

Application Note

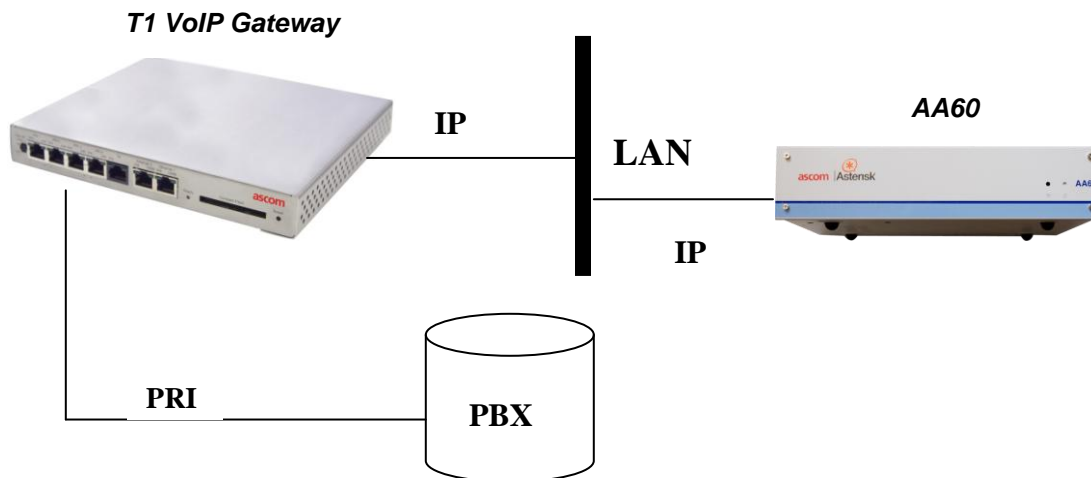


Product: Ascom AA60 Voice Appliance
Purpose: Configuration of Ascom AA60 as Conference Server with T1 VoIP Gateway
Date: January 28th, 2009

Introduction

This document outlines the steps for the configuration of the Ascom AA60 as a conference server with the Ascom T1 VoIP Gateway. This guide is intended for someone knowledgeable on the configuration of the Ascom T1 VoIP Gateway. The steps, screenshots, and guidelines depicted throughout this document are based upon Ascom VoIP Gateway software version 1.3.1 and AA60 software version C.2.1.1.

The Ascom T1 VoIP Gateway interfaces to the AA60 via TCP/IP using SIP trunking. A SIP “user” is created in the AA60 and the T1 VoIP Gateway registers to the AA60. A calling route is then created to the AA60 from the VoIP Gateway. In the example below, all dial strings starting with the digit 6 are routed to the AA60.



AA60 Initial Setup

The AA60 is initially configured to obtain an IP address using DHCP. Once booted, a connected monitor will display the AA60's IP Address on the main screen. Using Firefox, or similar browser, enter the IP address of the AA60 into the address bar (<http://XXX.XXX.XXX.XXX>) on another computer connected to the same LAN.

Ascom AA60 Network Configuration

The AA60 shall be configured with a static IP Address. Follow the below steps to change the IP Address to a static address.

1. With a compatible web browser, navigate to <http://AA60-IPADDRESS:8004> (where AA60-IPADDRESS is the IP Address of the AA60).

Application Note



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2. Sign in with Username *admin* and associated Password (i.e. password, changeme, etc.)
3. Click on the *Configuration* link and then the *Configure Networking* link.

4. Mouse over the eth0 box and click the *edit* link presented in the upper right corner. Set *Configure Manually* and enter the desired IP Address, Netmask, and Gateway. Click OK and then Apply.

Ascom AA60 Dial Plan Configuration

1. With a compatible web browser, navigate to <http://AA60-IPADDRESS> (where AA60-IPADDRESS is the IP Address of the AA60).
2. Login with Username *admin* and associated Password (i.e. changeme, password, etc.)
3. Navigate to the *Dial Plans* tab and click on *New DialPlan*. Provide an intuitive name to the Dial Plan and select only *conferences* in the "Include Local Contexts:" and press *Save*.

Application Note



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Edit DialPlan X

DialPlan Name:

Include Outgoing Calling Rules: Longdistance IAXTEL Local International 911 toward_igwp

Include Local Contexts: default parkedcalls conferences ringgroups voicemenus queues voicemailgroups directory

Ascom AA60 User Configuration

4. Navigate to the *Users* tab on the left and then click *Create New User*. Enter the following parameters and then press *Update*.
 - a. *Extension* – Value that the T1 VoIP Gateway will register towards
 - b. *Name* – Intuitive name for the T1 VoIP Gateway (i.e. IGWP)
 - c. *DialPlan* – Select the Dial Plan defined in the above section (i.e. DP_CONF)
 - d. *Enable SIP*
 - e. *Codec Preference* – Ensure that u-law is selected.
 - f. *SIP/IAX Password* – Secret that the T1 VoIP Gateway will use during registration (if applicable)
 - g. *DTMF Mode* – Select RFC2833
 - h. *Enable 3-Way Calling*
 - i. *Enable Call Waiting*

Application Note



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Edit User Extension - 6001

General :

Extension: 6001 Name: IGWP DialPlan: DP_CONF
CallerID: 6001 OutBound CallerID:

Enable Voicemail for this User
VoiceMail Access PIN code: Email Address:

Technology

SIP IAX Analog Station: None flash: rxfash:
Codec Preference : First : u-law Second : None Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : Line Number : 1 SIP/IAX Password: 6001
NAT: Can Reinvite: DTMF Mode: RFC2833 insecure: no

Other Options

3-Way Calling In Directory Call Waiting CTI Is Agent Pickup Group: 1

Cancel Update

Ascom AA60 Conference Configuration

5. Select the *Conferencing* tab on the left and create a new Conference Bridge Extension for each desired room.
 - a. *Extension* – Extension dialed to access conference room for common users
 - b. *Marked/Admin user Extension* – Extension dialed by admin or marked user to access conference room. This might be needed if conferences shall be closed or not started until the marked/admin user has exited / joined.
 - c. *Quiet Mode* – Enable to not play entry / exited tones.
 - d. *Close conference when last marked user exits* – Enable to close conference when a classified user(s) exits.
 - e. *Wait for marked user* – Enable to not begin conference until a classified user(s) joins.

Application Note



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Ascom T1 VoIP Gateway Configuration

Please note: The T1 VoIP Gateway must have the media gateway functionality enabled (licensed by IGW-LGW or IGW-LGK)

1. With a compatible web browser, navigate to the IP Address of the T1 VoIP Gateway.
2. Navigate to the *PBX* tab, select *Objects* and create a new *Gateway Object*. Fill in the following parameters and then press OK.
 - a. *Long Name* – Enter long name (i.e. AA60)
 - b. *Name* – Enter name (i.e. AA60)
 - c. *Number* – Enter the prefix of the conference rooms (6 means that all conference room extensions start with a 6). This number series shall be a series that does not need to be routed outbound toward the PBX or included in the Ascom handset number series.
 - d. *Password* – Enter the desired password for the SIP Gateway interface object.

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Long Name	<input type="text" value="AA60"/>		
Name	<input type="text" value="AA60"/>	Number	<input type="text" value="6"/>
URL	<input type="text"/>		
Hardware ID	<input type="text"/>		
Node	root ▼		
PBX	. ▼		
Local	<input type="checkbox"/>		
Send Number	<input type="text"/>		
Password	<input type="password" value="....."/>	retype Password	<input type="password" value="....."/>
Filter	normal ▼	Diversion-Filter	normal ▼
CFNR Timeout	<input type="text"/>		
Busy On	<input type="text"/>	Call(s)	
Group Indications	▼		
Gateway			
Enblock Count	<input type="text"/>		
Prefix	<input type="checkbox"/>		
OK Cancel Apply Delete			

Note: In this example, all conference rooms have the extensions 6X. Since there are no phones with extensions 6XXX in our dial plan, we can route any call starting with a 6 to AA60's conference rooms.

If the extension range 60XX-65XX were being used in the dial plan, the asterisk object number could be 66 instead. Each conference bridge extension would be 66X.

3. Navigate to the *Gateway* tab, select *Interfaces* and select the *SIP1* interface.

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VoIP Gateway

Configuration

- General
- IP
- ETH0
- ETH1
- LDAP
- PRI1
- PRI2
- PRI3
- PRI4
- TEL
- Administration
- PBX
- Gateway
- Download
- Upload
- Diagnostics
- Reset

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registration
PRI1	→	→	→	→	Up	EXTERN	→ 127.0.0.1
PRI2	+						
PRI3	+						
PRI4	+						
TEL	+						
TEST	+						
TONE	+						
HTTP	+						
ECHO	+						
CONF	+						
SIP1 AA60	+				172.20.106.106	AA60	→ 127.0.0.1
SIP2	+						
SIP3	+						
SIP4	+						

4. Fill in the following parameter and then press OK.
 - a. *Name* – Intuitive Name for this interface (i.e. AA60)
 - b. *Server Address (Primary)* – IP Address of the AA60
 - c. *ID* – SIP User extension defined for the T1 VoIP Gateway in the AA60
 - d. *@* - IP Address of the AA60
 - e. *Username* – SIP User extension defined for the T1 VoIP Gateway in the AA60
 - f. *Password* – SIP User extension password as defined for the T1 VoIP Gateway in the AA60
 - g. *Registration* – Select SIP
 - h. *Registration:Server Address* – Enter local loopback address (127.0.0.1)
 - i. *ID* – Enter the value of the Name field of the Gateway object previously created (i.e. AA60)
 - j. *Registration:@* - Enter local loopback address (127.0.0.1)
 - k. *Registration:Username* – Enter the value of the Long Name field of the Gateway object created (i.e. AA60)
 - l. *Registration:Password* – Enter the value of the password field of the Gateway object created

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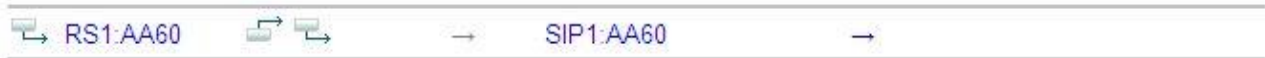
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Name	AA60	
Disable	<input type="checkbox"/>	
Server Address	172.20.106.106	(primary)
Server Address		(secondary)
ID	6001	@ 172.20.106.106
STUN Server		
Authorization		
Username	6001	
Password	●●●●●●	Retype ●●●●●●
SIP Interop Tweaks		
Proposed Registration Interval [s]		
Accept INVITE's from Anywhere	<input type="checkbox"/>	
From Header when Sending INVITE	Fixed AOR	
Identity Header when Sending INVITE	CGPN in user part of URI	
Registration	SIP	
Registration		
Server Address	127.0.0.1	(primary)
Server Address		(secondary)
ID	AA60	@ 127.0.0.1
STUN Server		
Username	AA60a	
Password	●●●●●●	Retype ●●●●●●
OK Cancel		

When successful, the T1 VoIP Gateway SIP interface shall be registered to the AA60 and to the local Gateway object.



5. Navigate to the *Gateway* tab and select *Routes*. Delete the route from the AA60 to the T1 VoIP Gateway (SIP1 to RS1). The only SIP1 route shall be RS1 to SIP1.



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Additional Information

If you have any questions or need additional information, please contact Ascom Technical Assistance Center at 1-877-71-ASCOM, option 3.