

I. Introduction

This Digium Asterisk AA-50 Integration Guide provides general instructions for integration of the **VOIP-600 Series Phone** with a Digium Asterisk AA-50. It is recommended to read this instruction set completely before starting any installation. For detailed VOIP-600 Series Phone setup instructions, please consult the **VOIP-600 Series Phone Manual**. This integration guide has been specifically developed for a Digium Asterisk AA-50, however the following instructions may apply to other asterisk installations.

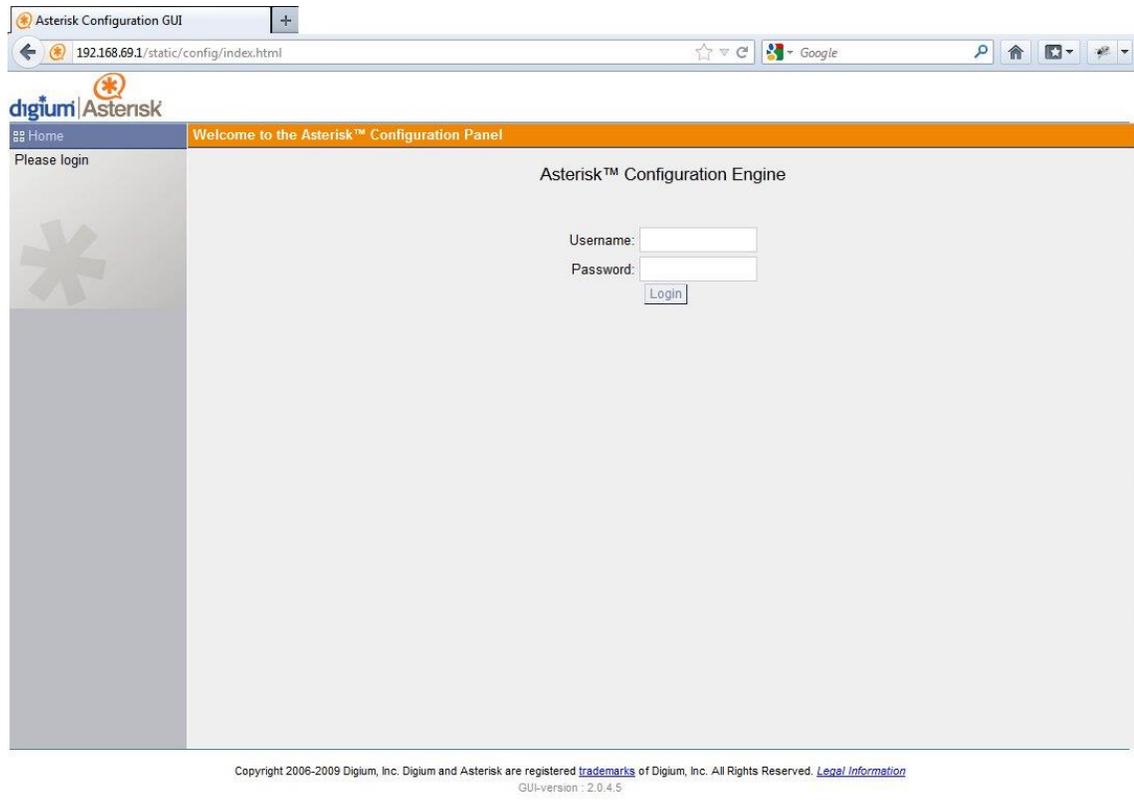
II. Prerequisites

- Digium Asterisk AA-50
- Network access to the AA-50, **VOIP-600 Series Phones** and all network services (SIP, TFTP, HTTP, FTP, DNS, RTP/SRTP)

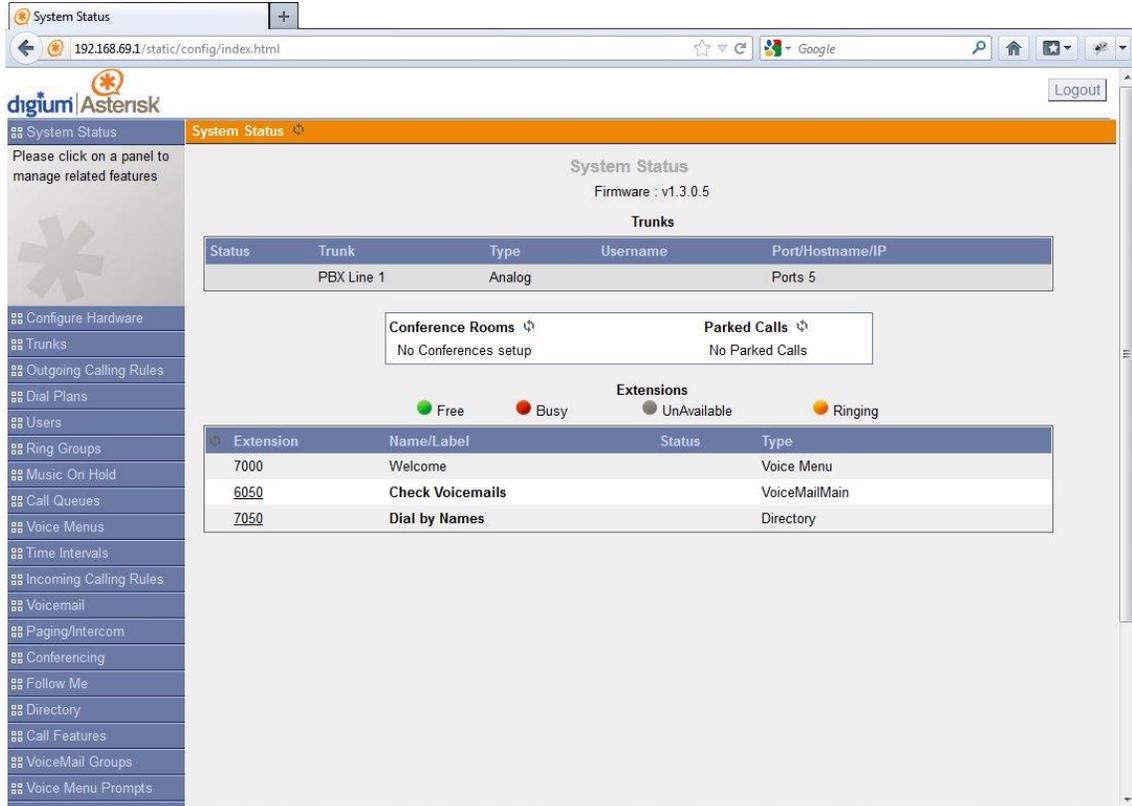
III. AA-50 Basic Configuration

Basic instructions for integrating a **VOIP-600 Series Phone** with a Digium Asterisk AA-50 are included. Advanced setup of AA-50 features is outside the scope of this document.

1. Using a web browser, enter the IP address (or FQDN if configured) of the AA-50 in the address bar:



2. Login to Digium Asterisk AA-50:



System Status
 Firmware : v1.3.0.5

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP
	PBX Line 1	Analog		Ports 5

Conference Rooms
 No Conferences setup

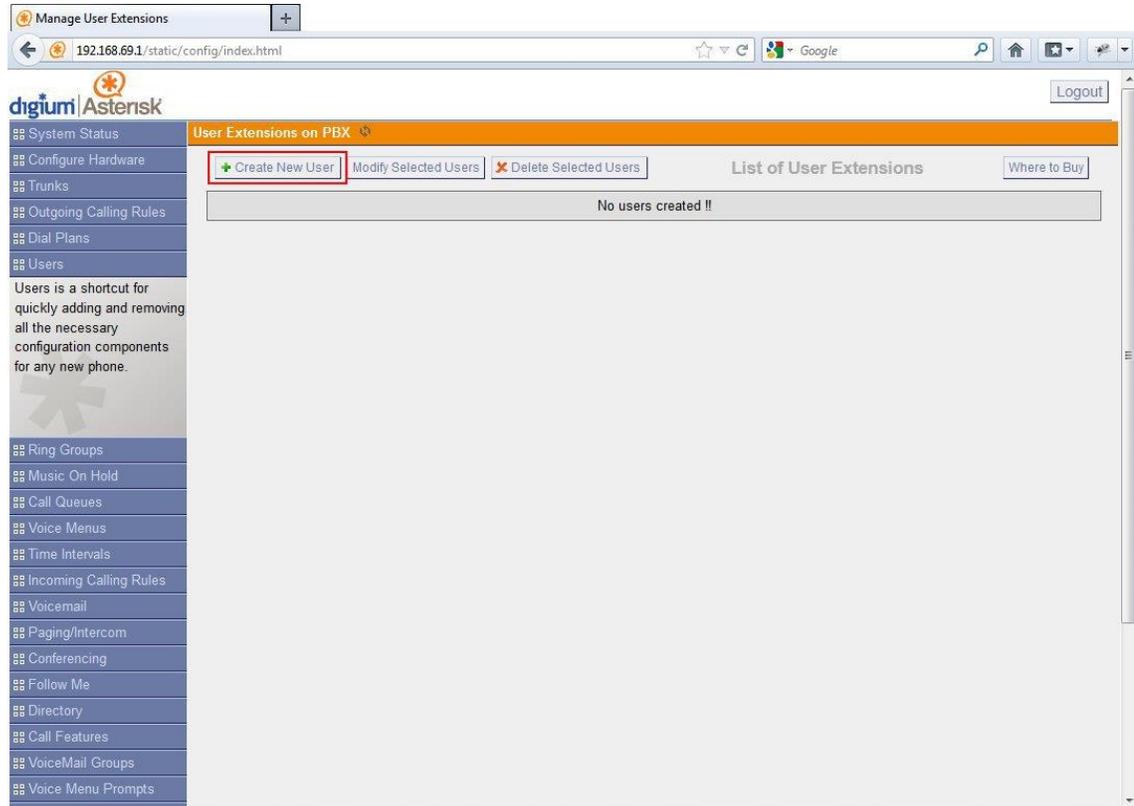
Parked Calls
 No Parked Calls

Extensions

Free Busy UnAvailable Ringing

Extension	Name/Label	Status	Type
7000	Welcome		Voice Menu
6050	Check Voicemails		VoiceMailMain
7050	Dial by Names		Directory

3. Each VOIP-600 Series Phone should have a unique User. In the Digium Asterisk AA-50 main menu, select **Users** then **Create New User**.



4. Enter the required fields to create a new User:

Extension: A unique Extension for each **VOIP-600 Series Phone**.

Name: A unique Name for each **VOIP-600 Series Phone**.

SIP: Ensure SIP (Session Initiation Protocol) is selected.

Codec Preference: Ensure u-law is selected.

MAC Address: Enter the MAC (Media Access Controller) address of the **VOIP-600 Series Phone**.

SIP/IAX Password: Enter a unique Password for the **VOIP-600 Series Phone**.

The screenshot shows the Asterisk web interface for managing user extensions. The 'Create New User' form is open, and several fields are highlighted with red boxes to indicate required information:

- General:** Extension: 6000, Name: VOIP-500-1, DialPlan: Default_DiaPlan.
- Technology:** SIP, IAX, Analog Station: None, flash: 750, rxfash: 1250, Codec Preference: First: u-law.
- VoIP Settings:** MAC Address: 001EEB0007D5, Line Number: 1, LineKeys: 1, SIP/IAX Password: 6000.
- Other Options:** 3-Way Calling, In Directory, Call Waiting, CTI, Is Agent, Pickup Group: 1.

5. If adding multiple **VOIP-600 Series Phones**, repeat Steps 3-4 for each device.

IV. VOIP-600 Series Phone Configuration

- Using a web browser, enter the IP address of the **VOIP-600 Series Phone** that you are programming. Login to the device with the configured Username and Password.
- In the VOIP-600 main menu, select **Network > SIP Settings**.
- Enter the following fields on the **SIP Settings** page. Then click **Apply**.
Assign a phone number:
Phone Number: Enter the Extension created in Step III-4

Specify domain name:

Domain Name: Enter the IP address of the Digium Asterisk AA-50

Enable/disable SIP registration:

Register: Checked

Specify SIP registrar:

Username: Enter the Extension create in Step III-4

Password: Enter the SIP/IAX Password created in Step III-4

IP Address: Enter the IP address of the Digium Asterisk AA-50

Port: (default: 5060)

Re-registration Time: (default: 3600)

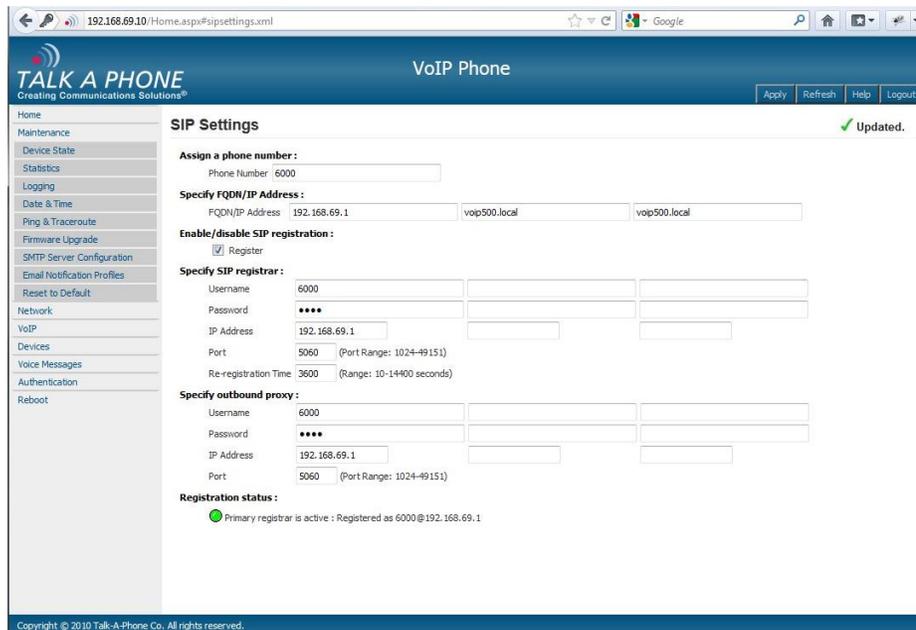
Specify outbound proxy:

Username: Enter the Extension create in Step III-4

Password: Enter the SIP/IAX Password created in Step III-4

IP Address: Enter the IP address of the Digium Asterisk AA-50

Port: (default: 5060)



The screenshot shows the 'SIP Settings' page of a VoIP phone. The interface includes a sidebar with navigation options like Home, Maintenance, Device State, Statistics, Logging, Date & Time, Ping & Traceroute, Firmware Upgrade, SMTP Server Configuration, Email Notification Profiles, Reset to Default, Network, VoIP, Devices, Voice Messages, Authentication, and Reboot. The main content area is titled 'SIP Settings' and contains several sections: 'Assign a phone number' with a text field containing '6000'; 'Specify FQDN/IP Address' with fields for FQDN/IP Address (192.168.69.1), domain (voip500.local), and port (voip500.local); 'Enable/disable SIP registration' with a checked 'Register' checkbox; 'Specify SIP registrar' with fields for Username (6000), Password (masked with dots), IP Address (192.168.69.1), Port (5060), and Re-registration Time (3600); 'Specify outbound proxy' with similar fields; and 'Registration status' showing a green dot and the text 'Primary registrar is active - Registered as 6000@192.168.69.1'. A 'Copyright © 2010 Talk-A-Phone Co. All rights reserved.' footer is at the bottom.

- Repeat steps 1-4 for any additional **VOIP-600 Series Phones**.