



Avaya Solution & Interoperability Test Lab

Application Notes for VoSKY Exchange Pro VIT1/E1 with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration required for VoSKY Exchange Pro VIT1/E1 to successfully interoperate with Avaya Communication Manager. Exchange Pro VIT1/E1 is a PBX to Skype™ gateway that connects to Avaya Communication Manager via an ISDN-PRI connection and is used to route calls between the enterprise and the Skype Voice over IP (VoIP) network.

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

1. Introduction

These Application Notes describe the configuration required for VoSKY Exchange Pro VIT1/E1 to successfully interoperate with Avaya Communication Manager. Exchange Pro VIT1/E1 is a PBX to Skype™ gateway that connects to Avaya Communication Manager via an ISDN-PRI connection and is used to route calls between the enterprise and the Skype Voice over IP (VoIP) network.

1.1. Interoperability Compliance Testing

The interoperability compliance testing consisted of placing calls through the Exchange Pro and exercising common PBX features. Calls were placed between the Avaya Communication Manager endpoints and Internet users running a Skype client; as well as between the Avaya Communication Manager endpoints and the Skype-connected PSTN. Interoperability with all major enterprise phone types (analog, digital, H.323 and SIP) was tested. See **Section 6** for complete test results.

1.2. Support

Contact VoSKY technical support via the following methods:

Phone: 719-884-7417

On-Line: <http://www.vosky.com/cms/index/support.php>

2. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows the Exchange Pro at the enterprise connected to an ISDN-PRI trunk on one side and the public Internet on the other. The public Internet connection provides access to the Skype service which allows the Exchange Pro to connect to other Skype users and the PSTN.

Located at the enterprise site is an Avaya SES and an Avaya S8300 Server running Avaya Communication Manager in an Avaya G700 Media Gateway. Avaya IA 770 Intuity Audix is also running on the Avaya S8300 Server. Endpoints include an Avaya 4600 Series IP Telephone (with SIP firmware), Avaya 9600 Series IP Telephones (with SIP and H.323 firmware), an Avaya one-X Desktop Edition, an Avaya 6408D Digital Telephone, and an Avaya 6210 Analog Telephone.

Skype users do not have phone numbers but instead are addressed via an alphanumeric Skype ID. In order for PBX endpoints to call these users, the Exchange Pro maps the Skype ID to a number that the PBX user can dial. This mapping is stored in the Exchange Pro phonebook. Similarly, inbound calls from Skype to Exchange Pro are addressed not by a number but by one of several Skype IDs/accounts assigned to the Exchange Pro. The Exchange Pro uses its Skype IDs as a pool of resources for all incoming calls. Calls to any of the Skype IDs can be answered by another if the addressed Skype ID is busy. All calls to any of the Skype IDs are directed to Avaya Communication Manager. Since Skype does not provide a destination phone number, all calls from Exchange Pro are directed to a single number on Avaya Communication Manager.

This number is typically the number of an automated attendant or other IVR application. This number is configurable on Avaya Communication Manager. The Exchange Pro VIT1/E1 supports both T1 and E1 interfaces. However, only the T1 option was tested as part of the compliance test.

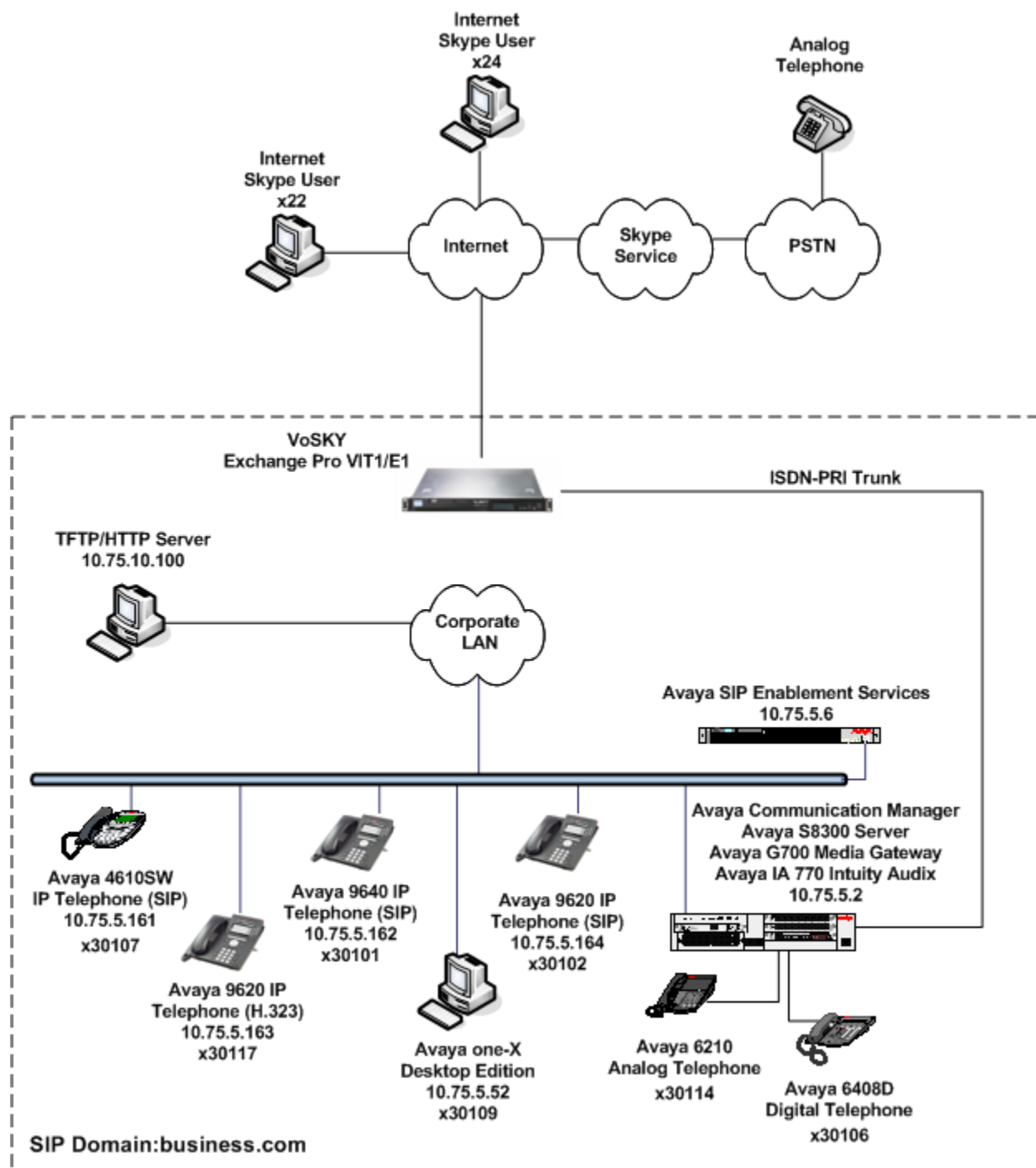


Figure 1: Exchange Pro VIT1/E1 Test Configuration

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300B Server	Avaya Communication Manager 5.1.1 Service Pack (01.1.415.16402) with Avaya IA 770 Intuity Audix
Avaya G700 Media Gateway	MGP: 28.18.0 VOIP: 76
Avaya S8500B Server	Avaya SIP Enablement Services (SES) 5.1.1
Avaya 9620 IP Telephone (H.323)	Avaya one-X Deskphone Edition 2.0
Avaya 4610SW IP Telephones (SIP)	2.2.2
Avaya 9620 IP Telephones (SIP) Avaya 9640 IP Telephones (SIP)	Avaya one-X Deskphone Edition SIP 2.0.5
Avaya one-X Desktop Edition (SIP)	2.1 Service Pack 2
Avaya 6408D Digital Telephone	-
Avaya 6210 Analog Telephone	-
Analog Telephone	-
Windows PC (TFTP/HTTP Server)	Windows XP Professional SP2
VoSKY Exchange Pro VIT1/E1	1.0

4. Configure Avaya Communication Manager

This section describes the Avaya Communication Manager configuration required to create the ISDN-PRI connection to the Exchange Pro and the associated routing. All other aspects of the network shown in **Figure 1** are assumed to already be in place.

The configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent.

Step	Description
1.	<p>DS1</p> <p>The ISDN-PRI connection to the Exchange Pro makes use of a DS1 circuit pack in the Avaya Media Gateway. After selecting which circuit pack will be used for the connection to the Exchange Pro, use the add ds1 x command to add this circuit pack to the system. The x parameter indicates the cabinet/slot location of the circuit pack to be added. For the compliance test, the circuit pack in location 1v4 was used with the settings shown below to support the T1 option. The E1 option was not tested.</p> <ul style="list-style-type: none"> ▪ Enter a descriptive name in the Name field. ▪ Set the Line Coding to b8zs. This must match the T1 Coding field on the Exchange Pro in Section 5, Step 12. ▪ Set the Framing Mode to esf. This must match the T1 Framing field on the Exchange Pro in Section 5, Step 12. ▪ Set the Signaling Mode to isdn-pri. This must match the Signaling setting of pri-net on the Exchange Pro in Section 5, Step 12. ▪ Set the Connect field to network. This value is typically used when connecting to a PSTN service provider. ▪ Set the Country Protocol to 1. The Country Protocol defines the country parameters. The value of 1 includes in the United States. The combination of the Country Protocol and the Protocol Version also defines the version of ISDN-PRI to be used. ▪ Set the Protocol Version to b. The value of b further specifies the ISDN-PRI version as Bellcore TR-1268. This corresponds to a Switchtype setting of national on the Exchange Pro in Section 5, Step 12. ▪ Set the Interface Companding to mulaw. For T1, companding is always set to mulaw. ▪ Default values can be used for all other fields. <div data-bbox="350 1276 1398 1797" style="border: 1px solid black; padding: 10px; margin-top: 20px;"> <pre> add ds1 1v4 Page 1 of 2 DS1 CIRCUIT PACK Location: 001V4 Name: 1v4 DS1 Bit Rate: 1.544 Line Coding: b8zs Line Compensation: 1 Framing Mode: esf Signaling Mode: isdn-pri Connect: network TN-C7 Long Timers? n Country Protocol: 1 Interworking Message: PROGress Protocol Version: b Interface Companding: mulaw CRC? n Idle Code: 11111111 DCP/Analog Bearer Capability: 3.1kHz T303 Timer(sec): 4 Slip Detection? n Near-end CSU Type: other Block Progress Indicator? n </pre> </div>

Step	Description
2.	<p>Signaling Group Create a signaling group for the ISDN-PRI connection to the Exchange Pro by using the add signaling-group <i>n</i> command where <i>n</i> is the number of an unused signaling group. For the compliance test, signaling group 2 was used. Signaling group 2 was configured using the parameters highlighted below.</p> <ul style="list-style-type: none"> ▪ Set the Group Type to <i>isdn-pri</i>. ▪ Set the Primary D-Channel to <i>001V424</i>. This represents the cabinet/slot/port location of the signaling channel (D-channel) of the DS1 circuit pack added in the previous step. In the previous step, the DS1 circuit pack was shown to be in cabinet 001, slot V4. For T1, port 24 is used for signaling. Thus, the complete value for the location is <i>001V424</i>. ▪ The Trunk Group for Channel Selection field is initially left blank. It can not be configured until the trunk group is created (see Step 3). Once the trunk group is created, use the change signaling-group command to set this value to the value of the trunk group. ▪ Default values can be used for all other fields. <div data-bbox="349 888 1398 1152" style="border: 1px solid black; padding: 10px; margin-top: 20px;"> <pre> add signaling-group 2 Page 1 of 5 SIGNALING GROUP Group Number: 2 Group Type: isdn-pri Associated Signaling? y Max number of NCA TSC: 0 Primary D-Channel: 001V424 Max number of CA TSC: 0 Trunk Group for NCA TSC: Trunk Group for Channel Selection: 2 TSC Supplementary Service Protocol: a </pre> </div>

Step	Description
3.	<p>Trunk Group Create a trunk group for the ISDN-PRI connection to the Exchange Pro by using the add trunk-group <i>n</i> command where <i>n</i> is the number of an unused trunk group. For the compliance test, trunk group 2 was used. Trunk group 2 was configured using the parameters highlighted below.</p> <ul style="list-style-type: none"> ▪ Set the Group Type to <i>isdn</i>. ▪ Enter a descriptive name for the Group Name. ▪ Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the TAC field. ▪ Set the Carrier Medium to <i>PRI/BRI</i>. ▪ Set the Service Type field to <i>tie</i>. ▪ Default values can be used for all other fields. <div data-bbox="342 741 1406 1045" style="border: 1px solid black; padding: 10px;"> <pre> add trunk-group 2 Page 1 of 21 TRUNK GROUP Group Number: 2 Group Type: isdn CDR Reports: y Group Name: PSTN COR: 1 TN: 1 TAC: 102 Direction: two-way Outgoing Display? n Carrier Medium: PRI/BRI Dial Access? n 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 </pre> </div>
4.	<p>Trunk Group – continued On Page 2, set the Disconnect Supervision – In and Out fields to <i>y</i>. This is required to allow outbound calls to Exchange Pro to be transferred to another user.</p> <div data-bbox="347 1230 1398 1614" style="border: 1px solid black; padding: 10px;"> <pre> add trunk-group 2 Group Type: isdn Page 2 of 21 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: 13 Incoming Calling Number - Delete: Insert: Format: Bit Rate: 1200 Synchronization: async Duplex: full Disconnect Supervision - In? y Out? y Answer Supervision Timeout: 0 Administer Timers? n CONNECT Reliable When Call Leaves ISDN? n </pre> </div>

Step	Description
5.	<p>Trunk Group – continued</p> <p>On Page 3:</p> <ul style="list-style-type: none"> ▪ Set the Send Name and Send Calling Number fields to y. This allows these values to be sent to the far-end. ▪ Set the Numbering Format field to public. This field specifies the format of the calling party number sent to the far-end. ▪ Default values can be used for all other fields. <div data-bbox="350 525 1398 1056" style="border: 1px solid black; padding: 10px; margin-top: 20px;"> <pre> add trunk-group 2 TRUNK FEATURES ACA Assignment? n Measured: none Wideband Support? n Internal Alert? n Maintenance Tests? y Data Restriction? n NCA-TSC Trunk Member: Send Name: y Send Calling Number: y Used for DCS? n Send EMU Visitor CPN? n Suppress # Outpulsing? n Format: public Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: n Hold/Unhold Notifications? n Modify Tandem Calling Number? n Send UUI IE? y Send UCID? n Send Codeset 6/7 LAI IE? y Dsl Echo Cancellation? n Apply Local Ringback? n US NI Delayed Calling Name Update? n Show ANSWERED BY on Display? y Network (Japan) Needs Connect Before Disconnect? n </pre> </div>

Step	Description
6.	<p>Trunk Group – continued</p> <p>On Page 5 in the Port Column of the Group Member Assignments, enter the port location of each port of the trunk used for the connection to the Exchange Pro. The Code column is filled in automatically. In the Sig Grp column, enter the corresponding signaling group for this connection from Step 2. In the case of the compliance test, all 23 bearer channels of the T1 trunk were used. Only the first 15 are shown in the example below.</p> <pre> add trunk-group 2 Page 5 of 21 TRUNK GROUP Administered Members (min/max): 2/23 GROUP MEMBER ASSIGNMENTS Total Administered Members: 23 Port Code Sfx Name Night Sig Grp 1: 001V401 MM710 2: 001V402 MM710 3: 001V403 MM710 4: 001V404 MM710 5: 001V405 MM710 6: 001V406 MM710 7: 001V407 MM710 8: 001V408 MM710 9: 001V409 MM710 10: 001V410 MM710 11: 001V411 MM710 12: 001V412 MM710 13: 001V413 MM710 14: 001V414 MM710 15: 001V415 MM710 </pre>
7.	<p>Public Unknown Numbering</p> <p>Public unknown numbering defines the calling party number to be sent to the far-end. Use the display public-unknown-numbering command to view the calling party entries. An entry was created for use by the trunk group defined in Step 3. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed across any trunk group (Trk Grp column is blank) will be sent as a 5-digit calling number.</p> <pre> 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 5 3 Total Administered: 1 Maximum Entries: 240 </pre>

Step	Description
8.	<p>Automatic Route Selection (ARS)</p> <p>Automatic Route Selection (ARS) was used to route outbound calls to the PSTN via the Exchange Pro. To dial PSTN numbers, enterprise users would first dial the ARS access code followed by the PSTN number. PSTN numbers beginning with 1732 were used for the compliance test. Use the change ars analysis command to create an entry in the ARS Digit Analysis Table to route 11-digit numbers beginning with 1732 to route pattern 2. Route pattern 2 will direct the call to the Exchange Pro trunk group.</p> <div> <pre>change ars analysis 1732</pre> <div> <div>ARS DIGIT ANALYSIS TABLE</div> <div>Location: all</div> <div>Percent Full: 3</div> <div> <div> <div>Dialed</div> <div>String</div> <div>1732</div> </div> <div> <div>Total</div> <div>Min</div> <div>Max</div> <div>11</div> <div>11</div> </div> <div> <div>Route</div> <div>Pattern</div> <div>2</div> </div> <div> <div>Call</div> <div>Type</div> <div>fnpa</div> </div> <div> <div>Node</div> <div>Num</div> </div> <div> <div>ANI</div> <div>Reqd</div> <div>n</div> </div> </div> </div> </div>

Step	Description
9.	<p>Route Pattern</p> <p>Create a route pattern for use by ARS when routing calls to the PSTN via the Exchange Pro. The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the change route-pattern n command, where n is the number of an unused route pattern to configure the parameters in the following manner. The example below shows the values used for the compliance test.</p> <ul style="list-style-type: none"> ▪ Pattern Name: Enter a descriptive name. ▪ Grp No: Enter the outbound trunk group for the Exchange Pro defined in Step 3. ▪ FRL: Set the Facility Restriction Level (FRL) field to a level that allows access to this trunk for all users that require it. The value of 0 is the least restrictive level. ▪ Pfx Mrk: Set the Prefix Mark to 1. This will prepend a 1 to any 10-digit numbers and leave numbers of any other length unchanged. This was not strictly necessary for the compliance test since only 11-digit PSTN dialing was tested. However, using a Prefix Mark of 1 is common practice when routing calls to the PSTN. ▪ Inserted Digits: 00 The Exchange Pro requires the prefix of 00 be inserted in front of the dialed number when directing a call to the PSTN. ▪ Default values can be used for all other fields. <pre> change route-pattern 2 Page 1 of 3 Pattern Number: 2 Pattern Name: PSTN SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC No Mrk Lmt List Del Digits QSIG 1: 2 0 1 00 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 M 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 </pre>

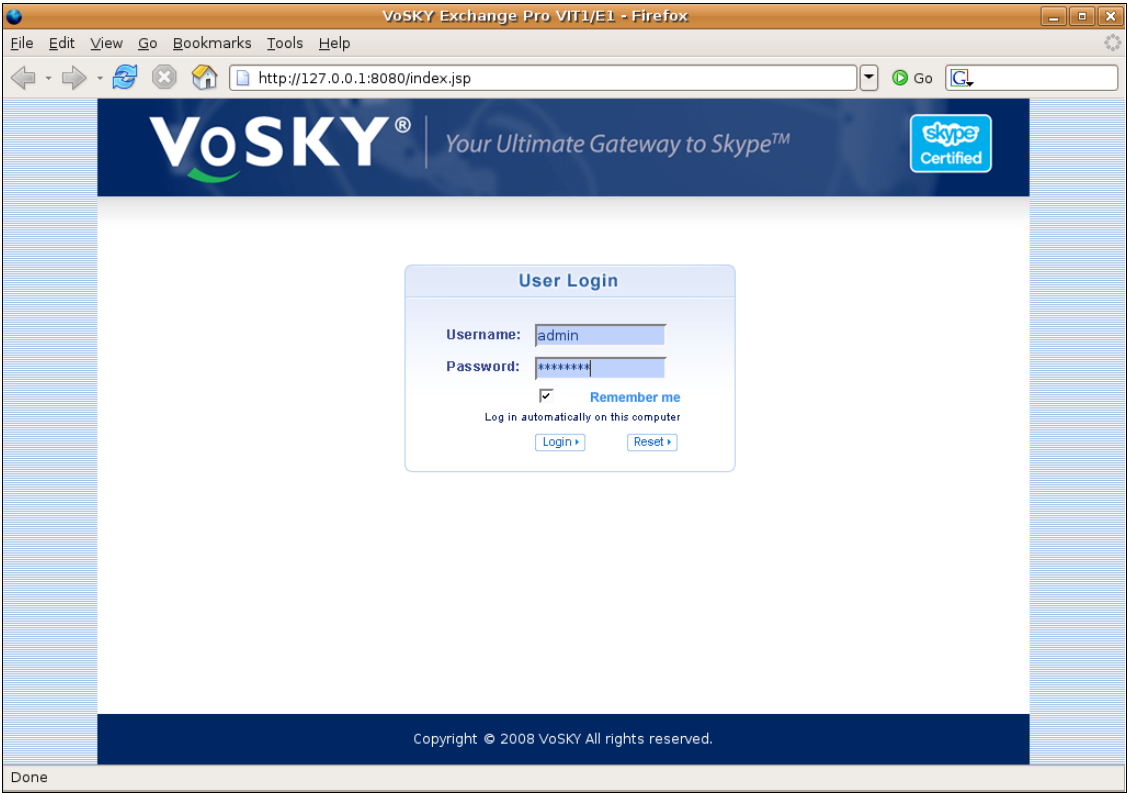
Step	Description
10.	<p>Automatic Alternate Routing (AAR) Automatic Alternate Routing (AAR) was used to route outbound calls to Skype users via the Exchange Pro. To dial the Skype users, enterprise users would first dial the AAR access code followed by the number assigned to the Skype user. For the compliance test, 2-digit numbers beginning with 2 were assigned to the Skype users. Use the change aar analysis command to create an entry in the AAR Digit Analysis Table to route 2-digit numbers beginning with 2 to route pattern 21. Route pattern 21 will direct the call to the Exchange Pro trunk group.</p> <pre> change aar analysis 2 AAR DIGIT ANALYSIS TABLE Location: all Percent Full: 3 Dialed Total Route Call Node ANI String Min Max Pattern Type Num Req'd 2 2 2 21 aar n </pre>
11.	<p>Route Pattern Create a route pattern for use by AAR when routing calls to the Skype users via the Exchange Pro. Create the route pattern in the same manner and using the same values as the route pattern configured in Step 9 with the following exceptions.</p> <ul style="list-style-type: none"> ▪ Pattern Name: Enter a unique name. ▪ Pfx Mrk: Leave the Pfx Mrk field blank. There is no need to set the Prefix Mark to 1 in this case since no 10-digit numbers will use this route pattern. ▪ Inserted Digits: Leave the Inserted Digits field blank. The Exchange Pro does not require any prefix be inserted in front of the dialed number of the Skype users. ▪ Default values can be used for all other fields. <pre> change route-pattern 21 Pattern Number: 21 Pattern Name: ExchangePro2 SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC No Mrk Lmt List Del Digits QSIG Intw 1: 2 0 2: 3: 4: 5: 6: 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 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 </pre>

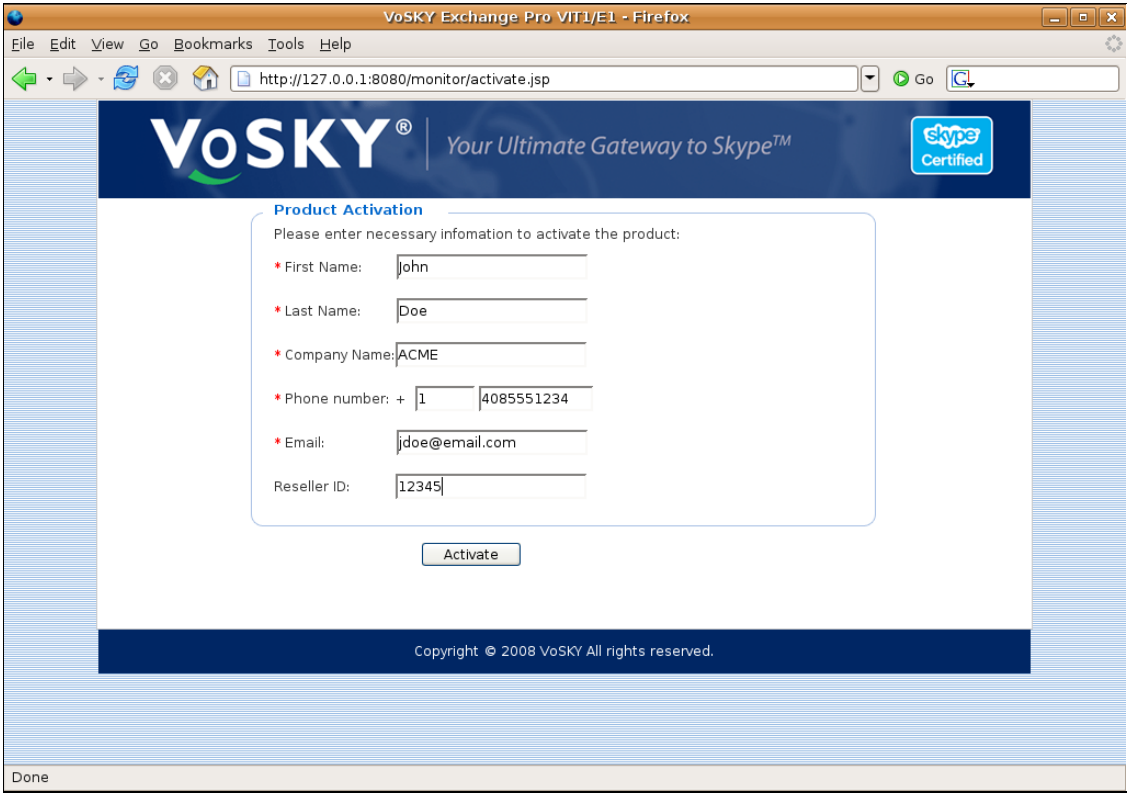
Step	Description
12.	<p>Inbound Calls</p> <p>Since incoming Skype calls do not have a destination number, then all of these calls should be directed to a single number on Avaya Communication Manager. This number is typically configured to be the extension of an automated attendant or other IVR application. Use the change inc-call-handling-trmt trunk-group 2 command to map all incoming dialed strings on trunk 2 to the extension of the auto-attendant 39100. The Called Len and Called Number fields are left blank which will match any dialed string. The Del field is set to 11 which will delete 11 digits of the dialed string which will be all the digits of any string of 11 digits or less. The Insert field is set to 39100 which is the extension of the auto-attendant.</p> <pre> change inc-call-handling-trmt trunk-group 2 INCOMING CALL HANDLING TREATMENT Service/ Called Called Del Insert Per Call Night Feature Len Number tie tie </pre>
13.	<p>Vector Directory Number (VDN)</p> <p>Extension 39100 is a vector directory number (VDN) which invokes a vector which implements a simple automated attendant. In the case of the compliance test, VDN 39100 invokes vector 1. To create a VDN, use the add vdn command. Enter any descriptive name for the Name* field. In the Vector Number field, enter the vector number to be invoked (see Step 14).</p> <pre> add vdn 39100 VECTOR DIRECTORY NUMBER Extension: 39100 Name*: AutoAttendant Vector Number: 1 Meet-me Conferencing? n Allow VDN Override? n COR: 1 TN*: 1 Measured: none Service Objective (sec): 20 1st Skill*: 2nd Skill*: 3rd Skill*: * Follows VDN Override Rules </pre>

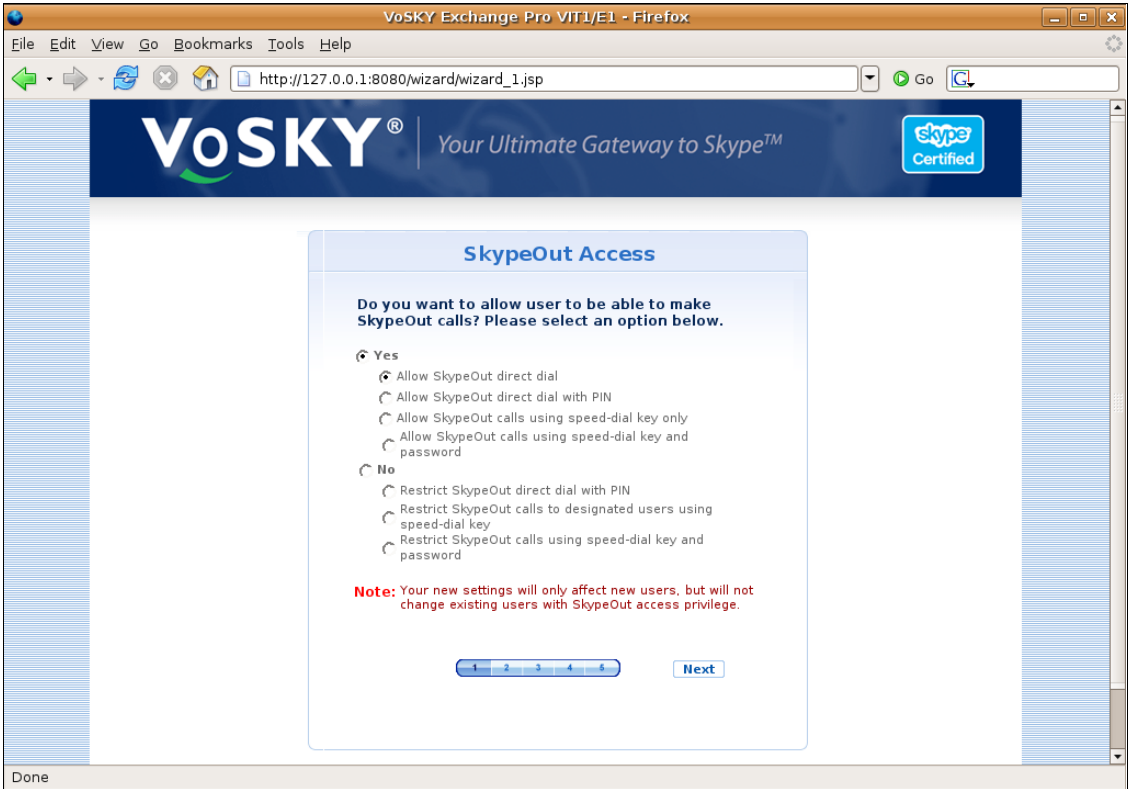
Step	Description
14.	<p>Automated Attendant Vector</p> <p>Vector 1 was used to provide an automated attendant for incoming calls. The configuration of vector 1 is shown below. A vector can be created with the change vector command.</p> <ul style="list-style-type: none"> • Name: Any descriptive name • Step 01: Collect 5 digits. No announcement is played. • Step 02: Route the calls to the extension collected in vector step 01 and if necessary proceed to coverage. <div data-bbox="350 569 1398 856" style="border: 1px solid black; padding: 10px; margin-top: 20px;"> <pre> change vector 1 Page 1 of 6 CALL VECTOR Number: 1 Name: AutoAttendant Meet-me Conf? n Lock? n Basic? y EAS? y G3V4 Enhanced? y ANI/II-Digits? y ASAI Routing? n Prompting? y LAI? y G3V4 Adv Route? y CINFO? y BSR? y Holidays? y Variables? n 3.0 Enhanced? y 01 collect 5 digits after announcement none 02 route-to digits with coverage y 03 </pre> </div>

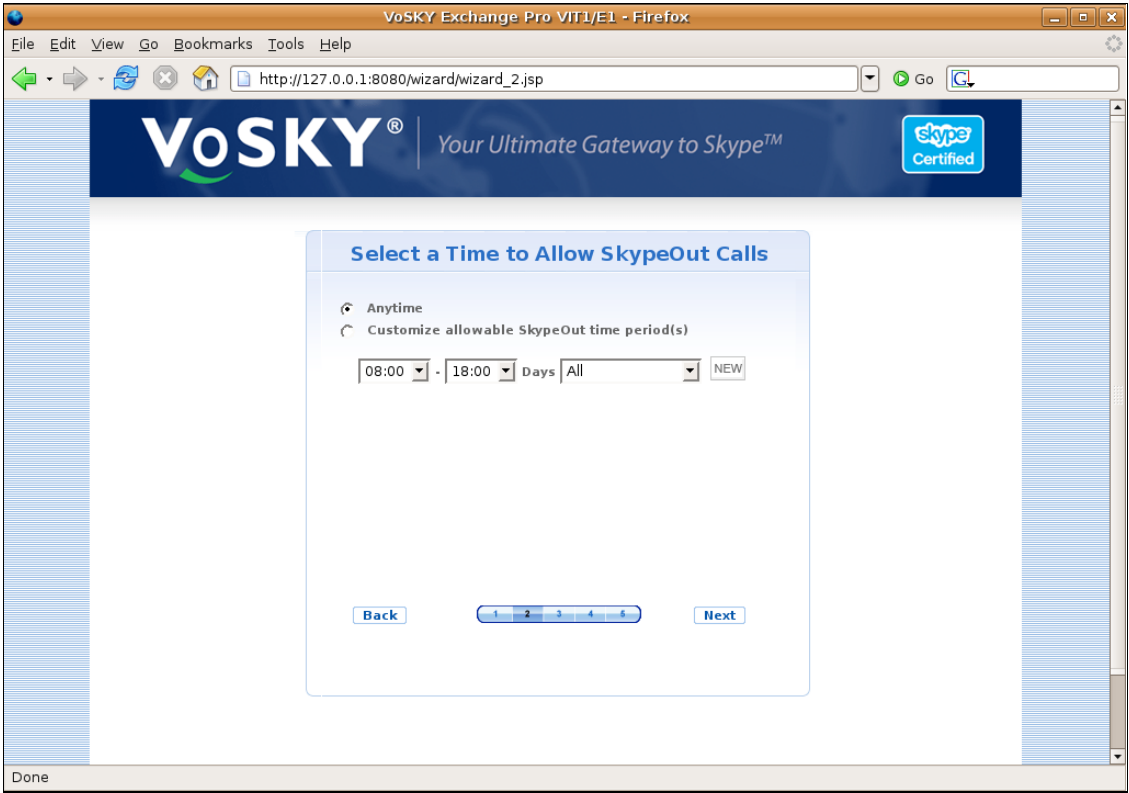
5. Configure Exchange Pro VIT1/E1

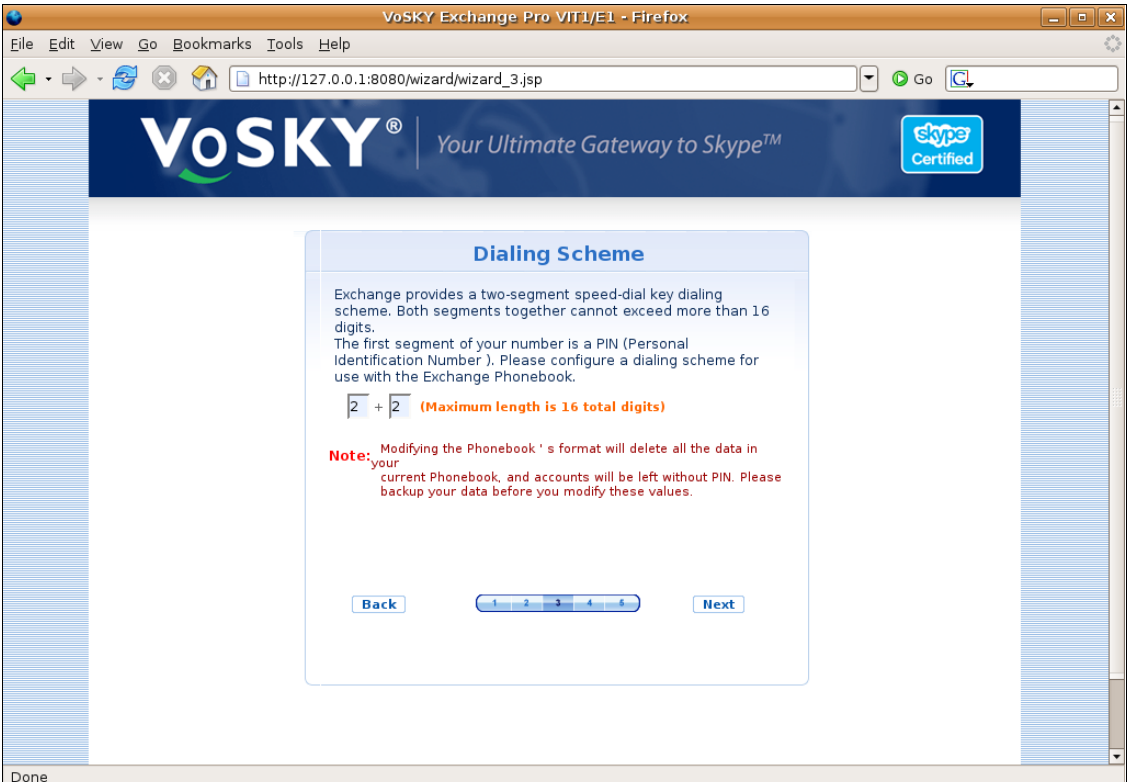
This section describes the configuration of the Exchange Pro. The Exchange Pro is configured via a web interface. On the Exchange Pro server, launch the Mozilla Firefox browser from the Linux GNOME desktop environment. The default home page opens to the login page.

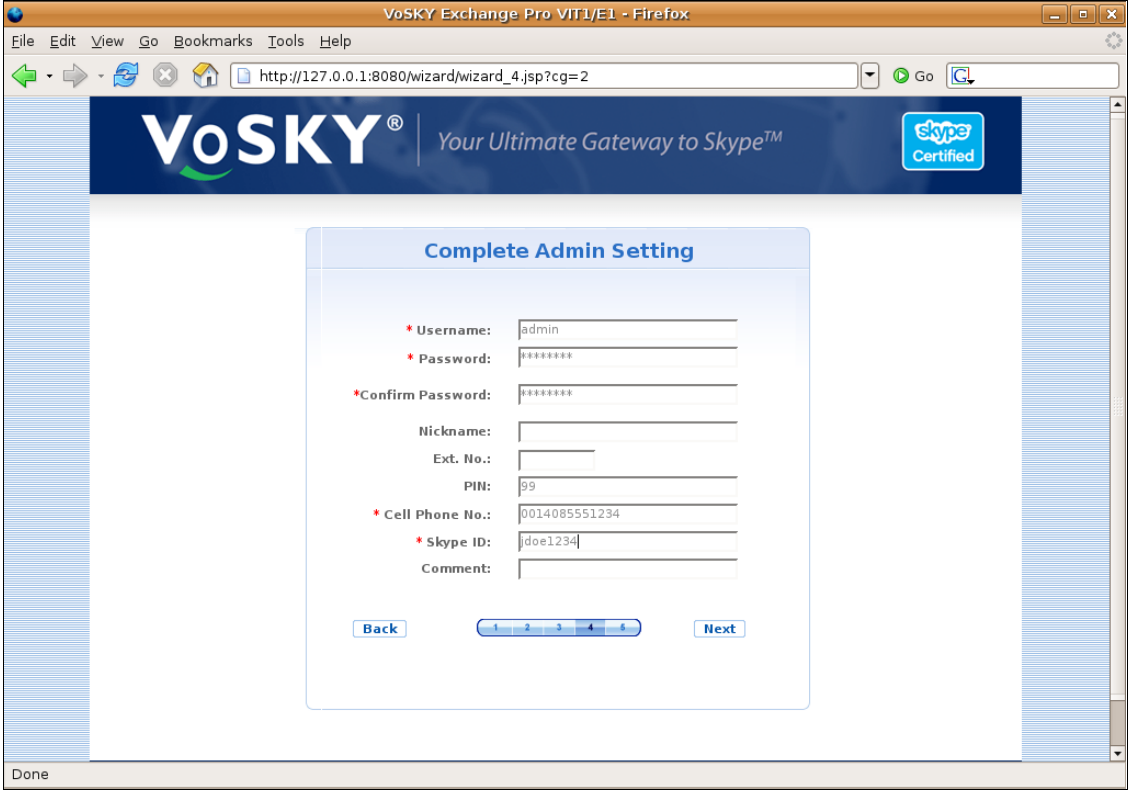
Step	Description
1.	<p>Login On the initial Login page, enter a proper Username and Password. Click Login.</p> 

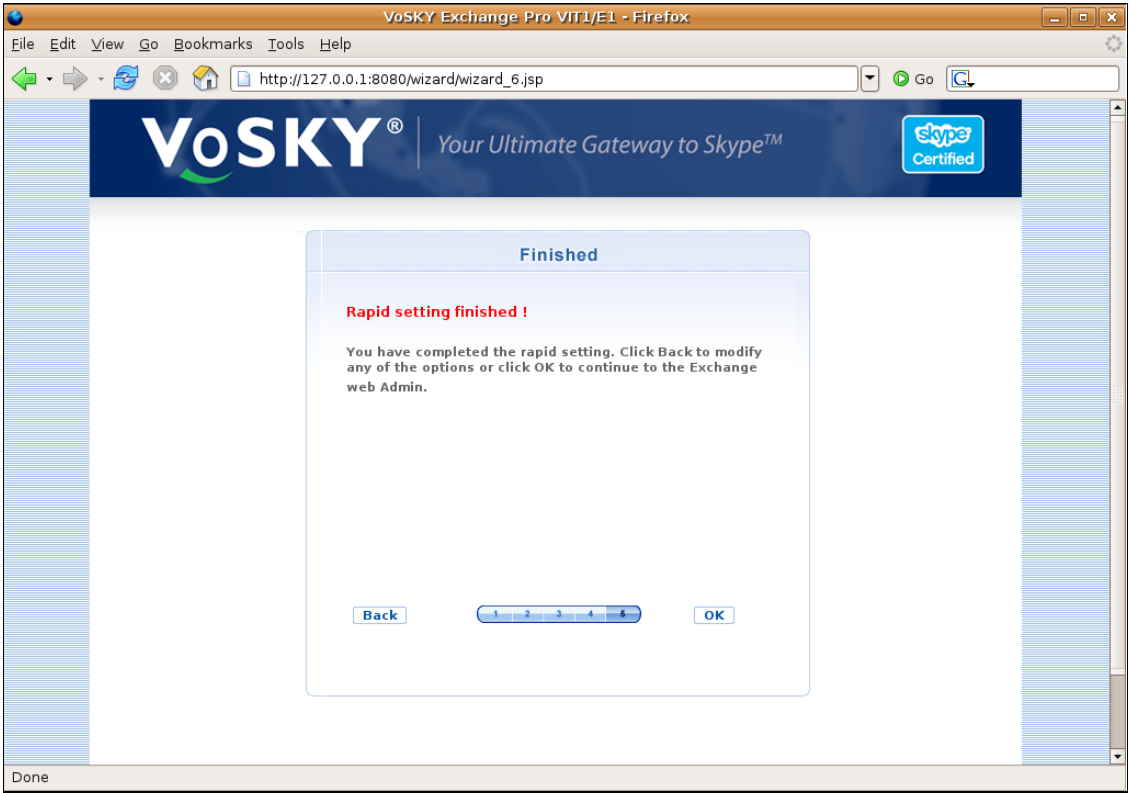
Step	Description
2.	<p>Product Activation</p> <p>The first time the web interface is accessed, the following Product Activation screen will appear. Enter all required fields indicated by a *, then click Activate.</p> 

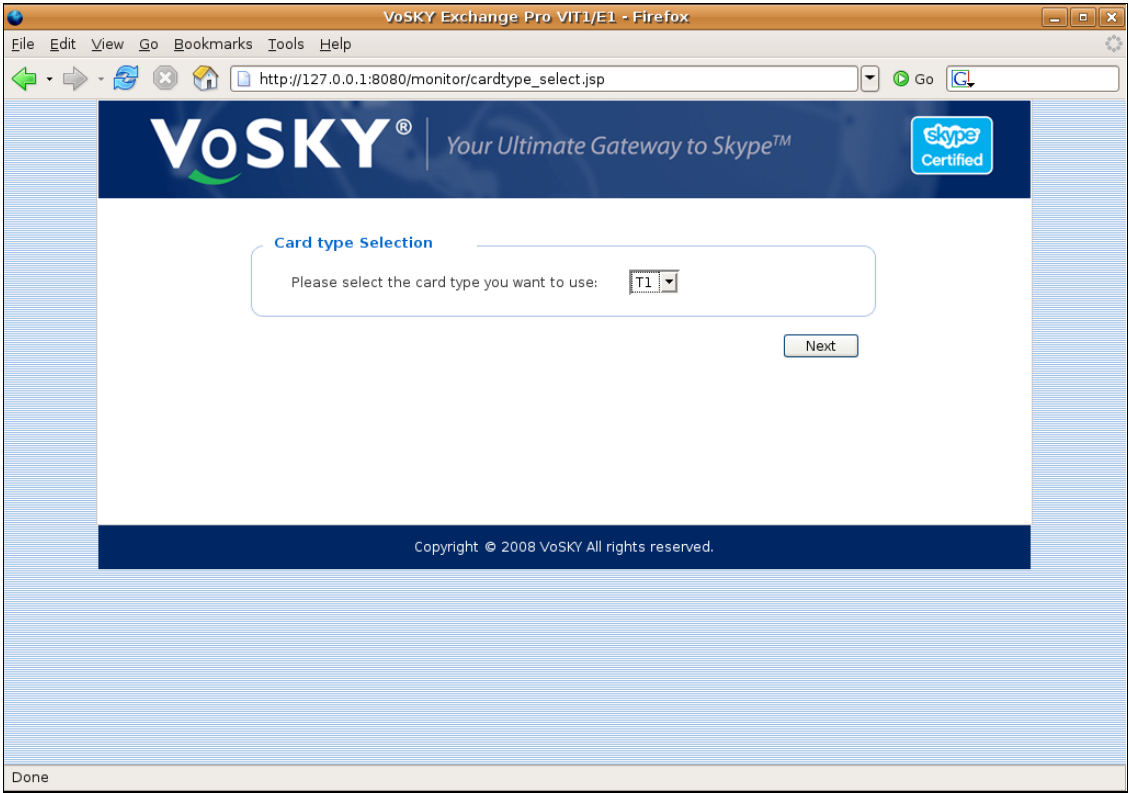
Step	Description
3.	<p>SkypeOut Access</p> <p>At this point, the configuration wizard will start automatically with the SkypeOut Access screen shown below. Select the appropriate options then click Next to proceed. For the compliance test, users were allowed to make outbound Skype calls. Thus, the Yes option was selected followed by the Allow SkypeOut direct dial option. This allows users to dial the outbound number directly without requiring a PIN, speed-dial key or password.</p>  <p>The screenshot shows a Firefox browser window titled 'VoSKY Exchange Pro VIT1/E1 - Firefox'. The address bar shows 'http://127.0.0.1:8080/wizard/wizard_1.jsp'. The page features the VoSKY logo and a 'Skype Certified' badge. The main content area is titled 'SkypeOut Access' and contains the following text: 'Do you want to allow user to be able to make SkypeOut calls? Please select an option below.' There are two main sections: 'Yes' and 'No'. Under 'Yes', there are four options: 'Allow SkypeOut direct dial' (selected), 'Allow SkypeOut direct dial with PIN', 'Allow SkypeOut calls using speed-dial key only', and 'Allow SkypeOut calls using speed-dial key and password'. Under 'No', there are three options: 'Restrict SkypeOut direct dial with PIN', 'Restrict SkypeOut calls to designated users using speed-dial key', and 'Restrict SkypeOut calls using speed-dial key and password'. A red note states: 'Note: Your new settings will only affect new users, but will not change existing users with SkypeOut access privilege.' At the bottom, there is a progress bar with 5 steps, the first of which is highlighted, and a 'Next' button.</p>

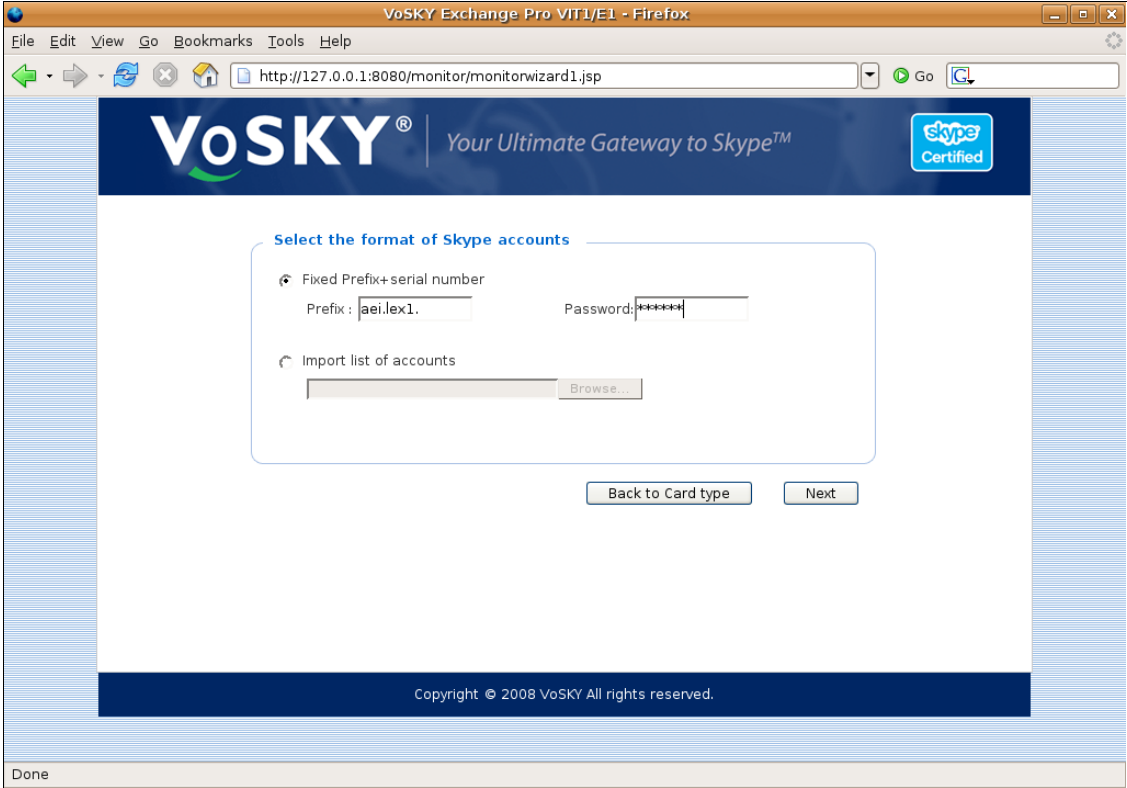
Step	Description
4.	<p>Time Outbound Calls Are Available</p> <p>Select which days and what time of day to allow outbound calls. Click Next to proceed. For the compliance test, the Anytime option was selected.</p>  <p>The screenshot shows a Firefox browser window titled 'VoSKY Exchange Pro VIT1/E1 - Firefox'. The address bar shows 'http://127.0.0.1:8080/wizard/wizard_2.jsp'. The page features the VoSKY logo and a 'skype Certified' badge. The main content area is a wizard titled 'Select a Time to Allow SkypeOut Calls'. It has two radio buttons: 'Anytime' (selected) and 'Customize allowable SkypeOut time period(s)'. Below the radio buttons, there are two time pickers set to '08:00' and '18:00', a 'Days' dropdown set to 'All', and a 'NEW' button. At the bottom of the wizard are 'Back' and 'Next' buttons, and a progress indicator with five steps, where the second step is highlighted.</p>

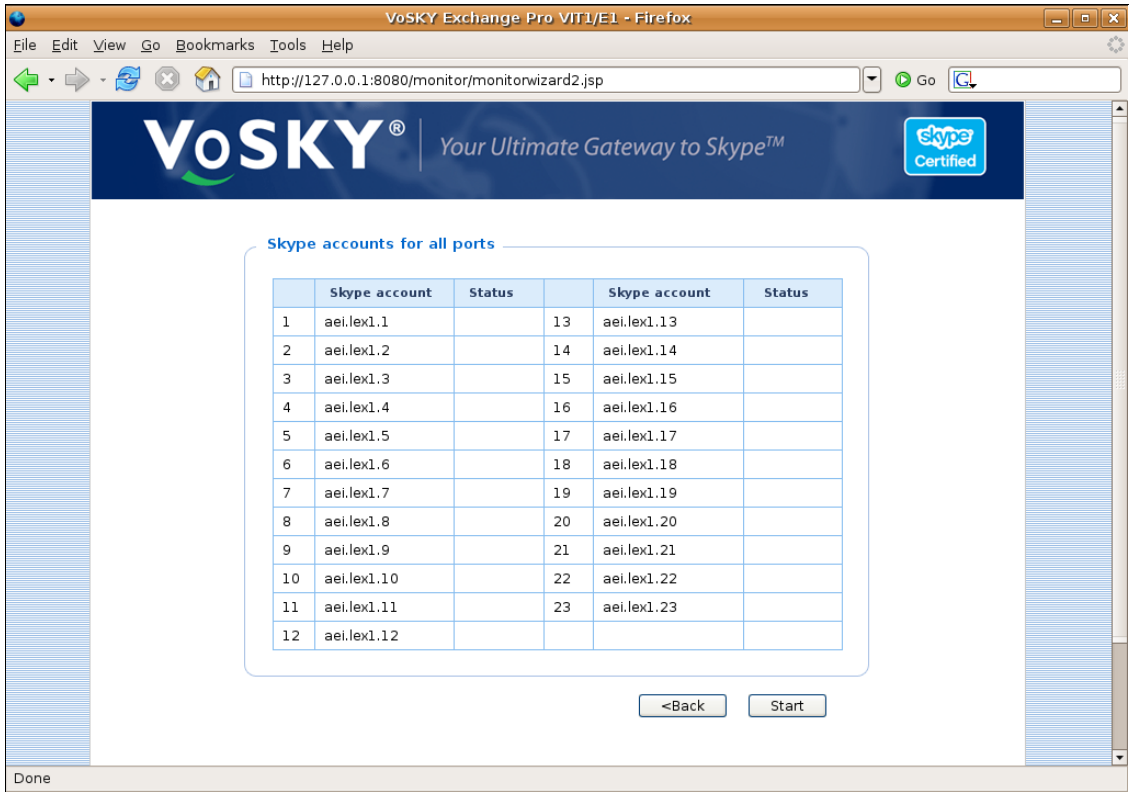
Step	Description
5.	<p>Dialing Scheme</p> <p>Define the dialing scheme for the speed-dial codes used to dial other Skype users. The speed-dial is comprised of a PIN (if required) plus the dialed string assigned to a particular user. Enter the digit length of the PIN in the first box and the digit length of the user code in the second box. For the compliance test, a 2-digit PIN and a 2-digit user code were defined. However, since a PIN is not required (as defined in Step 3) then all Skype users in the Exchange Pro phone book can be reached by dialing a 2-digit code.</p> 

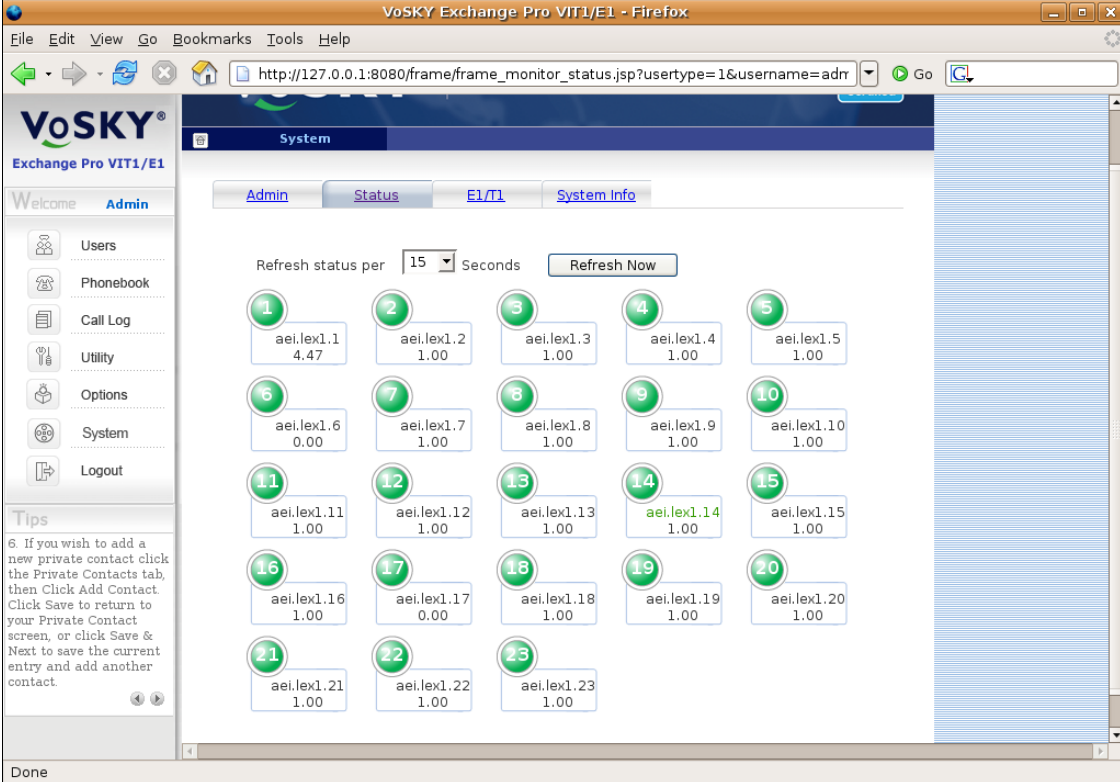
Step	Description
6.	<p>Admin Settings</p> <p>The Complete Admin Setting page that appears shows the default settings for the Exchange Pro administrator account. Enter any missing or update incorrect information in the required fields marked with a *. The cell phone number (Cell Phone No.) and Skype ID is the administrator contact information. Click Next to proceed.</p> 

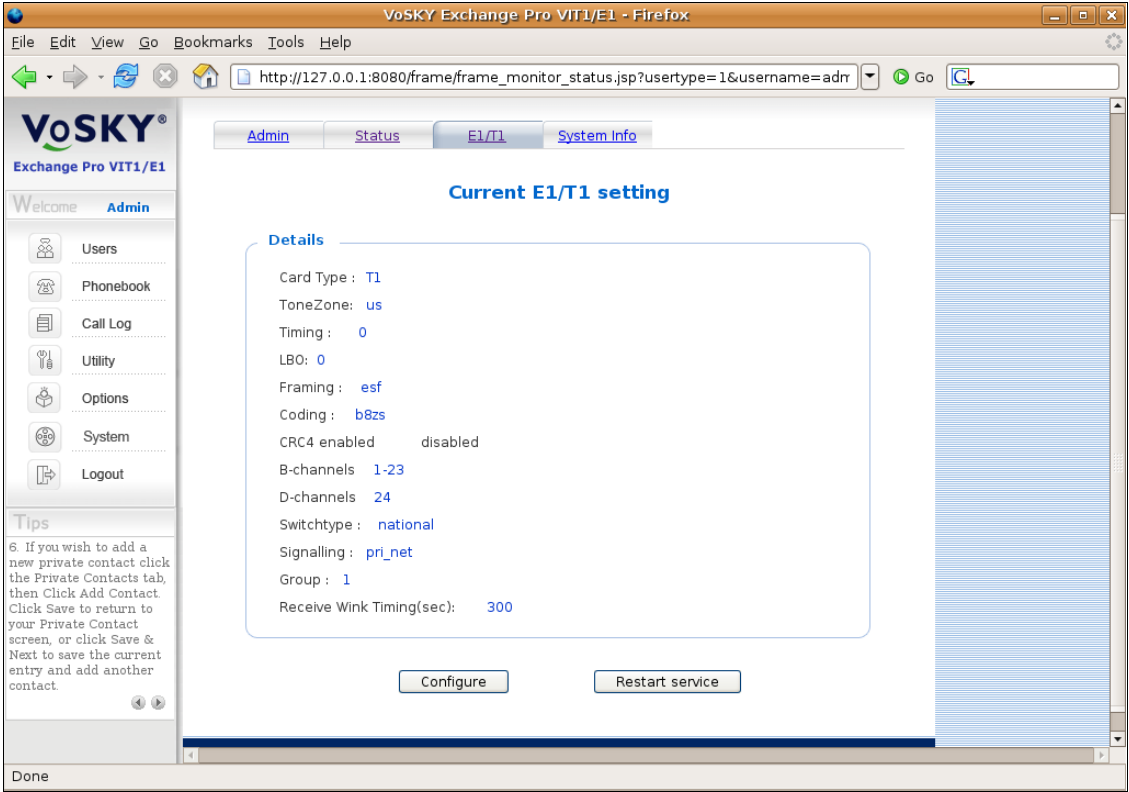
Step	Description
7.	<p>Wizard Setup Complete</p> <p>The Finished screen appears as shown below. Click OK to proceed to the Exchange Pro web administration.</p> 

Step	Description
8.	<p>Card Type On the Card Type Selection screen that appears, select T1 from the pull-down menu. Click Next to proceed.</p> 

Step	Description
9.	<p>Account Format</p> <p>Define the format of the Skype account names used by the Exchange Pro. Prior to Exchange Pro installation, the customer has been instructed to sign-up with the Skype service and obtain a set of accounts, one account for each license purchased with Exchange Pro. The customer selects the account names (Skype IDs) for these accounts so they may have a common format. If the accounts have a common format, it can be specified here. Otherwise, the account names can be imported from a list.</p> <p>One account is used for each active call involving the Exchange Pro. Thus, the number of accounts also represents the number of simultaneous calls supported by the Exchange Pro. In the case of the compliance test, each account started with the same prefix followed by a number. Thus, the Fixed Prefix + serial number option was selected. Enter the prefix in the Prefix field (<i>aei.lex.1</i>). Enter a password in the Password field. All accounts will share the same password. Click Next to proceed. Exchange Pro will start the serial number count at 1 and increment it for each available license. The compliance test used 23 licenses. Thus, Exchange Pro sets up accounts <i>aei.lex.1.1</i> through <i>aei.lex.1.23</i>.</p> 

Step	Description																																																																														
10.	<p>Account List</p> <p>A status screen appears, while the accounts defined in Step 9 are being set up. Once the set-up is complete, click the Start button to begin the process of logging in to each Skype account. The full administration GUI will open automatically to finish the configuration.</p>  <p>The screenshot shows a web browser window titled 'VoSKY Exchange Pro VIT1/E1 - Firefox'. The address bar shows 'http://127.0.0.1:8080/monitor/monitorwizard2.jsp'. The page header features the 'VoSKY' logo and the tagline 'Your Ultimate Gateway to Skype™', along with a 'skype Certified' badge. The main content area is titled 'Skype accounts for all ports' and contains a table with 24 rows of account information. Below the table are two buttons: '<Back' and 'Start'.</p> <table><tr><th></th><th>Skype account</th><th>Status</th><th></th><th>Skype account</th><th>Status</th></tr><tr><td>1</td><td>aei.lex1.1</td><td></td><td>13</td><td>aei.lex1.13</td><td></td></tr><tr><td>2</td><td>aei.lex1.2</td><td></td><td>14</td><td>aei.lex1.14</td><td></td></tr><tr><td>3</td><td>aei.lex1.3</td><td></td><td>15</td><td>aei.lex1.15</td><td></td></tr><tr><td>4</td><td>aei.lex1.4</td><td></td><td>16</td><td>aei.lex1.16</td><td></td></tr><tr><td>5</td><td>aei.lex1.5</td><td></td><td>17</td><td>aei.lex1.17</td><td></td></tr><tr><td>6</td><td>aei.lex1.6</td><td></td><td>18</td><td>aei.lex1.18</td><td></td></tr><tr><td>7</td><td>aei.lex1.7</td><td></td><td>19</td><td>aei.lex1.19</td><td></td></tr><tr><td>8</td><td>aei.lex1.8</td><td></td><td>20</td><td>aei.lex1.20</td><td></td></tr><tr><td>9</td><td>aei.lex1.9</td><td></td><td>21</td><td>aei.lex1.21</td><td></td></tr><tr><td>10</td><td>aei.lex1.10</td><td></td><td>22</td><td>aei.lex1.22</td><td></td></tr><tr><td>11</td><td>aei.lex1.11</td><td></td><td>23</td><td>aei.lex1.23</td><td></td></tr><tr><td>12</td><td>aei.lex1.12</td><td></td><td></td><td></td><td></td></tr></table>		Skype account	Status		Skype account	Status	1	aei.lex1.1		13	aei.lex1.13		2	aei.lex1.2		14	aei.lex1.14		3	aei.lex1.3		15	aei.lex1.15		4	aei.lex1.4		16	aei.lex1.16		5	aei.lex1.5		17	aei.lex1.17		6	aei.lex1.6		18	aei.lex1.18		7	aei.lex1.7		19	aei.lex1.19		8	aei.lex1.8		20	aei.lex1.20		9	aei.lex1.9		21	aei.lex1.21		10	aei.lex1.10		22	aei.lex1.22		11	aei.lex1.11		23	aei.lex1.23		12	aei.lex1.12				
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Step	Description
11.	<p>System Status</p> <p>The full administration GUI opens with the Status tab of the System page. It shows the status of each of the accounts previously created. The green light next to each account name indicates Exchange Pro was successfully able to login to the Skype service with that account.</p> 

Step	Description
12.	<p>T1 Settings</p> <p>To configure the T1 settings, click the System link in the left pane of the window. On the T1/E1 tab, the current settings are displayed. To change any setting, click the Configure button. The example below shows the settings used for the compliance test. These values must match the corresponding values used on Avaya Communication Manager (Section 4, Step 1).</p> <ul style="list-style-type: none"> ▪ Framing: <i>esf</i> ▪ Coding: <i>b8zs</i> ▪ B-channels: 1-23 ▪ D-channels: 24 ▪ Switchtype: national ▪ Signaling: pri_net 

6. General Test Approach and Test Results

The interoperability compliance testing consisted of placing calls through the Exchange Pro and exercising common PBX features. Calls were placed between the Avaya Communication Manager endpoints and the Skype users; as well as between the Avaya Communication Manager endpoints and the Skype-connected PSTN.

Exchange Pro passed compliance testing. The following features and functionality were verified. Any observations related to these tests are listed at the end of this section.

- Calls between an Avaya Communication Manager endpoint and a Skype user.
- Calls between an Avaya Communication Manager endpoint and a PSTN user via the Exchange Pro.
- Interoperability of the Exchange Pro with analog, digital, H.323, and SIP telephones.
- Interoperability of the Exchange Pro with Avaya one-X Desktop Edition (SIP soft client).
- Proper recognition of DTMF transmissions by navigating voicemail menus.
- Voicemail support
- PBX features including Hold, Transfer, Call Waiting, Call Forwarding and Conference.
- Proper system recovery after an Exchange Pro restart and loss of IP connection.

The following observations were made during the compliance test:

- If an enterprise user calls the PSTN and the called party is “busy”, the caller does not hear busy tone and the call is dropped. This was attributed to the operation of Skype and not related to an issue with interoperability between Exchange Pro and Avaya Communication Manager.
- Incoming Caller ID – When calling from an Internet Skype endpoint, the called party at the enterprise sees the caller’s name preceded by an unexpected character. When calling from the PSTN, the Exchange Pro sends 000000 as the calling number since the SkypeIN™ service does not support caller ID.

7. Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the ISDN-PRI signaling group is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the ISDN-PRI trunk group is in-service.
- Verify that calls can be placed between an Avaya Communication Manager endpoint and a Skype user.
- Verify that calls can be placed between an Avaya Communication Manager endpoint and a PSTN phone via the Exchange Pro.

8. Conclusion

These Application Notes describe the configuration required for VoSKY Exchange Pro VIT1/E1 to successfully interoperate with Avaya Communication Manager. VoSKY Exchange Pro VIT1/E1 successfully passed compliance testing.

9. Additional References

- [1] *Feature Description and Implementation For Avaya Communication Manager*, Doc # 555-245-205, Issue 6.0, January 2008.
- [2] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 4, January 2008.
- [3] *SIP support in Avaya Communication Manager Running on the Avaya S8xxx Servers*, Doc # 555-245-206, Issue 8, January 2008.
- [4] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005
- [5] *Installing, Administering, Maintaining, and Troubleshooting SIP Enablement Services*, Doc # 03-600768, Issue 6, June 2008.
- [6] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 11-300532, May 2005.
- [7] *4600 Series IP Telephone LAN Administrator Guide*, Doc # 555-233-507, July 2008.
- [8] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Installation and Maintenance Guide Release 2.0*, Doc # 16-601943, Issue 2, December 2007.
- [9] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide Release 2.0*, Doc # 16-601944, Issue 2, December 2007.
- [10] *Avaya one-X Desktop Edition Administration*, October 2006.
- [11] *Avaya one-X Desktop Edition Release 2.1 Quick Setup Guide*, Doc # 16-600974, Issue 2, October 2006.
- [12] *Avaya one-X Desktop Edition Getting Started*, Doc # 16-600973, Issue 2, September 2007.
- [13] *VoSKY Exchange Pro VIT1/E1 User Manual, Version 1.0*

Product documentation for Avaya products may be found at <http://support.avaya.com>.
Product documentation for Exchange Pro VIT1/E1 may be obtained from VoSKY.

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