Interactive Intelligence Messaging Interaction Center 2.4

PBX Configuration Note:

Avaya s8300 with MIC using SIP

Technical Reference

By Interactive Intelligence, Inc.

READ THIS BEFORE YOU PROCEED

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Customer Interaction Center®
Vonexus Enterprise Interaction Center®

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Content

This document describes the configuration required to setup Avaya S8300 to interoperate with Interactive Intelligence Messaging Interaction Center 2.4 (MIC) using SIP.

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Interaction Center Platform Statement

This document describes Interaction Center (IC) features that may not be available in your IC product. Several products are based on the IC platform, and some features are disabled in some products.

These products are based on the IC platform:

- Customer Interaction Center (CIC)
- Vonexus Enterprise Interaction Center (Vonexus EIC, or EIC)
- Message Interaction Center (MIC)

While all of these products share a common feature set, this document is intended for use with all IC products, and some of the described features may not be available in your product.

Table of Contents

Who should read this document

This document is intended for Systems Integrators with significant telephony knowledge.

Technical Support

The information contained within this document has been provided by Interactive Intelligence, its partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX. Improper configuration may result in the loss of service of the PBX. Interactive Intelligence is unable to provide support or assistance with the configuration or troubleshooting of components described within. Interactive Intelligence recommends readers to engage the service of an Interactive Intelligence MIC Certified Engineer or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Messaging Interaction Center.

Known Issues

REFER with Replaces messages will not release IP resources in this format, and present a dead air condition using the default handlers. This will be resolved in a future version of xIC, per SCRs 62854 and 62855.

The current workaround is to ensure that any handlers containing the Transfer tool step have the “Use Putback” input hardcoded to false.

Chapter 1: General Information

Components

PBX or IP-PBX

PBX Vendor	Avaya
Model	S8300
Software Version	Communication Manager R015x.01.0.414.3
Telephony Signaling	SIP
Additional Notes	None

Interactive Intelligence Messaging Interaction Center

Version	2.4 SU 28 + MIC SU 8

Prerequisites

PBX Prerequisites

- SIP option – Avaya SIP Enablement Services (SES) is NOT required

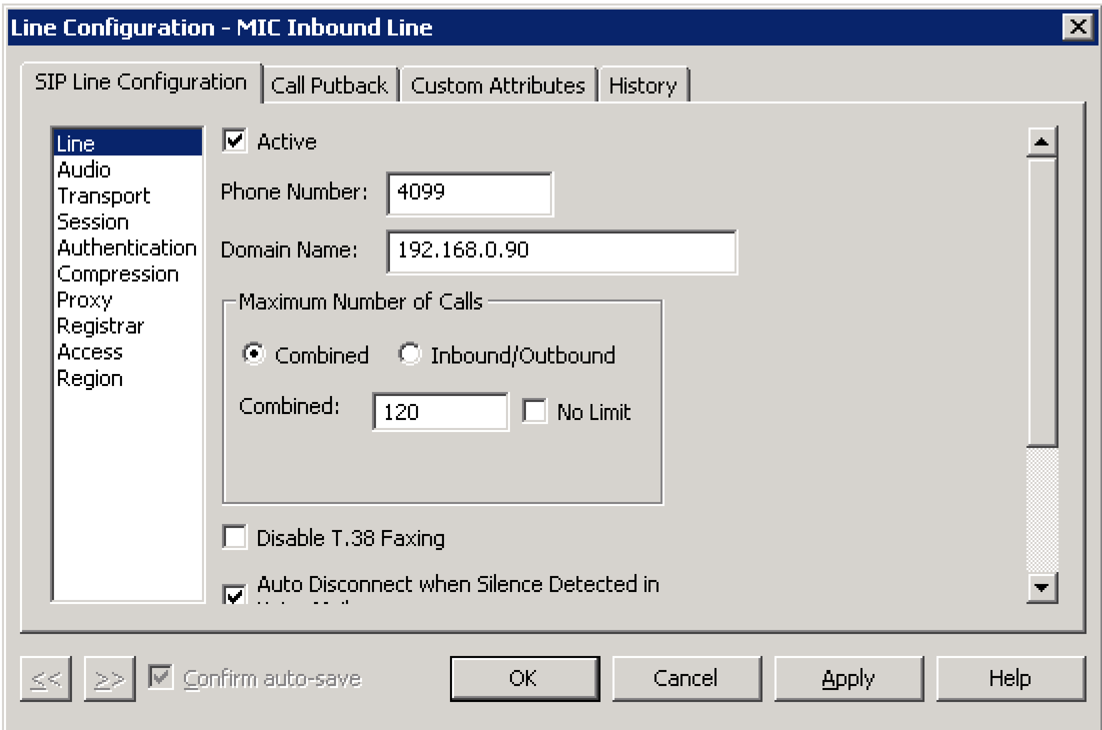
Summary and Limitations

REFER with Replaces messages will not release IP resources in this format, and present a dead air condition using the default handlers. This will be resolved in a future version of xIC, per SCRs 62854 and 62855.

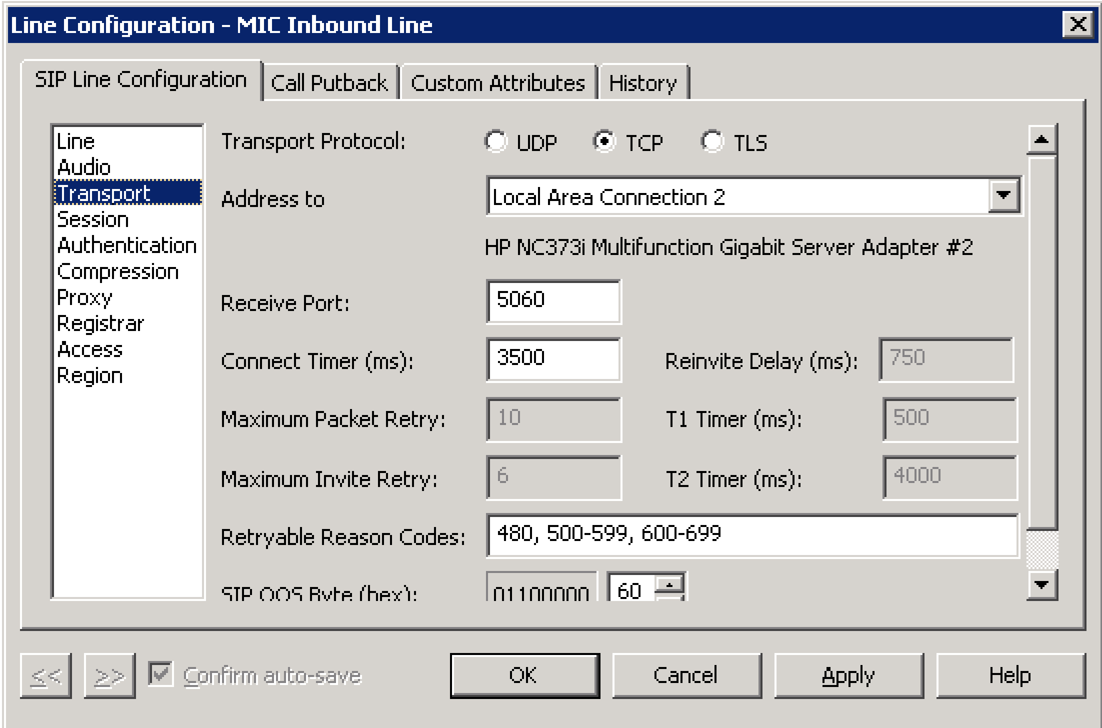
The current workaround is to ensure that any handlers containing the Transfer tool step have the “Use Putback” input hardcoded to false.

Chapter 2: xIC Setup

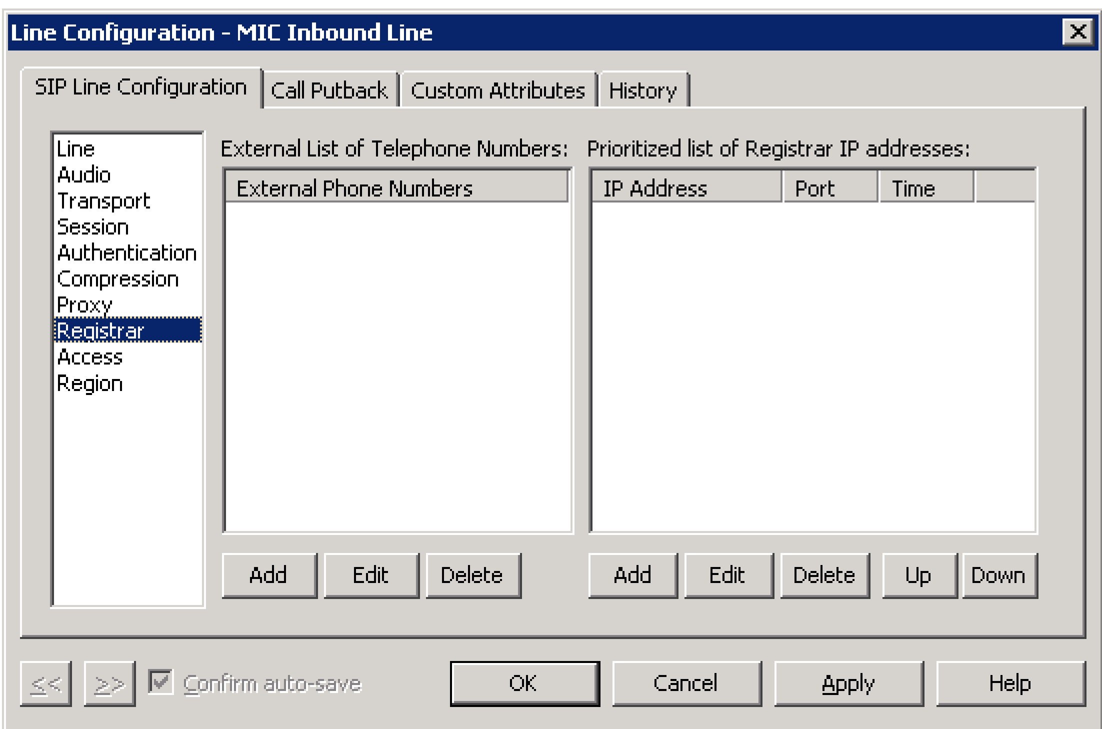
Step 1: Create the SIP Line



Take note of the Domain Name value here. It must match the value of the domain name in the Far-end Domain in the Avaya SIP Signaling Group.



The transport must be TCP to communicate with the Avaya.



The line does NOT need to register with the AVAYA.

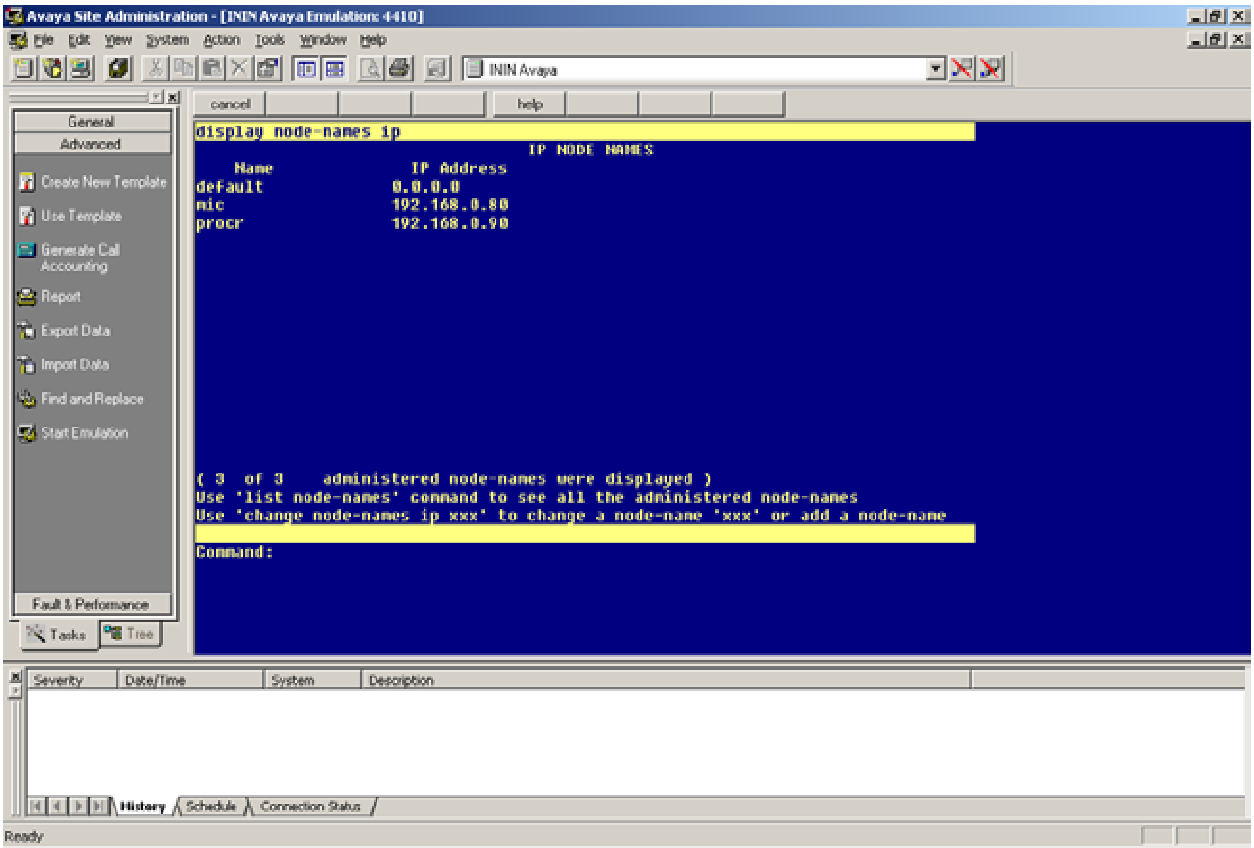
Chapter 3: PBX Setup

Information used for this test case:

- VoiceMail Hunt Group Pilot: ext. 4099

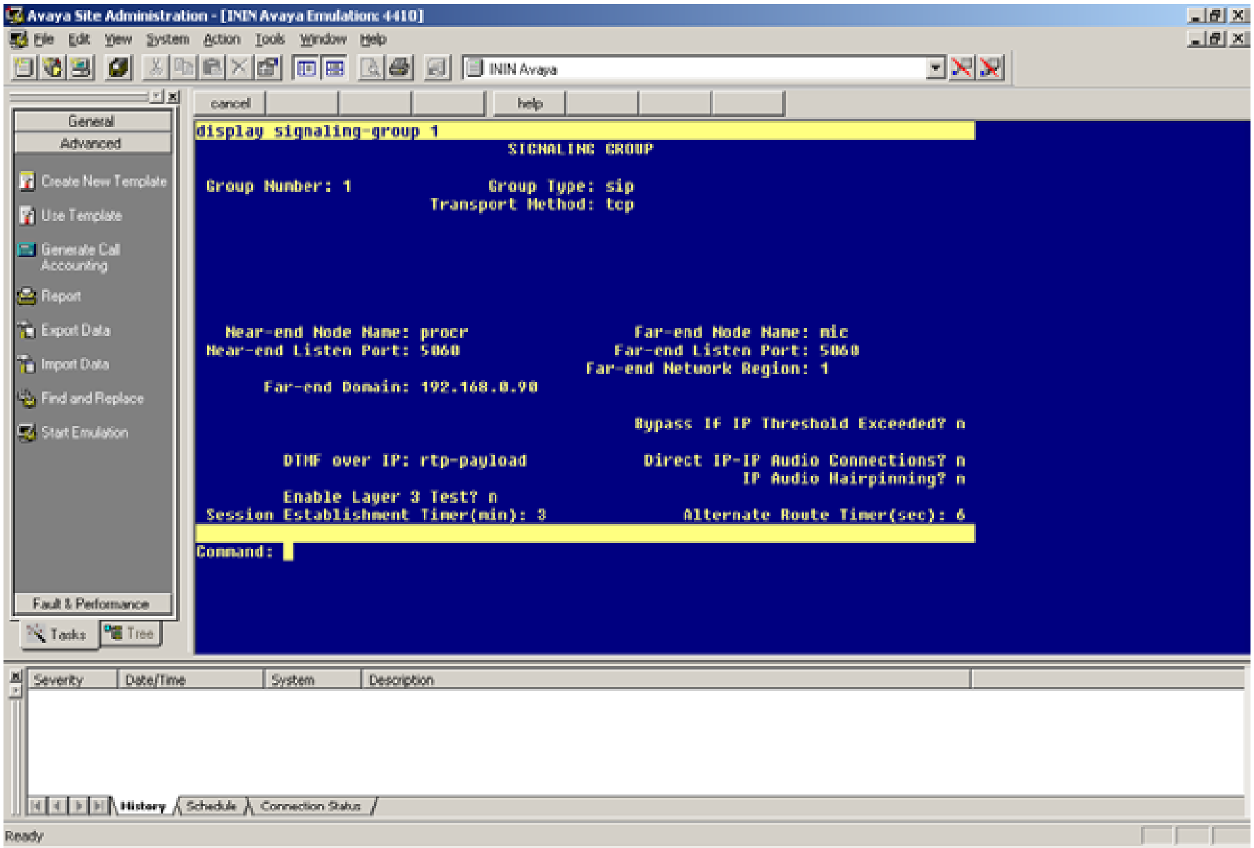
Step 1: Add xIC server as a valid IP Node

change node-names ip



Step 2: Configure the Signaling Group

add trunk-group X (X is the SIP trunk group number)

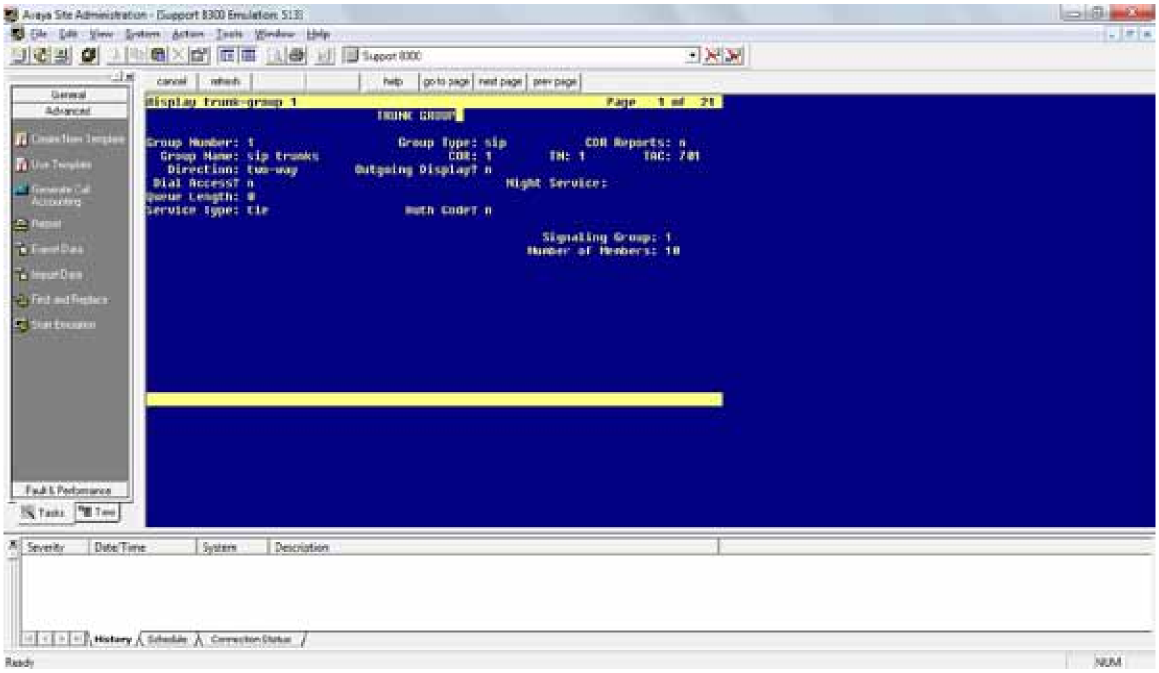


Notice that the Far-end Domain matches the domain name on the SIP line in xIC. This is also the value used in the MWIIPAddress Server Parameter (or the host portion of the MWI Extension for users where the complete URI is specified in Web Admin).

This value must be identical in all three places, or the Avaya will reject requests from the xIC server.

Step 3: Configure Trunk Group

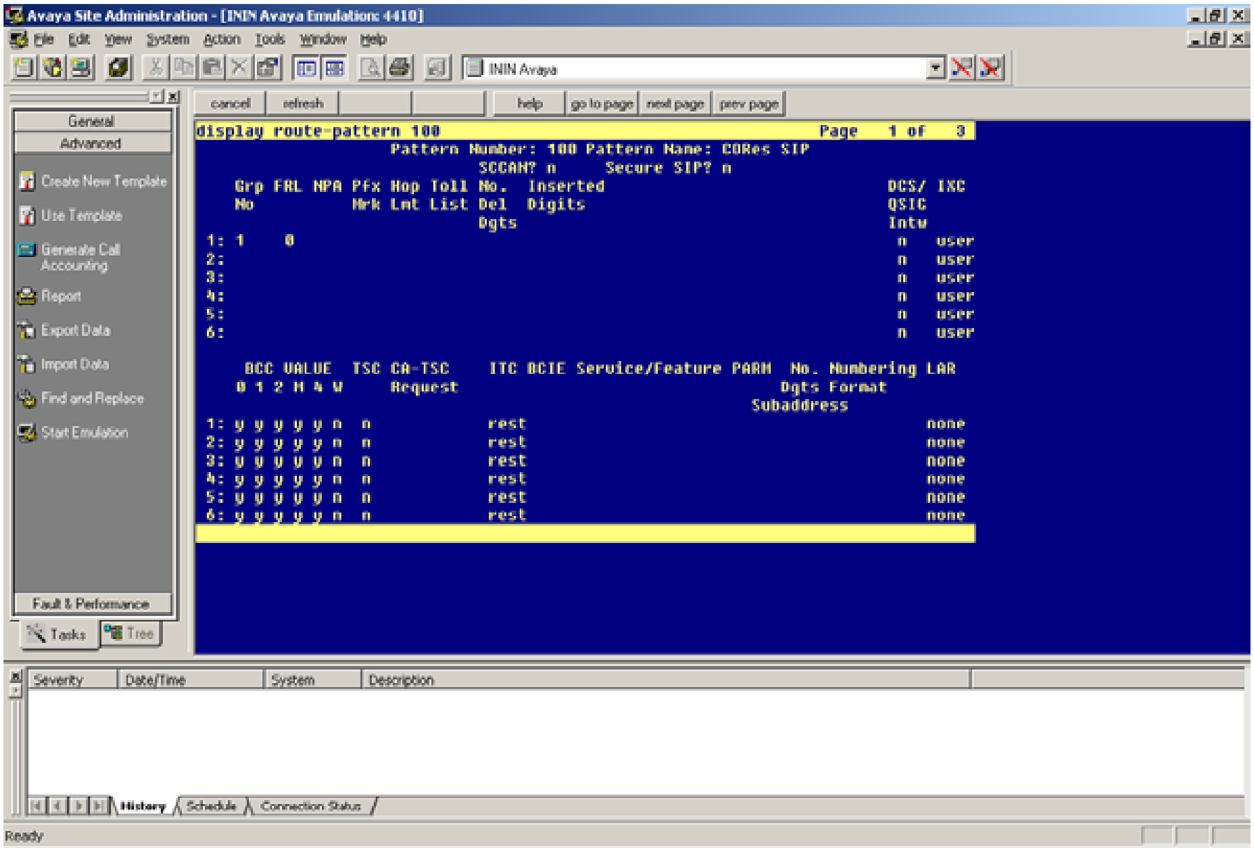
add trunk-group X (X is the route number you wish to use)



Notice the Service Type is tie, and that the Signaling Group is the Signaling Group setup in the previous step.

Step 4: Configure Route Pattern

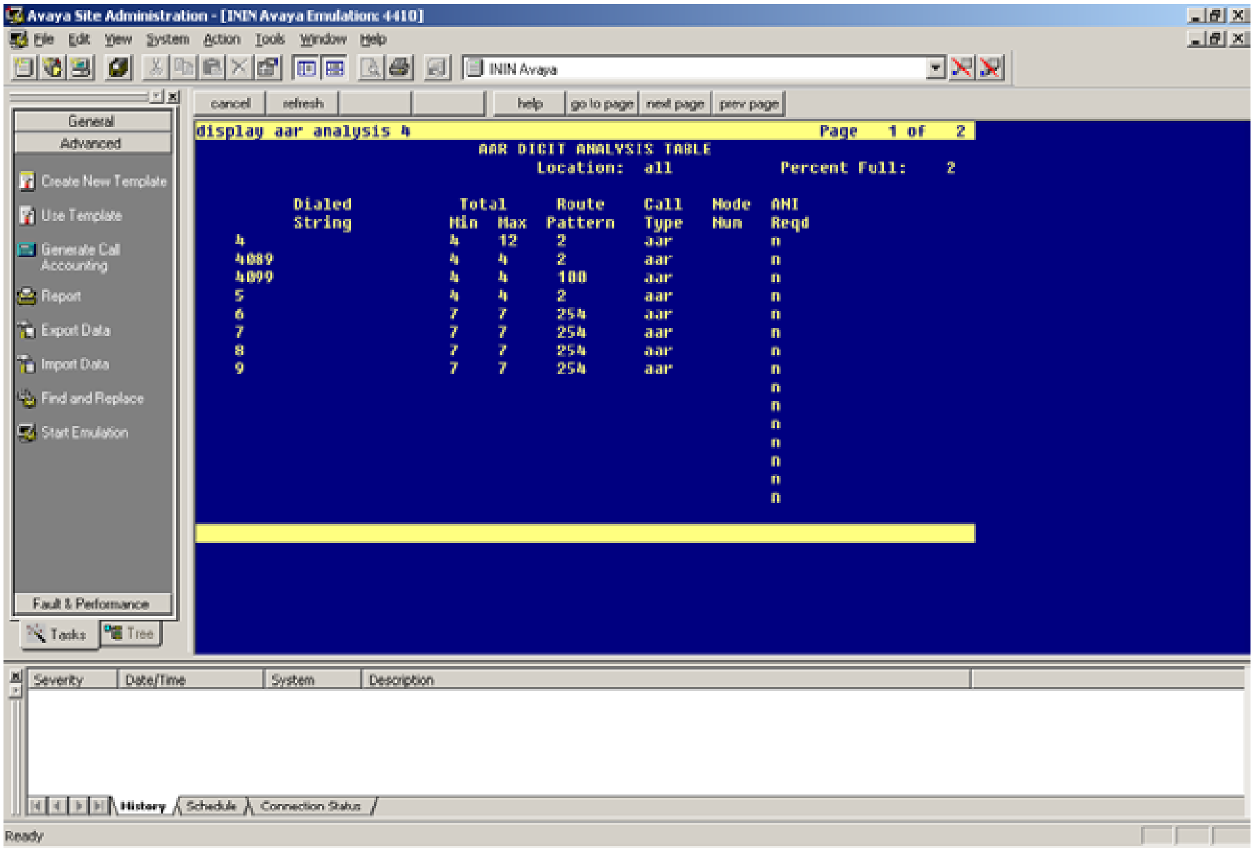
add route-pattern X (X is the route number you wish to use)



Notice the Group Number matches the Trunk Group Number.

Step 5: Configure AAR Analysis

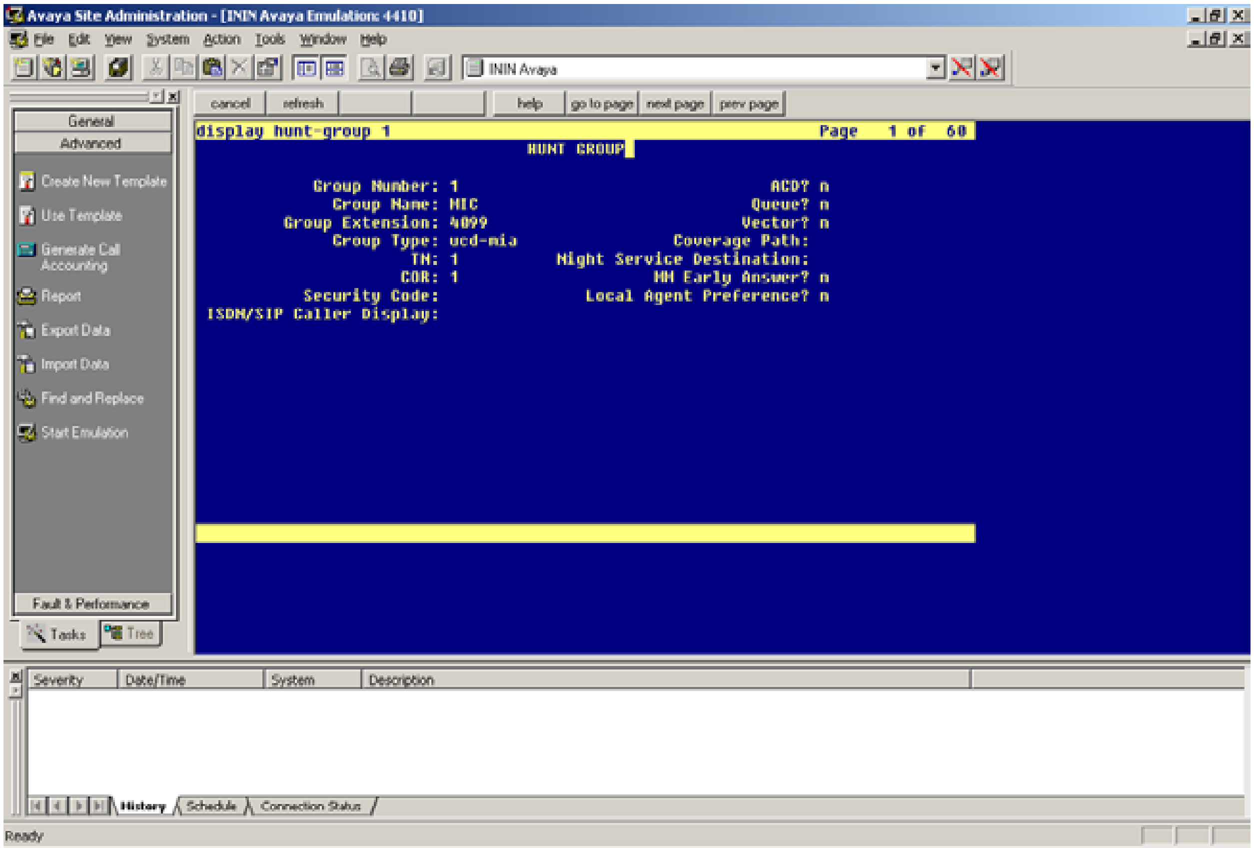
change aar analysis X (X is the leading digit of your VM Pilot Number)



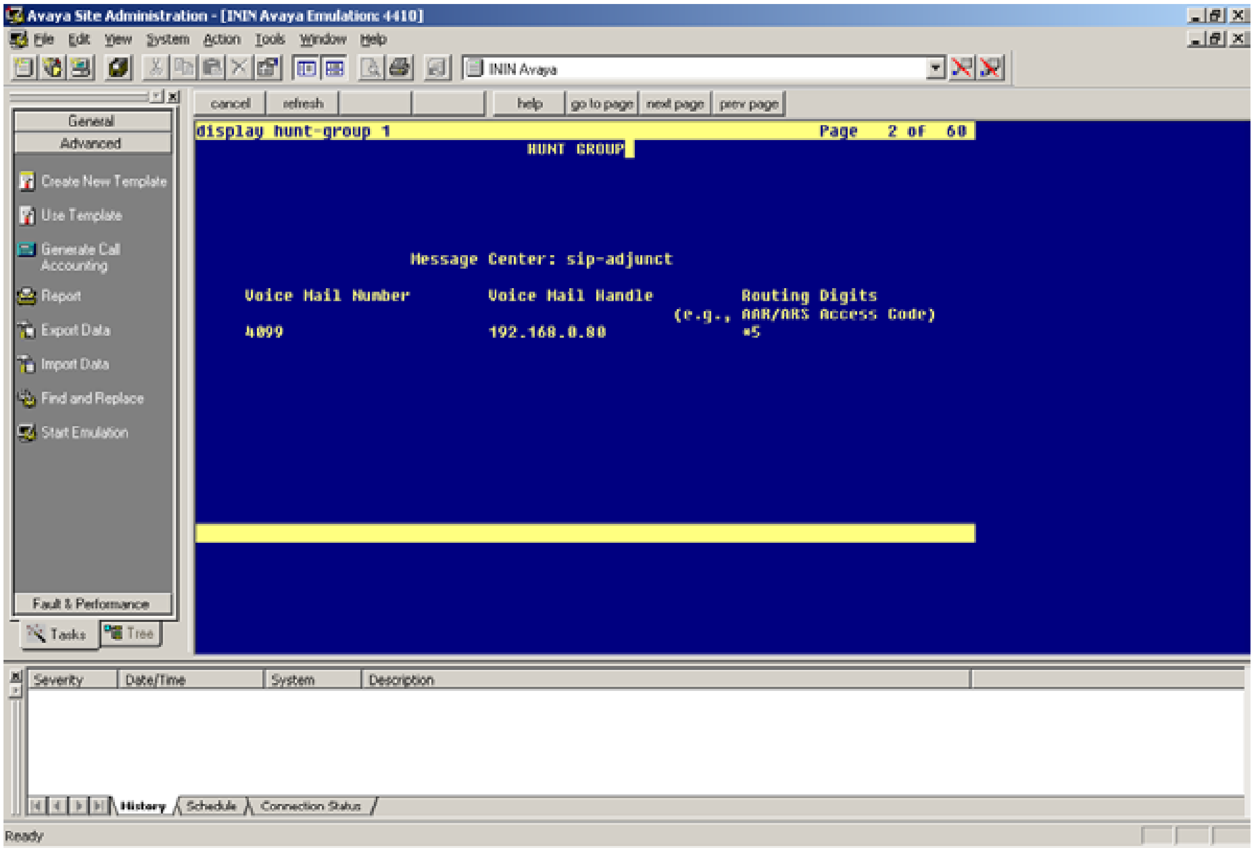
Notice extension 4099 is specifically directed at the SIP route pattern.

Step 6: Configure Hunt Group

add hunt-group X (X is the hunt group number you wish to use)



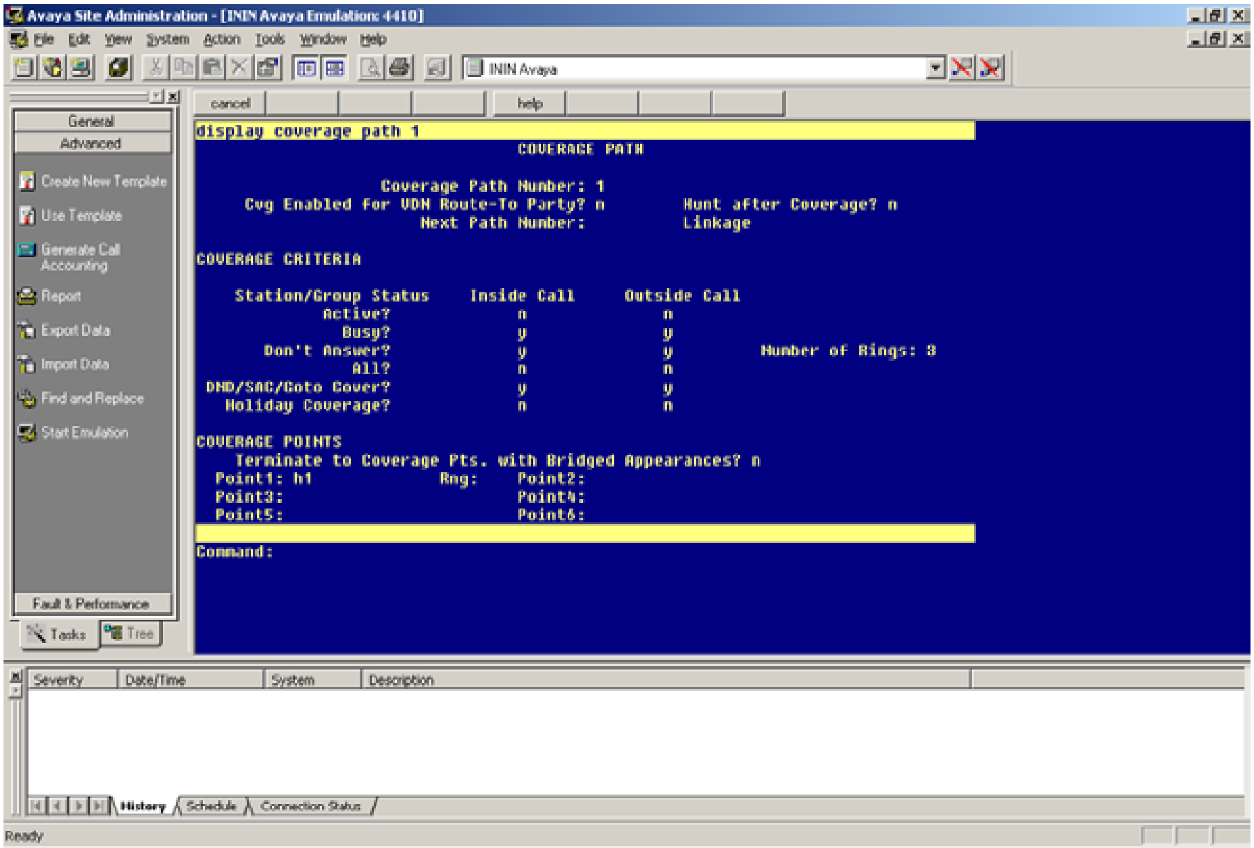
Notice the extension is the same as the value added to the AAR table to route over the SIP Signaling Group.



Once the Message Center value is set to sip-adjunct, the other settings will appear. The AAR/ARS Access Code is listed in your feature access codes (display feature-access-codes).

Step 7: Configure Coverage Path

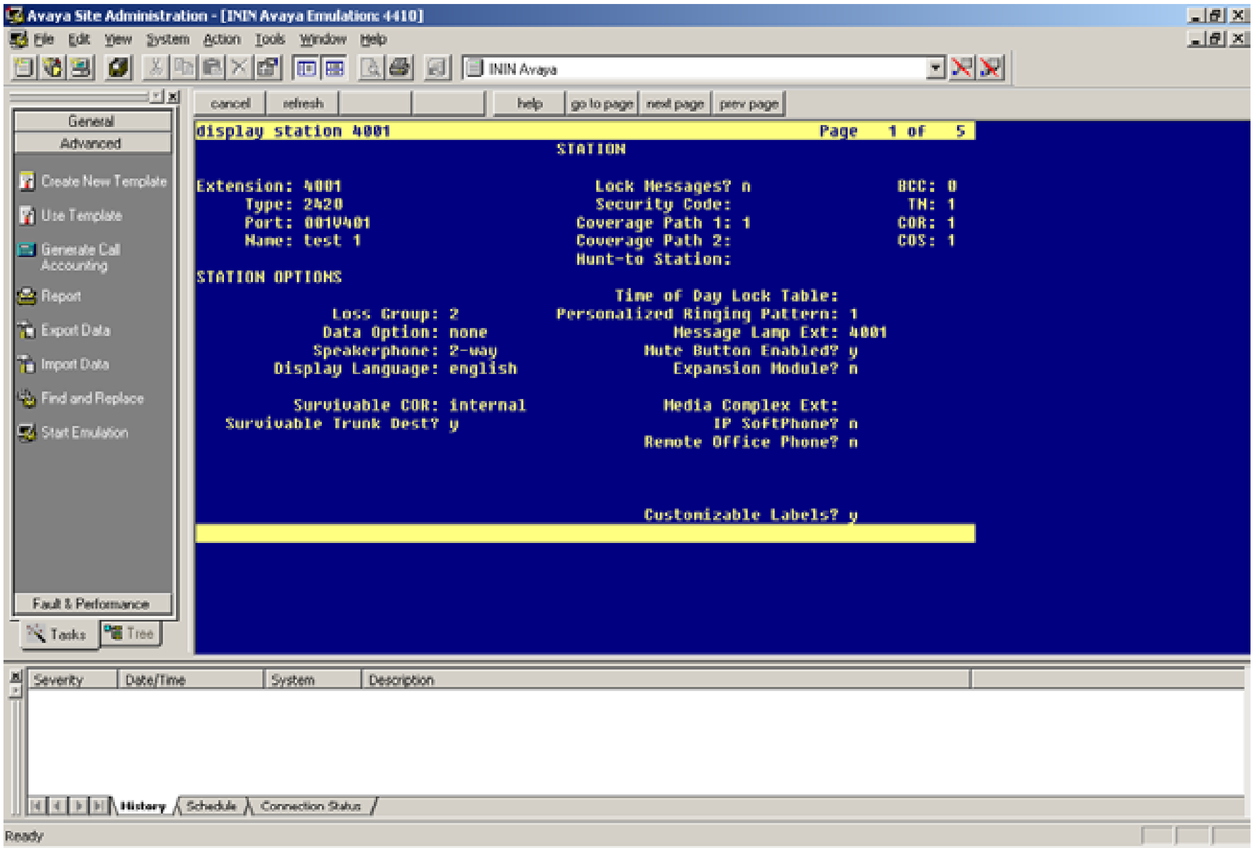
add coverage path X (X is the coverage path number you wish to use)



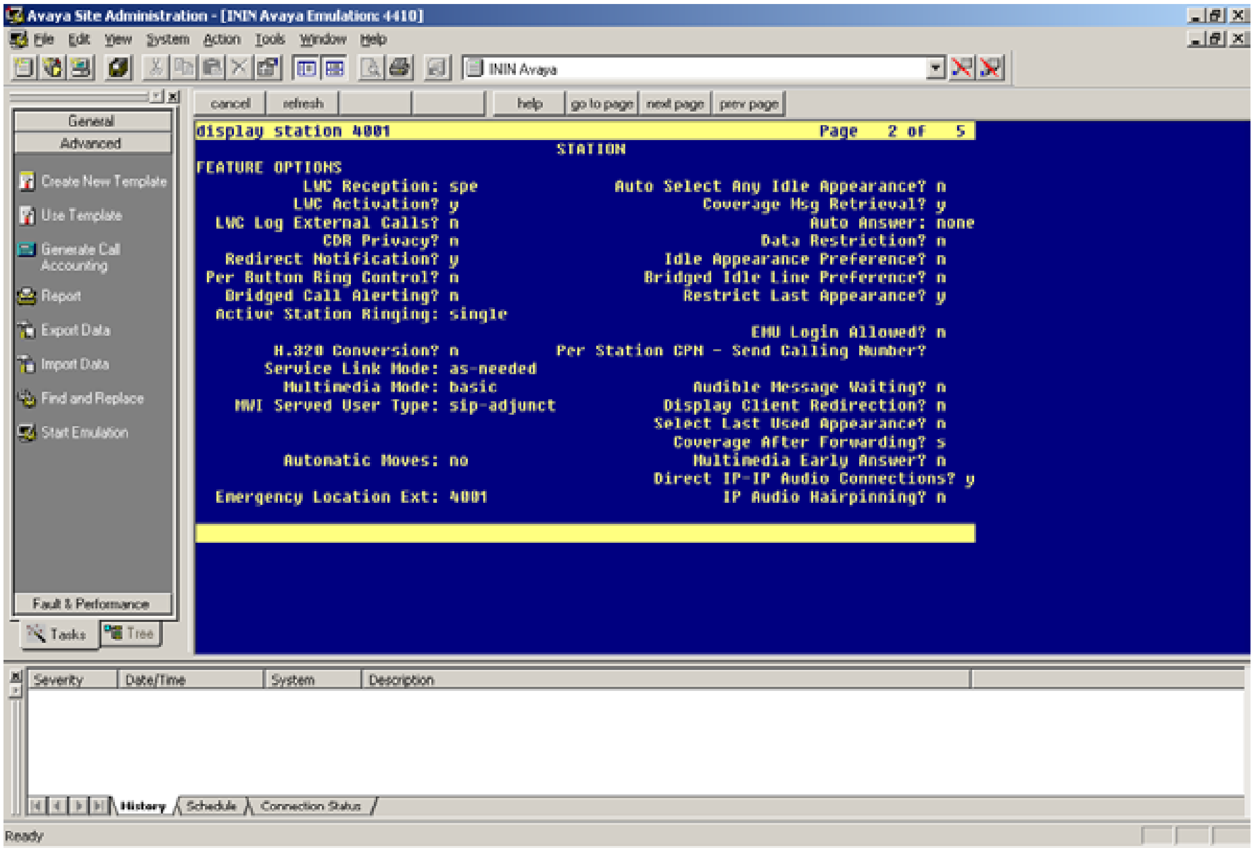
Notice the first and only point is the hunt group setup in the previous step.

Step 8: Configure Stations

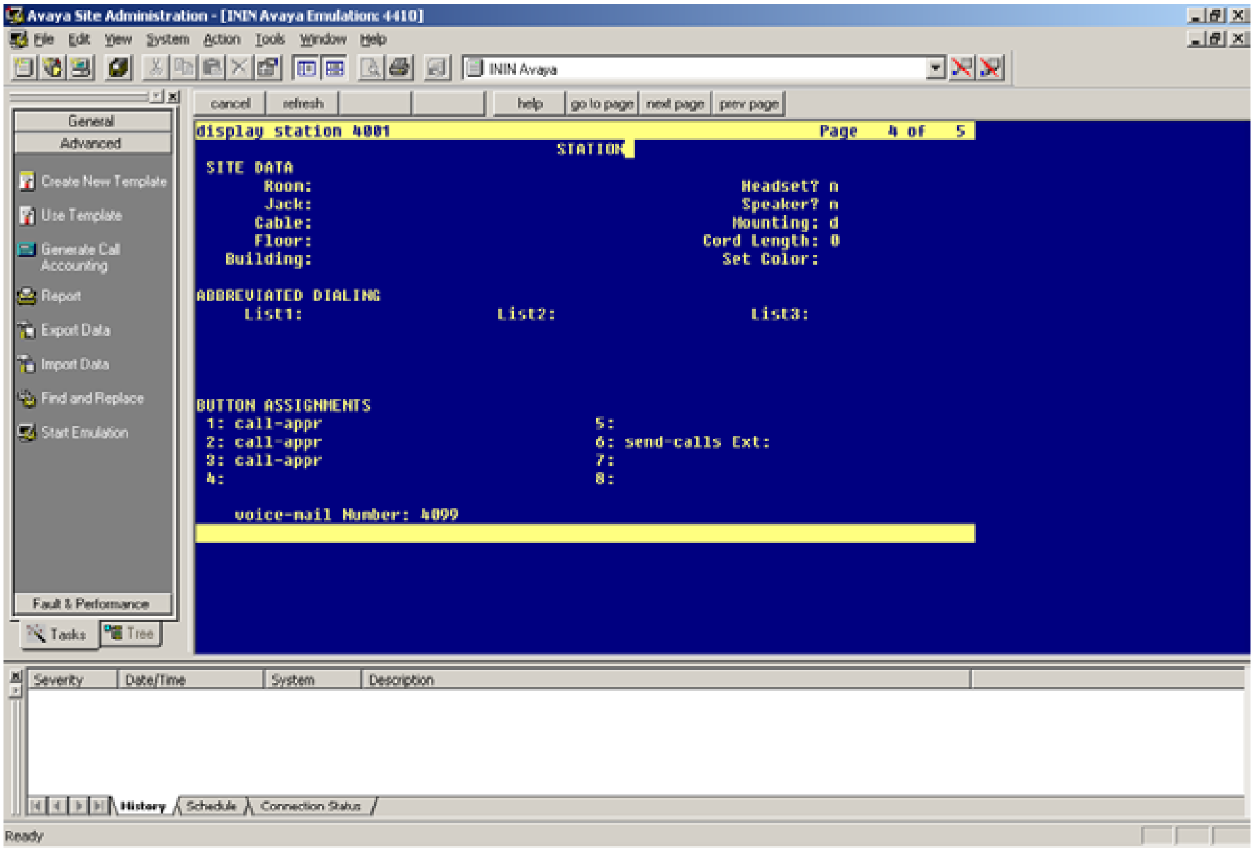
change station XXXX (XXXX is the station extension)



Notice the Coverage Path is set to the Coverage Path created in the previous step.



Notice the MWI Served User Type setting.



Notice the voice-mail Number setting.

Chapter 4: Messaging Interaction Center 2.4 Validation Test Matrix

Testing the Core Feature Set

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

No.	Call Scenarios (see appendix for more detailed instructions)	(P/CP/F/NT)	Reason for Failure (see 6.1 for more detailed descriptions)
1	Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox. Confirm hearing the prompt: "Welcome to Communité. To access your mailbox, enter your extension…"	**P**	
2	Navigate mailbox using Mobile Office	**NT**	
3	Navigate mailbox using the Telephony User Interface (TUI).	**P**	
4	Dial user extension and leave a voicemail.		
4a	Dial user extension and leave a voicemail from an internal extension.	**P**	
4b	Dial user extension and leave a voicemail from an external phone.	**P**	
5	Dial Auto Attendant (AA). Dial the extension for the AA and confirm the AA answers the call.	**P**	
6	Call Transfer by Dial By Name.		
6a	Call Transfer by Dial By Name and have the called party answer. Confirm the correct called party answers the phone.	**P**	

No.	Call Scenarios (see appendix for more detailed instructions)	(P/CP/F/NT)	Reason for Failure (see 6.1 for more detailed descriptions)
6b	Call Transfer by Dial By Name when the called party's phone is busy. Confirm the call is routed to the called party's voicemail.	**P**	
6c	Call Transfer by Dial by Name when the called party does not answer. Confirm the call is routed to the called party's voicemail.	**P**	
7	Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button. Confirm you are sent to the prompt: "Welcome to Communité. Please enter your passcode."	**P**	
8	Send a test Fax message to user extension. Confirm the Fax is received in the user's inbox.	**NT**	
9	Setup Message Waiting Indicator (MWI).	**P**	
10	Blind Transfer	**P**	
11	Consult Transfer	**CP**	Failed per description in Known Issues of this document. Resolved with SCRs 62854 & 62855.
12	Dynamic Audio	**P**	

Detailed Description of Limitations

Failure Point	None
Phone type (if phone-specific)	
Call scenarios(s) associated with failure point	
List of UM features affected by failure point	
Additional Comments	

Appendix

Dial Pilot Number and Mailbox Login

1. Dial the pilot number of the MIC server from an extension that is NOT enabled for Voicemail.
2. Confirm hearing the greeting prompt: "Welcome to Communité. Please enter your extension…"
3. Enter the extension, followed by the pound sign, and then the passcode of a Voicemail enabled user.
4. Confirm successful logon to the user's mailbox.

Navigate Mailbox using Mobile Office

1. Logon to a user's mailbox who is licensed for Mobile Office
2. Navigate through the mailbox and try out various voice commands to confirm that the Mobile Office is working properly.
3. This test confirms that the RTP is flowing in both directions and speech recognition is working properly.

Navigate Mailbox using Telephone User Interface (TUI)

1. Logon to a user's mailbox.
2. Navigate through the mailbox and try out the various key commands to confirm that the TUI is working properly.
3. This test confirms that both the voice RTP and DTMF RTP (RFC 2833) are flowing in both directions.

Dial User Extension and Leave Voicemail

Note: If you are having difficulty reaching the user's voicemail, verify that the coverage path for the user's phone is set to the pilot number of the MIC server.

From an Internal Extension

1. From an internal extension, dial the extension for a Voicemail enabled user and leave a voicemail message.
2. Confirm the voicemail message arrives in the called user's inbox.
3. Confirm this message displays a valid MIC user's name as the sender of this voicemail.

From an External Phone

1. From an external phone, dial the extension for a Voicemail enabled user and leave a voicemail message.
2. Confirm the voicemail message arrives in the called user's inbox.
3. Confirm this message displays the phone number as the sender of this voicemail.

Dial Auto Attendant (AA)

1. Create an Auto Attendant using the MIC Web Administrator:
2. Dial the extension of Auto Attendant.
3. Confirm the AA answers the call.

Call Transfer by Dial By Name

1. Dial the pilot number for the MIC server from a phone that is NOT associated with a MIC user.
2. To search for a user by name:
   - Press 2 to Dial By Name.
   - Call Transfer by Dial By Name by entering the name of an MIC user using the telephone keypad, last name first.

     Note: Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want. Ignore spaces and symbols when spelling the name.Called Party Answers
   - Call Transfer by Dial By Name to a user in the same dial plan and have the called party answer.
3. Confirm the call is transferred successfully.

Called Party is Busy

1. Call Transfer by Dial By Name to a user in the same dial plan when the called party is busy.
2. Confirm the calling user is routed to the correct voicemail.

Called Party does not Answer

1. Call Transfer by Dial By Name to a user in the same dial plan and have the called party not answer the call.
2. Confirm the calling user is routed to the correct voicemail.

Voicemail Button

1. Configure a button on the phone of a Voicemail enabled user to route the user to the pilot number of the MIC server.
2. Press the voicemail button.
3. Confirm you are sent to the prompt: “Welcome to Communité. Please enter your passcode…”

   Note: If you are not hearing this prompt, verify that the button configured on the phone passes the user’s extension as the redirect number. This means that the user extension should appear in the diversion information of the SIP invite.

Testing Fax Features

To test fax functionality:

1. Dial the extension for a fax-enabled MIC user from a fax machine.
2. Confirm the fax message is received in the user’s inbox.

Note: You may notice that the MIC server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for…"). When the MIC server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.

Note: MIC only supports T.38 for sending fax.

Message Waiting Indicator (MWI)

1. Enable MWI for a Voicemail enabled user.
2. Leave a message for that user.
3. Verify MWI goes on
4. Delete or Mark Saved that message
5. Verify MWI goes off

   Note: MWI doesn't go off until there are no more New messages in the Inbox.

Blind Transfer

1. Verify Putback is enabled on SIP Line.
2. Ring No Answer to a user's voicemail.
3. Zero Out to user's Operator.
4. Answer Operator phone.
5. Verify that all resources are released from IC.

Consult Transfer

1. Verify Putback is enabled on SIP Line, and that Follow Me is enabled for your test user.
2. Ring No Answer to a user's voicemail.
3. Press 2 to Follow Me.

   Note: Follow me should be setup to an internal extension.
4. Answer Follow Me call.
5. Verify that all resources are released from IC.

Dynamic Audio

1. Verify Putback is **NOT** enabled on SIP Line, Dynamic Audio is enabled on the SIP Line, and that Follow Me is enabled for your test user.
2. Ring No Answer to a user's voicemail.
3. Press 2 to Follow Me.

   Note: Follow me should be setup to an internal extension.
4. Answer Follow Me call.
5. Both legs of the call should be visible in Supervisor.
6. No IP resources should be in use.

Appendix A: Change Log

Change Log Date	Changes Made
09-03-2008	**Created document.**
11-25-2008	**Added Tests 10, 11, 12**
6-8-2009	**Added Section showing Avaya Trunk Group configuration**

Appendix B: Acronyms Used in This Document

Here are some of the most important acronyms used in this document.

CAS	Centralized Attendant Service
CNG	CalliNG tone sent by a fax machine
DTMF	Dual Tone Multi-Frequency
IA	Interaction Administrator
IC	Interaction Center
IP	Internet Protocol
PBX	Private Branch Exchange
SIP	Session Initiation Protocol
TDM	Time Division Multiplexing
VoIP	Voice Over IP (Voice Over Internet Protocol)