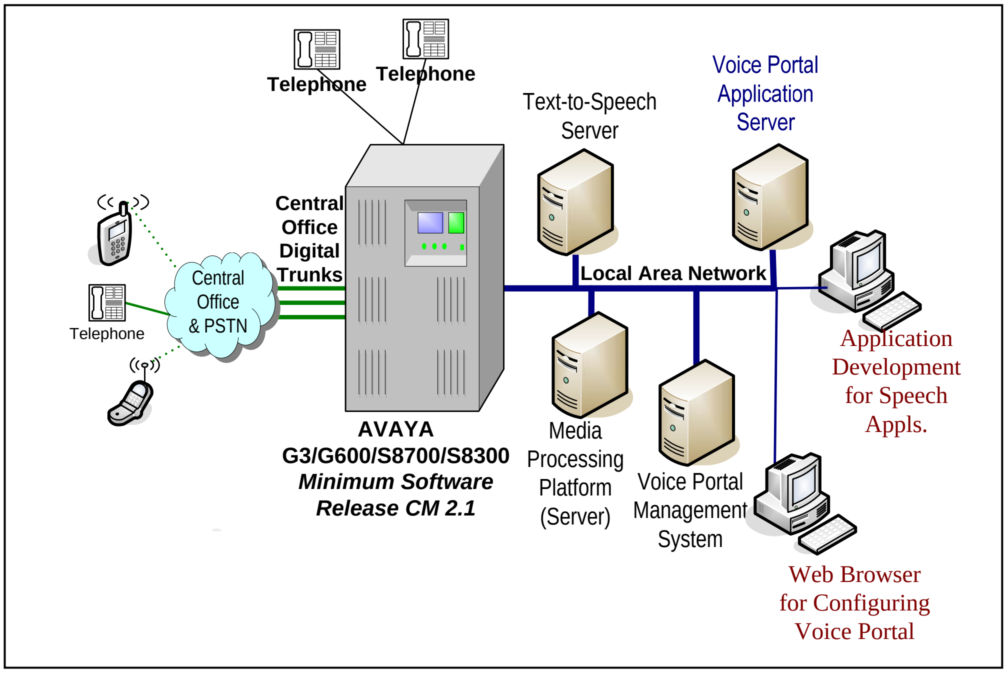
Avaya

Voice Portal

AVAYA

Configuration Note 3910 – Rev. F (1/08)

Avaya Voice Portal *(Software application)*



Important: This Configuration Note specifies the minimum configuration required by voice portal. It is advisable to consult with your application designer to ensure you have a configuration to meet your specific requirements.

An IP/CCMS connector, the link between the PBX and Voice Portal handles both call data information and voice communication

1.0 METHOD OF INTEGRATION

There is one IP Link (Ethernet) from the Voice Portal's Media Processing Platform (MPP) server to the PBX. The integration is done via the IP SoftPhone protocol with voice transmission and integration information carried over IP.

2.0 VOICE PORTAL ORDERING INFORMATION

For details on server hardware and software requirements please refer to:

- **Avaya Voice Portal 3 – Concepts and Planning**
- **Avaya Voice Portal 3 – Installation and Configuration Guide**

PBX hardware requirements

3.0 PBX HARDWARE REQUIREMENTS

For G3 and traditional Cabinet S87x0/S8500 PBXs

- **TN2302 AP Media Processor / TN2602 AP Media Resource** circuit packs (**Crossfire**). AP Media circuit packs are used 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.

*__FOR FAX Support__: **TN2302 Firmware 111 minimum /**
TN2602AP Firmware 24 minimum

- **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/S87x0/8500 using newer media modules

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

For connecting to PBX:

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

PBX Software Requirements

3.2 PBX SOFTWARE REQUIREMENTS

- Minimum software release CM2.1
- Minimum release Avaya CM 3.1 and Special Application (Green Feature) *SA8874 - Call status messages for 7434ND IP phones* is strongly recommended. SA8874 provides detailed call-progress indications to Avaya VP when the incoming or outgoing call is either a station-to-station call or is over a PRI trunk. This allows supervised transfer to fully integrate with Voice Portal for call progress and handling. Without this feature supervised transfers will function similar to blind transfers and callers may hear busy or similar tones that would otherwise be avoided. To order *SA8874* use Material Code #202775. Note: There is a cost to add this feature to your PBX.
- **AAS capability** – Avaya CM station type (7434ND) used by Avaya VP does not allow for Auto-Available Skill (AAS) capability to be configured. To resolve this issue install patch 11586, available through CM 3.0 Load 346.

Important: Account teams should check with Avaya Services to determine if any other patches may be needed for the Avaya CM release they will integrate with Avaya VP.

FEATURES

4.0 SUPPORTED FEATURES

The Voice Portal system provides either full or partial automation of telephone transactions that would otherwise be performed by an operator, attendant, or contact center agent. These automated transactions are driven by speech applications where each speech application is designed and developed to satisfy specific customer needs.

Note: *With Voice Portal 4.0 or higher, an alternate gatekeeper discovery address can be configured on the VPMS to be used if the primary address does not respond.*

5.0 CONFIGURING THE PBX TO INTEGRATE

Configuring the PBX to integrate

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

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

The tasks are as follows:

- Configure Switch System Settings
- Create Pilot number for the Voice Portal hunt group.
- Define the stations assigned to Voice Portal.

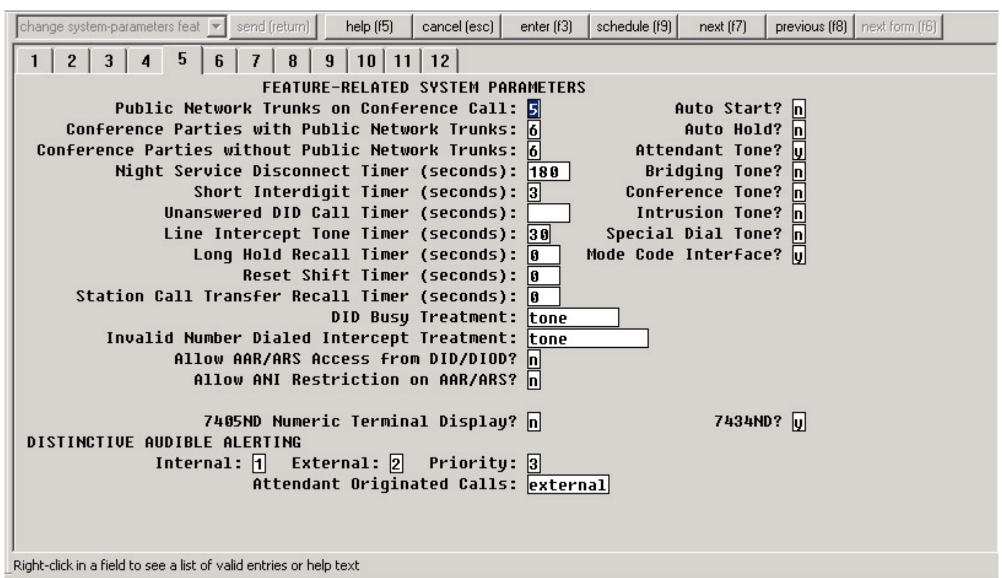
NOTE: *A number of configuration shows screens that follow are shown displayed in the Avaya Site Administration GUI, also known as GEDI (Graphically Enhanced DEFINITY interface). Screens and parameters may appear different depending on PBX software revision and load. Please note, for those using Definity 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



Ensuring the best audio quality.

"disp ip-codec-set 1"

5.1.1 Ensuring best-quality IP Audio

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

Voice Portal supports **Voip Audio Formats** set to **audio/basic**(default) or **audio/x-alaw-basic**. This selection is defined on the VPMS's web page under the **VoIP Settings**. Make sure the switch is configured to support the codec you select.
If you have selected **audio/basic** on the VPMS, then the codec set on the switch should include **G711MU**. If you have selected **audio/x-alaw-basic** on the VPMS, then the codec set on the switch should include **G711A**.

Note: When Voice Portal negotiates a codec from the switch at call setup, it typically will use the first codec on the list (below). A G711MU codec is typically used in the US and Japan, while a G711A is used in Europe and other parts of the world.

Figure 1a: Codec Sets

```
display ip-codec-set 1                                      Page 1 of 2
                              IP Codec Set

    Codec Set: 1

    Audio       Silence         Frames      Packet
    Codec       Suppression     Per Pkt      Size (ms)
1: G.711MU        n               3           30
2: G.711A         n               3           30
3:
4:
5:
6:
7:
Media Encryption: none
```

Important: This setting should be set to match the same setting on the Voice Portal system.

To work properly with Avaya Voice Portal, the required **minimum** ***Packet Size*** setting in the IP-Code-Set is **20** ms.

-continued on next page -

FAX: *Starting with Voice Portal 4.1 a fax tone detection feature is available. For this feature to work, you must configure the PBX correctly.*

Note: Turn **off** the special fax handling as shown here in this codec set. This allows the fax to be handled as an ordinary voice call.
With a codec set that uses G.711, this setting is required to send faxes to non-Avaya systems that do not support T.38 fax.

```
display ip-codec-set 1
Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


                    Mode               Redundancy
    FAX             off                0
    Modem           off                0
    TDD/TTY         US                 3
    Clear-channel   n                  0
```

5.1.2 Media Encryption

Voice Portal supports Media Encryption set to **Yes** (default) or **No**. This selection is defined on the VPMS's web page under the **H.323 Connections**. Be sure the switch has Media Encryption appropriately configured for **AES** (Encryption), **AEA** (Encryption) and/or **None** (No Encryption).

If Voice Portal has **Media Encryption** set to **Yes**, then the MPP will support Media Encryption for the switch set to **AES**(Encryption), **AEA** (Encryption) and/or None (No Encryption).

If Voice Portal has **Media Encryption** set to **No**, then the switch must have **None** (No Encryption) enabled (***see above - Figure 1a Codec Sets***). If this option is not set, then calls to the MPP will get dead air.

5.1.3 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.

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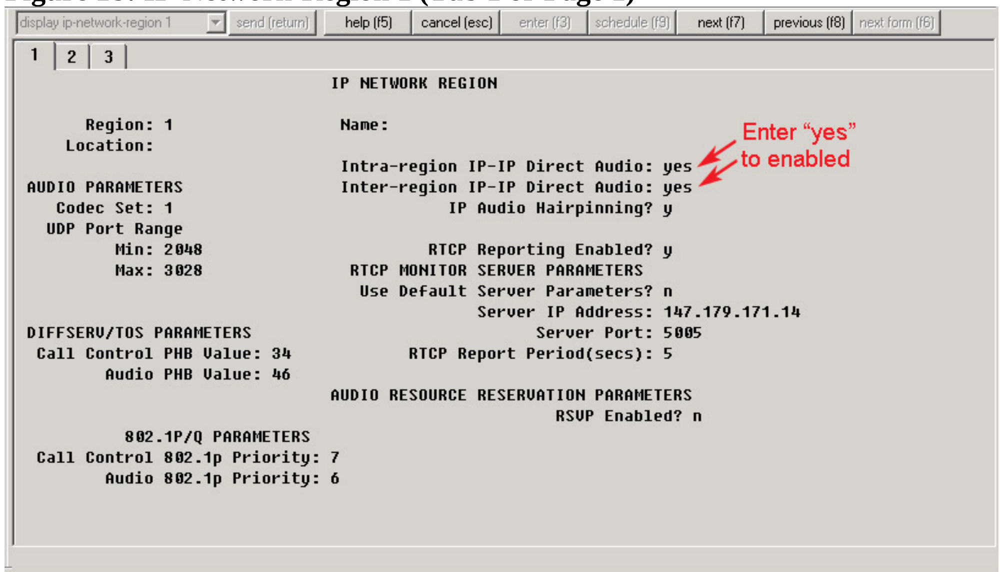
Verify Direct Audio settings are set to "yes"

"disp ip-network-region 1"

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




```
                              IP NETWORK REGION

     Region: 1                 Name:
   Location:
                               Intra-region IP-IP Direct Audio: yes
AUDIO PARAMETERS               Inter-region IP-IP Direct Audio: yes
   Codec Set: 1                           IP Audio Hairpinning? y
  UDP Port Range
          Min: 2048                     RTCP Reporting Enabled? y
          Max: 3028             RTCP MONITOR SERVER PARAMETERS
                                 Use Default Server Parameters? n
                                             Server IP Address: 147.179.171.14
DIFFSERV/TOS PARAMETERS                            Server Port: 5005
 Call Control PHB Value: 34             RTCP Report Period(secs): 5
        Audio PHB Value: 46
                              AUDIO RESOURCE RESERVATION PARAMETERS
                                                     RSVP Enabled? n
          802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
```

5.2 Configuration Requirements for Voice Portal Stations

- Create a **sequential** range of stations that will be used by Voice Portal. 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 8050" in our example below)*
    3. Display Module = *y*

       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: Voice Portals allows the security codes to be the same for every station, or incrementally set for stations in a given range.

    6. IP SoftPhone? = y

Figure 2: Station options – Page 1 of 5

Note: The following screens for configuring the **Station Options are shown using the standard terminal display**. If you are using the Avaya Site Administration GUI, also known as GEDI (Graphically Enhanced DEFINITY interface), the page numbers would be depicted graphically as tabs. Screens and parameters may appear different depending on PBX software revision and load.

Important: When configuring station options, the **Port** field is where you would enter **IP** (see red note below left). **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*** *(Station Options - page 2 of 5) will default to "****enhanced****" and cannot be changed.*

```
display station 57001                                          Page   1 of   5
                                    STATION

Extension: 57001                        Lock Messages? n          BCC: 0
     Type: 7434ND                       Security Code: *           TN: 1
     Port: S00400                     Coverage Path 1:            COR: 1
     Name: Elena1                     Coverage Path 2:            COS: 1
                                      Hunt-to Station:

STATION OPTIONS
              Loss Group: 2           Personalized Ringing Pattern: 1
             Data Module? n                 Message Lamp Ext: 57001
          Display Module? y
        Display Language: english             Coverage Module? n

                                           Media Complex Ext:
                                                IP SoftPhone? y
                                          Remote Office Phone? n
```

Figure 3: Station options – Page 2 of 5

The default feature options should be left unchanged.

```
display station 57001                                          Page   2 of   5
                                    STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer:
none
            CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                     Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n         Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced               Audible Message Waiting? n
     MWI Served User Type:                    Display Client Redirection? n
              AUDIX Name:                     Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 57001                        IP Audio Hairpinning? y
```

Figure 4: Station options – Page 3 of 5

Set the Button Assignments for the first two call appearances as shown.

```
display station 57001                                           Page   3 of   5
                                     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                          6:
 2: call-appr                          7:
 3:                                    8:
 4:                                    9:
 5:                                   10:
```

Figure 5: Station options – Page 4 of 5

The default feature options should be left unchanged.

```
display station 57001                                           Page   4 of   5
                                     STATION

        BUTTON ASSIGNMENTS

11:                                    23:
12:                                    24:
13:                                    25:
14:                                    26:
15:                                    27:
16:                                    28:
17:                                    29:
18:                                    30:
19:                                    31:
20:                                    32:
21:                                    33:
22:                                    34:
```

Figure 6: Station options – Page 5 of 5
Set the Button Assignments as shown below (Note: must be set as shown)

```
display station 57001                                            Page   5 of   5
                                       STATION

DISPLAY BUTTON ASSIGNMENTS

 1: normal
 2:
 3:
 4:
 5:
 6:
 7:
```

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5.3 CONFIGURE HUNT GROUP FOR VOICE PORTAL STATIONS

You will need to create a hunt group for the stations you defined for Voice Portal. 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, whatever name or number you enter in the *Group Name* field will appear on the telephones calling into the application.

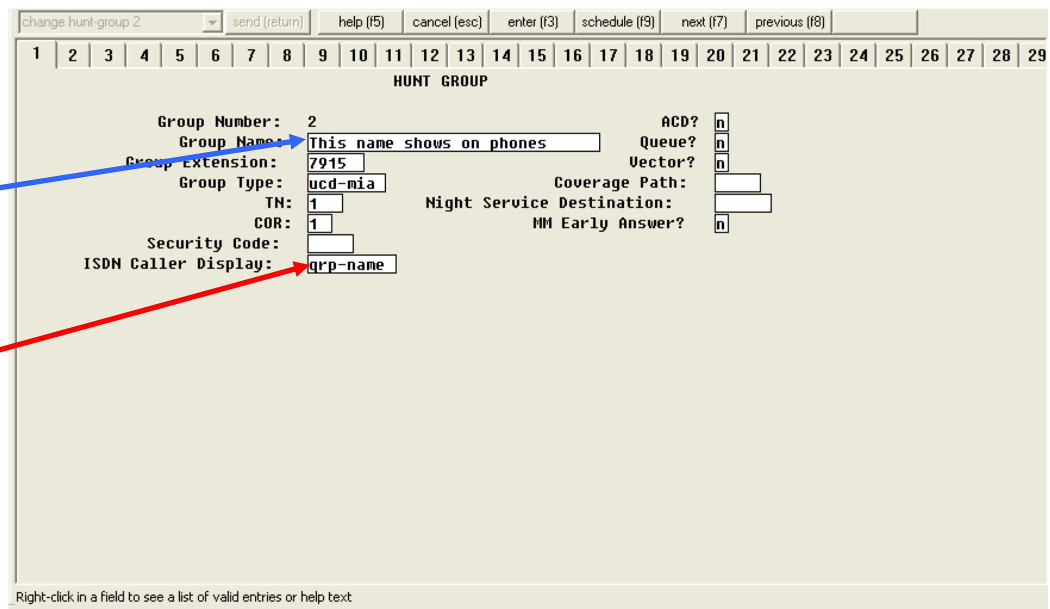
The screen below shows examples of a Group Name and Group Extention. The rest should be left as default.

Note: **Make sure you set the ISDN Caller Display Field to "*grp-name*" (see below). This enables you to display the name as entered in the Group Name field in the Hunt Group form (see below), on users' telephone displays when calling into Voice Portal.**

Establishing the Hunt Group/Pilot #

Enter whatever name you want here. This name will appear on the display of the phones calling Voice Portal.

Set ISDN Caller Display to "grp-name"

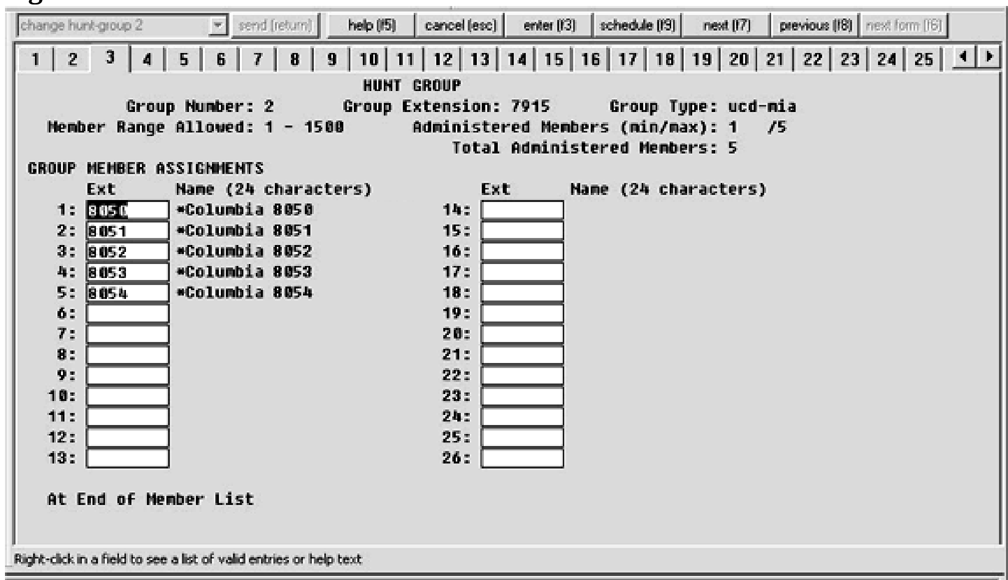


5.3.1 SETTING EXTENSION NUMBERS IN HUNT GROUP

Here is where you will enter the extension numbers created for Voice Portal. These are the group of numbers associated with the pilot number you defined earlier.

NOTE: Hunt Groups can contain as many as 1000 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



```
change hunt-group 2   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 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 | 23 | 24 | 25
                                   HUNT GROUP
          Group Number: 2       Group Extension: 7915       Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /5
                                           Total Administered Members: 5
GROUP MEMBER ASSIGNMENTS
      Ext       Name (24 characters)           Ext       Name (24 characters)
  1: 8050     *Columbia 8050             14:
  2: 8051     *Columbia 8051             15:
  3: 8052     *Columbia 8052             16:
  4: 8053     *Columbia 8053             17:
  5: 8054     *Columbia 8054             18:
  6:                                     19:
  7:                                     20:
  8:                                     21:
  9:                                     22:
 10:                                     23:
 11:                                     24:
 12:                                     25:
 13:                                     26:

  At End of Member List
Right-click in a field to see a list of valid entries or help text
```

5.4 ESTABLISHING THE LICENSE FILE

A LICENSE FILE " IP_API_A" MUST BE DESIGNATED FOR VOICE PORTAL. BELOW IS THE SCREEN THAT SHOWS IP REGISTRATIONS BY ID.

CHECK TO ENSURE IP-API_A IS LISTED. OTHERWISE IT MAY BE NECESSARY TO ADD THE LICENSE TO CUSTOMER OPTIONS TABLE.

```
display system-parameters customer-options                      Page  10 of  11
                     MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID  Rel. Limit          Used
IP_API_A       : 1000           124
IP_API_B       : 1000           0
IP_API_C       : 1000           0
IP_Agent       : 2000           117
IP_IR_A        : 2000           0
IP_Phone       : 12000          11
IP_ROMax       : 12000          0
IP_Soft        : 1000           0
IP_eCons       : 10             0
               : 0              0
               : 0              0
               : 0              0
               : 0              0
               : 0              0
               : 0              0
```

6.0 CONFIGURING SERVER

See the **Avaya Voice Portal – Installation and Configuration Guide**

7.0 TESTING

7.1 Testing the Application

Testing the Application

- Call the pilot number for your system. Validate the functionality.

8.0 CONSIDERATIONS

8.1 Although we indicated Group Type on a Hunt Group as the selected method of call distribution, users may opt to modify this according to their own needs.

8.2 **A Maintenance Port is optional** and configured on the switch as a non-maintenance port. The port should be in its own hunt group, or in a hunt group with other maintenance ports. It is used for testing applications or MPPs (Media Processing Platform). Each allocated MPP should have one maintenance port.

8.3 **ANI/DNIS sent to Voice Portal -** The ANI or DNIS sent to the Voice Portal should contain numbers only. Non-numerical values may cause the MPP improperly map the ANI/DNIS to the application to run.

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Document History		
Revision	**Issue Date**	**Reason for Change**
A	**11/10/05**	GA release
B	**03/23/06**	Added note regarding IP packet size setting in ip-codec-set (see page 4)
C	**09/01/06**	Added TN2602AP (a.k.a. Crossfire) circuit pack to Section 3.0 (PBX Requirements) as an alternate to the TN2302AP.
D	**03/08/07**	Updated CN with new Green Feature requirement Section 3.2
E	**10/8/07**	Updated section 3.0 with AAS/7434ND info and added SA8874 number to clarify the green feature noted in same section.
F	**1/18/08**	Added new screen (page 2 of ip-codec-set) and sidebar in section 5.1.1 for fax tone detection that is available starting in VP 4.1

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