AudioCodes™

MP-11x, Mediant 1000 & Mediant 2000

Application Note

AudioCodes Media Gateway's SAS and ENUM Capabilities for Central Routing Decision Making

In a

Microsoft™ Office Communications Server 2007 Environment

Document #: LTRT-08051

March 2009

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Notice

This document describes the configuration of AudioCodes' Mediant 1000, Mediant 2000 and MediaPack SIP VoIP media gateways providing SAS and ENUM features in a Microsoft™ Office Communications Server 2007 deployment.

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual.

Typographical Conventions

The typographical convention used throughout this guide is described in the table below:

Table 1-1: Typographical Conventions

Items	Convention Used	Example
Screen names, field names, parameter values	Enclosed by single quotation marks.	Open the 'Coders' screen.
Path to screens	Bolded with the path given as: Menu name from menu bar > submenu name from submenu bar > command under submenu bar (if any).	Access the 'Coders' screen (**Protocol Management** menu > **Protocol Definition** > **Coders**).
Command buttons	Bolded.	Click the **OK** button.
Values entered by typing	Enclosed by double quotation marks.	In the 'Gateway Name' field, enter "10.0.0.10".

Related Documentation

Document #	Manual Name
LTRT-688xx (LTRT-68801)	Mediant 2000 & TP-1610 & TP-260-UNI SIP User's Manual
LTRT-690xx	Mediant 3000 & Mediant 2000 & TP Series SIP Gateways Release Notes
LTRT-833xx	Mediant 1000 SIP User's Manual
LTRT-831xx	Mediant 1000 SIP Release Notes
LTRT-654xx	MP-11x & MP-124 SIP User's Manual
LTRT-656xx	MP-11x & MP-124 SIP Release Notes
LTRT-260xx	Mediant 1000 & 2000 & MS OCS 2007 Quick Guide

Note: Throughout this guide, whenever the term *gateway* refers to AudioCodes' Mediant 1000, Mediant 2000 and MP-11x media gateways.

1 Introduction

Currently, Microsoft's Office Communications Server 2007 Mediation Server (*Mediation Server*) and AudioCodes Voice-over-IP (VoP) media gateway implement a peer-to-peer connection in which each Mediation Server routes outbound calls to its peer media gateway and vice versa. In other words, outbound calls from the Microsoft Office Communicator (*OC*) client cannot be routed by Mediation Server to various media gateways or any other SIP entity such as IP phones or IP-PBX's. Moreover, most of the IP phones need to perform registration to start calls, however Mediation Server does not support registration methods.

AudioCodes Stand-Alone Survivability (SAS) feature, hosted on AudioCodes media gateways (MP-11x, Mediant 1000, and Mediant 2000), can provide call connectivity between enterprise SIP devices (such as IP phones, ATA devices or other media gateways) and Mediation Server, as well as between the SIP devices themselves. This interface enables incoming and outgoing calls from and to OC clients towards the SIP devices as well as inbound calls between the SIP devices even in scenarios where Mediation Server is down.

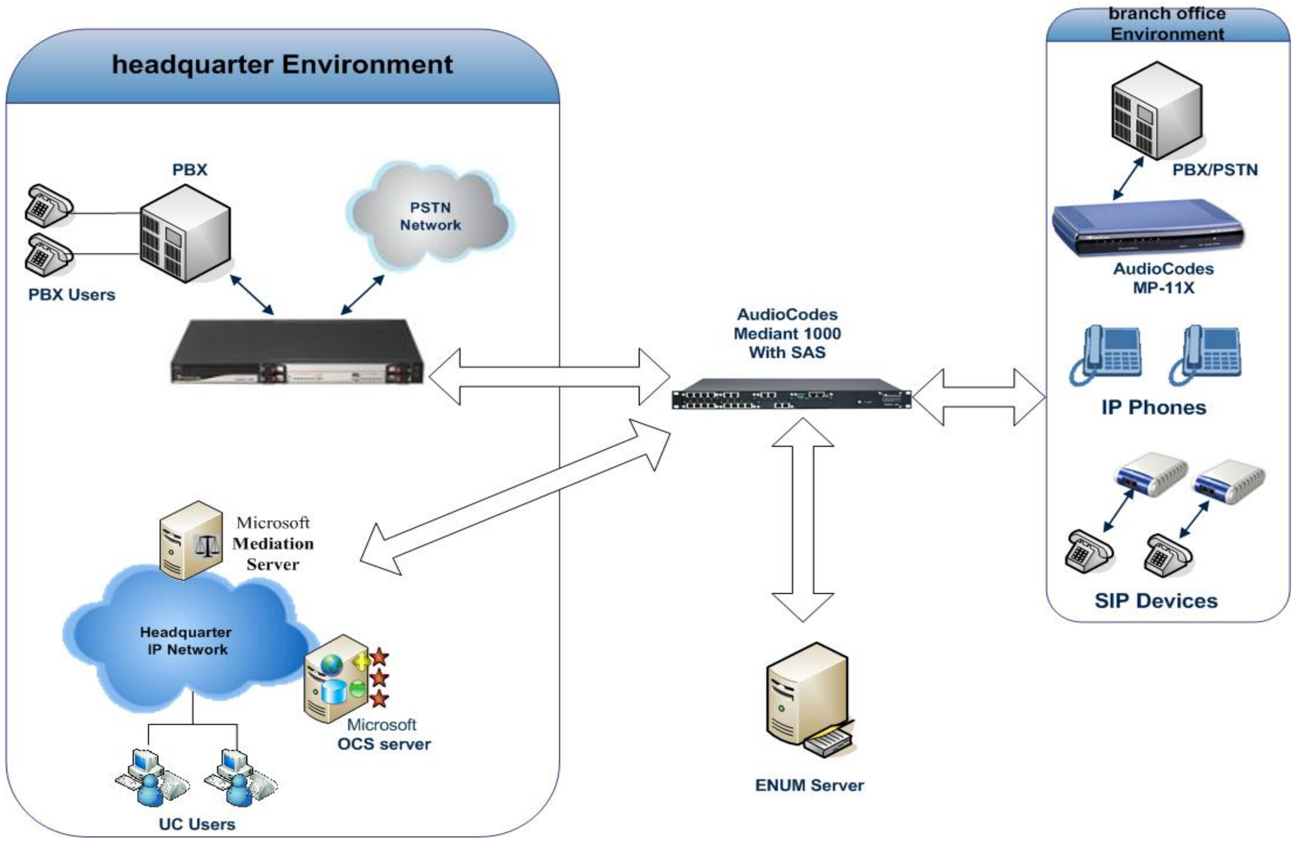
In addition, the SAS feature enables the gateway to make call routing decisions based on information stored on an E.164 Number Mapping (ENUM) server. Therefore, incoming calls from and to Microsoft's Office Communication Server 2007 (*OCS 2007*) can be routed according to user location (configured in the ENUM server). This enables using one common, popular database to manage and maintain information regarding user location, allowing a smooth users migration.

To better understand the advantages of implementing the SAS feature in an OCS 2007 environment, let's use an example scenario with the following assumptions:

- The enterprise has a deployed OCS 2007 at its headquarters and OCS clients connected to the OCS 2007 server.
- The enterprise has a deployed Mediant 2000 media gateway that is connected with several trunks to the legacy PBX (with users that have still not migrated to the OCS 2007 platform) and the PSTN,
- The enterprise has a remote branch office with several IP phones and an MP-11x FXO gateway connected to the remote PSTN.
- The enterprise wants the ability to receive and make calls between all their devices located at the headquarters, branch office and the PSTN.
- The enterprise wants the ability to smoothly migrate users from their legacy PBX's to the OCS 2007 platform and any-to-any connection between all users (OCS, PBX, IP phones, PSTN).

AudioCodes feature-rich SAS with ENUM server query ability offers the ideal solution for serving as a central routing decision maker for the environment.

Figure 1-1: Connecting SIP Devices to OCS 2007 Environment Using SAS AudioCodes Gateway



1.1 Theory of Operation

The theory of operation is based on the gateway's SAS feature operating in Emergency mode. For a detailed explanation on SAS, refer to the gateway's *User's Manual*.

Referring to the example scenario illustrated in Figure 1-1, the SAS theory of operation is as follows:

All SIP user devices such as IP phones update their location dynamically by performing registration to the Mediant 1000 gateway's SAS database. All other users such as PBX users or UC (OCS 2007) users will have a permanent location listed in the ENUM server.

The following calls are routed to the Mediant 1000 SAS device:

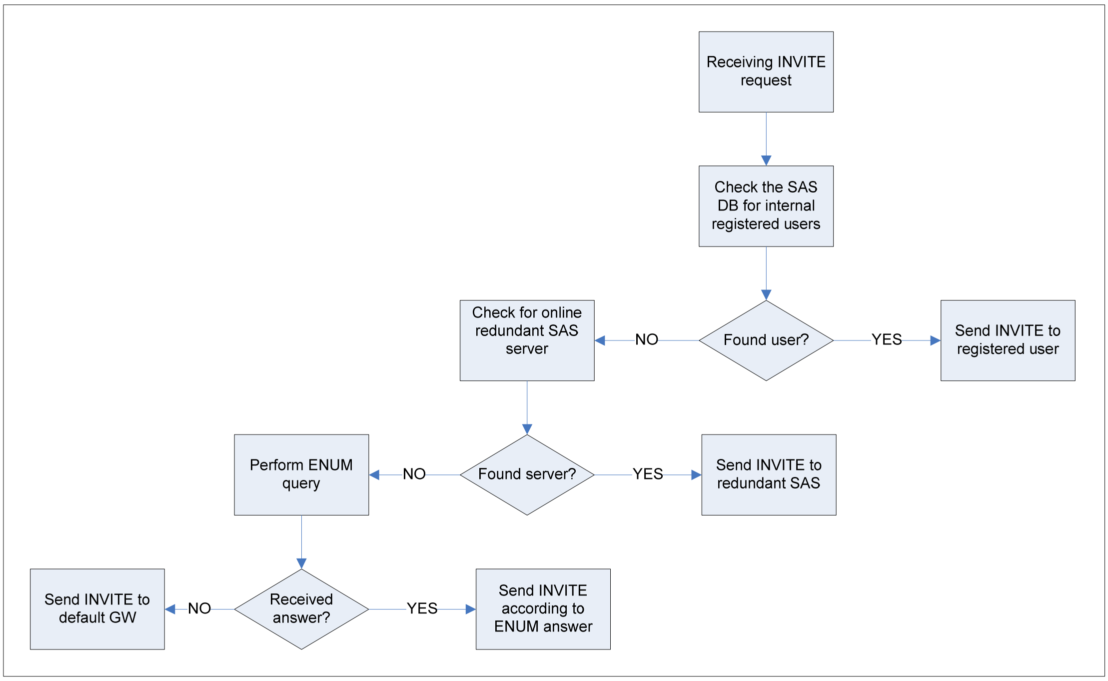
- Calls between two registered users (for example, two IP phones)
- Calls between registered users to OCS/PBX user and vice versa
- Outbound calls from/to PSTN to any enterprise user (IP phones, PBX, OCS 2007)

The Mediant 1000 SAS device acts as the Central Routing Decision Maker. The routing decisions are first based on existing information in its internal database (i.e., SAS database), and then, if not located, then using an external database (i.e., ENUM database).

Once an INVITE is received, the Mediant 1000 SAS database of registered users is searched for a matching user and destination (Address of Record). If not found, the Redundant SAS servers are searched. If there is still no match, an ENUM query is performed and the response is used to correctly route the INVITE. If no response is received from the ENUM server, the INVITE is routed to the default gateway.

The work flow of the Mediant 1000 SAS routing decisions is illustrated below:

Figure 1-2: SAS Work-Flow when Receiving INVITE Requests in Emergency Mode



1.2 Call Scenario Examples

For a better understanding of the gateway's SAS and ENUM capabilities, below are a few examples of call scenarios (illustrated in Figure 1-1):

- **UC (OCS 2007) user from headquarters want to call an IP phone user at a branch office:**

  Mediation Server is configured with the Mediant 1000 SAS entity as the next hop gateway. When the SAS receives the INVITE, it searches its dynamic database for a registered user. Once the IP phone user is located in the database, the SAS routes the call directly to the user IP phone.

- **UC (OCS 2007) user from the headquarters want to call a PBX user at headquarters:**

  Mediation Server routes the call to the Mediant 1000 SAS. Since this user is not registered in the SAS database, the SAS queries the ENUM server for the user's location. Upon the ENUM server reply, the call is routed to the Mediant 2000, which is connected to the PBX at the headquarters.

Note: The ENUM resolver class supplies the following variables (after parsing the ENUM answer):

- User part of the SIP request URI
- Host part of the SIP request URI
- Port
- Transport type (UDP,TCP or TLS)

- **UC (OCS 2007) user from headquarters wants to call a PSTN number:**

  Mediation Server routes the call to the Mediant 1000 SAS. Since this number is not registered in the SAS database and does not have an entry in the ENUM server, the call is routed to a user-defined default gateway IP address, which in our case is the Mediant 2000 connected to the PSTN.

- **A PBX user migrates from the PBX to the OCS 2007 environment:**

  The only change required is in the ENUM server entry where the host part of the SIP URI must be changed to be represent Mediation Server instead of Mediant 2000. Any call destined to this user is now routed to Mediation Server instead of Mediant 2000.

2 Gateway Configuration

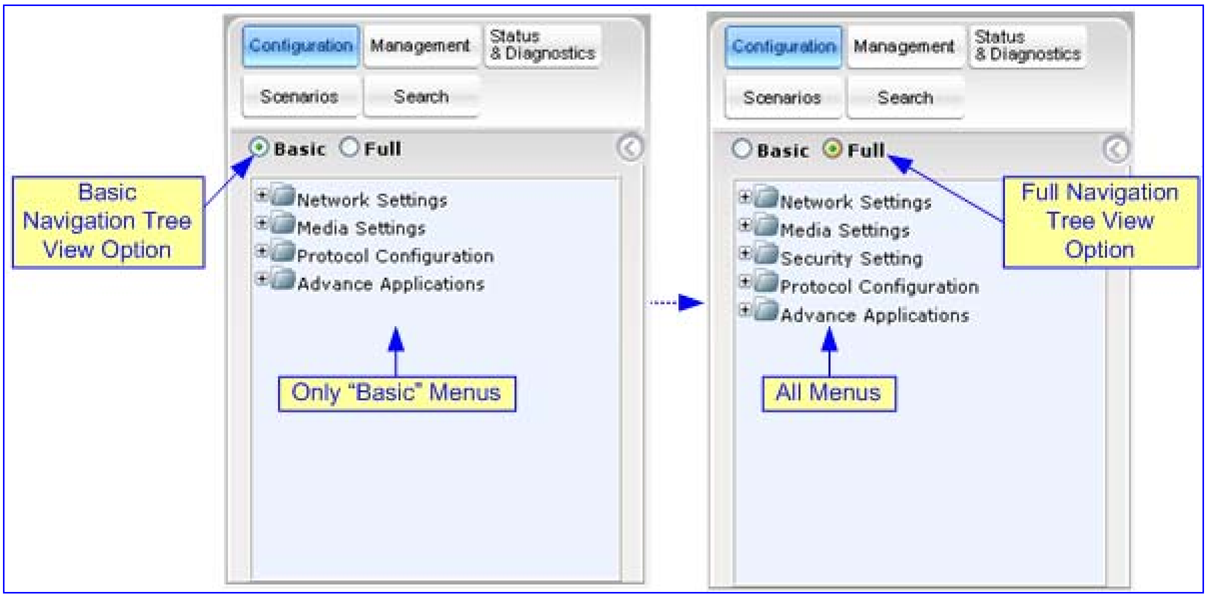
The procedures described in this section relate only to AudioCodes' gateway parameters that are relevant to the SAS and ENUM capability feature. As described previously, the same gateway can be used to connect between Mediation Server and the PBX/PSTN in the OCS 2007 environment.

To fully configure the gateway for interoperability with OCS 2007, refer to *LTRT-26004 Mediant 1000 & 2000 & MS OCS 2007 Quick Guide* for Mediant Gateway and *LTRT-26302 MP-11x FXO & MS OCS 2007 Quick Guide* for the MP-11x gateway.

Note: To enable the SAS capabilities on the AudioCodes gateway, the gateway must be loaded with the feature key that contains SASurvivability options, and the gateway must be running SIP version 5.4 or later.

The procedures described in this section are performed using the gateway's Web-based management tool (i.e., embedded Web server). Before you begin configuring the gateway, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as shown below:

Figure 2-1: Web Interface Showing Basic/Full Navigation Tree Display



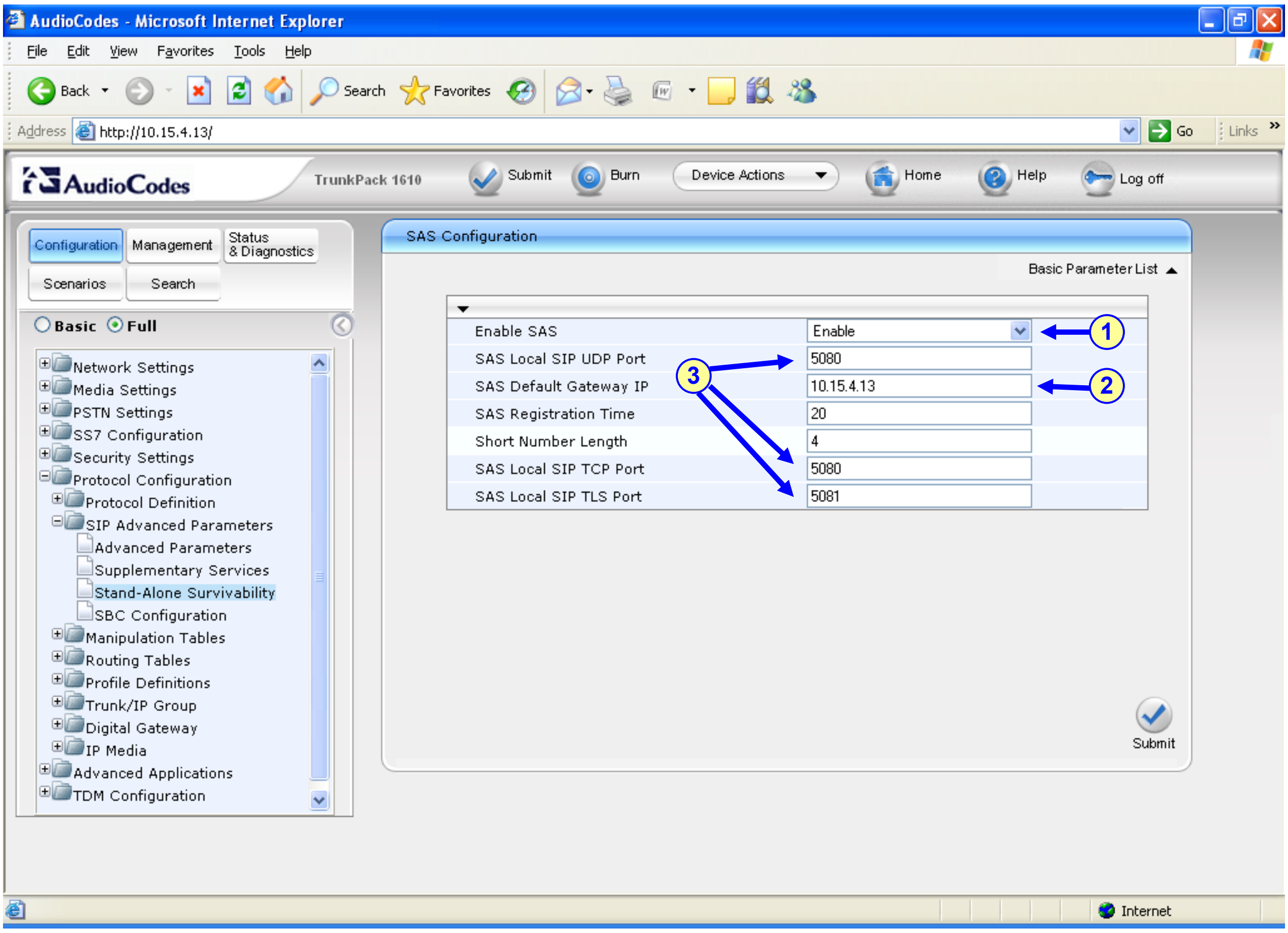
2.1 Step 1: Configure SAS Parameters

This step defines how to configure the gateway's SAS parameters.

➢ To configure SAS parameters:

1. Open the 'Stand-Alone survivability' page (**Protocol Configuration** menu > **SIP Advanced Parameters** submenu > **Stand-Alone Survivability**).

Figure 2-2: SAS Configuration Page



2. From the 'Enable SAS' drop-down list, select "Enable" to enable the Stand-Alone Survivability (SAS) feature.

3. In the 'SAS Default Gateway' field, enter the IP address of the SAS default gateway, which is used in SAS 'Emergency Mode'. When an incoming SIP INVITE is received and the destination Address-Of-Record (AOR) is not in the SAS database, the request is immediately sent to this default gateway. When using AudioCodes gateway for the SAS feature as well as to connect between Mediation Server and PBX/PSTN, the SAS default gateway IP address can be the gateway itself (i.e., same IP address as the gateway) to route external calls to PSTN.

4. The ‘SAS Local SIP Ports’ field represents the local ports for sending and receiving SIP messages for SAS. The SIP entities in the local network need to send the Invitation\Registration requests to this port. According to our example scenario (described in Section 1), you should configure Mediation Server and all other SIP devices in the enterprise to send SIP requests to this SAS local port, according to transport type (UDP, TCP, and TLS).

> **Note:** The SAS local ports must be different from the ports configured for the gateway’s local SIP ports.

5. All other parameters on this page can be left with their default value. For more details about the SAS parameters, refer to the gateway’s *User’s Manual*.

2.2 Step 2: Configure Additional SAS Parameters

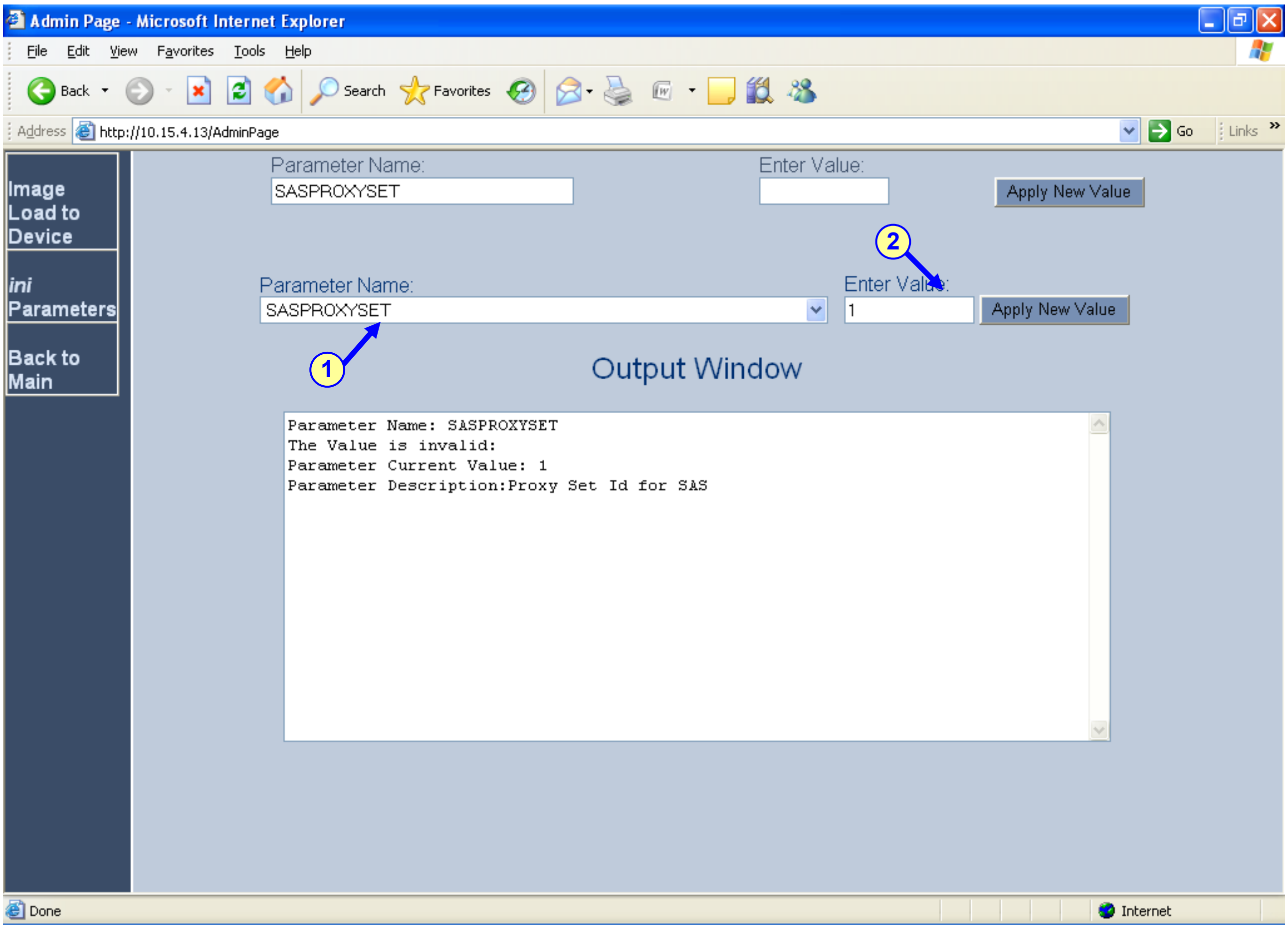
This step defines how to configure additional SAS parameters that must be configured in the gateway to enable the gateway to serve as a central routing decision maker for the entire environment. These parameters can be set using the *ini* file or the AdminPage as described below.

- **SASSurvivabilityMode:** determines the Survivability mode used by the SAS application. In our case study, it should be set to 1 (i.e. “Always Emergency”) so that the SAS application always operates in Emergency mode.
- **SASEnableENUM:** Determines whether the SAS application uses ENUM queries to route incoming INVITE requests when in Emergency mode. In our case study, it must be to 1 (Enable).

➢ To configure the parameters via the AdminPage:

1. Open the 'Admin" page, by appending the case-sensitive suffix ‘AdminPage’ to the gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.13/AdminPage).

Figure 2-3: Admin Page



2. In the Admin Page, on the left pane, click **ini Parameters**.
3. From the 'Parameter Name' drop-down list, select the appropriate parameter that you would like to configure
4. In the 'Enter Value' field, enter right value.
5. Click **Apply New Value**.

Reader's Notes

MP-11x, Mediant 1000 & Mediant 2000

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