Avaya Solution & Interoperability Test Lab

Application Notes for Digital Dynamics IP Telephony Application Suite with Avaya IP Telephones - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of the Digital Dynamics IP Telephony Application Suite with Avaya IP Telephones and Avaya Communication Manager. The Digital Dynamics IP Telephony Application Suite is comprised of multiple workplace productivity utilities and applications designed to run on Avaya IP Telephones. These IP Telephony applications leverage the advantages of a converged IP telephony network and are designed to increase workplace productivity by enabling greater collaboration and communication between application users.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of the Digital Dynamics IP Telephony Application Suite with Avaya IP Telephones and Avaya Communication Manager. The Digital Dynamics IP Telephony Application Suite is comprised of multiple workplace productivity utilities and applications designed to run on Avaya IP Telephones. Once IP Telephones are registered with the Digital Dynamics IP Telephony Application Suite, each IP Telephone will receive applications pushed from the IP Telephony Application Suite. These IP Telephony applications leverage the advantages of a converged IP telephony network and are designed to increase workplace productivity by enabling greater collaboration and communication between application users.

For additional information on Digital Dynamics, please refer to Digital Dynamics IP Telephony Application Suite documentation [3].

Figure 1 illustrates the network configuration used to verify the Digital Dynamics solution. The configuration details provided in these Application Notes focus on the interfaces between Avaya Communication Manager, the IP Telephony Application Suite, and Avaya IP Telephones.

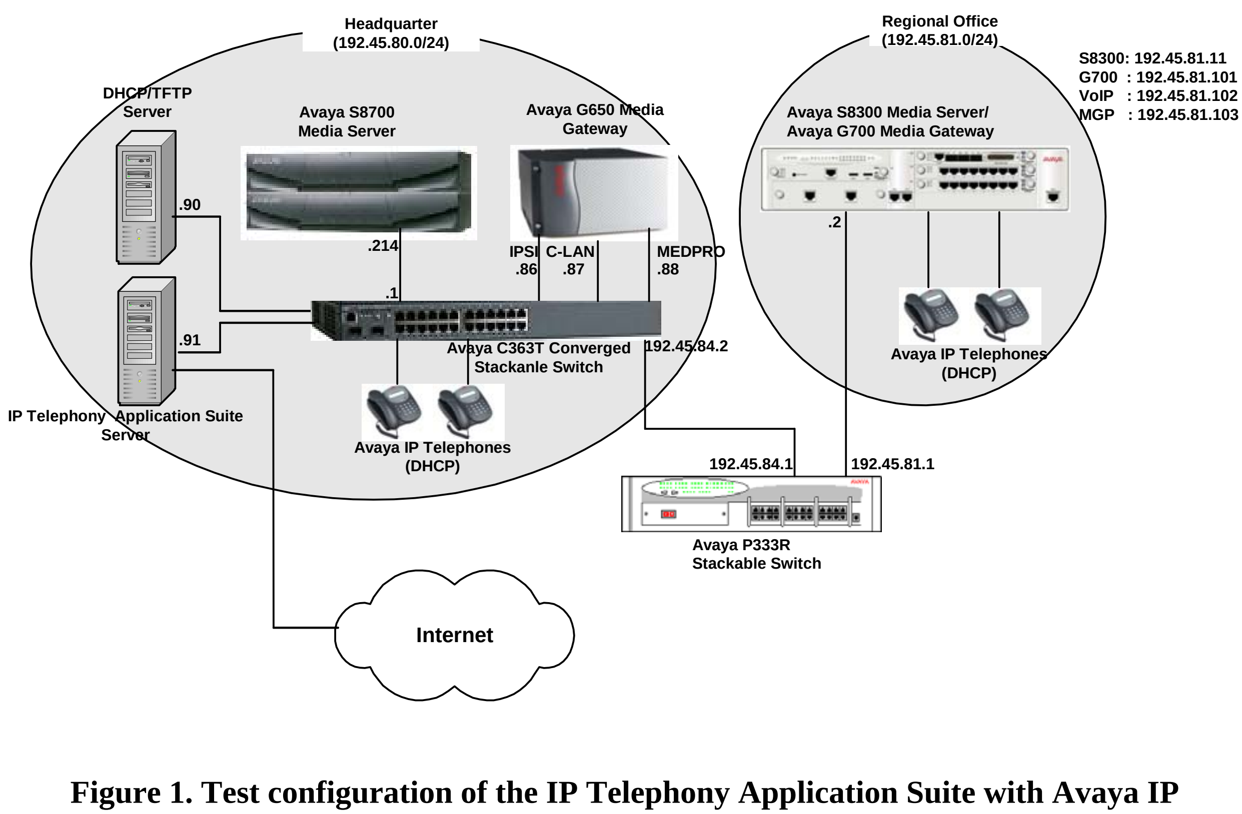


Figure 1. Test configuration of the IP Telephony Application Suite with Avaya IP Telephones

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya S8700 Media Server with Avaya G650 Media Gateway	R013x.00.0.346.0
Avaya S8300 Media Server with Avaya G700 Media Gateway	R013x.00.0.346.0
Avaya C363 T Converged Stackable Switch	4.5.14
Avaya P333R Multilayer Stackable Switch	4.0.8
Avaya IP Telephones 4620 4625	 2.2.3 2.5
Digital Dynamics IP Telephony Application Suite	1.0

3. Configuring Avaya Communication Manager

The following sections show the relevant configuration screens for Avaya Communication Managers (Headquarter and Regional Office). **The screen shots included in this section focused only on the configuration of the IP trunk-group, signaling-group, and route pattern setting**. The difference between the two Avaya Communication Manager configurations was the dialing plan. The 5-digit 5xxxx extension range was used for the headquarter side, and 6xxxx extension range was used for the regional Office. Therefore, screen shots included in this section only cover the Avaya S8700 Media Server with Avaya G650 Media Gateway side. For the purpose of showing the final configuration, the "**display**" command was used through out the document, instead of "**add**" or "**change**".

From the Avaya Communication Manager System Access Terminal (SAT) interface, use the **add signaling-group X** command, where **X** is signal-group number. Configuring the signaling-group is a two-step procedure:

- Create a signaling-group and provide the following information:
  - o Group Type : **h.323**
  - o Near-end Node Name : **CLAN**
  - o Far-end Node Name : **S8300**
  - o Near-end Listen Port / Far-end Listen Port :**1720**
- After the trunk-group is created, specify the **Trunk Group for Channel Selection** field in the signaling group.

The following screen shot shows the first step. The important signaling-group related parameters that were different from the default values are highlighted.

```
display signaling-group 50                                     Page   1 of   5
                                SIGNALING GROUP

 Group Number: 50             Group Type: h.323
                           Remote Office? n          Max number of NCA TSC: 0
                                     SBS? n           Max number of CA TSC: 0
                                IP Video? n        Trunk Group for NCA TSC:
       Trunk Group for Channel Selection:
          Supplementary Service Protocol: a
                         T303 Timer(sec): 10

   Near-end Node Name: CLAN                  Far-end Node Name: S8300
 Near-end Listen Port: 1720                Far-end Listen Port: 1720
                                        Far-end Network Region: 1
         LRQ Required? n                 Calls Share IP Signaling Connection? n
         RRQ Required? n
                                             Bypass If IP Threshold Exceeded? n
                                                      H.235 Annex H Required? n
         DTMF over IP: out-of-band            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
                                                 Interworking Message: PROGress
                                         DCP/Analog Bearer Capability: 3.1kHz
```

To add a trunk group, enter **add trunk-group Y**, where **Y** is the trunk-group number. The following two screens (Page 1 and Page 4) show the trunk configuration. The important trunk related parameters that were different from the default values are highlighted.

```
display trunk-group 50                                         Page   1 of  20
                                TRUNK GROUP

Group Number: 50                   Group Type: isdn         CDR Reports: y
  Group Name: DigDyn                      COR: 1        TN: 1        TAC: 103
   Direction: two-way        Outgoing Display? y        Carrier Medium: IP
 Dial Access? y                Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 18
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Two ports were enabled for the IP trunk, since two IP Telephones were connected to each Avaya Communication Manager. Specify each port to the signaling-group that the IP trunk will utilize.

```
display trunk-group 50                                          Page   4 of  20
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/2
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   2

       Port    Code Sfx Name        Night            Sig Grp
  1: IP                                               50
  2: IP                                               50
  3:
  4:
```

Enter **change signaling-group X**. This is the second step for configuring the signaling-group. The following screen shown below is the signaling-group. Set the Trunk Group for Channel Selection to the trunk group number.

```
display signaling-group 50                                      Page   1 of   5
                                SIGNALING GROUP

 Group Number: 50             Group Type: h.323
                           Remote Office? n          Max number of NCA TSC: 0
                                     SBS? n           Max number of CA TSC: 0
                                IP Video? n         Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 50
          Supplementary Service Protocol: a
                         T303 Timer(sec): 10

   Near-end Node Name: CLAN                  Far-end Node Name: S8300
 Near-end Listen Port: 1720                Far-end Listen Port: 1720
                                        Far-end Network Region: 1
         LRQ Required? n                 Calls Share IP Signaling Connection? n
         RRQ Required? n
                                             Bypass If IP Threshold Exceeded? n
                                                      H.235 Annex H Required? n
         DTMF over IP: out-of-band             Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
                                                 Interworking Message: PROGress
                                         DCP/Analog Bearer Capability: 3.1kHz
```

Enter **change aar analysis A**, where **A** is the Automatic Alternate Routing (AAR) number. Automatic Alternate Routing (AAR) was used to route calls to the appropriate route pattern. When a user from Headquarter dials 6xxxx, AAR will use the route pattern 50.

```
display aar analysis 6                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                       Percent Full:    1

          Dialed            Total     Route    Call   Node  ANI
          String           Min  Max  Pattern   Type   Num   Reqd
    6                       5    5     50      aar          n
    7                       7    7    999      aar          n
    8                       7    7    999      aar          n
    9                       7    7    999      aar          n
```

Enter **change uniform-dialplan U**, where **U** is the first digit of the extension. The Uniform Dial Plan screen is shown below. The 5-digit 6xxxx extension range was used for the Avaya Communication Manager side (Regional Office).

```
display uniform-dialplan 6                                      Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching          Insert          Node    Matching          Insert          Node
  Pattern   Len Del Digits Net Conv Num     Pattern   Len Del Digits Net Conv Num
  6          5   0          aar  n                                         n
                                 n                                         n
                                 n                                         n
                                 n                                         n
                                 n                                         n
```

Enter **change route-pattern R**, where **R** is the route-pattern number. Route Pattern 50 routes calls to trunk group 50, and no digit will be deleted.

```
display route-pattern 50                                        Page   1 of   3
                    Pattern Number: 50  Pattern Name: 50
                                              Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 50   0                    0                                        n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n             rest                                        none
 2: y y y y y n  n             rest                                        none
 3: y y y y y n  n             rest                                        none
 4: y y y y y n  n             rest                                        none
 5: y y y y y n  n             rest                                        none
 6: y y y y y n  n             rest                                        none
```

4. Configuring the Avaya IP Telephones

The following steps describe the registration process of an Avaya IP Telephone. To configure an Avaya IP Telephone, the following should be configured in the DHCP Server:

- DHCP IP address range / Network Subnet Mask
- Default Gateway of the network
- Create Avaya Option 176, and provide the following info:
    - Call Server IP address (CLAN / S8300 Media Server)
    - Call Server Port Number (1719)
    - File server (TFTP Server)

The Avaya IP Telephone will contact the DHCP Server and obtain the following information:

- IP address of the IP Telephone / Network Subnet Mask

- Default Gateway of the IP Telephone
- Call Server IP address (CLAN / S8300 Media Server)
- Call Server Port Number (1719)
- File server IP Address

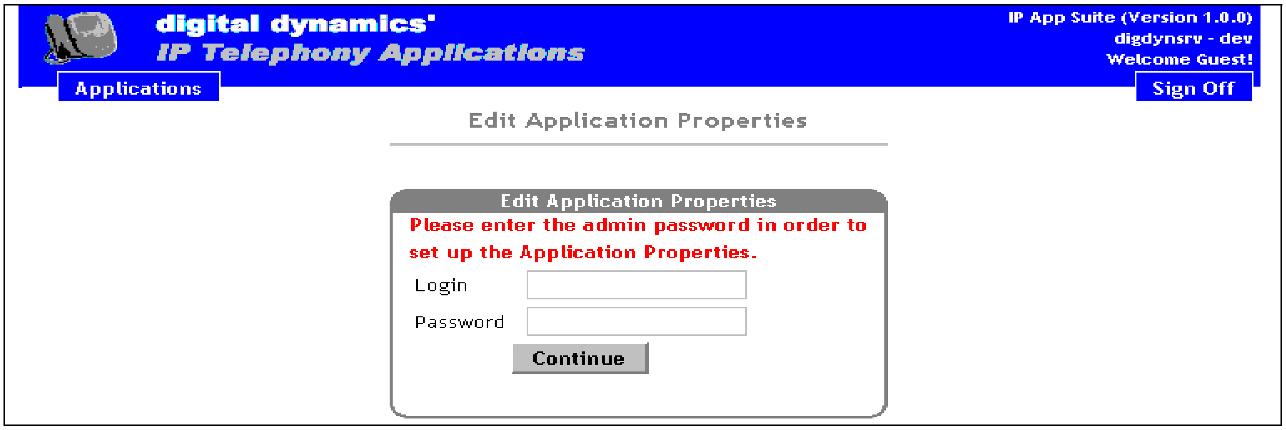
For the Avaya IP Telephone to register to the IP Telephony Application Suite, add the following line to the 46xxsetting.txt file for each IP Telephone type.
SET WMLHOME HTTP:// IP Telephony Application Suite IP Address:8888/digdynavaya/wml/main.jsp
The next screen shows the Web setting for each IP Telephone type.

```
.
.
##############
# SETTINGS4620
##############

######## Web Settings for 4620 IP Phone ########
### Please refer to the description on these terms above in SETTINGS4610

##SET WMLHOME http://192.45.80.112
SET WMLHOME HTTP://192.45.80.91:8888/digdynavaya/wml/main.jsp
##SET WMLCODING ASCII
## SET WMLPROXY my.proxy.company.com
## SET WMLPORT 8080
## SET WMLEXCEPT 192.45.80.112
##SET TPSLIST 192.45.80.112
##SET SUBSCRIBELIST http://192.45.80.112:9999/subscribe
goto END
.
.
.
####### END OF 4620 IP Phone Settings #######
##############

# SETTINGS4625
##############

######## Web Settings for 4625 IP Phone ########

### Please refer to the description on these terms above in SETTINGS4610
SET WMLHOME HTTP://192.45.80.91:8888/digdynavaya/wml/main.jsp
##SET WMLHOME http://192.45.80.112
##SET WMLCODING ASCII
## SET WMLPROXY my.proxy.company.com
## SET WMLPORT 8080
##SET WMLEXCEPT 192.45.80.112
##SET TPSLIST 192.45.80.112
##SET SUBSCRIBELIST http://192.45.80.112:9999/subscribe
goto END
```

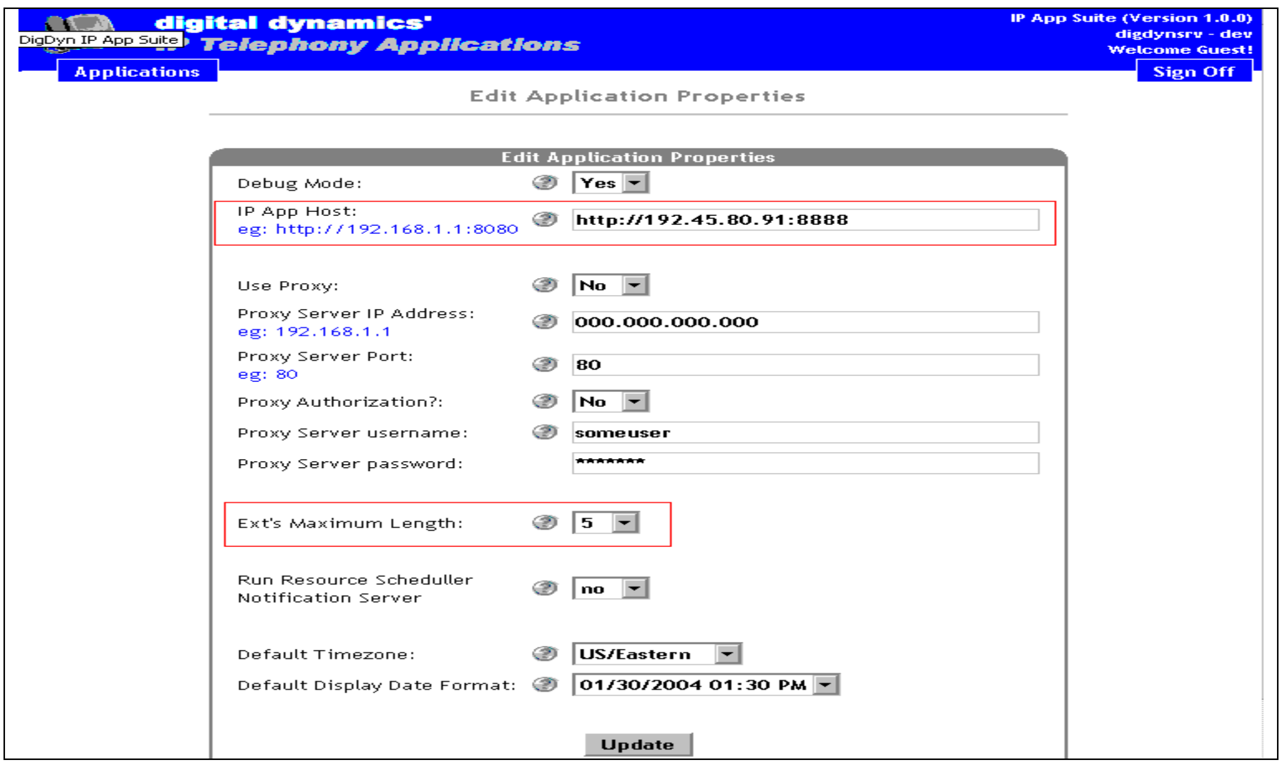
When an IP Telephone completes the initialization process, a Web button will be added on the Avaya IP Telephone screen. If a user clicks the Web button and provides the Login Name and Password, the user will have access to all applications that are assigned to the user.

5. Configure Digital Dynamics IP Telephony Application Suite

The Digital Dynamics IP Telephony Application Suite can be configured using a web based console interface. The following screen shows the Login screen. Enter a proper administrator's Login Name and Password, then click **Continue**.



Once the Authentication process is completed, the **Edit Application Properties** window appears. The **IP App Host** field should be configured properly using the IP Address of the IP Telephony Application Suite. Note that the **Ext's Maximum Length** is set to **5**, meaning 5-digit extensions were utilized. After the change, click the **Update** button to save the change.

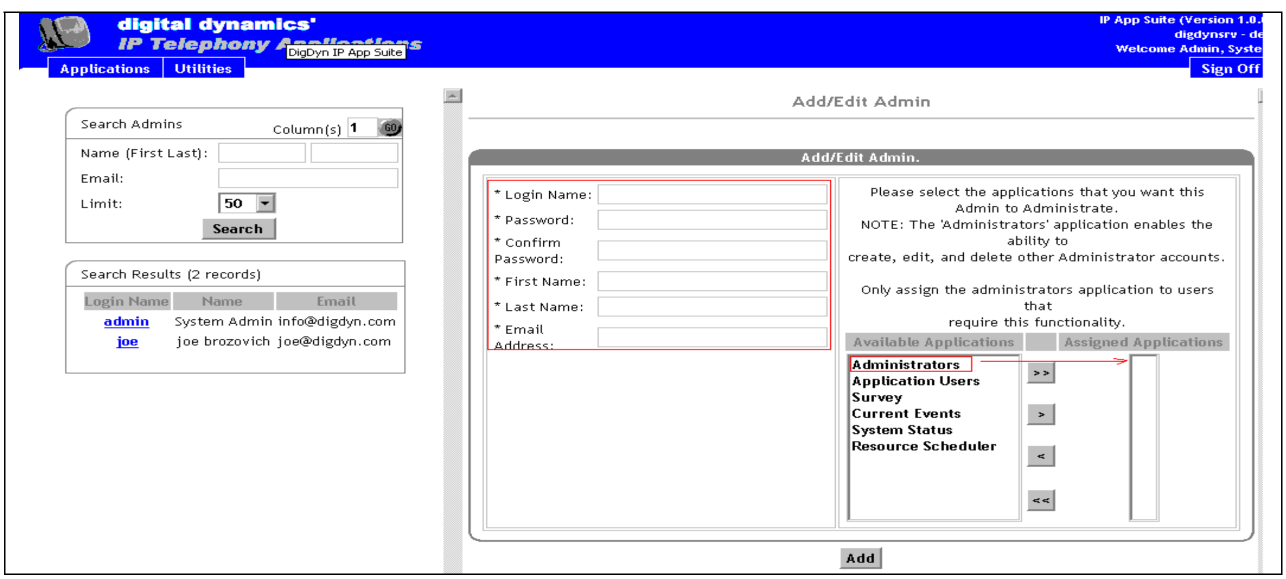


After the update is completed, click the **Applications** button. To add administrator, select the **Administrators** button.

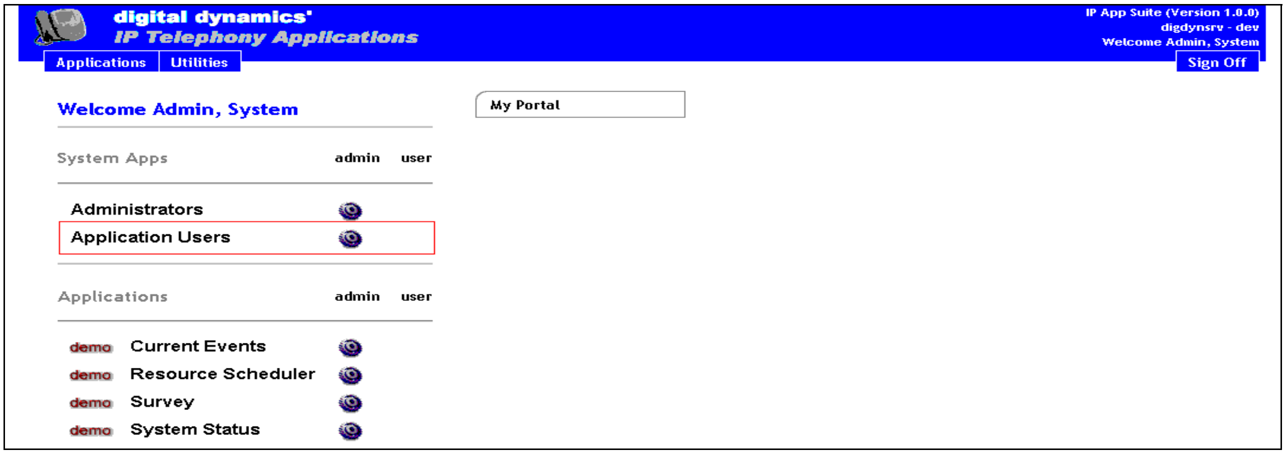
From the Add/Edit Admin window, provide the following:

- **Login Name**
- **Password**
- **Confirm Password**
- **First Name**
- **Last Name**
- **E-mail Address**
- Move the **Administrators** application from the Available Applications folder to the Assigned Applications folder.

Click the **Add** button to add the administrator. Click the **Applications** button to go back to the Applications window.



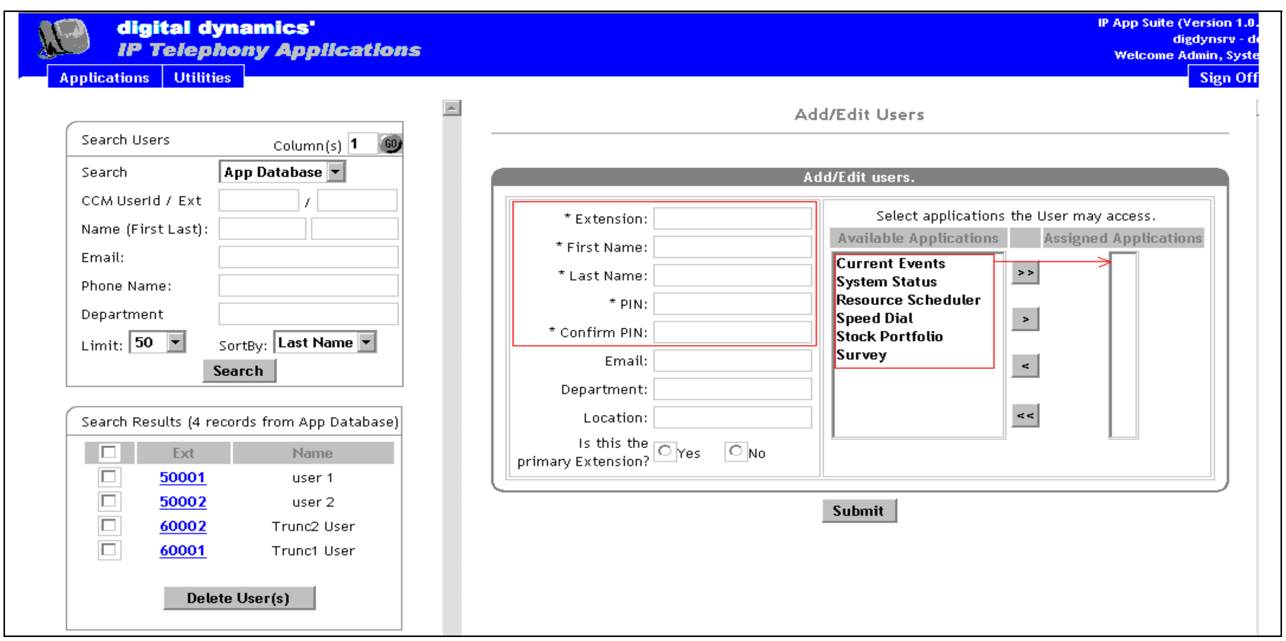
The following two screens show how to add Application users. To add an Application user, select the **Application Users** button.



From the Add/Edit Users window, provide the following:

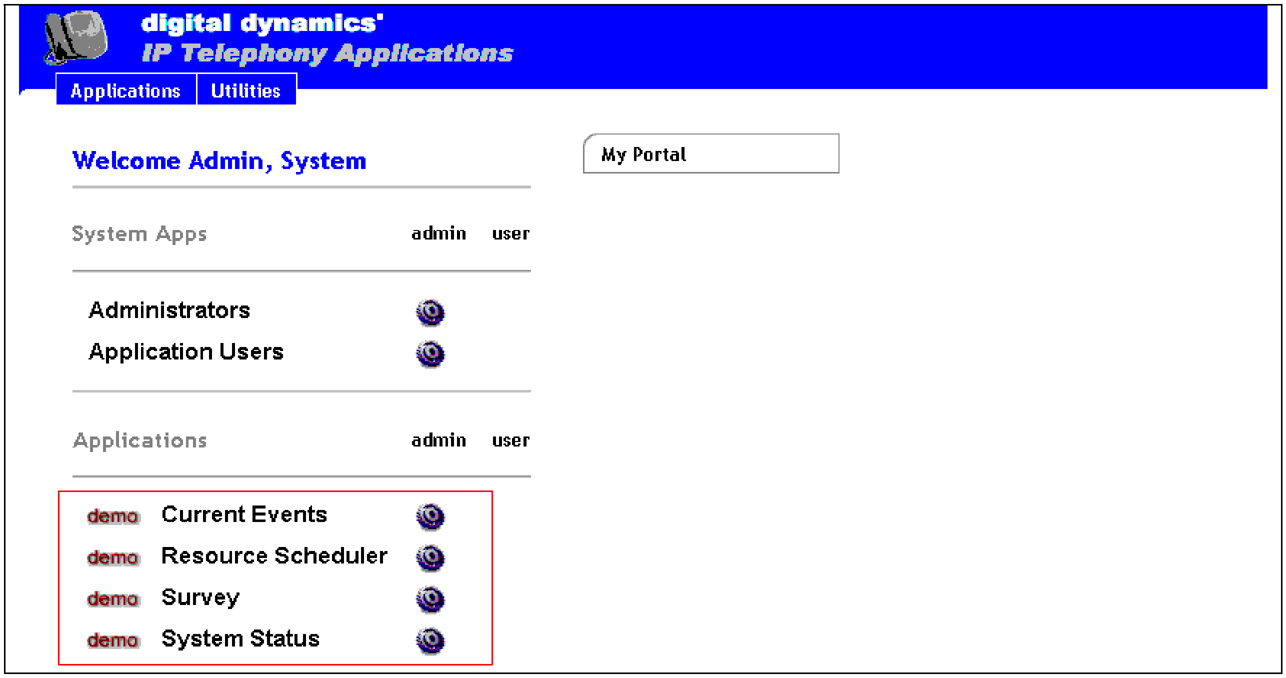
- **Extension**
- **First Name**
- **Last Name**
- **Pin**
- **Confirm Pin**
- Move all applications from the Available Applications folder to the Assigned Applications folder.

After the completion, click the **Submit** button to add a user. Click the **Applications** Button to go back to the Applications window.

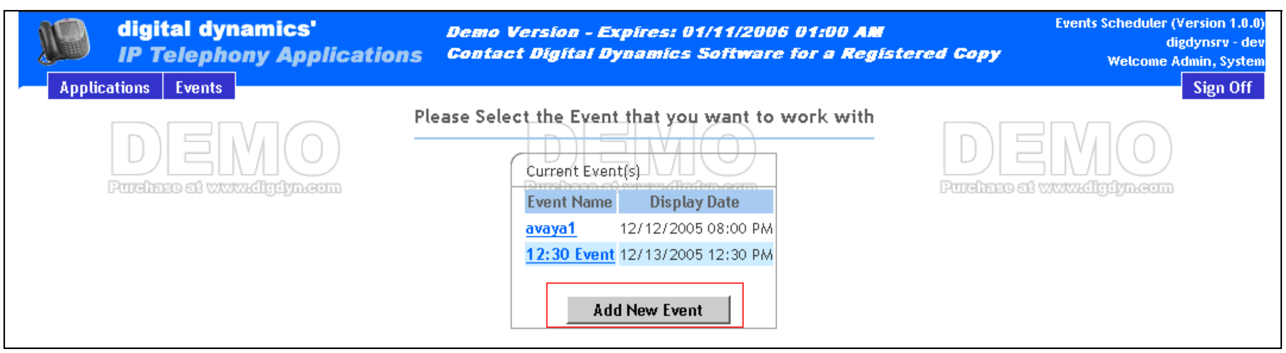


5.1. Configure the Current Events Application

To configure the Current Event application, log in to the IP Telephony Application Suite as Administrator. As shown below, the Administrator can configure four applications. Select the **Current Events** button to configure.



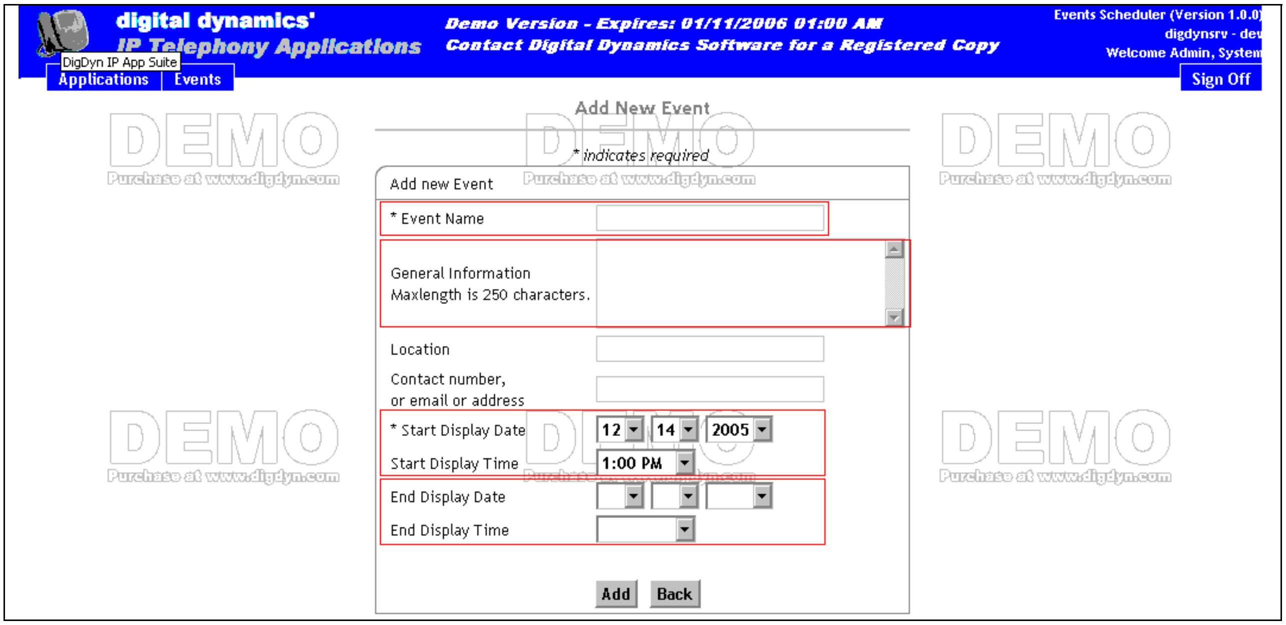
The Current Events window appears. Click the **Add New Event** button to create an event.



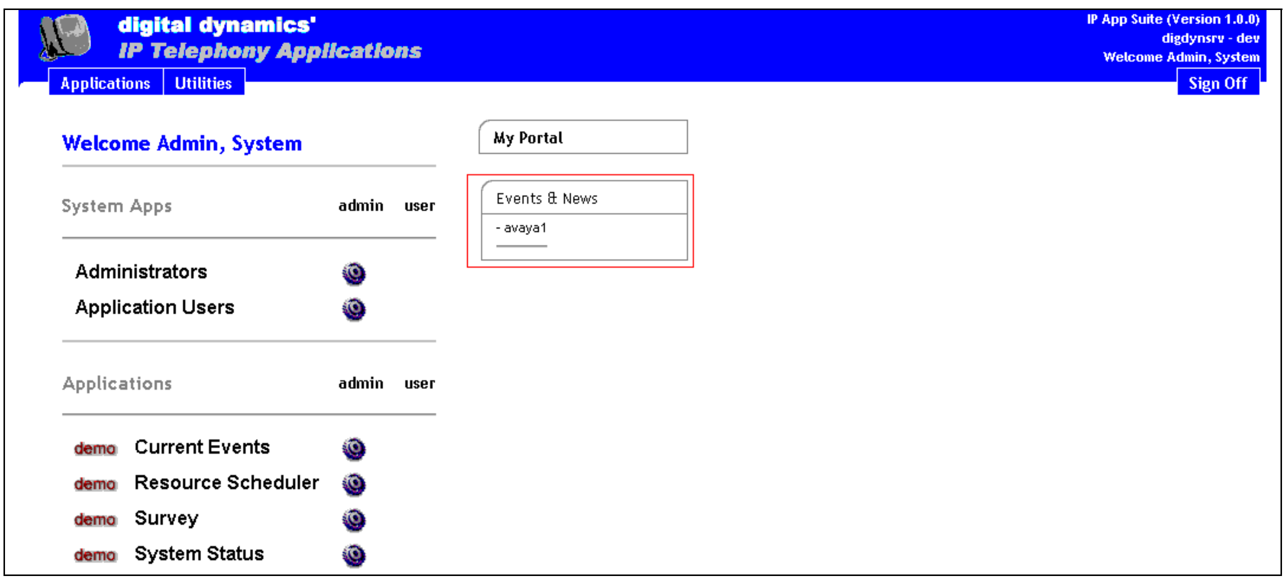
From the Add New Event page, supply the following info:

- **Event Name**
- **General information**
- **Start Display Date & Time** (This is the time the server will push the current event to IP Telephones
- **End Display Date & Time** (This is the time the server will stop pushing the current event message)

Click **Add** button to add the New Event.



Once configured correctly, the event message will appear in the Applications window.

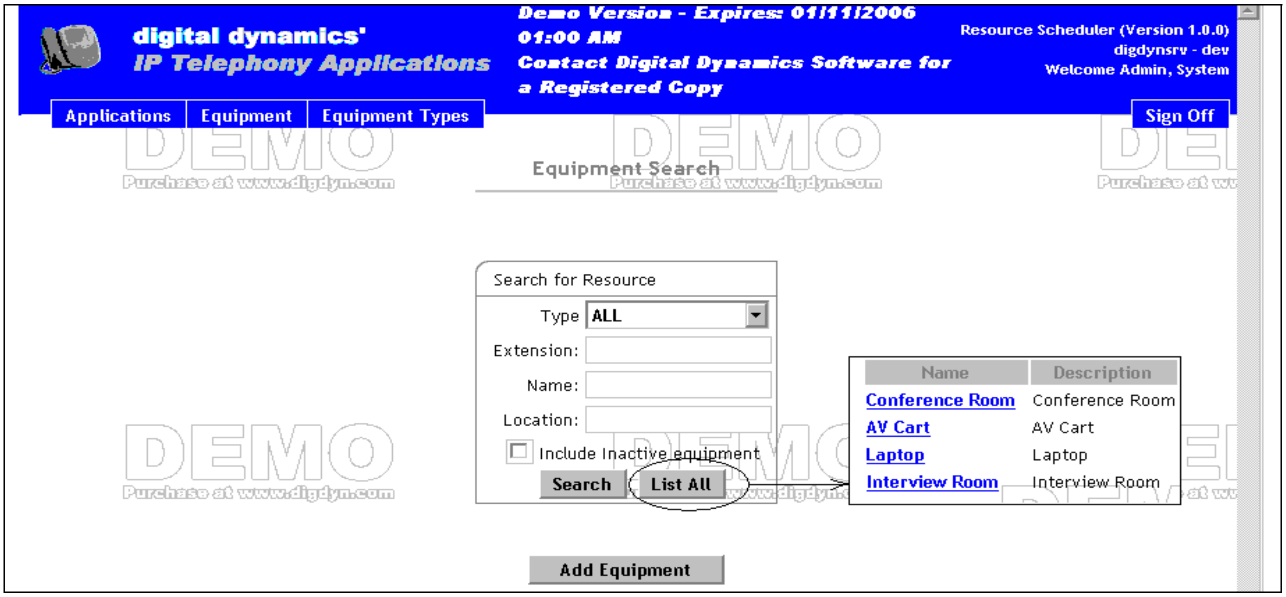


5.2. Configure the Resource Scheduler Application

To configure the Resource Scheduler application, log in to the IP Telephony Application Suite as Administrator and select the **Resource Scheduler** button to configure. The Equipment Search page appears. The following describes the terms used in the Resource Scheduler.

- **Equipment Type** is any resource that can be scheduled or shared (Conference Room, Interview Room).
- **Equipment** is a schedule (or reservation) of the resource.

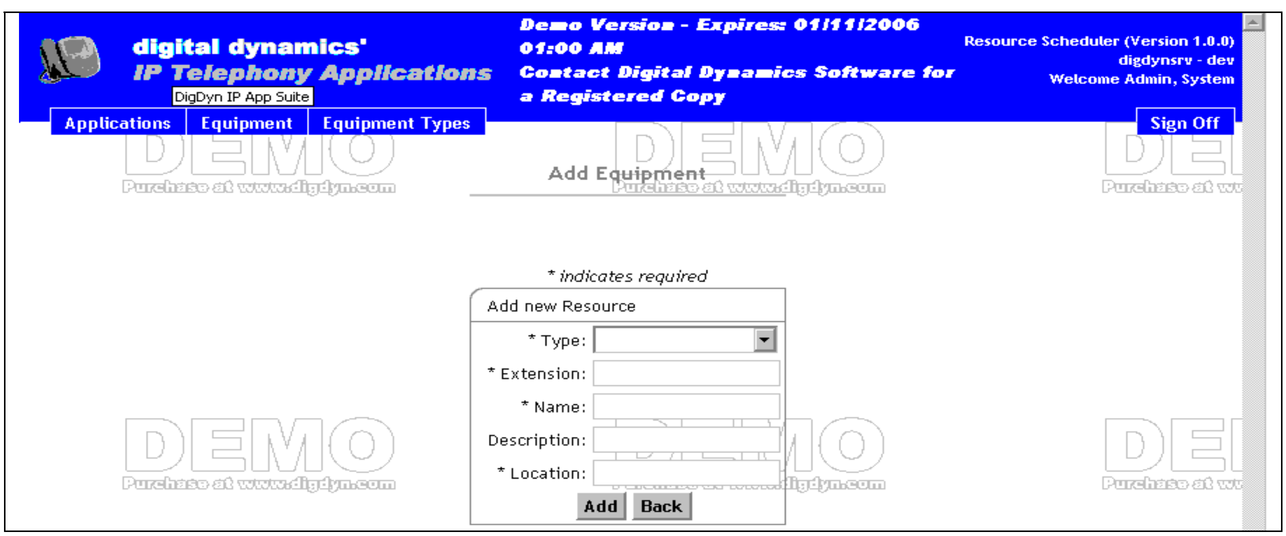
Select the **Add Equipment** button to schedule the resource. By clicking the List All button, the administrator can select an Equipment Type which has been already created, but not activated.



From the Add Equipment page, which schedules for the Equipment Types, provide the following info:

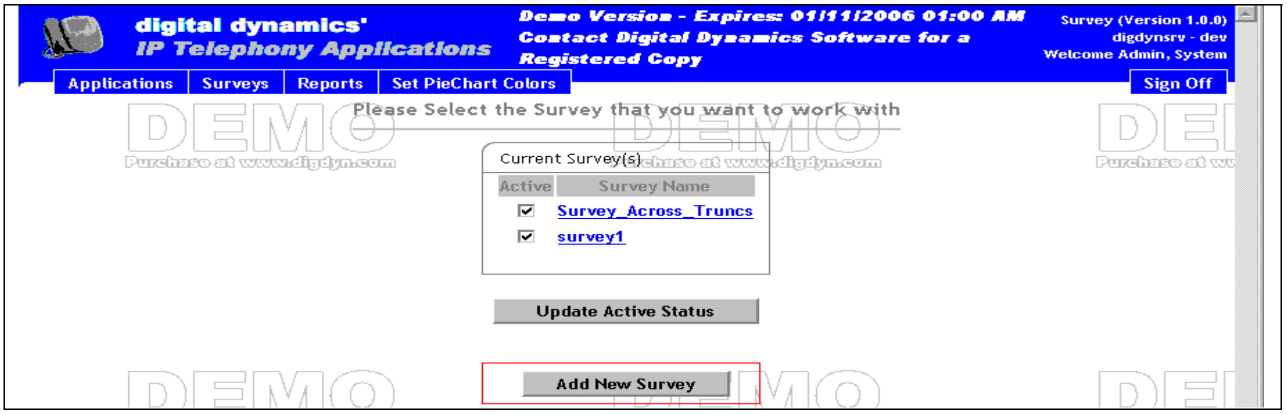
- **Equipment Type** (Resource)
- **Extension of Equipment Type**
- **Name**
- **Location**

After the completion, click the **Add** button to activate the resource. Once the **Add** button is selected, the IP Telephony Application Suite will push the Resource Scheduler to the assigned users.

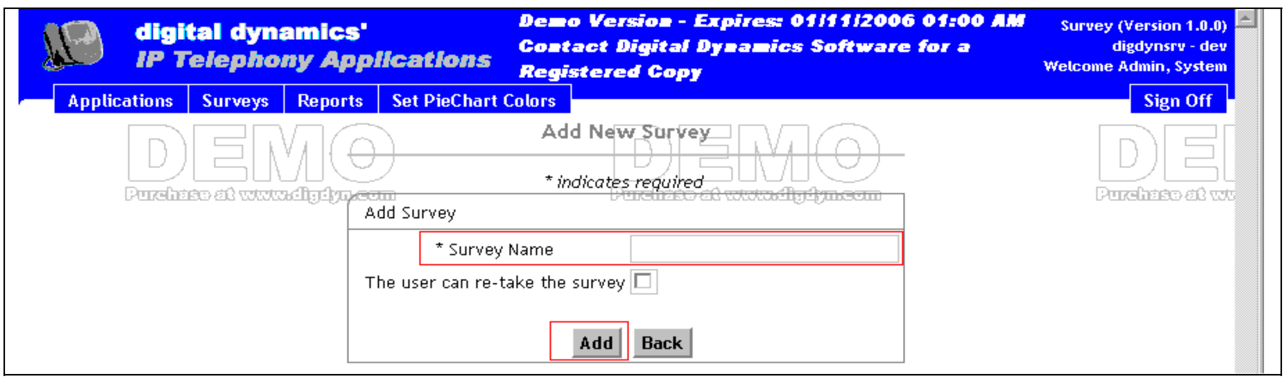


5.3. Configure the Survey Application

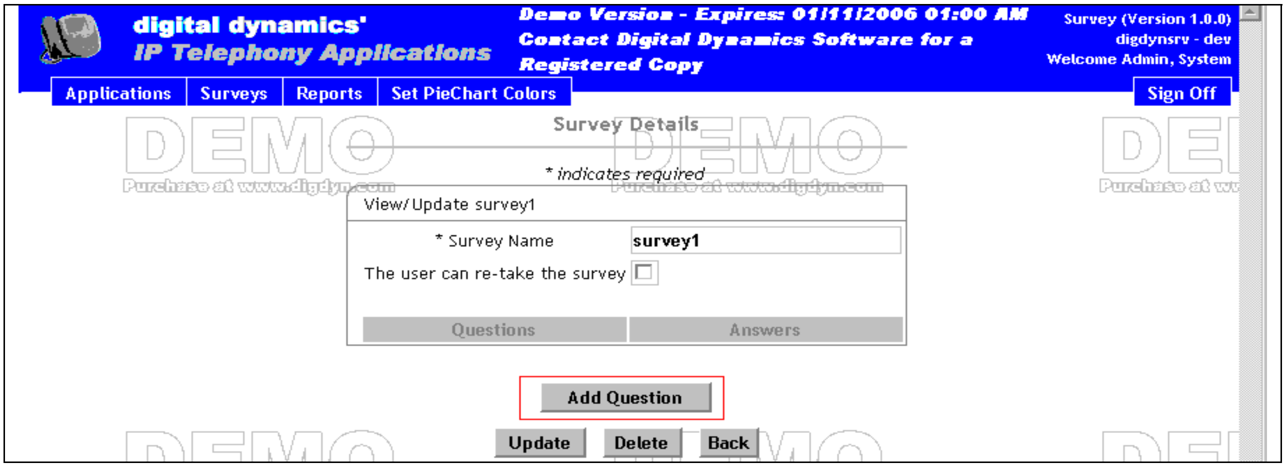
To configure the Surveys application, log in to the IP Telephony Application Suite as a user with administrator privileges. Select the **Surveys** button from the Applications window, and the **Current Survey(s)** page displays. Click the **Add New Survey** button to create a new survey.



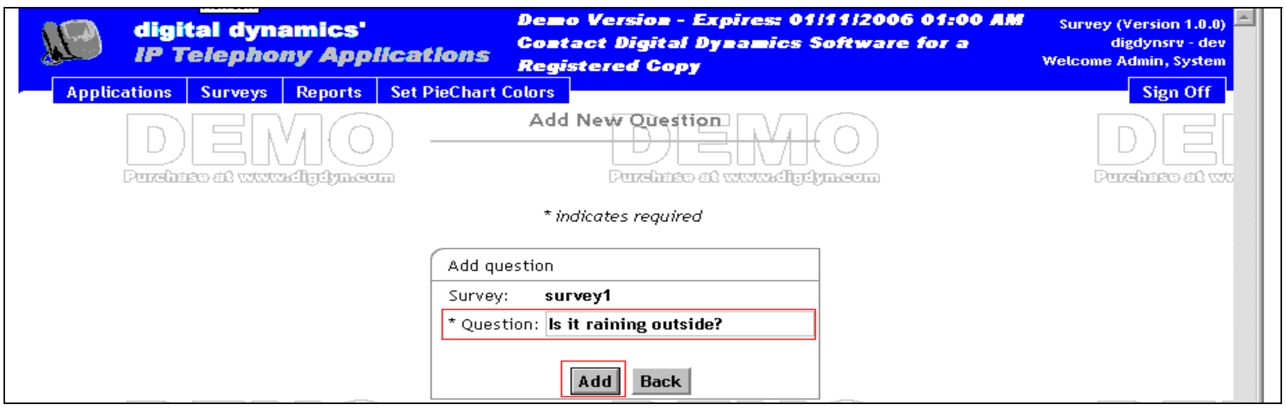
From the Add New Survey page, provide the **Survey Name,** and click the **Add** button.



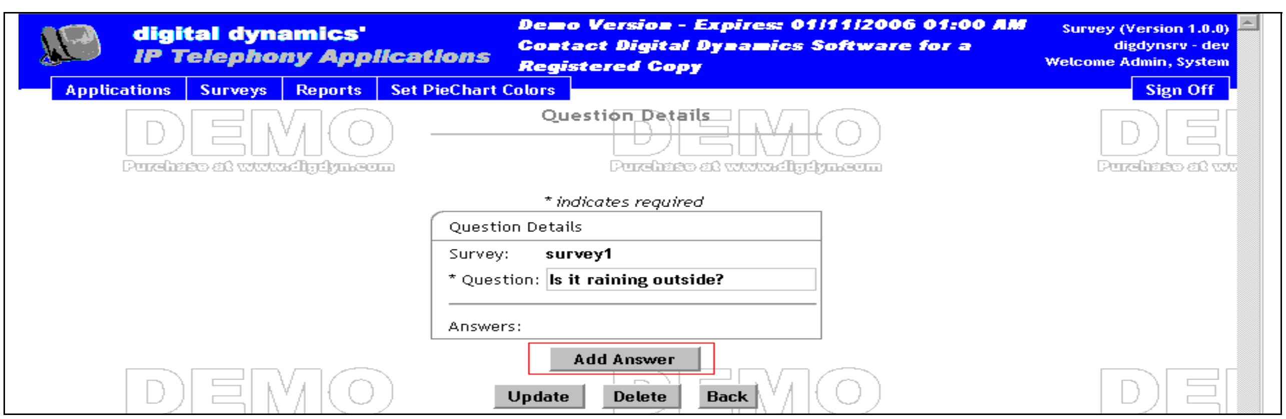
From the Survey Details page, click the **Add Question** button.



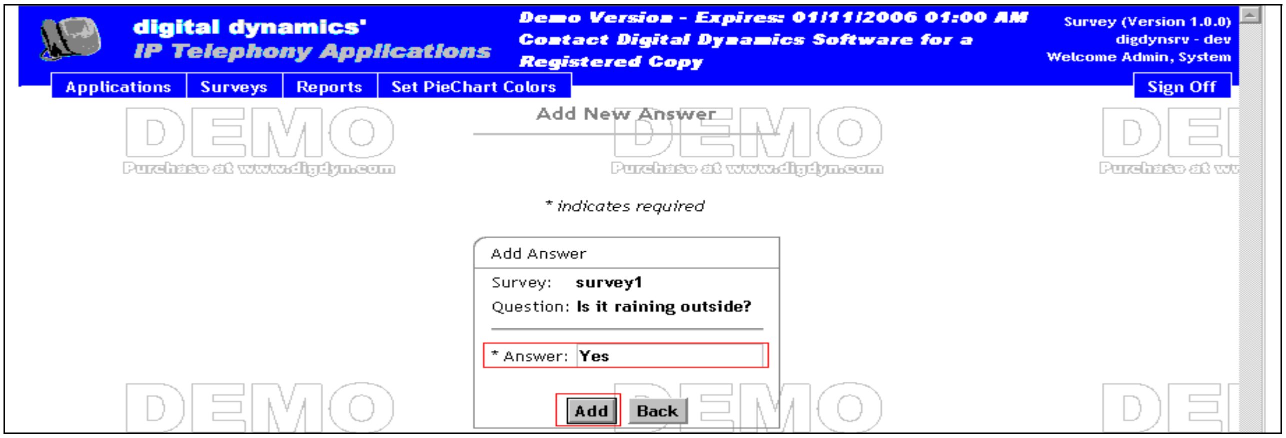
From the Add New Question page, provide a survey questionnaire for users to participate. Click the **Add** button to continue.

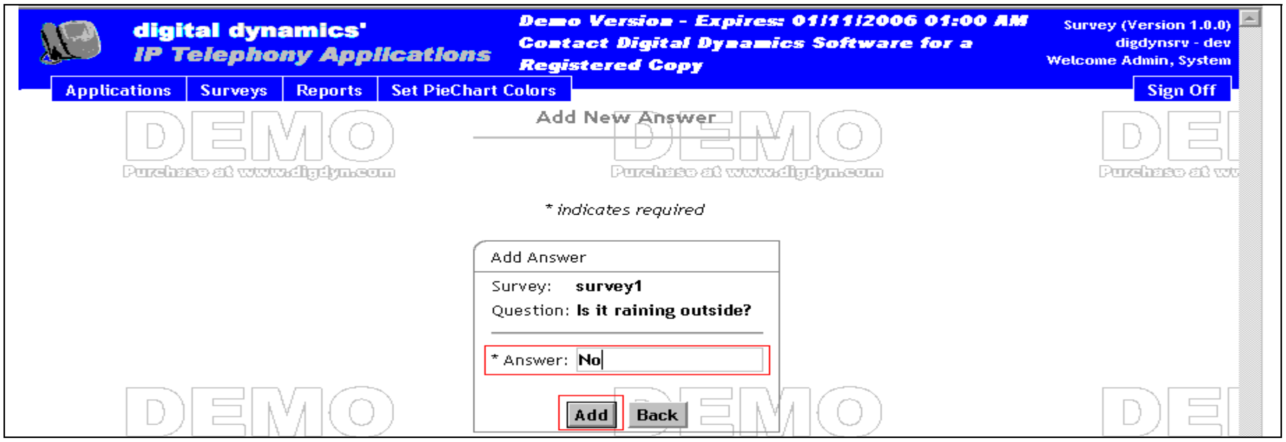


The next screens show the user's possible response type to the question. Click the **Add Answer** button to continue.

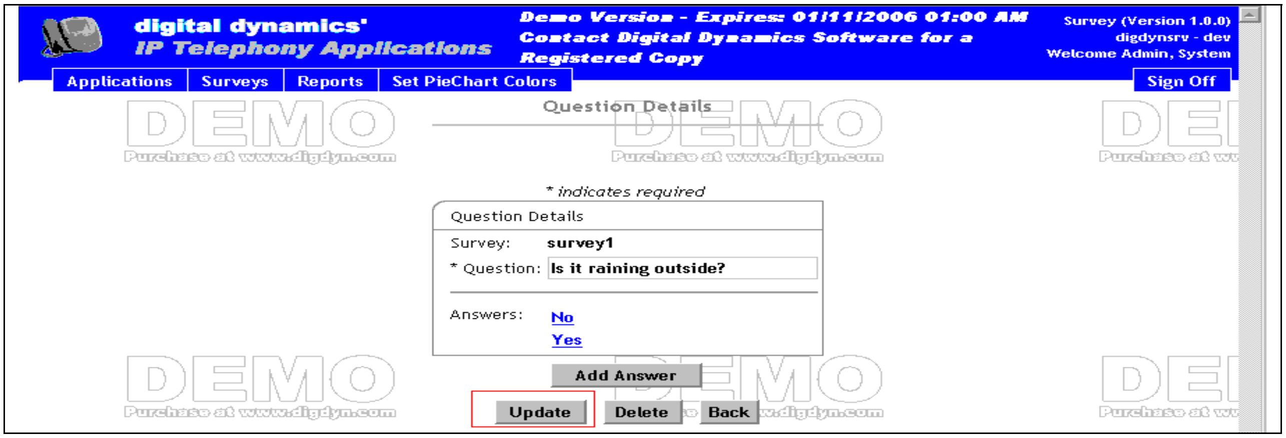


From the **Add New Answer** page, provide the response type to the survey question. For this sample survey, the response types were “**Yes”** or “**No”**. Click the **Add** button.



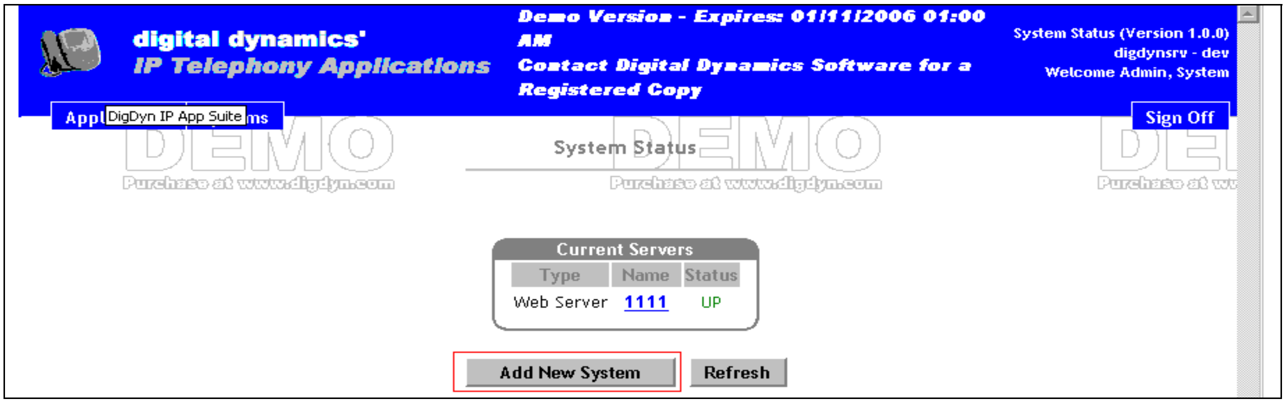


After adding all possible response types, click the **Update** button. The IP Telephony Application Suite will push the Survey application to the assigned users.



5.4. Configure the System Status Application

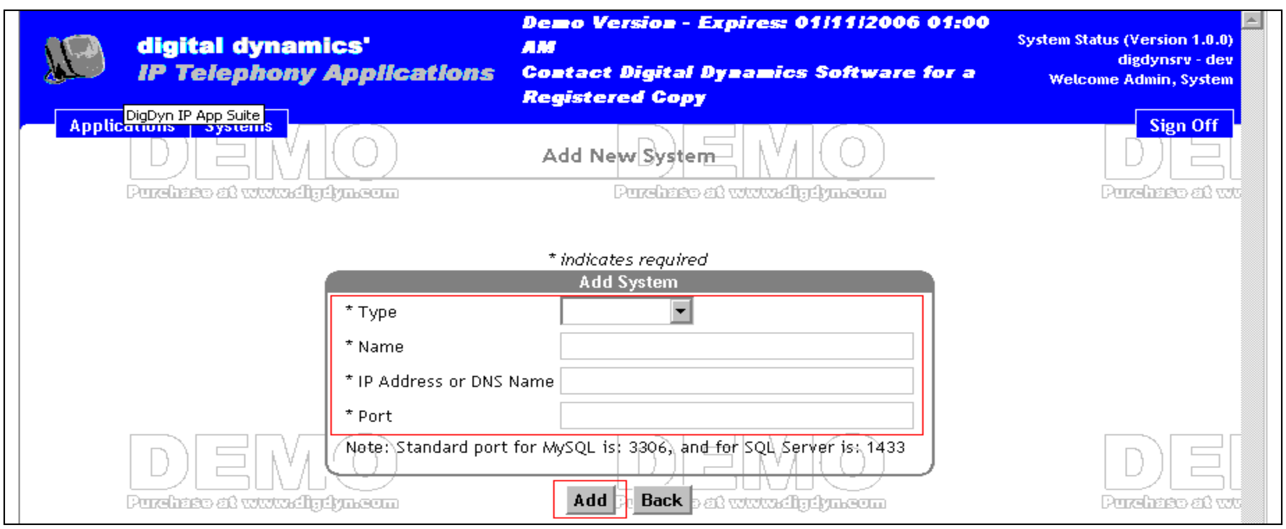
To configure the System Status application, log in to the IP Telephony Application Suite as a user with administrator privileges. Select the **System Status** button from the Applications page, and the System Status page appears. Click the **Add New System** button.



From the **Add New System** page, provide the following info:

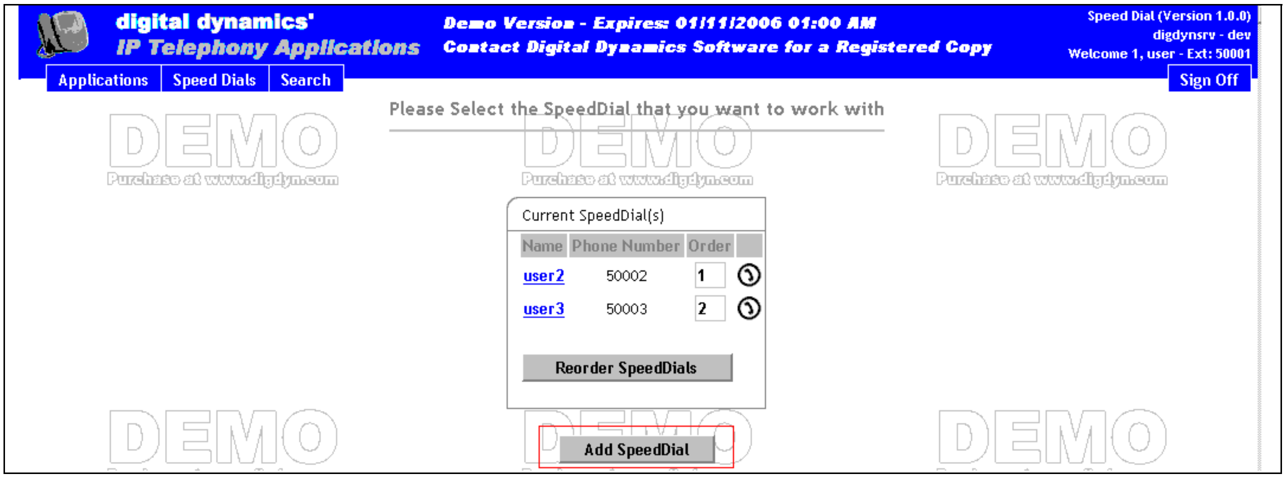
- **Type** of the system
- **Name** of the system
- **IP address** of the system
- **Port**, which the system uses

After the completion, click the **Add** button and the IP Telephony Application Suite will push the System Status application to the assigned users.

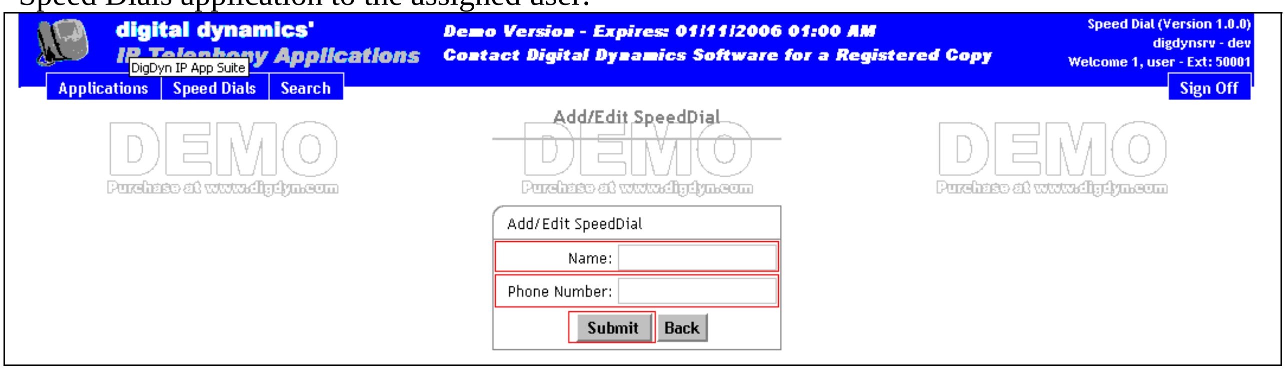


5.5. Configure the Speed Dial Application

To configure the Speed Dial application, log in to the IP Telephony Application Suite as an Application User. Select the **Speed Dials** button, and the Current SpeedDial(s) page appears. Click the **Add SpeedDial** button.

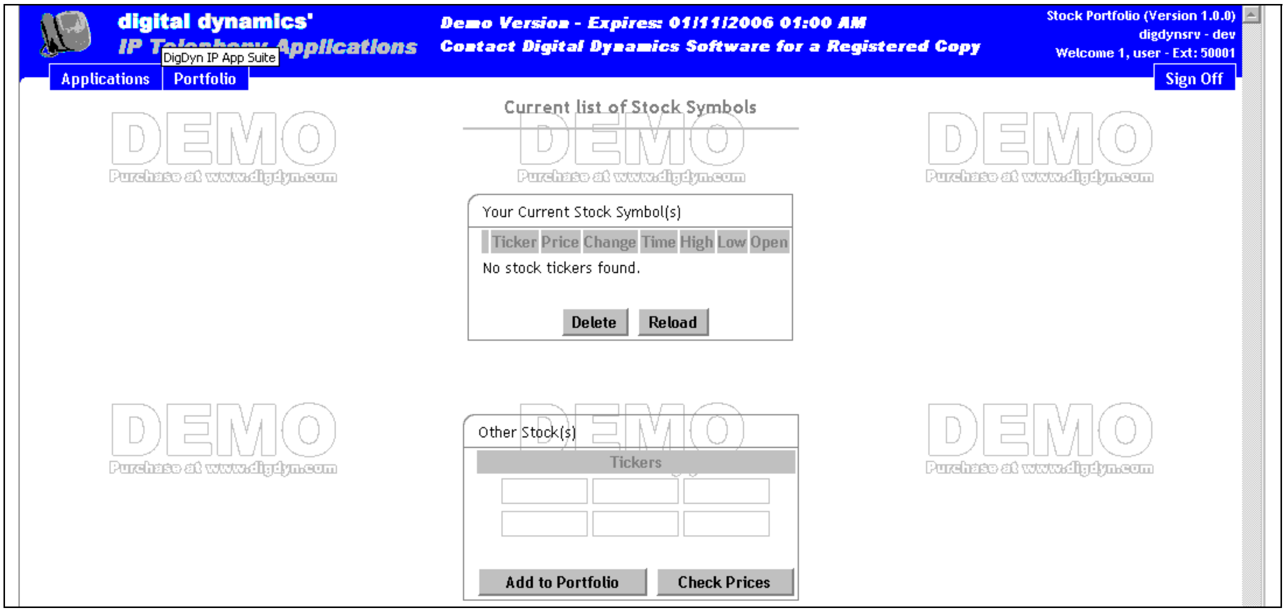


From the **Add/Edit SpeedDial** page, provide the **Name** and **Phone Number** to create a speed dial. Once the **Submit** button is selected, the IP Telephony Application Suite will push the Speed Dials application to the assigned user.



5.6. Configure the Stock Portfolio Application

To configure the Stock Portfolio application, log in to the IP Telephony Application Suite as an Application User. Select the **Stock Portfolio** button and the Current List of Stock Symbols page appears. Provide **Stock Symbols,** and click the **Add to Portfolio** button. After the completion, the IP Telephony Application Suite will push the Stock Portfolio application to the assigned user.



6. Verification Steps

The following steps were used to verify the configuration

- Use the **ping** command to verify connectivity from the IP Telephony Application Suite to all devices.
- Each Avaya IP Telephone was registered and verified that the **Web** softkey is added.

- After pressing the Web softkey, verified the login screen is displayed.
- Press the softkey under the Ext label, and enter the extension. This is a number field. Use the **Num** softkey to enter the Extension number.
- Press the softkey under the Pin label, and enter the Pin number. This is a number field. Use the **Num** softkey to enter the Pin number.

- Press the **Login** button to finish.

- The IP Telephony application list will be displayed after a user login.

- All applications were checked and verified from each Avaya IP Telephone.

7. Interoperability Compliance Testing

Interoperability compliance testing covered connectivity and feature functionality. Feature functionality testing verified that the Digital Dynamics IP Telephony Application Suite communicates with Avaya IP Telephones reliably.

7.1. General Test Approach

All test cases were performed manually. The test was conducted two phases. The first phase of the test was accomplished using Intra-Switch scenario. The second phase of the test was performed using an IP trunk between PBXs (Inter-Switch scenario). The second phase test was needed to simulate a user from the Headquarter coming to the regional office, and utilizing the same Avaya IP Telephone setting that the user used to have in the Headquarter. The following features and functionality were verified from the Avaya IP Telephones by accessing each application:

- Current Events Application
- Resource Scheduler Application
- Survey Application
- System Status Application
- Speed Dial Application
- Stock Portfolio Application

7.2. Test Results

All test cases passed. Avaya IP Telephones were successfully registered to the IP Telephony Application Suite, and did not find any flaw during the compliance test. The Digital Dynamics IP Telephony Application Suite worked reliably with Avaya IP Telephones.

8. Support

For technical support on the Digital Dynamics IP Telephony Application Suite, call Digital Dynamics Support at (800) 330-3830 or send email to support@digdyn.com.

9. Conclusion

These Application Notes describe the configuration steps required for integrating the Digital Dynamics System with Avaya IP Telephones. The systems interoperated successfully, providing a suitable solution to increase workplace productivity by enabling greater collaboration and communication between application users.

10. References

This section references the Avaya and Digital Dynamics documentation that are relevant to these Application Notes.

The following Avaya product documentation can be found at http://support.avaya.com.

[1] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 8, June 2004, Document Number 555-233-504.
[2] *Administrator's Guide for Avaya Communication Manager*, Issue 8, June 2004, Document Number 555-233-506.

The following Digital Dynamics product documentation is provided. For additional product and company information, visit http://www.digdyn.com.

[3] *Digital Dynamics IP Telephony Application Suite Administrator Manual*, Version 1.0

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.