

**Avaya Solution & Interoperability Test Lab**

# Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and the Cisco VG224 Voice Gateway - Issue 1.0

## Abstract

These Application Notes describe the configuration steps required to connect the Cisco VG224 Voice Gateway to a SIP infrastructure consisting of an Avaya SIP Enablement Services (SES) server and an Avaya S8300 Media Server with G700 Media Gateway running Avaya Communication Manager. The configuration steps described are also applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.


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# 1. Introduction

With the introduction of the SIP standard that supports telephony as well as a wide range of other communication modes, there is a much broader range of SIP telephones and gateways available to customers. There will be sales opportunities involving customers who have a significant investment in analog telephones and wish to purchase the Avaya SIP offer, but already own analog gateways other than those offered by Avaya. Customers may be interested in replacing their existing telephony infrastructure (e.g., Cisco Call Manager) with Avaya servers, but wish to re-use the existing non-Avaya gateways.

These Application Notes describe the configuration steps for using the Cisco VG224 Voice Gateway with Avaya SIP Enablement Services (SES), S8300 Media Server, and G700 Media Gateway. Only the configuration steps pertinent to interoperability of Cisco and Avaya equipment are covered. General administration information can be found in the product documentation as well as the specific references listed in Section 10. The configuration described should be applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

## 1.1. Configuration

The configuration used as an example in these Application Notes is shown in **Figure 1**. The diagram indicates logical signaling connections. With the exception of the Avaya 6408D Digital Telephone, the analog telephones, and the fax machines, all components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch and are administered on a single subnet. The Avaya IA770 INTUITY™ AUDIX® Messaging Application resides on the S8300 Media Server and is used to support voice messaging. The VG224 supports 24 Foreign Exchange Station (FXS) analog ports. An analog telephone and a fax machine are connected to two of the ports.

The configuration steps described in the following sections cover the routing of calls by Avaya SES, Avaya Communication Manager, and the VG224. The VG224 is configured as a peer gateway that routes calls between its analog and SIP interfaces, as opposed to registering with Avaya SIP Enablement Services as a SIP endpoint on behalf of each analog telephone. This type of configuration is also known as "SIP trunking." **Table 1** profiles the network management capabilities of the VG224.

| | |
|---|---|
| **Administration mechanisms** | Command line interface |
| **Administration levels** | Craft, Administrator |
| **File transfer server** | TFTP |
| **Error logs** | Standard logs supported by Cisco IOS |
| **SNMP support** | Yes |

**Table 1: Network Management Capabilities of Cisco VG224 Voice Gateway**


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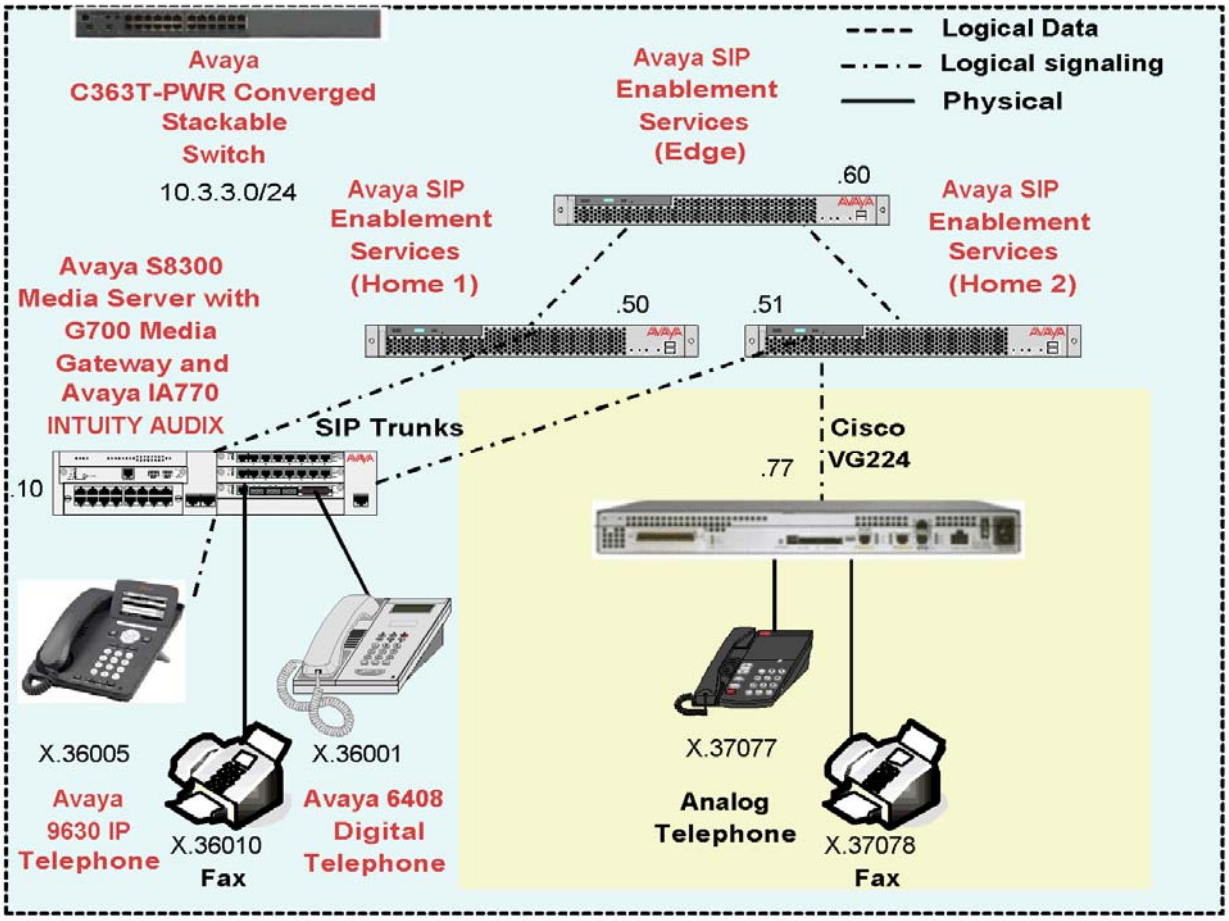



**Figure 1: Avaya SIP Test Configuration with Cisco VG224 Voice Gateway**

## 2. Equipment and Software Validated

The following equipment and software were used in the configuration.

| Equipment | Software |
|---|---|
| Avaya SIP Enablement Services (SES) Server | 3.1.1 |
| Avaya C363T-PWR Converged Stackable Switch | 4.3.10 |
| Avaya S8300 Media Server with G700 Media Gateway | Avaya Communication Manager 3.1.2, Patch 12866 |
| Avaya IA770 INTUITY®AUDIX™ | N3.1-26.0 |
| Avaya 6408 Digital Telephone | - |
| Avaya 9630 IP Telephone | S1.2.1 |
| Cisco VG224 Voice Gateway | 12.4 (11)T |

**Table 2: Equipment and Software Versions Used**


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# 3. Supported Features

**Table 3** gives a summary of the features available on the Cisco VG224 when configured to use SIP trunking with Avaya SES. Note that these features are a subset of those available to SIP endpoints that register with Avaya SES and supported by Avaya Communication Manager via Off-PBX Station (OPS) mapping.

| Feature | Comments |
|---|---|
| Extension-to-extension call | G.711, G.729AB |
| Basic calls to legacy phones | Digital, H.323, Analog |
| Intercept tones/displays | Tones only |
| Call Waiting | |
| Call Hold | |
| Fax (G.711 voice mode) | |
| Fax (T.38) | |
| Caller ID | |

**Table 3: SIP Telephony Feature Support**

# 4. Call Routing Requirements

To support extension to extension calls as well as off-premises dialing, the sample configuration implements the following call routing:

1. Calls from Avaya-supported[1] telephones to VG224-supported analog telephones must be routed by Avaya Communication Manager through the SIP trunk to Avaya SES, which must in turn route the calls to the VG224.
2. Calls from VG224-supported telephones to Avaya-supported telephones must be routed by the VG224 to Avaya SES.[2]

The following sections address these requirements.

[1] The term *supported* refers to telephones that are either directly connected to or are logically registered with the SIP telephony product (e.g., Avaya G700 Media gateway, Avaya SES, Cisco VG224 Voice Gateway).
[2] This technique can also be used to support calls to telephones connected to another VG224 or to an off site telephone via a SIP service provider. Such calls can be routed to Avaya SES, where host address maps can be used to route to the destination. This serves to centralize the routing rules of all Cisco VG224 Voice Gateways in one place. This type of configuration was not tested and is beyond the scope of these Application Notes.


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# 5. Configure Avaya Communication Manager

This section focuses on the commands required to route calls from Avaya-supported telephones to analog telephones connected to the VG224. For complete documentation on SIP administration, see References [2-3]. Use the System Access Terminal (SAT) interface to perform the following steps. Log in with the appropriate credentials.

## 5.1. Verify SIP Trunk Capacity

Verify that an adequate number of SIP trunk members are administered for the system using the **display system-parameters customer-options** command. Note that one trunk group member is consumed for each SIP endpoint on a call, so that the maximum number of simultaneous SIP-SIP calls supported is equal to half the total number of members (two members per call), whereas SIP-H.323 or SIP-digital calls only consume one trunk member per call. When sizing the number of SIP trunks required to support the VG224, keep in mind that calls between analog telephones on different VG224 gateways will require 2 members (SIP-SIP). Calls within the same gateway don't require any members, since the VG224 handles the call internally.

```
display system-parameters customer-options                    Page   2 of  10
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 5     0
           Maximum Concurrently Registered IP Stations: 5     1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 5     0
                 Maximum Video Capable H.323 Stations: 0     0
                  Maximum Video Capable IP Softphones: 5     0
                      Maximum Administered SIP Trunks: 100   0

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 0     0
                    Maximum G250/G350/G700 VAL Sources: 5     1
          Maximum TN2602 Boards with 80 VoIP Channels: 0     0
         Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

## 5.2. Define Dial Plan

Avaya Communication Manager applies dial plan and routing rules to determine where a call should be routed. In the sample configuration, the Uniform Dialing Plan (UDP) and Automatic Alternate Routing (AAR) features will be configured to route calls to VG224-supported telephones via the SIP trunk to Avaya SES. The extension numbers for these telephones will be organized to make these types of routing rules easy to specify. For the sample configuration, these analog telephones will have 5-digit extensions beginning with 37.


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Use the **display system-parameters customer-options** form to verify that **Uniform Dialing Plan** is set to "y". If it must be enabled, remember to log off and log into the SAT session for this option to take effect.

```
display system-parameters customer-options                    Page   5 of  10
                              OPTIONAL FEATURES

                Multinational Locations? n           Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? y     Station as Virtual Extension? n
                     Multiple Locations? n
                                              System Management Data Transfer? n
          Personal Station Access (PSA)? n             Tenant Partitioning? n
                       Posted Messages? n      Terminal Trans. Init. (TTI)? n
                       PNC Duplication? n             Time of Day Routing? n
                  Port Network Support? n           Uniform Dialing Plan? y
                                             Usage Allocation Enhancements? y
               Processor and System MSP? y      TN2501 VAL Maximum Capacity? y
                     Private Networking? y
                     Processor Ethernet? y              Wideband Switching? n
                                                               Wireless? n
                         Remote Office? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y


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

Use the **display dialplan analysis** command to make sure that the desired extension range is covered.

```
change dialplan analysis                                      Page   1 of  12
                           DIAL PLAN ANALYSIS TABLE
                                                        Percent Full:    0

     Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
     String  Length Type      String  Length Type      String  Length Type
       0       3    fac
       1       3    fac
       3       5    ext
       4       5    ext
       5       5    ext
       7       5    ext
       9       1    fac
```


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Use the **change uniform-dial-plan** command to specify that extensions beginning with 37 will use AAR.

```
change uniform-dialplan 3                                    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
  37        5   0         aar  n                                    n
```

## 5.3. Define AAR Routing

Use the **change aar analysis** command to configure route selection. The entry below defines the number format to be matched and the route pattern that will be selected. For 5-digit numbers beginning with 37, route pattern 12 is used. This route pattern (not shown) specifies the trunk group that corresponds to the standard SIP trunk for SES Home 2 (see **Figure 1**). Steps to configure a SIP trunk can be found in Reference [2].

```
change aar analysis 3                                        Page   1 of   2
                           AAR DIGIT ANALYSIS TABLE
                                                        Percent Full:    0

         Dialed           Total     Route    Call   Node  ANI
         String          Min  Max  Pattern   Type   Num   Reqd
    37                   5    5     12       aar          n
```

## 5.4. Define Codec Set

Define the audio codec set to be used. In the sample configuration, local calls were tested with both G.711 mu-law and G.729AB. Use the **change ip-codec-set** command to define these two codec sets (Page 1).

```
change 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         2        20
 2: G.729AB          n         2        20
 3:
```


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On Page 2, specify the fax mode. Set **FAX Mode** to "t.38standard" or "off", depending on whether the VG224 is configured for T.38 or G.711 voice mode, respectively. See Reference [7] for more information on selecting fax modes. The value shown below corresponds to the VG224 configuration shown in Section 7.1. Default values can be used for the remaining fields.

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

                         IP Codec Set

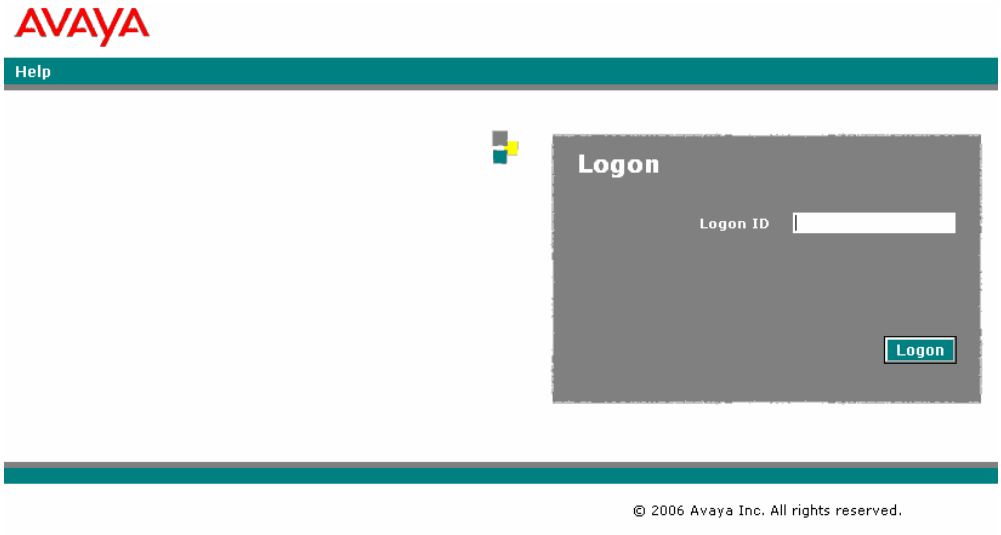
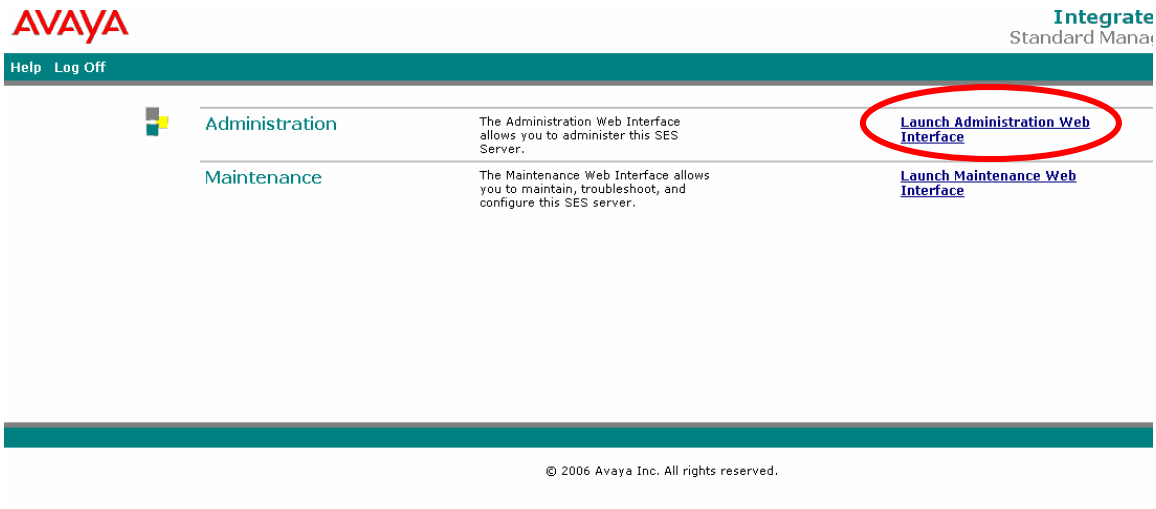
                             Allow Direct-IP Multimedia? n


                   Mode                Redundancy
    FAX            t.38-standard        0
    Modem          off                  0
    TDD/TTY        US                   3
    Clear-channel  n                    0
```


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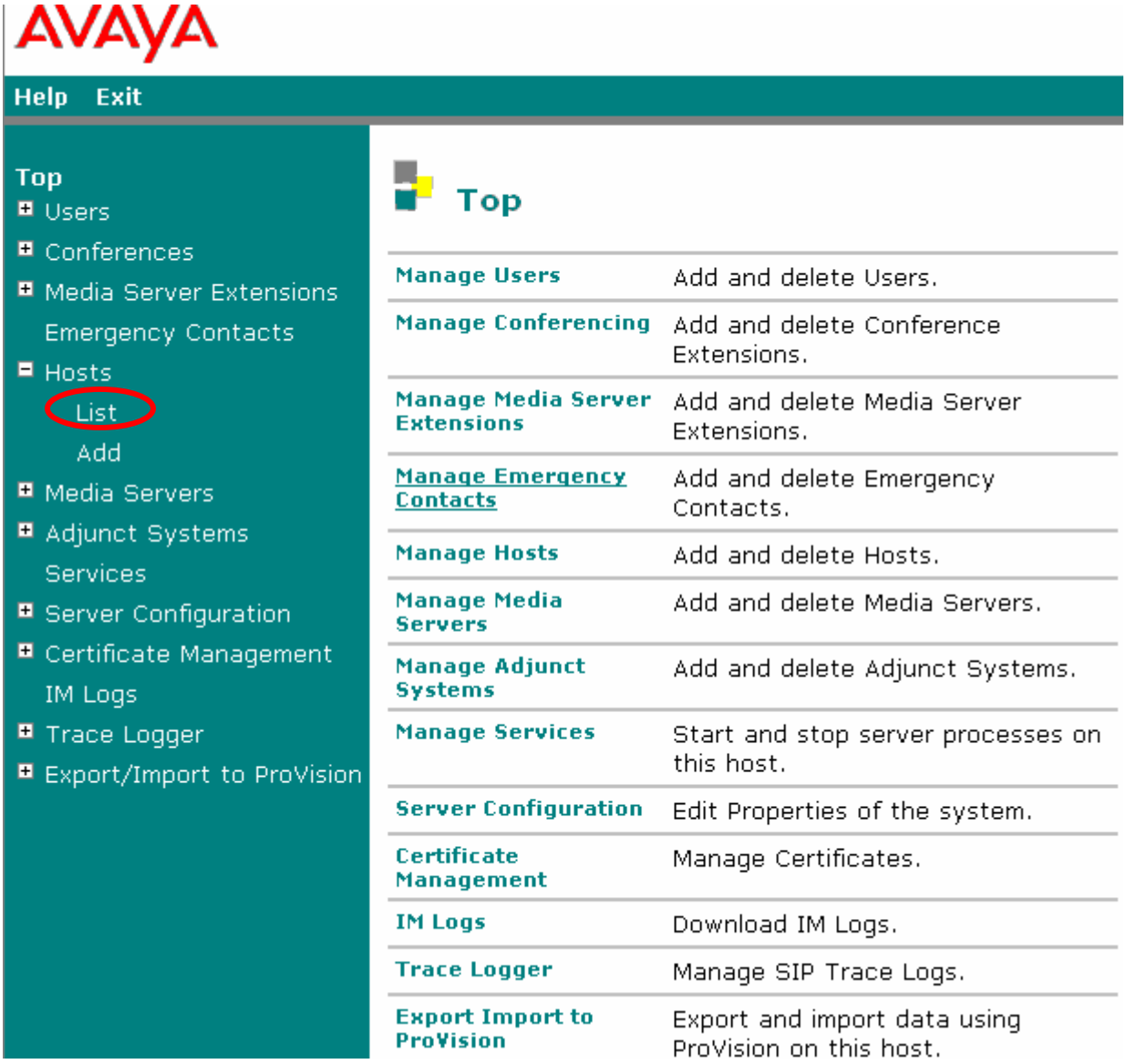
# 6. Administer Avaya SIP Enablement Services

Avaya SES must be configured to route calls to the VG224. This will be accomplished by using a *host address map*, which specifies one or more dial strings to be matched and the SIP destination to which the call is to be routed if a match is found. Other standard administration functions are covered in Reference [1].

| Steps | Description |
|---|---|
| 1. | Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where *IP-address* is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server. On the *Logon* screen, enter the appropriate **Login ID** and click on **Logon**. When prompted, enter the password.<br> |
| 2. | The main administration screen will be displayed after logging in. Click on **Launch Administration Web Interface.**<br> |


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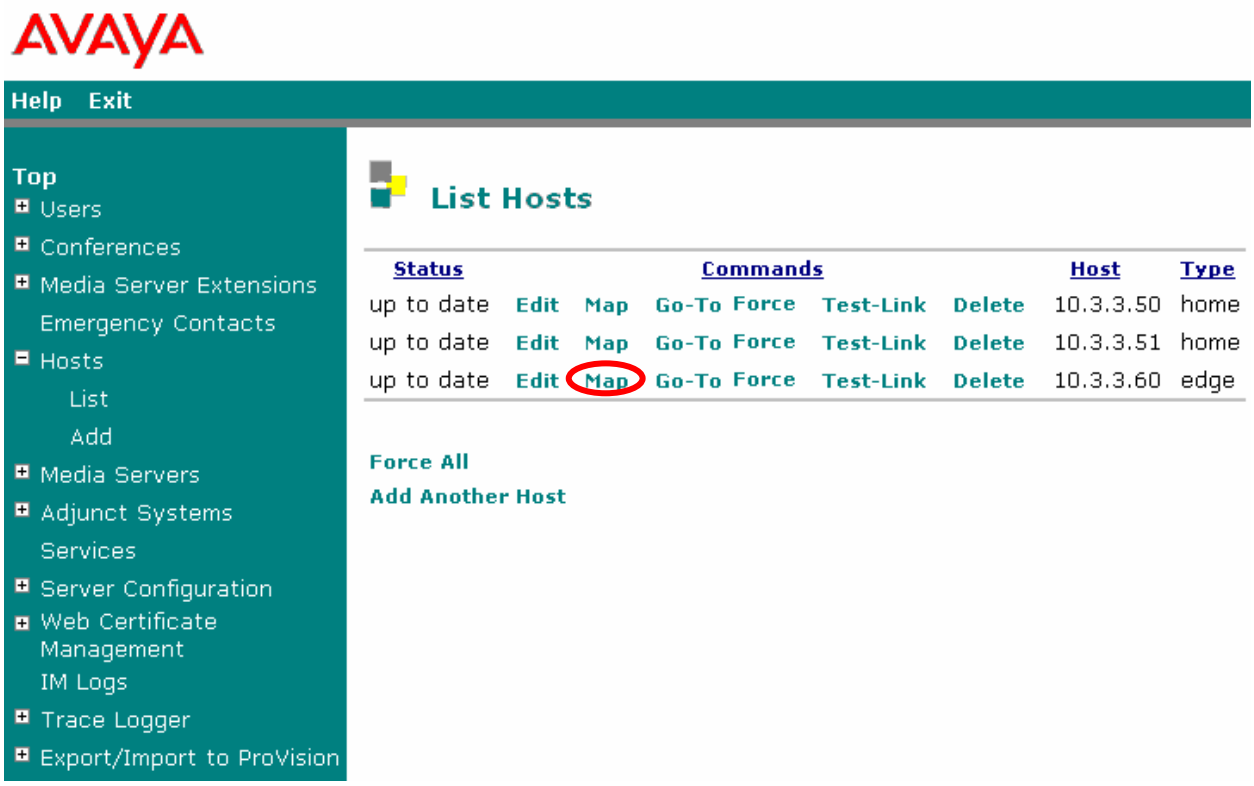
| Steps | Description |
|---|---|
| 3. | Expand the **Hosts** link on the left side of the main administration page, and then click on **List**. |





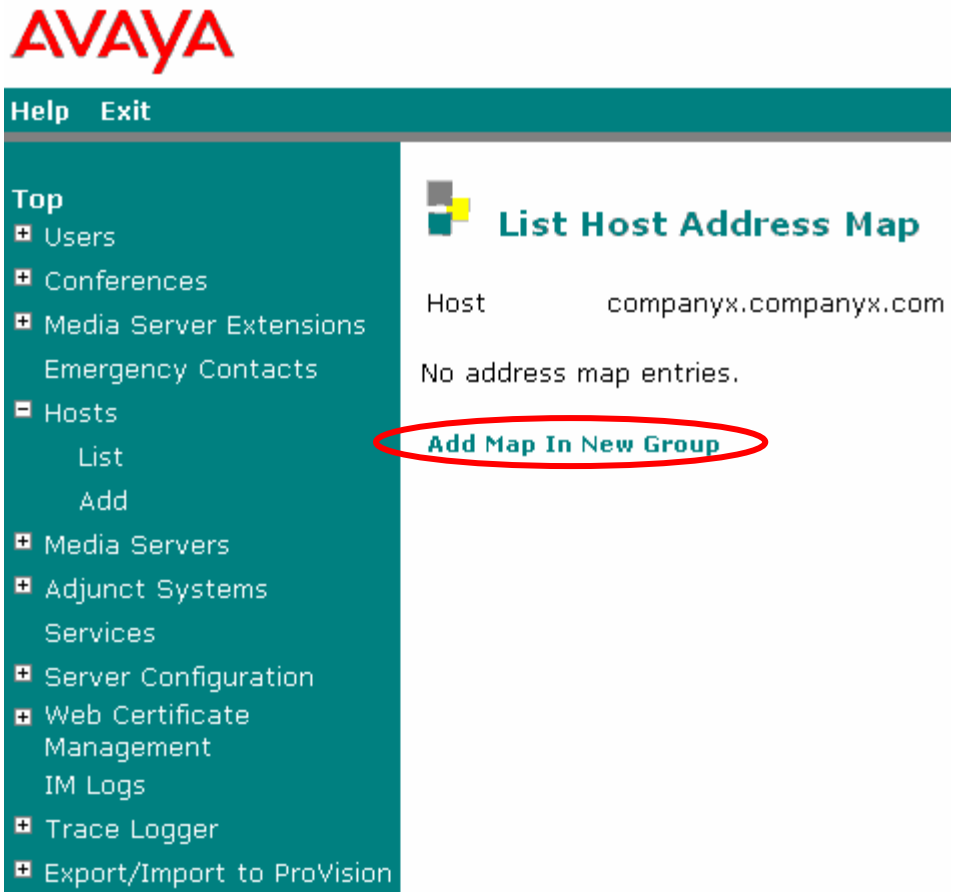
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| Steps | Description |
|---|---|
| 4. | The existing home and edge servers will be displayed. The address map will be defined once for the edge server, and will be active in all home servers when they route calls. In the row corresponding to the edge server, click on **Map** under *Commands*. |



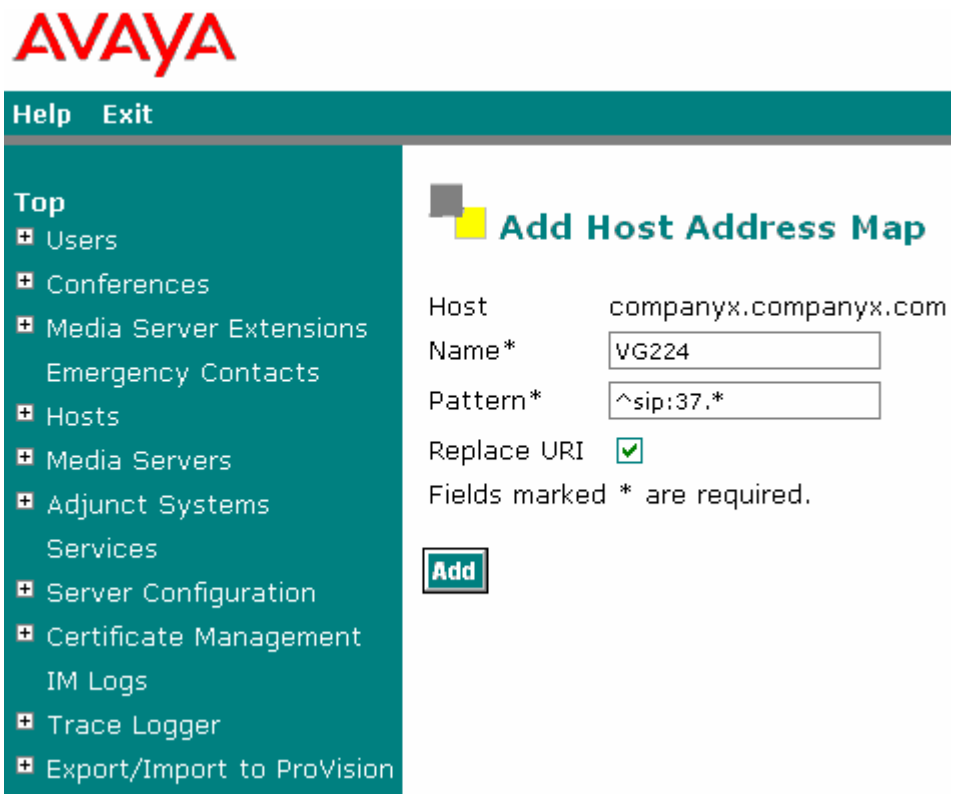


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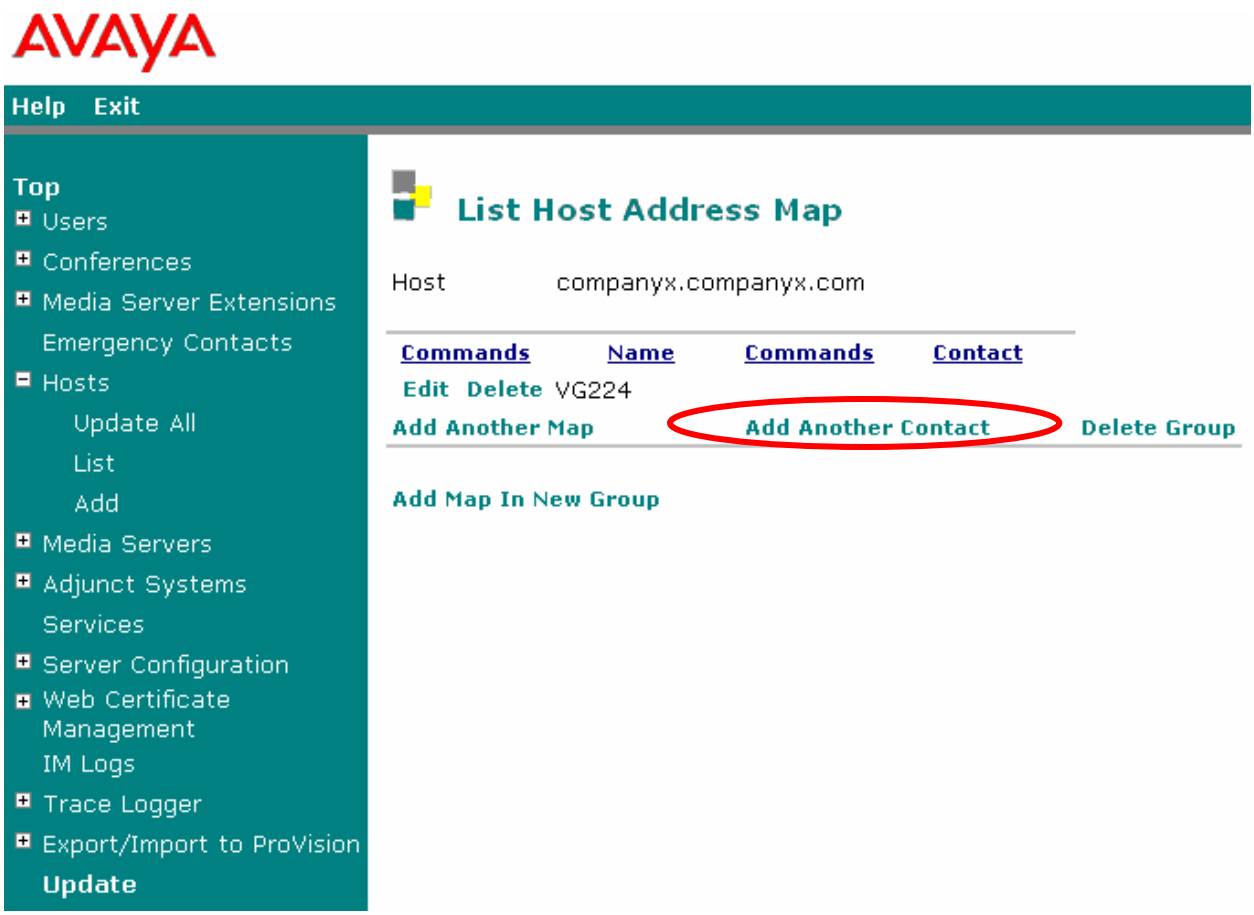
| Steps | Description |
|---|---|
| 5. | The *List Host Address Map* page is displayed. Click on **Add Map in New Group**.  |




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| Steps | Description |
|---|---|
| 6. | The *Add Host Address Map* page will be displayed. Specify a **Name** for the address map, and the **Pattern** match specification. In the sample configuration, 5-digit extensions beginning with 37 are to be routed to the Cisco VG224 Voice Gateway. The pattern match specification is applied to the Uniform Resource Identifier (URI) field of incoming INVITE messages. The URI usually takes the form sip:user@domain, where *domain* can be a domain name or an IP address. In the sample configuration, *user* is actually the telephone number of the phone. An example of a URI sent by a local SIP telephone would be sip:37077@companyx.com.<br><br>The specification below means "match on the characters 'sip:37' if they occur at the beginning of the URI, followed by any number of characters." This will cover the extension range within which VG224-supported analog telephones are assigned. See Appendix A for a more detailed description of the syntax for address map patterns.<br><br>Check **Replace URI** (default). When routing the incoming INVITE, Avaya SIP Enablement Services will replace the URI with the URI specified in the contact (see Steps 7-8).<br><br>Click on **Add**; then click on **Continue** on the confirmation page.<br><br> |


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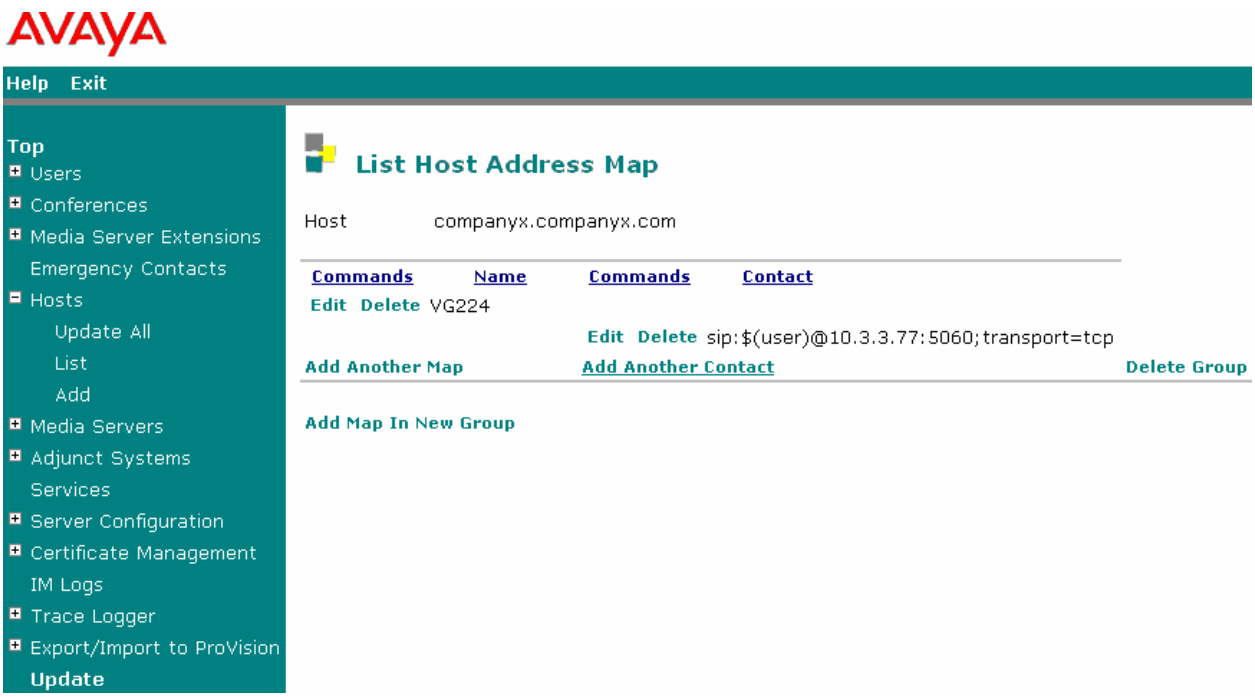
| Steps | Description |
|---|---|
| 7. | The current host address map list is displayed. To specify the routing for the address map just defined, click on **Add Another Contact**.  |




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| Steps | Description |
|---|---|
| 8. | Fill in the **Contact** field as shown. In this case, the IP address and port number used by the VG224 are specified, as well as the transport protocol used to send SIP signaling messages. Avaya SIP Enablement Services will substitute "$(user)" with the user field (i.e., the dialed number) in the incoming INVITE message.<br><br>Click on **Add**; then click on **Continue** on the confirmation page.<br><br> |


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| Steps | Description |
|---|---|
| 9. | The *List Host Address Map* page is then displayed with the updated contact information. This completes configuration of the host address map for routing calls to the VG224.<br> |
| 10. | All VG224 gateways to which calls can be routed should be administered as *trusted hosts* in each of the Avaya SIP Enablement Services home servers. This will permit far-end call clearing and other supplementary call service requests by the gateways to be handled without authentication challenges. This is performed at the Linux CLI of the edge server (or home/edge if only one server is used). Use the standard methods to access the command line interface of the edge server, and log in with the appropriate credentials. The format of the **trustedhost** command is:<br><br>`trustedhost –a host –n server –c comment`<br><br>For the sample configuration, the following **trustedhost** commands are executed:<br><br>`trustedhost –a 10.3.3.77 –n 10.3.3.50 –c Cisco VG224 (Home 1)`<br>`trustedhost –a 10.3.3.77 –n 10.3.3.51 –c Cisco VG224 (Home 2)` |
| 11. | To apply the administration in the above steps, *including Step 10*, click on **Update** on the left side of the page on the web browser interface. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed to that point. |


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# 7. Configure Cisco VG224 Voice Gateway

The following sections provide annotated configuration steps for supporting analog telephones and voice/fax call routing in the Cisco VG224 Voice Gateway. They are not a complete set of configuration steps for the product; only configuration commands relative to the sample configuration are shown.

## 7.1. Define VoIP Interfaces

The following commands define the Ethernet interfaces and associated VoIP services. Note that the codec and fax configuration commands are global and apply to all analog ports. Access the VG224 via telnet or the console interface.

| | |
|---|---|
| `voice service voip` | *VoIP service definition* |
| ` fax protocol t38 ls-redundancy 0 hs-redundancy 0` | *T.38 Fax (replace with* |
| ` !` | *"*`fax protocol pass-through g711ulaw`*" for voice mode)* |
| ` !` | |
| ` !` | |
| ` sip` | *SIP support* |
| `  bind control source-interface FastEthernet0/0` | *Interface for signaling* |
| `  bind media source-interface FastEthernet0/0` | *Interface for media* |
| `  session transport tcp` | *Signaling transport* |
| `!` | |
| `voice class codec 1` | *Global Codec set in* |
| ` codec preference 1 g711ulaw` | *priority order* |
| ` codec preference 2 g729br8` | See Section 5.4 - |
| `!` | corresponds to G.729AB |
| `!` | in ip-codec-set |
| `interface FastEthernet0/0` | |
| ` ip address 10.3.3.77 255.255.255.0` | *Network parameters for the* |
| ` duplex full` | *IP interface* |
| ` speed auto` | |
| `!` | |
| `ip route 0.0.0.0 0.0.0.0 10.3.3.1` | |
| `!` | *Default route (i.e.,* |
| `!` | *gateway)* |
| `sip-ua` | |
| ` sip-server ipv4:10.3.3.51` | *Global SIP parameters for* |
| ` offer call-hold conn-addr` | *peer* |
| `!` | *IP address of SES Home 2.* |
| `!` | *Signal HOLD via setting* |
| `!` | *Connection IP to 0.0.0.0* |
| `!` | *in Session Description* |
| `!` | *Protocol (SDP) portion of* |
| | *SIP INVITE message* |


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## 7.2. Define Analog Telephone Interfaces

The following commands define the analog interfaces and associated Plain Old Telephone Service (POTS) parameters.

```
voice service pots
!
application
 service dsapp                          Define Call features
  param callWaiting TRUE                Define as "dsapp" service
!                                       Include Call Waiting
 global
  service default dsapp                 Apply above services by default to
 !                                      all POTS interfaces
voice-port 2/0
 timeouts call-disconnect 5             Port for analog phone (Port 0)
 station-id name VG224 P1               Timeout for disconnect tone
 station-id number 37077                Set name ("display name" in SIP)
 caller-id enable                       Set Extension  #
!                                       Caller-ID signaling to phone
voice-port 2/1
 timeouts call-disconnect 5             Port for fax machine (Port 1)
 station-id name VG224 P2
 station-id number 37078                (similar to Port 0)
 caller-id enable
```

## 7.3. Define Call Routing

The following commands specify call routing to the analog ports and to Avaya SIP Enablement Services via the VoIP interface. Note that the VG224 will match a dialed number to the **destination-pattern** specified in the dial peers to determine which dial peer to use for routing the call. In the sample configuration, the extensions for the individual analog ports match the VoIP dial peer (30000) as well as the POTS dial peers (37077, 37038), but dial peer hunting will prefer the longest match without the use of wildcard characters. Therefore, incoming calls to an analog extension will route to the corresponding POTS rather than the VoIP dial peer.

```
dial-peer voice 30000 voip              VoIP dial peer
 destination-pattern 3....              5-digit extensions, 1st digit is 3
 voice-class codec 1                    Codec list defined in Section 7.1
 session protocol sipv2                 SIP signaling
 session target sip-server              Route to Avaya SIP Enablement
 !                                      Services (see Section 7.1)
 dtmf-relay rtp-nte                     Use RFC 2833 for signaling DTMF
!
dial-peer voice 37077 pots              Analog phone dial peer
 destination-pattern 37077              Extension #
 port 2/0                               Physical port
!
dial-peer voice 37078 pots              Fax dial peer
 destination-pattern 37078              Extension #
 port 2/1                               Physical port
```


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# 8. Verification Steps

All features shown in **Table 3** were tested using the sample configuration. The following steps can be used to verify and/or troubleshoot installations in the field.

1. Place a call between two analog telephones on the same VG224 gateway. If the call fails, check the analog port and POTS dial peer configurations.
2. Place a call from a VG224-supported analog telephone to an Avaya-supported telephone. If the call fails:
    a. Verify that the SIP INVITE message is being routed by the VG224 to Avaya SES. If not, check the VoIP dial peer on the VG224.
    b. If the INVITE is being routed properly from the VG224, make sure that the telephone being dialed has the proper administration in Avaya SES.
3. Place calls from an Avaya-supported telephone to a VG224-supported telephone. If calls fail, use the Avaya Communication Manager "list trace" command on the SIP trunk and verify that the call is being routed through that trunk. If not, check the UDP and AAR configuration.
4. Connect fax machines to an analog port on the VG224 and to analog ports on Avaya Media Gateways, as applicable. Confirm successful calls and accurate fax transmissions. If they fail, use the previous verification steps to verify that the calls are properly routed. Note that G.711 voice mode fax may not work in networks experiencing excessive packet loss and jitter. Make sure that if compressed codecs such as G.729 are in use, T.38 is configured in both the VG224 and Avaya Communication Manager.

# 9. Support

For technical support of Cisco products:

Internet: http://www.cisco.com/en/US/support/index.html
Email: tac@cisco.com
Telephone: 1-800-553-2447

# 10. Additional References


[1] *Installing and Administering SIP Enablement Services R3.1*, Issue 1.4, Doc ID 03-600768, February, 2006, available at http://support.avaya.com.

[2] *SIP Support in Release 3.1 of Communication Manager Running on the S8300, S8400, S8500, S8500B, S8700, and S8710 Media Server*, Issue 1.4, Doc ID 555-245-206, February, 2006, available at http://support.avaya.com.

[3] *Avaya Extension to Cellular and OPS Installation and Administration Guide*, Version 6.0 Issue 9, DocID 210-100-500, June 2005, available at http://support.avaya.com.

[4] *Administrator Guide for Avaya Communication Manager*, Issue 2, Doc ID 03-300509, February 2006, available at http://support.avaya.com.

[5] *Cisco VG224 Voice Gateway Software Configuration Guide*, available at http://www.cisco.com

[6] *Cisco IOS Fax Services over IP Roadmap*, available at http://www.cisco.com.

[7] *Avaya FoIP, MoIP, and TTYoIP*, Issue 2, available at: http://www.avaya.com/master-usa/en-us/resource/assets/whitepapers/ef-lb2626.pdf.




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# Appendix A

## Address Map Pattern Strings in Avaya SIP Enablement Services

The syntax for the pattern is a Linux regular expression that will match the extensions to be mapped. Regular expressions are a textual way to describe pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s). For example, **[0-9]** represents any single digit and ***** represents any number of digits or characters. For example, ^**sip:538[0-9]*** would match any SIP INVITE message (^ matches the beginning of a line) for any extension 3 or more digits in length, beginning with the digits **538**, and ending with any other sequence of digits. Square brackets contain a selection of characters to be matched, with a hyphen indicating a range. In the example, **[0-9]** matches any digit, or as another example, **[13579]** matches odd-numbered digits. Braces containing a whole number match that number of instances of the preceding item. For example, **[0-9]{4}** matches any four digits. Note that the braces may require preceding \ escape characters: **\{4\}**. Another helpful *metacharacter* is dot (period), which matches any single character. For example, the regular expression **.*** matches any quantity (including zero) of any character(s).


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