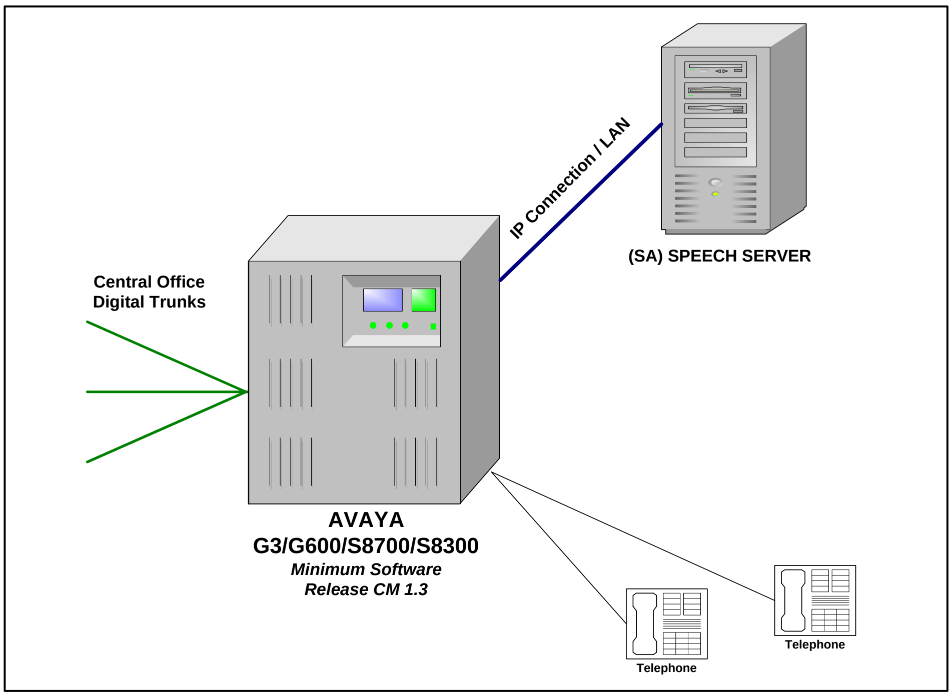
Configuration Note 3900 – Version C (7/05)

NOTE: This Config Note is available for field support only. This is a discontinued product/integration and new sales are not supported. Contact product management for additional questions.

Speech Access for Communication Manager

(Software application)



NOTE: Software patch #5558 is required for SA for CM and must be installed on your PBX for this application to work properly. Please review Section 8.0 in this CN for other considerations.

Using a pure IP/CCMS connector, the link between the PBX and the Speech server handles both call data information and voice communication

1.0 METHOD OF INTEGRATION

There is one IP Link (Ethernet) from the SA Server to the PBX. The integration is done via IP SoftPhones with voice transmission and integration information carried over IP.

Remote registered subscribers will be prompted for their account number and password to gain access to the speech server, while registered desk users registration information will automatically pass to the server. Non-subscribers (anonymous mode) will only have access to the corporate directory, and will hear the prompt “who would you like to call?”

2.0 SA ORDERING INFORMATION

Refer to the **Speech Access for Communication Manager Installation Guide (part number 101-xxxx-000)** for details on server hardware and software requirements.

3.0 PBX HARDWARE REQUIREMENTS

PBX hardware requirements

For G3 and traditional Cabinet S8700 PBXs

- **TN2302** AP Media Processor circuit pack to convert the audio levels for the IP telephone to audio levels for DCP phones when IP phones are used in a call with non-IP telephones.
- **TN799** Control-LAN (CLAN) circuit pack for the signaling capability (either the B or C vintage) on the csi, si, and r platforms.

For S8300/S8700 using newer media modules

- Use of EXT 2 port on the front panel of G700 to connect to SA Server

For connecting SA Server to PBX:

- Category 5 wiring or line cord to connect SA Server to PBX.

3.2 PBX SOFTWARE REQUIREMENTS

PBX Software Requirements

- Minimum software release CM1.3
- REQUIRED FEATURES

NOTE: SA for CM is an application that provides registered users with speech accessible telephony features that are determined as allowed by that individual's Class of Service (COS). If PBX features such as conference calling, transfers, etc. fail to function, that individual's COS may need to be changed.

4.0 SUPPORTED FEATURES

Speech Access features

Speech Access (ASA) provides a telephone speech user interface for Avaya PBX users. SA communicates with callers in spoken English and provides a natural language interface for individuals to access PBX related functionality. SA subscribers can call the system on a telephone and perform functions such as the following:

- Place telephone calls
- Manage conference calls
- Call forwarding

5.0 CONFIGURING THE PBX TO INTEGRATE

The following tasks must be completed when programming the PBX to integrate.

IMPORTANT NOTE: Although the integration uses IP Softphones for connectivity, they are defined within the PBX as 4-wire 7434ND (numeric display) sets.

Configuring the PBX to integrate

The tasks are as follows:

- Create Pilot number for the SA for CM hunt group.
- Define the stations assigned to the SA for CM application.
- Define the "registered" user stations
- Establish an OSSI login for the application to use for automated data retrieval and administration.

Note: The following configuration shows screens as displayed in the Avaya Site Administration GUI, also known as GEDI (Graphically Enhanced DEFINITY interface). Screens and parameters may appear different depending on PBX software revision and load.

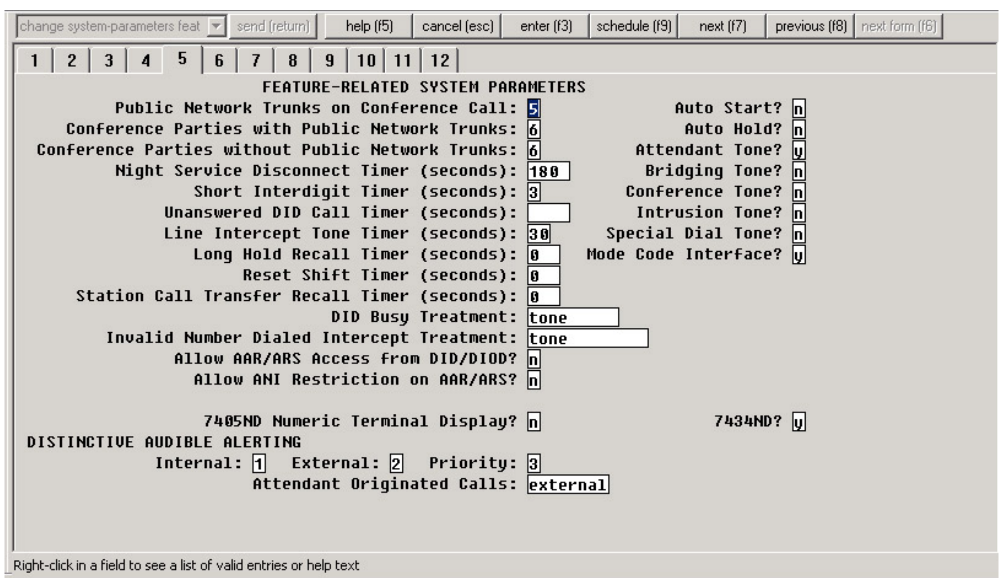
<u>IMPORTANT:</u> Those using Definity System Administration, the tabs on the following screens correlate to the page #s as displayed on the admin screens.

5.1 SYSTEM PARAMETER FEATURES

Under *Parameters* double-click change system-parameters features.

Select tab 5 and verify that the field 7434ND? is set to "y"

Figure 1: System Parameters Features - Tab 5


change system-parameters feat | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8) | next form (f6)

1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12

FEATURE-RELATED SYSTEM PARAMETERS

Public Network Trunks on Conference Call: 5 — Auto Start? n
Conference Parties with Public Network Trunks: 6 — Auto Hold? n
Conference Parties without Public Network Trunks: 6 — Attendant Tone? y
Night Service Disconnect Timer (seconds): 180 — Bridging Tone? n
Short Interdigit Timer (seconds): 3 — Conference Tone? n
Unanswered DID Call Timer (seconds): — Intrusion Tone? n
Line Intercept Tone Timer (seconds): 30 — Special Dial Tone? n
Long Hold Recall Timer (seconds): 0 — Mode Code Interface? y
Reset Shift Timer (seconds): 0
Station Call Transfer Recall Timer (seconds): 0
DID Busy Treatment: tone
Invalid Number Dialed Intercept Treatment: tone
Allow AAR/ARS Access from DID/DIOD? n
Allow ANI Restriction on AAR/ARS? n

7405ND Numeric Terminal Display? n — 7434ND? y

DISTINCTIVE AUDIBLE ALERTING
Internal: 1 External: 2 Priority: 3
Attendant Originated Calls: external

Right-click in a field to see a list of valid entries or help text

5.1.1 Ensuring best-quality IP Audio

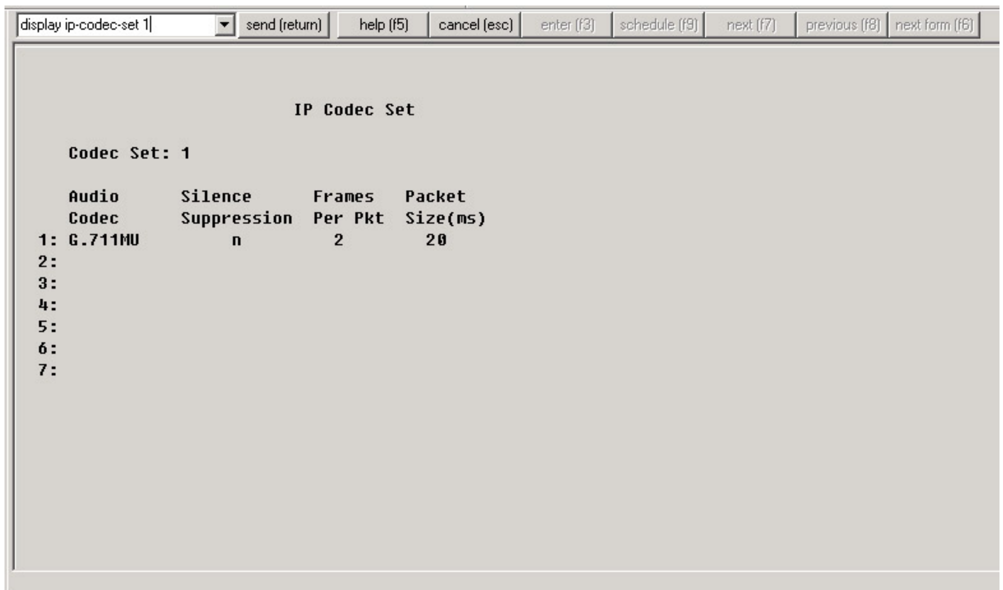
Ensuring the best audio quality.

"disp ip-codec-set 1"

To ensure best audio quality when calling the SA4CM server from an IP phone, use the "**display ip-codec-set 1"** command and verify that a G711MU codec is shown at the first position.

Note: When SA4CM negotiates a codec from the switch at call setup, it typically will use the first codec on the list (below). A G711A codec is typically used in Europe, not in the US. SA4CM was only tested with the G711MU codec.

Figure 1a: Codec Sets



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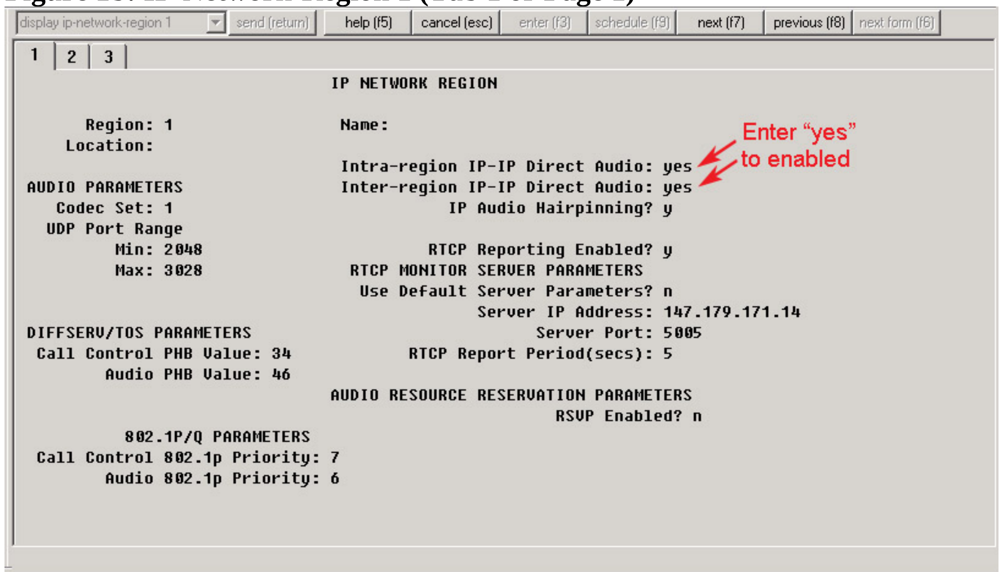
Verify Direct Audio settings are set to "yes"

"disp ip-network-region 1"

5.1.2 IP-IP Direct Audio settings

The **IP-IP Direct Audio** settings must be enabled. Figure 1b (below) shows how to verify this. Use the "**display ip-network-region 1**" command and review.

Figure 1b: IP-Network-Region 1 (Tab 1 or Page 1)



5.2 Configuration Requirements for SA for CM Stations

- Create a **sequential** range of stations that will be used by the SA for CM Server. Each station should have the following five fields set as shown below.

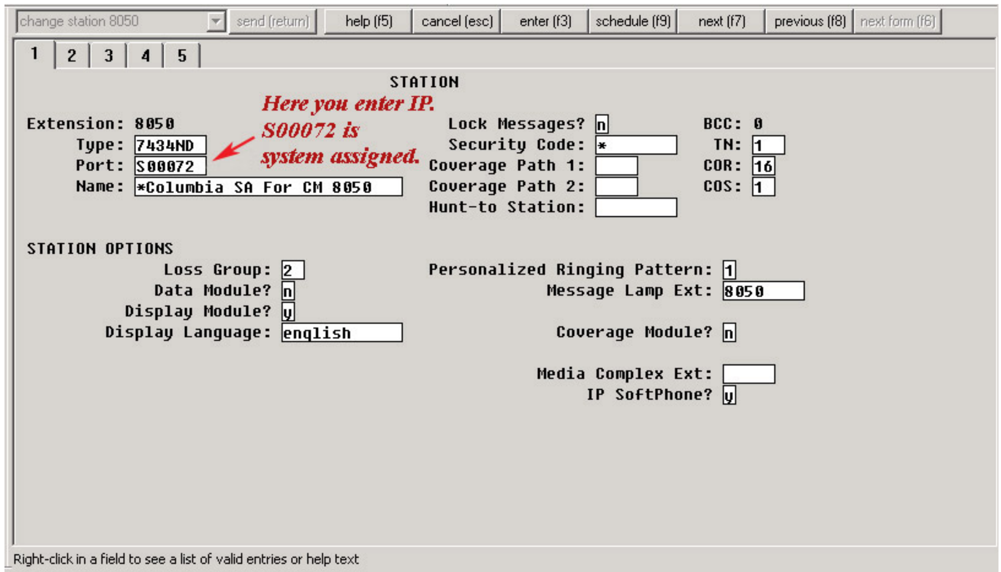
1. Type = *7434ND*
2. Display Name = Speech Access *(this name would replace "*Columbia SA for CM 8050" in our example below)*
3. Display Module = *y (only when you select*

   Note: only after you select "y" will you see Display Language appear
4. Display Language = *English*
5. **Security Code = Set to a valid value.**

   Note: All SA for CM stations must have the same Security Code.
6. IP SoftPhone? = y

Figure 2: Station options - Tab 1

Important: When configuring station options, the **Port** field is where you would enter **IP** (see red note on screen below). **Once you have added the user, the system then assigns a port.**



Note: *When the* **IP SoftPhone** *option is set to "y" as shown in* **Figure 2** *then the multimedia mode as shown in* ***Figure 3*** *will default to "enhanced" and cannot be changed.*

Figure 3: Station options - Tab 2

The default feature options should be left unchanged.

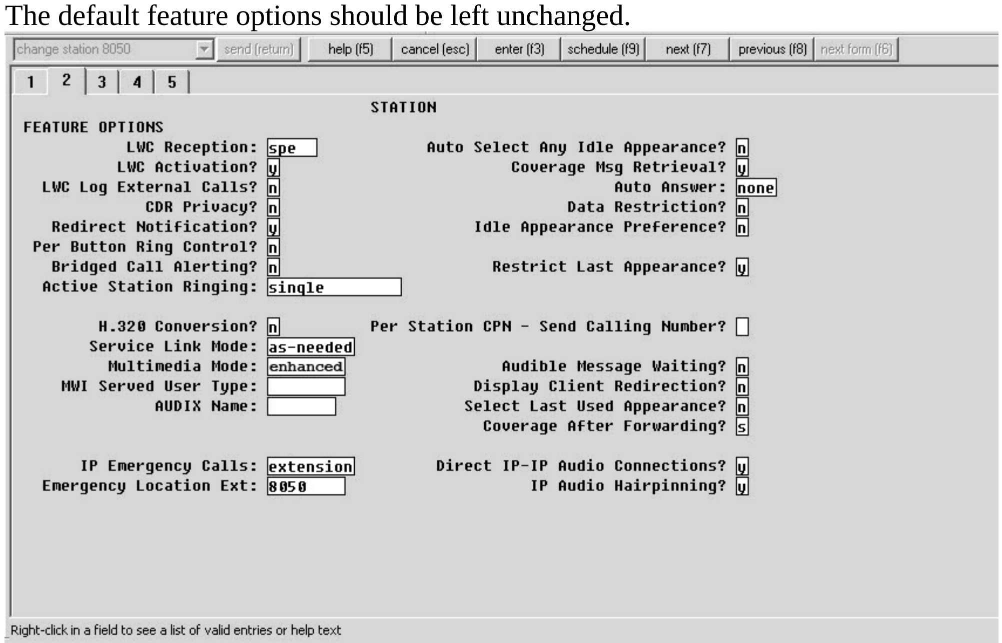


Figure 4: Station options - Tab 3
Set the Button Assignments for the first two call appearances as shown.

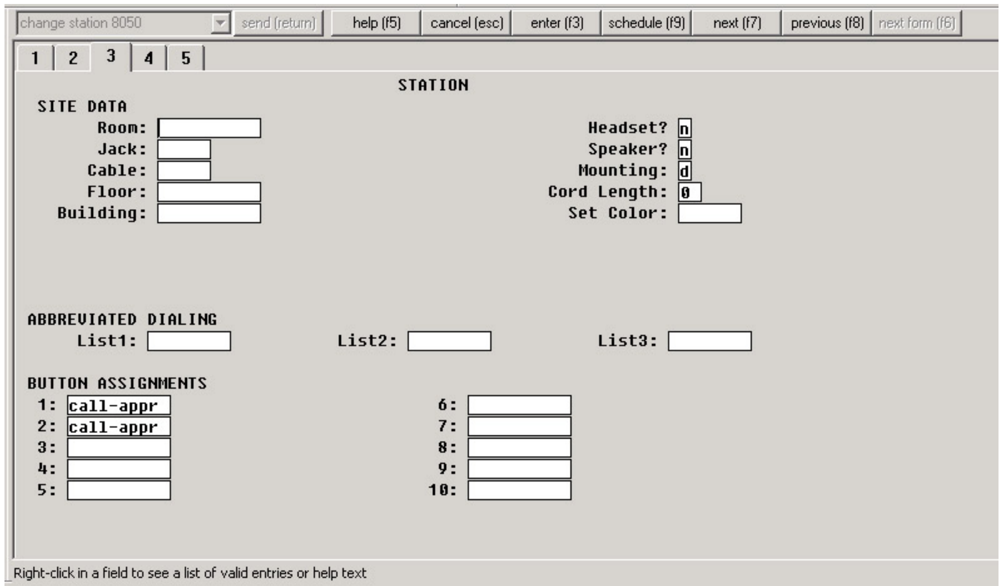


Figure 5: Station options - Tab 4
The default feature options should be left unchanged.

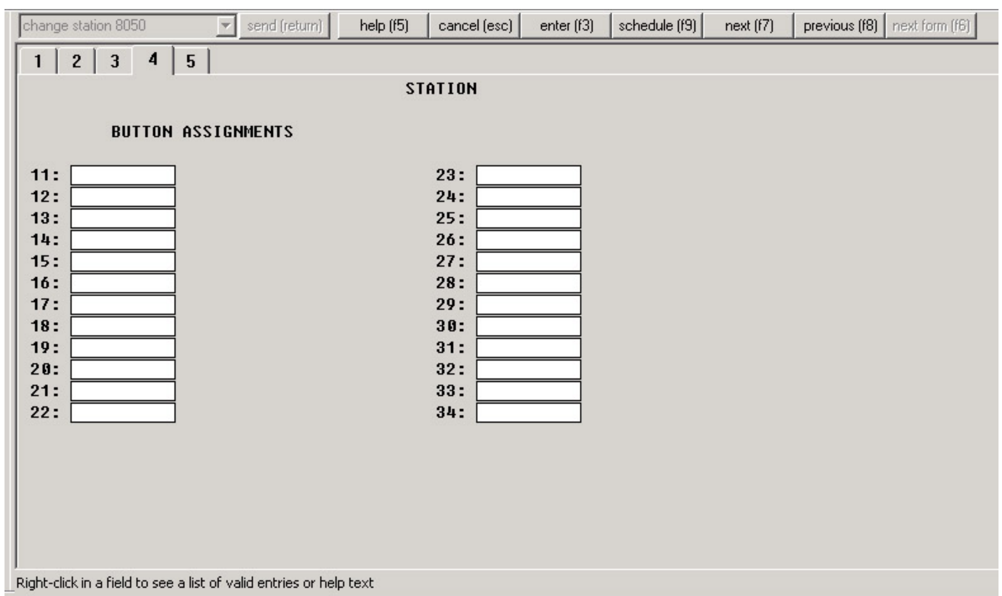


Figure 6: Station options - Tab 5
Set the Button Assignments as shown below (<u>Note</u>: must be set as shown)

5.3 CONFIGURING *REGISTERED* USER STATIONS

Configuring the stations for users of SA for CM

- These are the user stations for "registered" users of SA for CM. Each existing station should have IP SoftPhone enabled.

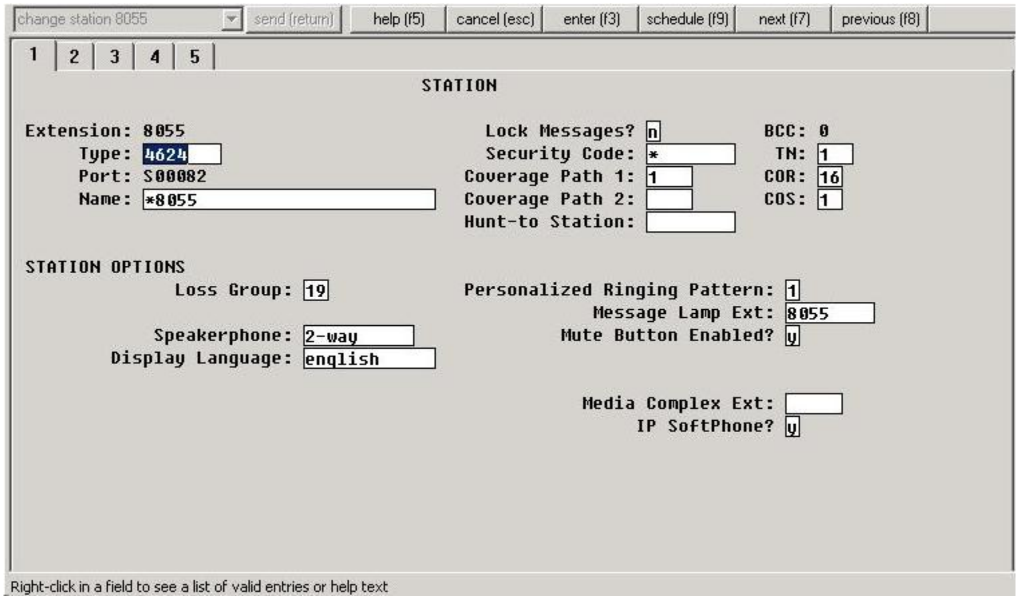
<u>NOTE</u>: **When a registered user calls the Pilot number from telephone that is "not provisioned," in other words that extension is not set up in the application for anonymous or registered access, then the user will need to log into the SA for CM application. Once logged in the user will then access the functions of their desk phone using an IP SoftPhone controlled by their voice.** Functionality is based on their Class Of Service.

5.3.1 EXPRESS LOGIN FEATURE

EXPRESS LOGIN FEATURE - The field on the bottom right side in Figure 7 (below) "IP SoftPhone" should be set to "y". A security code will need to be set. The security code will work as the user's password when logging in.

<u>NOTE</u>**: The Express Login feature denotes remote usage.**

Figure 7: Station Options [for users] - Tab 1



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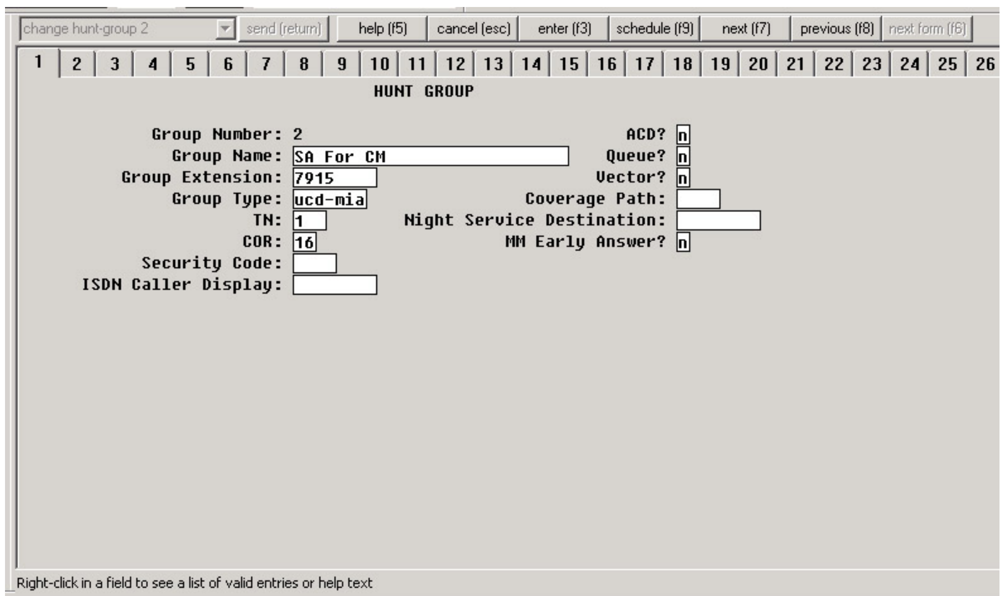
5.4 CONFIGURE HUNT GROUP FOR THE SA FOR CM STATIONS

You will need to create a hunt group for the stations you defined for SA for CM. The group extention is the pilot number that is used to call the application. In Figure (below) the Group Extension shown as 7915 is only an example.

Figure 8: HUNT GROUP - Tab 1

When creating a Hunt Group, provide a *name* you can easily remember and a group extension (which is your pilot #). The screen below shows examples of a Group Name and Group Extention. The rest should be left as default.

Establishing the Hunt Group/Pilot #



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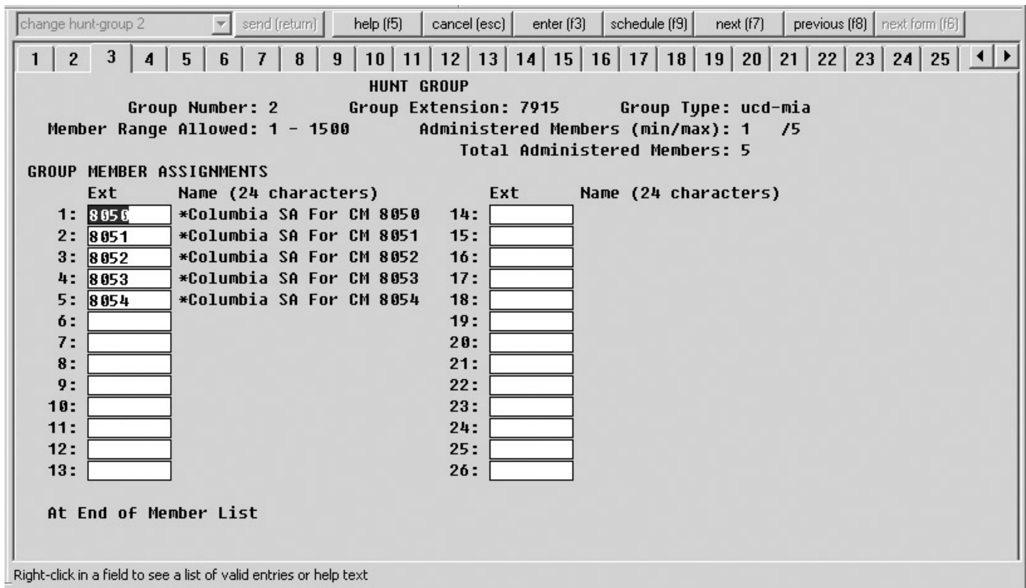
5.4.1 SETTING EXTENSION NUMBERS IN HUNT GROUP

Here is where you will enter the extension numbers created for the SA for CM application. These are the group of numbers associated with the pilot number you defined earlier.
NOTE: Hunt Groups can contain as many as 1500 numbers. Enter only the numbers you have defined for your application (see section 5.2 in this Config Note).

Figure 8: HUNT GROUP - Tab 3



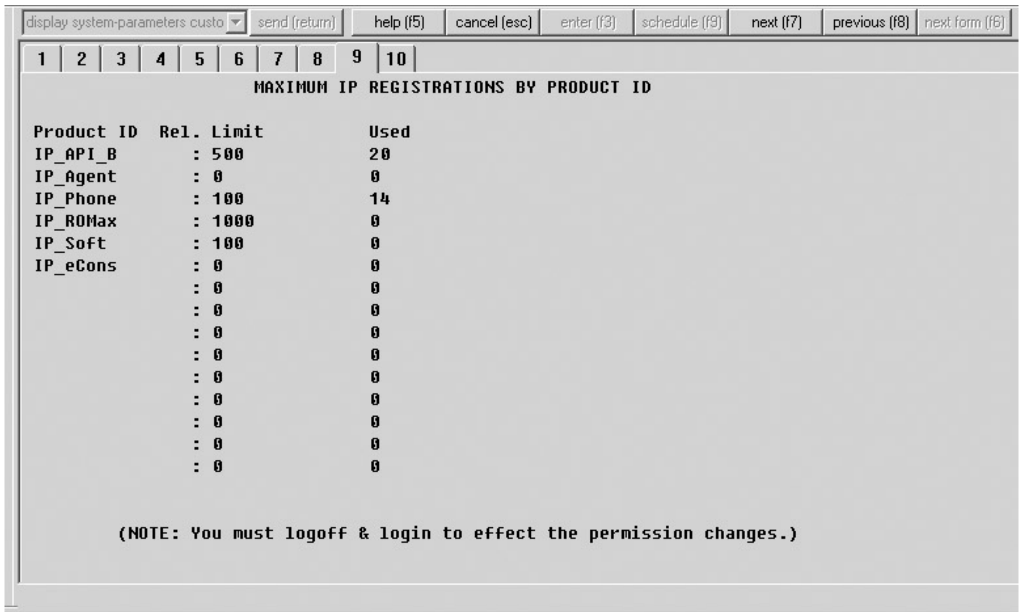
5.5 ESTABLISHING THE APPLICATION LOGIN CONFIGURATION

Establishes a Login for the Application to use for automated data retrieval and admin.

- A login will need to be created for the application (SA for CM) that it will use to retrieve a list of all stations on the PBX. This station list will be used to establish which stations are "registered," "anonymous," with all undefined as "unknown." The OSSI (Operations Support Systems Interface) login will permit the application to do a scheduled retrieval/automated retrieval of this list and subsequent administration as required for the SA for CM application.
- The switch administrator then needs to change the permissions for the Super User and then using *Change Permissions command* set the "Display Admin and Maintenance Data" to "y." Please note that "password aging" should be disabled.

5.6 ESTABLISHING THE LICENSE FILE

A LICENSE FILE "IP_API_B" MUST BE DESIGNATED FOR SA FOR CM. BELOW IS THE SCREEN THAT SHOWS IP REGISTRATIONS BY ID. CHECK TO ENSURE IP-API_B IS LISTED. OTHERWISE IT MAY BE NECESSARY TO ADD THE LICENSE TO CUSTOMER OPTIONS TABLE.



MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID	Rel. Limit	Used
IP_API_B	: 500	20
IP_Agent	: 0	0
IP_Phone	: 100	14
IP_ROMax	: 1000	0
IP_Soft	: 100	0
IP_eCons	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0
	: 0	0

(NOTE: You must logoff & login to effect the permission changes.)

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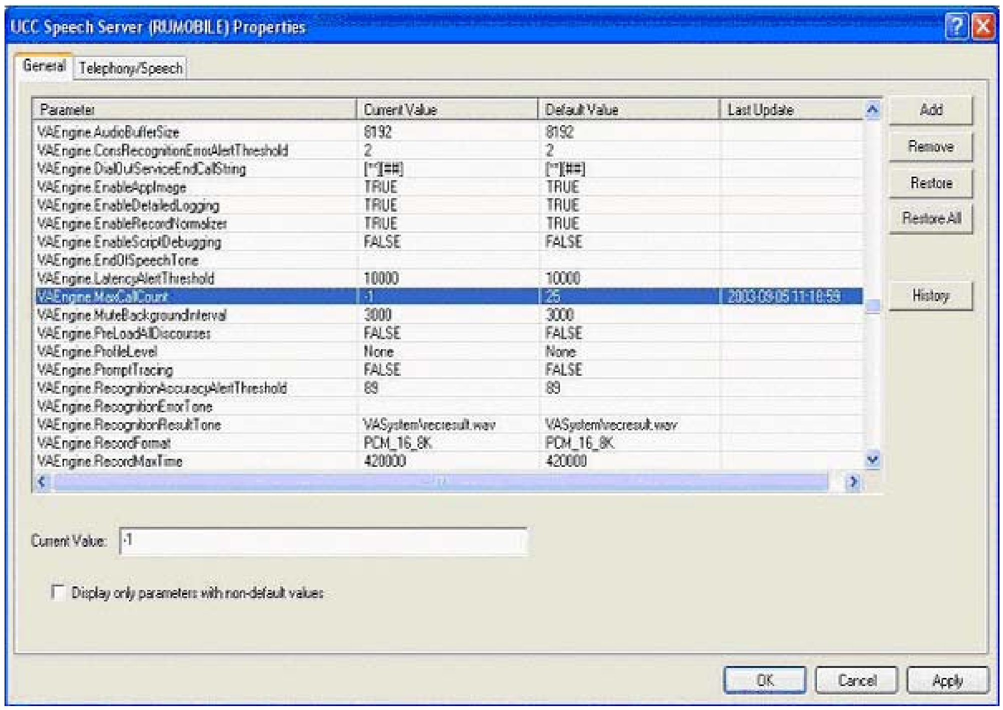
6.0 CONFIGURING the SPEECH SERVER to DISABLE RESTARTS

Avoiding potential engine restarts

The global parameter that controls engine restarts is **VAEngine.MaxCallCount**. Normally the default value is 25 but to avoid engine restarts you will need to change this value to -1.

The screen below shows the **VAEngine.MaxCallCount parameter** and the **current value as –1** to disable restarts.



7.0 TESTING

7.1 Testing the Application

Testing the SA Application

- Call the ASA pilot number for your system from a station defined as a registered user. Enter your account number and password to complete the login sequence. Validate telephony functionality including placing a call, transfer, conferencing, etc.
- Call the ASA pilot number for your system from an “anonymous user” station. You will hear “Who would you like to call?” You should be able to reach the proper party using voice commands.
- Call the ASA pilot number for your system from an “unknown user” station. You should be greeted with “please log in”.

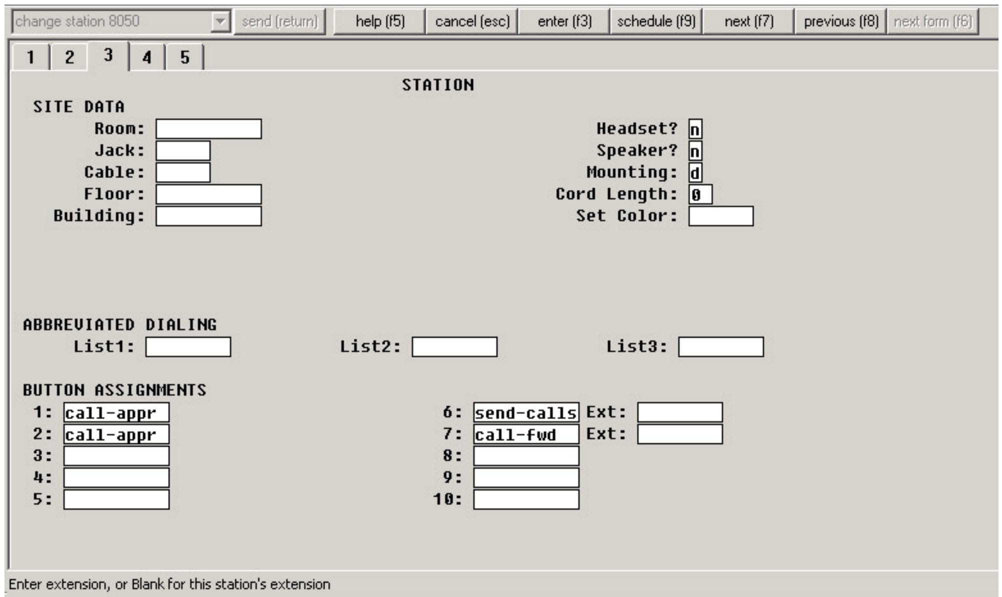
8.0 CONSIDERATIONS

8.1 Remote users must have *send-calls* and *call-fwd* configured as "button assignments" on their "desk/office phones" to allow the application to send or forward calls when instructed by the user calling into the application.

Note: **When using the change station command to configure the desk phones, assign the features *send-calls* and *call-fwd* as "button assignments" on tabs/pages 3 and 4. (See #s 6 & 7 in Figure 8.1A below)**

DO NOT set them as "display button assignment" (effectively a "soft key") on tab/page 5.

Figure 8.1A: STATION OPTIONS - Tab 3



8.2 All telephony functionality for registered users (*see Important Note below*) is based on that user's Class of Service level and Class of Restriction. The application basically uses the functionality of the IP SoftPhone with your voice serving as the dial pad.

What you need to consider with SA for CM

Note: *Consider verifying the Class Of Restrictions to limit access to long distance or international dialing to prevent toll fraud.*

Important: For registered users in desktop mode, dialing is restricted by Deny International Calling on following COR's, Deny Toll Calling on following COR's, Deny Local Calling on following COR's located in the MMC -> Telephony Setup -> Communication Manager

8.3 Registered users can access the system both locally and remotely. To access the system remotely, the user will need to have an IP SoftPhone configured for their station.

If the registered user calls from a station on-PBX other than their own registered station, they can log into their SA account and perform remote functionality. They can also login from an off-PBX station and perform remote functionality. In the remote case, the IP SoftPhone will disable the user's desk phone and provide access to the phone functionality using a combination of IP SoftPhone and voice commands.

Access from a phone with Anonymous access will grant all other users the ability to dial a contact that is listed on the corporate directory. The corporate directory can be either a specified LDAP directory or consist of CM switch users.

Callers will not be able to gain access to SA from an "unknown user station" without an ID and password.

8.4 Audio Channel Issue - After completing initial install, all processes successfully start in the Speech Server Management Console but calls made to the system get dead air. This may be due to the Speech Server sending an incorrect RTP port when the PBX's IP address has a ".0." in one of the octets. (i.e., 156.138.0.15). To correct this you will need to install the *Speech Access for Communication Manager Hot Fix 0001*, which can be found on the http://support.avaya.com website, under the *Speech Access for Communication Manager Product* section.

Document History

Revision	Issue Date	Reason for Change
A	**09/10/03**	Initial release
B	**02/11/05**	Changed Multimedia mode to Enhanced on tab 2 for station.
C	**07/22/05**	Added Consideration 8.4 regarding potential Audio Issue.

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