.
    b. SM adapts R-URI to E.164 format +19952250222
    c. SM adapts PAI header to E.164 format +17203564567

2. Processing by SM prior to imsterm call leg to CM

a. SM does a lookup of R-URI from SBC of +19952250222 and sees that it is a registered user and forwards the call to CM based on termination sequence administration in System Manger using preferred handle 3212002.
b. SM sends PAI to CM as E.164

3. Processing by CM prior to termdone call leg to SM

    a. The call flows through origination mapping and there is no match with PAI +17203564567 and sees no match.
    b. The call next flows through ICHT on SIP telephone trunk group (TG110/130) to convert R-URI from private long form 3212002 to public short form 2002.
    c. Call processing now proceeds through Dialing Analysis which includes: dialplan analysis, uniform dialplan, and/or calltype analysis and in this case ARS analysis.
    d. The call flows through termination mapping for station 2002 for short to long form processing and the extension number 2002 maps to phone 3212002.
    e. CM AAR routes the call to the proper route pattern based on terminating phone number 3212002
    f. CM sends PAI in E.164 format and R-URI in private long format to SM

4. SM now matches the R-URI with the called user profile and sends the call to the phone
5. SIP Phone displays number 17203564567

10.3.2 CM-ES and CM-FS Administration

Following is administration for the incoming call handling treatment (ICHT) table for trunk group 110 (and 130)

```
display inc-call-handling-trmt trunk-group 1                          Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
tie             7   321             3
```

CM-ES and CM-FS both use the +1995225 delete 6 digits translation for long to short form administration added to TG 110/130 based on call flows described in Section 10.3.1.

11 Domain Considerations

Domains also come into play in CM call processing. The handle of the SIP user in System Manager contains a domain as well as a phone number. The domain is used to populate SIP messages during Registration and Subscription and in all subsequent call flows.

The domain specified in the phone must match the domain administered in System Manager User Profile for successful registration. In the examples shown a single domain of avaya.com is used. The domain for the phone can be set in the 46xxsetting.txt file using the following:

```
SET SIPDOMAIN    "avaya.com"
```

This setting can be also administered on the SIP phone through the craft interface. It is important that the authoritative domain be set on the ip-network-region of interface used on the near end of the signaling group used by the SIP phones. In most cases this will be the NR associated with PROCR. This is necessary for successful subscription of the SIP phone. In addition, the far end domain needs to be set on the signaling group form.