

VOICEGENIE

VoiceGenie 7 SIP Proxy Users' Guide

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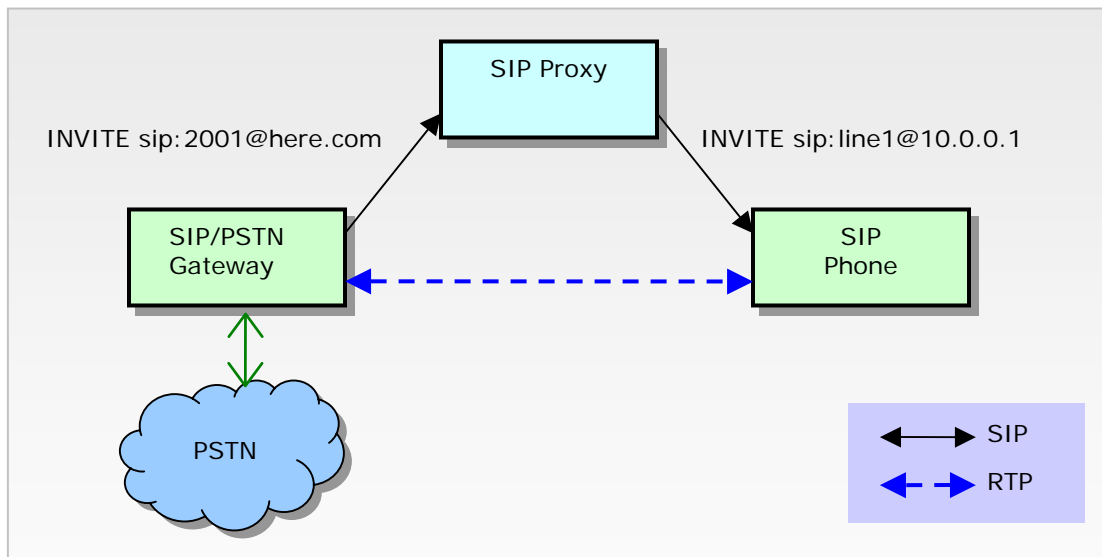
Revision History

Version	Date	Change Summary	Author/Editor
1.0	January 25 th , 2005	Initial release	Henry Lum
1.1	March 4 th , 2005	Updated for VoiceGenie 7 release	Andrew Ho
1.2	April 13 th , 2005	Revised Version	Andrew Ho

1 Introduction

1.1 Overview

SIP proxies provide a variety of services in VOIP networks that are based on the use of SIP, such as authorization and access control, validation and security, call routing, accounting, user location, and others. A variety of SIP proxies are available, some of which are commercial (e.g. Cisco) and some of which are open source (e.g. Vovida, iptel.org). Fundamentally, any SIP proxy delivers services by controlling how requests and responses are routed between a SIP client (UAC) and a SIP server (UAS), generally by deciding on how requests get routed, as well as by manipulating headers in the request or response. A simple example of this is shown in the figure below:



VoiceGenie's SIP proxy provides a very specific set of services that are not available from general-purpose SIP proxies that are widely available. These services are specific to providing media-centric SIP services, such as VoiceXML dialogs, and conferencing capabilities. VoiceGenie's SIP Proxy is also designed for scalability and has redundant architecture to protect against server failures.

1.2 Features and Capabilities

The VoiceGenie SIP proxy focuses on the following capabilities:

SIP Service mapping capabilities, where incoming calls are mapped to particular services with support for transformations on service parameters. For instance, an INVITE request processed by the proxy with the Request-URI set to `sip:dialog.vxml.http%3A//a/test.vxml@test.com` could be remapped to `sip:service1234@box1.server.com;voicexml=http://a/test.vxml`.

Active monitoring of SIP resources, ensuring that requests for the service will be routed around any unavailable SIP resources.

Load balancing of requests/calls to a particular SIP service across the SIP resources that provide that service, such as distributing a set of incoming calls to a collection of VoiceGenie media platforms.

High availability, in which multiple clustered VoiceGenie SIP proxies handle the failure of individual proxies within the cluster, maintaining availability of SIP services managed by the cluster.

Conference management, in which conferences are allocated to available conferencing resources, with all calls to access a conference being routed to the appropriate resource. This alleviates conference users from having to be aware of which conferences exist on what resources.

High availability is very closely related to the way that the proxy is deployed and is discussed below under the Deployment Architecture section. The other capabilities listed above are described in greater detail in the following sections.

1.3 Terminology

The follow table list essential terminologies used throughout this document.

Terminology	Description
SIP Service	<p>A proxy manages access to SIP Services via SIP requests, which are delivered using one or more SIP Resources managed by the proxy.</p> <p>Examples: VoiceXML Attendant Service Conference Server CCXML Interpreter Media Server SIP/PSTN gateway</p>
SIP Resource	<p>An instance of server that provides one or more SIP Services. A SIP Resource is addressed by a specific hostname and port of the server that accepts SIP requests. It may have a number of other properties such as an assigned SIP resource type.</p> <p>Example: MediaPlatform1 { Host = mediaplatform.voicegenie.com:5060 ip address = 10.0.0.123 SIP Resource Type = VG-MediaPlatform-7 }</p>
SIP Resource Type	<p>A collection of properties that describe specific types of resources, and provide the information necessary for the SIP proxy to use that resource. For instance, a "VG-MediaPlafrom-7" resource type could be defined as one that supports 300 ports of VoiceXML or Conferencing sessions.</p> <p>All SIP resources have a single SIP Resource Types.</p> <p>Example Properties for defining a SIP Resource Type: Registration Method Monitoring Method Capability</p>

	Number of ports Maximum conference size
Capability	Defines the type of capability that a SIP Resource can provide. A SIP Resource can provide one or more capabilities. Examples: Conferencing VoiceXML CCXML
Registration Method	Each SIP Resource must be registered with the SIP Proxy in order to properly forward requests to the resource. A resource can register to the SIP via SIP REGISTER method, or if the component is part of VoiceGenie's OA&M framework, the component can register through the CMP interface. Examples: SIP REGISTER CMP no registration methods (assumed registered)
Monitoring Method	SIP Proxy is required to monitor the availability of the resources in order to forward requests to active resources. SIP Proxy can periodically send SIP OPTIONS message to a resource to check the livelihood of the server, or use VoiceGenie's OA&M framework to monitor the resource. SIP Proxy only forwards to resources that are active. Examples: SIP OPTIONS CMP no monitoring methods (assumed active)
SIP Transport	SIP Proxy uses different types of transport to receive and deliver the SIP messages. Each transport protocol has different properties and provides different advantages. Examples: UDP (unicast) UDP (multicast) TCP SCTP (not supported in this release)

2 Deployment Architecture

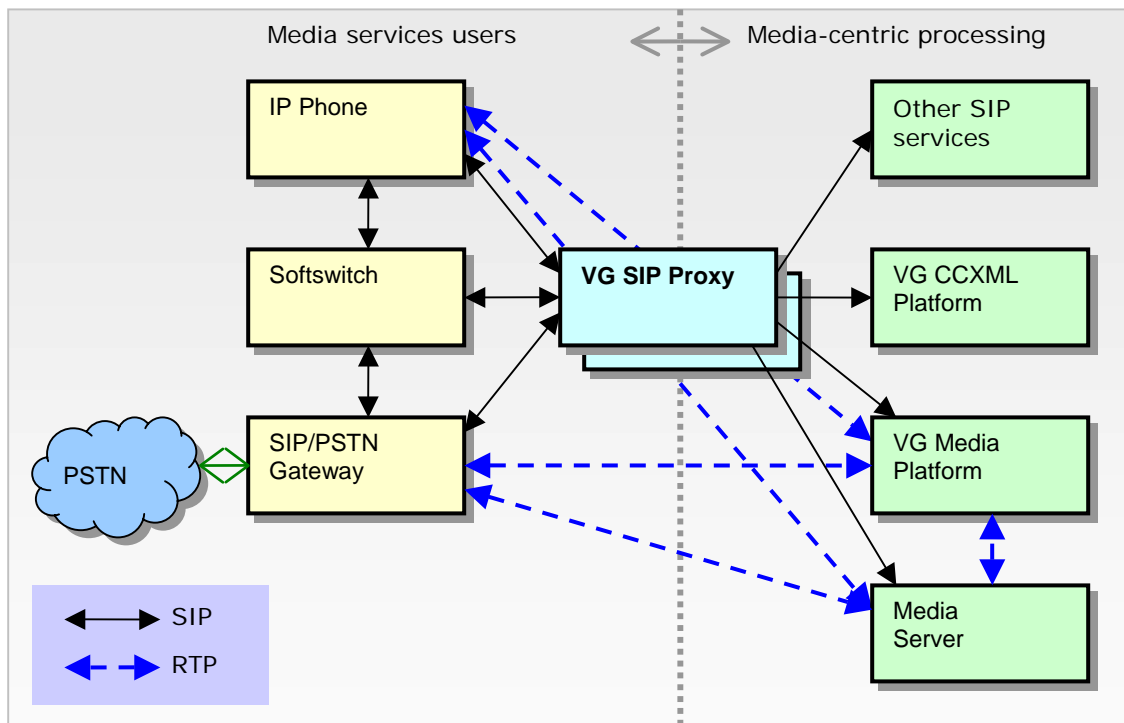
There are a number of options with respect to how VoiceGenie's SIP proxy can be deployed. These options are explained by the following sections:

The **Logical Architecture** of a deployment using VoiceGenie's SIP proxy is first discussed, showing how the proxy interacts with other elements in the network;

The **Network Configurations** section shows more concretely how VoiceGenie's SIP proxy is connected to the network or networks that exist in a typical deployment.

2.1 Logical Architecture

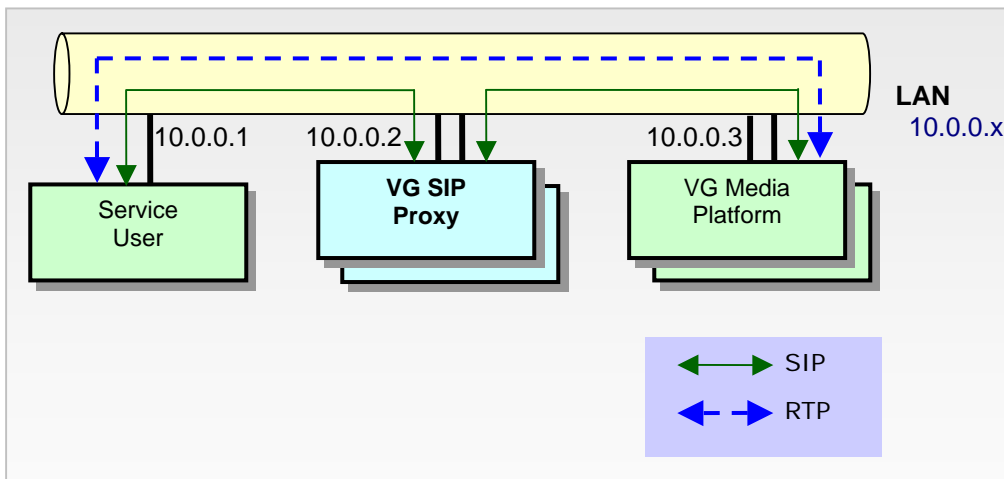
From a logical perspective, the purpose of VoiceGenie's SIP proxy is to act as an interface to a collection of media processing resources, such as VoiceGenie's media platform, CCXML platform, audio conferencing or other resources. SIP devices and applications can then make use of media-centric services through the proxy, without having to know the actual location of those resources or how to manage various routing decisions. The services provided by the VoiceGenie SIP proxy are used not only by clients such as media gateways or softswitches, but may also be used by internal media resources to co-ordinate interactions with one another. For instance, VoiceGenie's CCXML platform offers the ability to manage a VoiceXML dialog that actually executes on VoiceGenie's media platform; the CCXML platform may make use of proxy capabilities to locate an appropriate VoiceXML platform. The following diagram illustrates this logical architecture:



Although the above diagram shows a number of elements, both users of media services managed by the proxy as well as the underlying media-centric services that the proxy routes to, the actual configuration used in a deployment is typically much simpler. For instance, in a contact centre environment providing automated self-service or call routing, the deployment might consist solely of a number of SIP/PSTN gateways to handle incoming calls, a cluster of VoiceGenie media platforms that provide the actual treatment of calls through touch-tone or speech applications, and a redundant pair of VoiceGenie SIP proxies that provide load balancing across the available VoiceGenie platforms. In more complex next-generation network architectures, the VoiceGenie SIP proxy essentially acts as a **logical media server**, aggregating the capabilities of heterogeneous array of processing resources and acting as a single interface point for media services users.

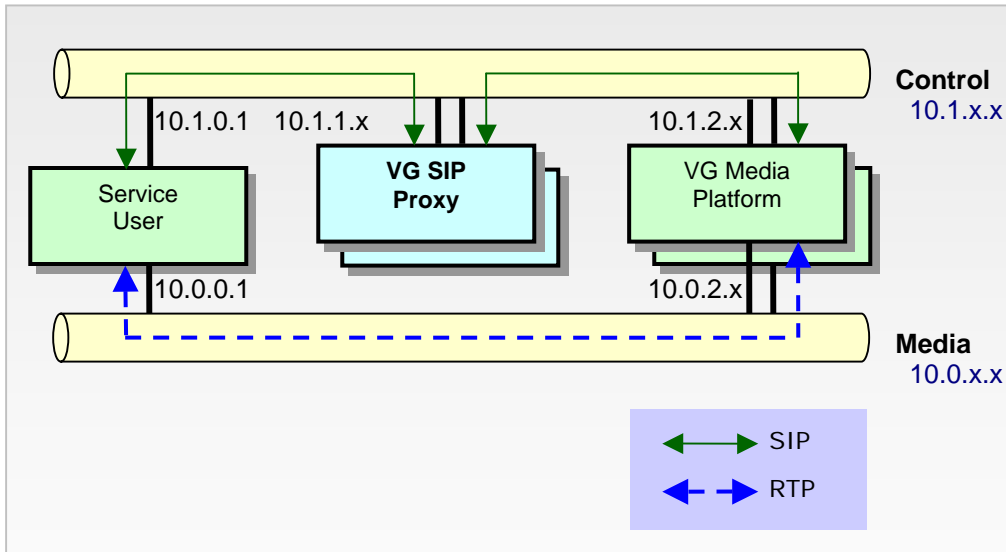
2.2 Network Configurations

VoiceGenie's SIP proxy can be deployed in three basic network configurations, which vary with respect to the degree of isolation between SIP control and RTP media, and in the separation of "external" SIP interfaces – which face the users of the SIP proxy – from "internal" SIP interfaces – which face the SIP resources being managed. The simplest of these is the **integrated network** configuration, in which all SIP and RTP traffic between elements make use of a single physical network:



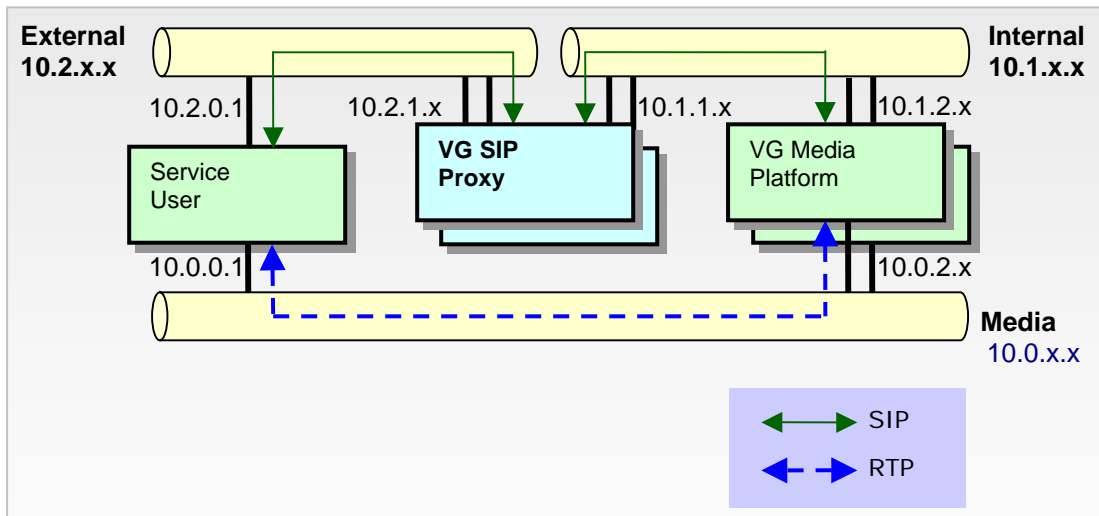
The single integrated network shown is a **logical** network; the physical realization of this may employ the use of redundant network connections with Ethernet bonding, or other similar approaches to ensure that there is no single physical point of failure.

In some environments, it is desirable to separate control and media, typically to ensure priority and quality of service of the control channel, but also sometimes to ensure the privacy of data exchanged on the control channel. This **separated network** configuration is shown below:



As mentioned above, each of the control/media networks may themselves involve multiple redundant physical connections. However, it is also possible that the two networks (control and media) may be logically separated, but might share the same physical network infrastructure. As an example, a single physical switch might provide routing for two separate VLANs, with the service user and VoiceGenie media platform each being configured to support two IP addresses (each on a different VLAN) on a single physical interface.

The final configuration, a **partitioned network**, hides the internal media processing resources – at least from a control (SIP) perspective – behind the proxy, using separate logical networks for communication to the SIP proxy from the service user and from internal media resources.

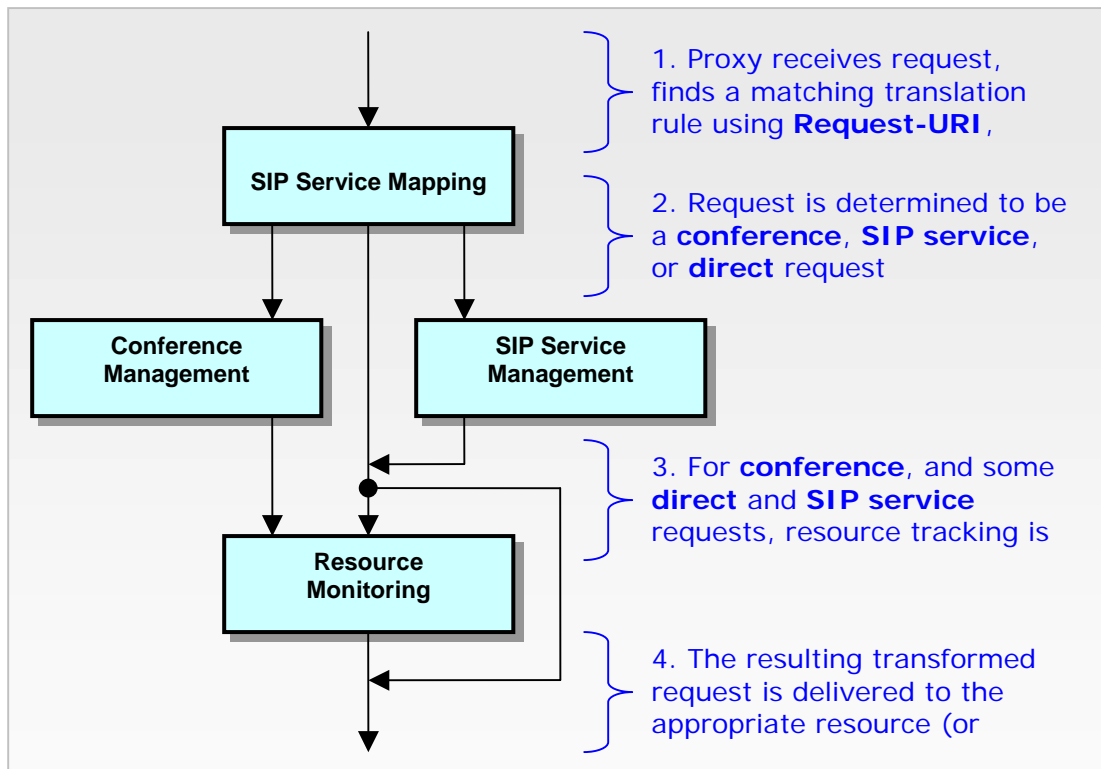


An option with the partitioned network configuration shown above is the use of an RTP firewall/proxy that allows separate internal and external media networks. This completely isolates the internal infrastructure, and hides any internal topology from the outside. This also facilitates internal media redirection without visibility to the service user. This RTP firewall/proxy would typically operate under the control of the SIP proxy, although for mid-call media redirection, this element could be controlled by a different entity (such as a SIP-based CCXML platform, which acts as a user agent).

3 Features

3.1 Request Processing

The first step involved when VoiceGenie's SIP proxy handles a request for service (which currently is for INVITE requests only) is to perform service translation and matching to transform the incoming request and match that request to a service offered by the proxy. In addition to the translation services that the proxy offers, it may also provide SIP service, conference, and resource management. The following diagram illustrates the process of handling a request:



SIP Service Mapping provides rule-based matching of a received request, and allows for general-purpose transformations of the Request-URI and any parameters of the Request-URI. The example given above was the use of this service to convert a URI such as `sip:dialog.vxml.http%3A//a/test.vxml@test.com` (from the Rosenberg IETF draft) to

`sip:service1234@box1.server.com;voicexml=http://a/test.vxml` (consistent with the burger-netann IETF draft). The rules, which are supplied by the proxy user, make use of regular expression matching and manipulation to allow any combination of input parameters to be mapped to any combination of output parameters. In addition, the rules allow control over whether the request will be further handled by conference service management, the more general SIP service management, or whether the request is directly routed to a particular target URI.

Conference Management is described separately below, and thus will not be covered here.

SIP Service Management allows for requests to be routed to higher level SIP services. As part of defining these SIP services, the administrator defines the resources that can be used to fulfill these services. This allows for the identification of a collection of resources that could potentially fulfill a given request. The capabilities described in the sections below on resource monitoring and load balancing are then used to filter and prioritize the resources that are actually used as the destination for the request.

SIP Resource Monitoring is an optional service that is typically applied for resources managed by the proxy, but skipped for external resources such as media gateways which may be shared by other users. Resource tracking allows the proxy to keep track of the level of utilization of a particular resource, based on the requests that are active against that resource. Far from a simple tracking table, the resource tracking module provides synchronization across a cluster of proxies, so that each proxy is capable of making a globally optimal decision with respect to resource routing.

The above steps in the request handling process do not necessarily result in the selection of a single resource or destination for the request; rather, a prioritized list is generated of resources that could handle the needs of a given request. This is described in greater detail in the load balancing section below.

3.2 SIP Resource Monitoring

In order to make optimal decisions, VoiceGenie's SIP proxy needs to have up-to-date information concerning the status of resources under the management of the proxy. The proxy uses a variety of mechanisms to monitor resources:

SIP Registration. The proxy allows resources to use it as a SIP registrar – however, this is not required. Resources that require registration but do not hold an active registration are considered to be unavailable.

Periodic OPTIONS Requests. The proxy can monitor SIP resources by sending a periodic OPTIONS request to validate that the SIP resource is responding correctly. This allows the proxy to be aware within a relatively short period (configurable with a default of two minutes) if a resource fails.

OA&M Integration. When the proxy is used with VoiceGenie products that are integrated into VoiceGenie's common OA&M infrastructure, the proxy gets real-time status information about the resource, supported by a frequent heartbeat between the OA&M infrastructure and the resource.

When the proxy determines the set of resources that are capable of servicing a particular request, it will restrict that set to the resources that are currently detected as being available. This ensures that requests are only routed to resources that are available.

In addition to using monitoring information to make optimal routing decisions, the VoiceGenie SIP proxy also feeds this information into VoiceGenie's underlying OA&M infrastructure, allowing the System Management Console (SMC) to display status information about the resources in the cluster as perceived by the VoiceGenie SIP proxy. This provides the operator with a single view over all resources being used in conjunction with the proxy.

3.3 Load Balancing

Once VoiceGenie's SIP proxy has determined the resources that are capable of servicing a particular request, and validates that those resources are available (based on actively monitoring resources as described above), it load balances requests across those resources. Two kinds of load balancing are supported by the proxy:

Round Robin load balancing, which rotates requests amongst available resources without using additional information about the resource;

Least Utilized routing, in which the proxy makes use of information about the services currently being provided using each resource in order to select the resource that has the most available capacity.

The mechanism used for load balancing serves as a means for prioritizing across available resources. Thus, if the resource that the proxy believes is the least utilized turns out not to be available to service the request, the proxy will not fail the request and return an error – it will try the next least utilized resource in the set it identified was capable of servicing the request. From a more technical perspective, VoiceGenie's SIP proxy thus acts as a sequentially forking proxy.

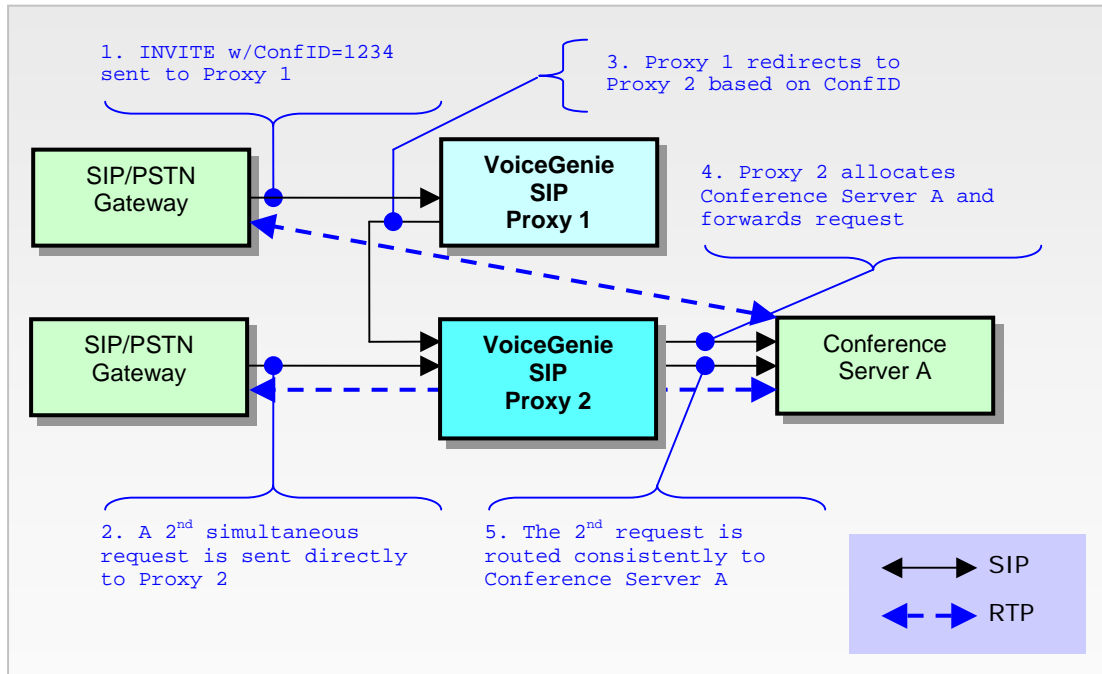
The proxy also supports customization of the timeouts used when sending requests, so that it is possible to failover to an available second resource quickly in the event that the initial resource selections are not available.

3.4 Conference Management

A key capability of the proxy that is relative orthogonal to other capabilities is the management of conferencing/mixing resources. Whereas general SIP services generally have the property that independent instances (separate calls) do not interact with one another, conferences have the property that multiple callers to the same conference need to be routed to the same physical resource for mixing. In addition, even if a request is received simultaneously on two or more proxy instances in a clustered environment, it is necessary that the decision be consistent across the entire cluster. That is to say, if proxy # 1 routed conference 1234 to conferencing resource X, then proxy # 3 must also route any request for conference 1234 to conferencing resource X.

VoiceGenie's SIP proxy provides consistent **conference routing** through two internal capabilities. The first is **transparent request redirection**, where one specific available proxy is always guaranteed to handle requests to a particular conference. Thus, if proxy # 2 is designated as the handler for conference 1234, then all requests related to conference 1234 will be handled by proxy # 2. Through supporting low-level message redirection or UDP multicast (described below), the proxy is able to achieve this very efficiently. UDP multicast in particular adds very little processing overhead, since an extra network transmission is not necessary – the proxy chosen to handle the incoming SIP request is always the right one. The second capability is **conference state mirroring**, in which conferences that are set up

successfully are broadcasted to all proxies in the cluster. In this way, if one particular proxy in the cluster was to fail, the surviving members of the cluster could route conference requests in a manner that was consistent with the failed proxy. This is shown in the diagram below:



In addition to managing the routing of conferences, the proxy also manages **conference resource allocation**. Conference resources are typically rated to handle a particular number of simultaneous conference participants; they are also typically further subject to limitations on the maximum size of any individual conference being hosted. VoiceGenie's SIP proxy makes use of information about the currently available resource level, coupled with minimum and maximum conference size information that is derived from the conference request, to allocate conferences to the optimal resource with the highest chance of being able to serve the request successfully. The ability to specify a minimum conference size allows the proxy to reserve a certain amount of capacity on a conference resource, such that even if other calls arrive in the interim, it is guaranteed that sufficient capacity will be available to deliver the required minimum number of participants on existing conferences using that resource.

Conference management requires the following 3 request-URI parameters for all SIP requests to conferences:

- 1) confinstid – a cluster-wide unique identifier that identifies the conference.
- 2) confmaxsize – the maximum size of the conference. If there are more participants joining the conference than the max size, SIP Proxy will reject the callers above the maximum conference size.
- 3) confreserve – the number of ports reserved for this conference when the first caller starts a conference.

Note that the caller does not have to provide all 3 parameters except for the unique identifier. SIP Service Mapping can use translation rules to provide the URI parameters. Here is an example:

Caller sends:

```
sip:conf=1234@10.0.0.1;conftype=threewaycall
```

SIP Service Mapping translates the URI to:

```
sip:conf=1234@10.0.0.1;sipservice=conf;confinstid=1234;confmaxsize=3;confreserve=3
```

The represents a 3-way call where the caller reserves 3 ports on the conference resource. SIP Proxy only allows a maximum of 3 concurrent participants to this conference. The 4th caller will be rejected by the SIP Proxy.

4 Installation

Like all other VoiceGenie 7 components, the SIP Proxy should be installed/deployed/configured using VoiceGenie OA&M framework, which must be installed before any other VoiceGenie software can be deployed.

This section covers the installation process for the SIP Proxy module, assuming that the operating system, all third party software and the VoiceGenie OA&M Framework software are installed; if they are not installed please refer to the document:

[VoiceGenie 7 Installation Guide](#)

Required files:

Component	Filename
SIP Proxy	ccp-proxy-7.0.0-1.tar.gz
CCP Redundancy Manager	ccp-rm-7.0.0-1.tar.gz
Linux Bonding Driver (Linux only)	bonding-7.0.0-3.tar.gz

4.1 Installation Steps

1) Product Manager

Open a web browser and go to the System Management Console homepage;
 Login and click on "Configuration" tab;
 On the left hand column click on "Product Manager";
 Click Browse and select one of the filenames listed above and click OK;
 Click Upload to upload the component to Product Manager;
 Repeat until all files are uploaded.

2) Config Profile Manager

In System Management Console, go to "Configuration" tab;
 On the left hand column click on "Config Profile Manager";
 At the Product drop-down list, select "SIP Proxy – 7.0.0";
 Enter a name for this profile;
 Click Create, and you will notice the profile name will show up at the table below;

Repeat the above steps for product "Call Control Platform Redundancy Manager".

3) Deployment Manager

In System Management Console, go to "Configuration" tab;
 On the left hand column click on "Deployment Manager";
 Click on the radio button that corresponds to "SIP Proxy 7.0.0";
 Select the configuration profile from the drop-down list that corresponds to this row;
 Select the servers where SIP Proxy will be deployed;
 Click Deploy button to start deploying;
 Check Deployment History to make sure the product is deployed successfully;

Repeat the above steps for the Call Control Platform Redundancy Manager product.

4.2 Configuring the SIP Proxy

4.2.1 General

Before starting SIP Proxy, there are 6 configuration tables that need to be completed in order for SIP Proxy to properly forward incoming SIP requests:

- 1) SIP Service Table
- 2) SIP Services Mapping
- 3) SIP Resource Types Table
- 4) SIP Resources Table
- 5) SIP Proxy Configuration
- 6) Redundancy Manager Cluster Mapping

Please refer to the [Provisioning the SIP Proxy](#) section for instructions for provisioning the first 4 tables.

Please refer to the [VoiceGenie 7 SIP Proxy System Reference Guide](#) for the full lists of configuration parameters for "SIP Proxy Configuration" and "Redundancy Manager Cluster Mapping".



Note about enabling TCP Support

The sip.transport.X parameters configures the SIP stack's transport settings. By default sip.transport.0 has the value "transport0 udp:any:5060".

To enable TCP transport:

1. Enable sip.transport.1 and assign its value as "transport1 tcp:any:5060"
2. Enable sip.route.default.udp and assign its value as 0
3. Enable sip.route.default.tcp and assign its value as 1.

4.2.2 2 network IPs with bonding driver setting

The following are the instructions to configure 2 network IPs with bonding driver, and necessary SIP Proxy configuration changes to make this architecture work.

4.2.2.1 Network configuration changes

Bonding configuration files are located under /etc/sysconfig/network-scripts.

To configure 2 bondings, there should be two files such as ifcfg-bond0, ifcfg-bond1.

The first file ifcfg-bond0:

```
DEVICE=bond0
BOOTPROTO=none
IPADDR="10.0.0.174"
```

```
NETMASK="255.255.255.0"
BROADCAST="10.0.0.255"
ONBOOT="yes"
USERCTL=no
```

The second file ifcfg-bond1:

```
DEVICE=bond1
BOOTPROTO=none
IPADDR="10.0.1.174"
NETMASK="255.255.255.0"
BROADCAST="10.0.1.255"
ONBOOT="yes"
USERCTL=no
```

Network configuration files

This totally depends on how many cards are installed. The example here is four. The file names are ifcfg-eth[0-3]. If eth0 and eth1 are bonding to bond0, the file content should be:

```
DEVICE=eth0
USERCTL=no
MASTER=bond0
SLAVE=yes
BOOTPROTO=none
ONBOOT="yes"
```

Device name will be different for eth1. If different bonding driver the NICs are bonding to the 'MASTER' will be different:

```
DEVICE=eth2
...
MASTER=bond1
...
```

Change /etc/modules.conf:

Below is an example:

```
alias eth0 tg3
alias eth1 tg3
alias eth2 e1000
alias eth3 e1000
alias scsi_hostadapter mptbase
alias scsi_hostadapter1 mptscsih
alias usb-controller usb-ohci
alias bond0 bonding
options bond0 -o bond0 miimon=100 mode=1 multicast=1
alias bond1 bonding
options bond1 -o bond1 miimon=100 mode=1 multicast=1
```

4.2.2.2 CCP Proxy configuration change.

Second IP should be set in the respaddr parameter:

```
proxy.sip.proxy.respaddr = 10.0.0.174,10.0.1.174
```

1 more transport is added for second IP and 2 more are added if TCP is configured as well:

```
proxy.sip.transport.0 = transport0 udp:10.0.0.174:5061  
proxy.sip.transport.1 = transport1 udp:10.0.1.174:5061  
proxy.sip.transport.2 = transport0 tcp:10.0.0.174:5061  
proxy.sip.transport.3 = transport0 tcp:10.0.1.174:5061
```

Related route destinations also need to be added:

```
proxy.sip.route.dests = 0 1 2 3  
proxy.sip.route.dest.0 = 10.0.0.0 255.255.255.0 0 0  
proxy.sip.route.dest.1 = 10.0.1.0 255.255.255.0 1 0  
proxy.sip.route.dest.2 = 10.0.0.0 255.255.255.0 2 0  
proxy.sip.route.dest.3 = 10.0.1.0 255.255.255.0 3 0
```

5 Operations, Administration, Maintenance

5.1 Starting and Stopping

Like all other VoiceGenie 7 components, the System Management Console offers a dedicated page to start or stop the SIP Proxy. Click on the Operations tab and click on "Start/Stop Software" on the left hand column. Click on the server(s) that you want to start/stop and then click the Start/Stop button.

Note that the "Suspending" a SIP Proxy is not supported.

The following is a screenshot of the start/stop page:

System Management Console
millhouse.voicegenie.com | Connected to CMP Proxy | User Name: pw | v7.0.0 Cluster

Start/Stop Server

This page allows you to *start, stop, suspend, or resume* services on a set of VoiceGenie servers.

Please select the Clusters or Nodes or Components that you would like to perform the operation on. Then click Start, Stop, Suspend, or Resume to perform the desired operation.

Operations
 Start/Stop Software ■
 Get Platform Info ■

Web Cache
 Start/Stop Cache ■
 View Cache ■
 Cache List Manager ■
 Perform Preload/Purge ■
 View Access Log ■
 View Event Log ■

Licensing
 Platform Licensing ■

Logout ■

Network
 10.0.0.190 [properties] ⚠

- ✓ Call Manager
- ✓ Command Line Console
- ✓ Redundancy Manager
- ✓ SIP Proxy
- ✓ System Management Console
- ✓ VoiceXML Interpreter
- ✓ Web Proxy

Start Stop Suspend Resume

5.2 Health Status

Health status summary can be made available through the command line console (CLC) or the SMC. Individual health status items can be retrieved through SNMP traps.

In CLC, type: health ccproxy

In SMC, go to "Monitoring" tab and click on the SIP Proxy item.

Here is a sample output:

```
Health for SIP Proxy (ccproxy) on 10.0.0.212
```

```
Started: 2005-01-26/17:40:38.913
Proxy State: Active
Requests forwarded: 0
Conferences created: 0
Requests forwarded in past 5 minutes: 0
Peak requests forwarded: 0
Bad Messages Received: 0
SIP Transports: UDP:10.0.0.212:5061
  Resource(10.0.0.212:5060) Ports:0/100 Registered:No Status:Up
  Resource(10.0.0.99:5060) Ports:0/100 Registered:No Status:Down
```

The following are explanations of each parameter in the health status summary:

Started – The time which SIP Proxy was started

Proxy State – possible states are Initialized, Unavailable, Activating, Active, Standby, Soft Shutdown, or Unknown

Requests forwarded – total number of SIP requests forwarded

Conferences created – total number conference created

Requests forwarded in part 5 minutes – total number of SIP requests forwarded in the past 5 minutes

Peak requests forwarded – peak number of SIP requests forwarded in any 5 minute period

Bad messages received – number of bad SIP messages received and discarded

SIP Transports – lists the proxy SIP transports. Each transport has transport type (UDP or TCP), IP address, port number that it listens on.

SIP Resources – lists the SIP resources and their operational state on each line. It contains the host and port, current ports used/total ports available, registered state, and operational status.

The health status parameters are available through SNMP get. The complete list of parameters and description is listed in Appendix A. Please also refer to the OA&M Framework User Guide for details on retrieving the information.

5.3 Logging

SIP Proxy generates logging information using the VoiceGenie OA&M Framework. All logs of level Critical (LOG_0), Error (LOG_1), and Warning (LOG_2) sent upstream and to the log file. The default log file is in /usr/local/ccp-proxy/log/CMP.log.ccp-proxy.

Log levels for Notice (LOG_3) and Information (LOG_4) are stored in the log file.

Trace logs (LOG_5) is disabled by default. Enabling it will trace all SIP messages sent and received by the SIP Proxy. Trace is not recommended for deployment environment as trace will flood the trace files quickly and decreases system performance.

To enable trace, go to SIP Proxy Configuration and select false for `cmp.trace_flag`. Click Update to submit the configuration change.

5.4 Cluster Management

Details for how to create and administer Clusters can be found in the following document:
[VoiceGenie 7 OA&M Framework – SMC Guide](#)

6 Provisioning the SIP Proxy

One can provision the SIP Proxy to properly forward incoming SIP requests through changing the following 4 configuration tables:

- 1) **SIP Service Table** defines all available SIP services within the cluster. Load balancing scheme can be defined for each SIP Service.
- 2) **SIP Services Mapping Table** defines a set of rules that maps incoming requests to a SIP Service. This table can also translate SIP Request URI based on regular expression rules.
- 3) **SIP Resource Types Table** defines a template for SIP resources that share common attributes.
- 4) **SIP Resources Table** defines the list of SIP resources. Each SIP Resource provides one or more SIP Service and each SIP Resource belongs to only one SIP Resource Type.

Note that these provisioning pages do not require targeting; clicking on the "Create" or the "Update" button will apply the changes automatically to all configured SIP Proxy in the cluster.

The following subsections describe the details of setting up the tables.

6.1 SIP Service Table

The SIP Service Table defines all the SIP Services in the cluster. The table can be accessed through SMC in the Configuration tab->SIP Services. The following is a screen shot of the table:

System Management Console
millhouse.voicegenie.com | Connected to CMP Proxy | User Name: pw | v7.0.0 Cluster

SIP Service Configuration

To create a new SIP Service Entry enter the SIP Service Name, select a Load Balancing Scheme and the Required Capability, then click on *Create*.

SIP Service Name:
Load Balancing Scheme: Least Used
Required Capability: VoiceXML

The following entries already exist.
 To update an entry change the value in the text box and click on *Update*.
 Click on *Delete* to delete an entry.

SIP Service Name: conf ID: 8
Load Balancing Scheme: Least Used
Required Capability: VoiceXML

SIP Service Name: mp ID: 4
Load Balancing Scheme: Least Used
Required Capability: VoiceXML

Each Provisioning entry represents a row in the table. Each entry contains the following items:

SIP Service Name is a string (no space) that identifies the SIP Service.

Load Balancing Scheme defines the scheme of which the SIP Resource is selected. There are two schemes available:

- Least Used – the SIP Resource with the least number of active dialogs will be selected.
- Round-Robin – selects a SIP Resource in a round-robin fashion regardless of the load of the resource

Required Capability – there are only two types of capabilities available: VoiceXML and Conference.

6.2 SIP Services Mapping Table

6.2.1 General

The SIP Service Mapping Table list rules for mapping incoming SIP requests into a SIP Service to forward to. The rules are sorted by precedence and each SIP request is compared in order until finding a matching rule.

The table is accessible through SMC by clicking the Configuration tab->"SIP Service Mapping".

SIP Service Mapping Configuration

To create a new SIP Service Mapping Entry enter the matching rule and the SIP Service that it maps to, then click on *Create*.
For more Advanced capabilities click on *Advanced*.

Precedence: <input type="text"/>	
SIP Request URI Matching Rule: <input type="button" value="Advanced"/>	
Userinfo Regex:	Parameters to Match:
sip: (*) @	;*
Userinfo:	SIP Service:
sip: \1 @	mp
	Parameters to Add:
	;*
<input type="button" value="Create"/>	

The following entries already exist.
To update an entry change the value in the text box and click on Update.
Click on Delete to delete an entry.

Precedence: <input type="text" value="2"/>	ID: 32
SIP Request URI Matching Rule: <input type="button" value="Advanced"/>	
Userinfo Regex:	Parameters to Match:
sip: ([^@]*) @	;*
Userinfo:	SIP Service:
sip: \1 @	mp2
	Parameters to Add:
	;*
<input type="button" value="Update"/> <input type="button" value="Delete"/>	

Precedence: <input type="text" value="1"/>	ID: 19
SIP Request URI Matching Rule: <input type="button" value="Advanced"/>	
Userinfo Regex:	Parameters to Match:
sip: ([^@]*) @	;*
Userinfo:	SIP Service:
sip: \1 @	mp
	Parameters to Add:
	;*
<input type="button" value="Update"/> <input type="button" value="Delete"/>	

Each provisioning entry is a rule in the SIP Service Mapping table. There are 6 fields in each rule:

Precedence defines the order in the table in ascending order. The numbers need not be consecutive but must be unique. If there is more than one entry that contains the same precedence value, an alarm will be generated and order will not be guaranteed.

Matching rules defines two rules; both of them must be met in order for the entry to be considered a match.

- 1) **Userinfo Regex** defines a regular expression. If the regular expression matches the SIP Request URI, the request is considered a match;
- 2) **Parameters to Match** define a list of parameters that must be included in the SIP Request URI in order to consider a match. Multiple parameters can be defined in the field; it is space delimited. Use * to match any parameters.

Service Mapping defines 3 fields to translate the Request URI. The resulting Request URI will be used to forward the SIP message to. Regular expression replace can be used in the fields. Matched groups can be expressed in \1, \2, \3, etc. tokens, while matched parameters in "Parameters to Match" can be replaced with [parameter] tokens.

- 1) **Userinfo** portion of the Request URI will be replaced. Regex replacement tokens can be used.
- 2) **SIP Service** selects one SIP Service defined in the SIP Service Table.
- 3) **Parameters to Add** places URI parameters. Besides allowing regex replacement tokens, "*" can be used to place all matched parameters in "Parameters to Match".

6.2.2 Advanced Configuration

Due to the flexibility of the SIP Service Mapping Table, complex matching rules can be created to match anything that regular expression can offer. The Advanced button allows the user to go into advanced mode that shows the raw form of the rules.

This is the text format of the mapping rule:

```
[order]
[resource name]
[Request URI]
[URI parameters]
[translated userinfo]
[translated hostport]
[translated URI parameters]
```

[order] is the order number of the entry;

[resource name] is the SIP Service name where the SIP request must originate from;

[Request URI] is the regular expression that must match the SIP request URI;

[URI parameters] is a space delimited string of URI parameters that must exist in the SIP Request URI;

[translated userinfo] is the userinfo portion of the Request URI to be replaced. Regex replacement tokens can be used;

[translated hostport] can be a SIP service name or an actual hostname;

[translated URI parameters] is a space delimited string of URI parameters that will be in the replaced Request URI. Regex replacement tokens can be used.

6.3 SIP Resource Types Table

The SIP Resource Types table defines common attributes for each SIP Resource Type. The following is a screenshot of the table:

SIP Resource Type Configuration

To create a new SIP Resource Type Entry enter the SIP Resource Name and all other pertinent details, then click on *Create*.

SIP Resource Name:	<input type="text"/>		
Ports:	<input type="text"/>	Capabilities:	VoiceXML Conference
Registration Method:	CMP	Monitoring Method:	CMP
General Parameters:	<input type="text"/>		
<input type="button" value="Create"/>			

The following entries already exist.
To update an entry change the value in the text box and click on Update.
Click on Delete to delete an entry.

SIP Resource Name:	MP-7.0		ID: 26
Ports:	300	Capabilities:	VoiceXML Conference
Registration Method:	CMP	Monitoring Method:	CMP
General Parameters:	<input type="text"/>		
<input type="button" value="Update"/> <input type="button" value="Delete"/>			

SIP Resource Name:	Conf-Server		ID: 23
Ports:	60	Capabilities:	VoiceXML Conference
Registration Method:	SIP	Monitoring Method:	Options
General Parameters:	confresourcetype-maxsize=10		
<input type="button" value="Update"/> <input type="button" value="Delete"/>			

There are 6 fields for each SIP Resource Type:

- 1) SIP Resource Name – the name of the SIP Resource Type

- 2) Ports – the number of ports available for the SIP Resource belonging to this SIP Resource Type
- 3) Capabilities – defines the type of capabilities that this SIP Resource Type can provide.
- 4) Registration Method – defines how the SIP Resource registers with SIP Proxy. Registering provides up to date information to the SIP Proxy about how to forward SIP messages to the resource. There are 3 methods supported:
 - a. SIP – The resource will send SIP REGISTER message periodically. SIP Proxy expects an expiry period of 60 seconds and will modify the Contact header with an expiry parameter of 60.
 - b. CMP - VoiceGenie products that are integrated into VoiceGenie's common OA&M infrastructure will select this value in order for the proxy to get real-time status information about the resource.
 - c. None – If the non-VoiceGenie resource type does not support SIP REGISTER, select this parameter.
- 5) Monitoring Method – SIP Proxy monitors the resource with one of the following 3 methods:
 - a. OPTIONS - The proxy can monitor SIP resources by sending a periodic OPTIONS request to validate that the SIP resource is responding correctly. This allows the proxy to be aware within a relatively short period (configurable with a default of two minutes) if a resource fails.
 - b. CMP – SIP Proxy receives a periodic heartbeat from VoiceGenie products via the VoiceGenie's common OA&M infrastructure.
 - c. None – the resource is incapable of handling SIP OPTIONS message and SIP Proxy assumes the resource is always available.
- 6) General Parameters – any other attributes that belongs to the resource type will go into this text field. The field accepts a comma-delimited string of attribute-value pairs. An example of the parameter is "confresourcetypemaxsize=10". Please refer to Appendix C for a complete list of SIP Resource Type parameters.

6.4 SIP Resources Table

The SIP Resource Table lists all the SIP Resources available in the cluster. The following is a screenshot of the table:

SIP Resource Configuration

To create a new SIP Resource Entry enter the SIP Resource Name and all other pertinent details, then click on *Create*.

SIP Resource Name:	<input type="text"/>
<input checked="" type="radio"/> CMP Enabled Resource: <input type="radio"/> Non-CMP Enabled Resource:	
<input type="text" value="10.0.0.190"/>	Enter Hostname: <input type="text"/> Enter Port: <input type="text"/>
Associated SIP Resource Type:	MP-7.0
Associated SIP Services:	<input type="text" value="mp"/> <input type="text" value="mp2"/>
<input type="button" value="Create"/>	

The following entries already exist.

To update an entry change the value in the text box and click on Update.

Click on Delete to delete an entry.

SIP Resource Name:	<input type="text" value="millhouse"/>	ID: 31
<input checked="" type="radio"/> CMP Enabled Resource: <input type="radio"/> Non-CMP Enabled Resource:		
<input type="text" value="10.0.0.190"/>	Enter Hostname: <input type="text" value="10.0.0.190"/> Enter Port: <input type="text" value="5060"/>	
Associated SIP Resource Type:	MP-7.0	
Associated SIP Services:	<input type="text" value="mp"/> <input type="text" value="mp2"/>	
<input type="button" value="Update"/> <input type="button" value="Delete"/>		

There are 4 fields in a SIP Resource entry:

- 1) **CMP Enabled Resource** – click the radio button to select a VoiceGenie resource. When this radio button is selected, "Enter Hostname" field will be grayed out. Use the drop-down list to select the host of the resource. "Enter Port" field is the SIP port defaults to 5060.

- 2) **Non-CMP Enabled Resource** – click the radio button to select a non-VoiceGenie resource. The drop-down list will be grayed out. Enter the hostname of the SIP resource in “Enter Hostname” field. Enter the SIP port as well; the port defaults to 5060.
- 3) **Associated SIP Resource Type** – this is a drop-down list that contains the names of all SIP Resource Types. Select one type to make this resource as part of this SIP Resource Type.

Associated SIP Services – this is a multi-select field that shows all SIP Services available. Ctrl-click on all SIP services that you wish the SIP Resource to be part of.