Genesys Application Note

Grandstream GXP SIP Phones With Genesys SIP Server

Version 1.0

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1 Summary

This document details the configuration to successfully integrate Genesys SIP Server with Grandstream GXP1105, 1165, 1405, 1450, 2100, and 2124 SIP phones. As these phones use the same firmware, a full test run was executed only for the GXP1450 phones, followed by a limited selection on the other phones.

Genesys Media Server was used to handle all media interactions such as, call treatment announcements, collect digits and music on hold.

2 SIP End Point Features

2.1 Feature Chart

Feature Name	
General features (1PCC)	**Supported**
Agent Login from the Phone	No
Auto-Answer	Yes
Alternate Ringtones	No
Caller ID	Yes
Call Forward	Yes
Do Not Disturb	Not Tested
DNS-based redundancy (SIP Proxy, SIP Cluster)	Not Tested
DTMF tones generation	Yes
IPv6 support	Yes
Multiple calls on one extension	Yes
Message Waiting Indicator	No
SIP authentication	Yes
TLS/SRTP	Yes
Call Control with 1PCC	**Supported**
Basic calling	Yes
Conference	Yes
Hold/Retrieve	Yes
Unattended transfer (Genesys Single-Step Transfer)	Yes
Semi-attended transfer (Genesys Blind Transfer)	Yes
Attended transfer (Genesys Two-Step Transfer)	Yes
Call Control with 3PCC	**Supported**
Answer Incoming Call	Yes
Conference	Yes
Hold/Retrieve	Yes
Make Outgoing Call	Yes
Remote Auto-Answer	No
Unattended transfer (Genesys Single-Step Transfer)	Yes
Semi-attended transfer (Genesys Blind Transfer)	Yes
Attended transfer (Genesys Two-Step Transfer)	Yes
Video Support	**Supported**
Basic Video Calls	No
Push Video	No
Video Call on Hold / Retrieve	No
Video Call Transfer	No
Video Conference	No
Support of Genesys Solutions	**Supported**
Genesys Business Continuity	Yes
Genesys Voice Mail Solution Support	Not Tested

* See section 3.3 for known limitations

2.2 Feature Chart Glossary

2.2.1 General features

1pcc: First-Party Call Control is a method to handle calls using the phone keypad.

3pcc: Third-Party Call Control is a method to handle calls using T-Library desktop connected to SIP Server.

Agent Login from the Phone: Agent sets login/logout from the phone. Agent state ready/not ready can be set from the phone or it can be pushed from the server to the phone after agent logs in from the phone. Functionality is supported based on RFC3863 using presence states open/closed.

Alternate Ringtones: Phone provides distinctive ringtones requested by SIP Server. Functionality is supported based on RFC3261 using the Alert-Info header.

Auto-Answer: Phone can be configured to answer all calls automatically.

Caller ID: Phone is able to display the number and name of the calling party.

Call Forward: Phone can forward calls unconditionally or based on internal state (e.g. 'busy').

Do Not Disturb: Phone can reject all incoming calls.

DNS-based redundancy: Phone can toggle between SIP Servers provisioned by single FQDN if current SIP Server becomes unavailable. This functionality is required to deploy a phone with Genesys SIP Proxy. It also may be used for Genesys Business Continuity.

DTMF tones generation: Phone can pass DTMF tones in-band (RFC2833, RFC4733) or using SIP INFO messages.

IPv6 support: Phone can support the IPv6 protocol.

Message Waiting Indicator: SIP MWI support (RFC3842).

Multiple calls on one line: Phone can process multiple incoming/outgoing calls simultaneously on the same line.

SIP authentication: Phone can authenticate with SIP Server using the HTTP Digest algorithm (RFC3261 and RFC2617).

TLS/SRTP: Phone supports secure SIP environment that uses TLS and SRTP.

2.2.2 Call Control with 1PCC

Basic calling: Incoming and outgoing calls.

Conference: Phone can bridge two or more calls without using MCU.

Hold/Retrieve: Phone can put a call on hold and then to retrieve it.

Transfer:

- **Unattended transfer**: Call transfer using REFER.
- **Semi-attended transfer**: Completing the transfer when one party is on hold and the other party is ringing, using REFER with Replaces.
- **Attended transfer**: Completing the transfer using REFER with Replaces when one party is on hold and the other party has answered the call.

2.2.3 Call Control with 3PCC

Answer Incoming Call: Phone can answer the call using Broadsoft extension 'talk' passed in SIP NOTIFY.

Conference: Phone supports server side single-step or two-step conference.

Hold/Retrieve: Phone can put a call on hold and retrieve it using Broadsoft extension 'hold' and 'talk' passed in SIP NOTIFY.

Make Outgoing Call: Phone can make outgoing call initiated by SIP Server through the Genesys T-Library interface.

Remote Auto-Answer: Phone can answer a call automatically based on Auto-Answer (RFC5373) or Alert-Info headers.

Transfer:

- **Unattended transfer (Genesys Single-Step Transfer)**: Phone supports unattended transfer initiated by SIP Server using REFER or re-INVITE.
- **Semi-attended transfer (Genesys Blind Transfer)**: Phone supports completion of two-step transfer initiated by SIP Server when one party is on hold and the other party is ringing.
- **Attended transfer (Genesys Two-Step Transfer)**: Phone supports completion of two-step transfer initiated by SIP Server when one party is on hold and the other party has answered the call.

2.2.4 Video Support

Basic Video Calls: Incoming and outgoing video calls.

Push Video: Agent can show a video clip to the customer.

Video Hold/Treatment: Playing video file when call is put on hold or treatment is applied from routing strategy.

Video Call Transfer: Transferring video calls

Video Conference: Video Conference with active speaker detection using Genesys Media Server

2.2.5 Support of Genesys Solutions

Genesys Business Continuity: Phone is certified to be used in Genesys Business Continuity environment in one of two modes. It either can switch between the two geo-redundant sites or it can stay connected to both of them at the same time.

Genesys Voice Mail Solution: Phone is certified to be used with the Genesys Voice Mail solution. Optional advanced features support group Voice Mail Boxes, enable multiple Voice Mail Boxes to be configured for one line, and provide easy access to all configured Voice Mail Boxes.

3 Software and Hardware Versions

The following Genesys components and Grandstream phones were validated for reference configuration examples.

3.1 Genesys Components

Genesys Components		
Component	**Version**	**Release Type**
SIP Server	8.1.100.64	GA
Media Server	8.1.603.45	GA
Universal Routing Server	8.1.300.16	GA
Stat Server	8.1.000.32	HF

3.2 Non Genesys Components

3rd Party Hardware Components		
Component	**Version**	**Release Type**
Grandstream GXP 1105	1.0.5.23	Unknown
Grandstream GXP 1165	1.0.5.24	Unknown
Grandstream GXP 1405	1.0.5.15	Unknown
Grandstream GXP 1450	1.0.5.15	Unknown
Grandstream GXP 2100	1.0.5.23	Unknown
Grandstream GXP 2124	1.0.5.23	Unknown

For a full listing of 3rd party hardware/software supported by Genesys, see the Genesys Supported Media Interface Guide (SMI).

3.3 Known Issues and Limitations

3.3.1 Issues and Limitations Identified with Genesys Products

Genesys Product Issues		
Description	**Product**	**Version**
None found		

3.3.2 Issues and Limitations Identified with Third Party Products

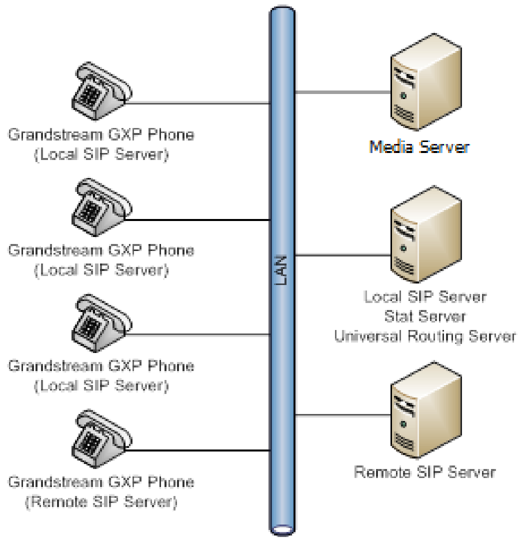
3rd Party Product Issues		
Description	**Product**	**Version**
In a scenario where a GXP phones have a primary call on hold and an established consultation call, and if the primary call is released, both the primary call and consultation call are torn down.	Grandstream GXP Phone	1.0.5.xx

4 Integration and Configuration Section

This section describes the various components involved with integrating the Grandstream GXP Phones and Genesys SIP Server for general interoperability.

The "Integration Points" section describes at a high level the functionality of each of the components involved in the solution.

The configuration sections cover the configuration of the respective Genesys and non-Genesys components.



4.1 Integration Points

This section details each component and the role it plays within the solution.

4.1.1 Grandstream GXP SIP Phones

Grandstream GXP Phones are next generation enterprise grade IP phones.

4.1.2 Genesys SIP Server

Genesys SIP Server is the interface between the Grandstream GXP SIP phones and the Genesys software components. All messaging and call control passes through SIP Server via a mix of SIP and Genesys T-Library signaling/messaging.

4.1.3 Genesys Media Server

It is used to handle media interactions such as, Call Progress Detection for outbound campaigns, call treatments (Play application, collect digits etc.), and music/video on hold.

4.2 Genesys Configuration Section

This section covers the configuration of DN objects in Genesys Configuration Server and phone configuration screenshots.

Please note that basic configuration is not covered (i.e. how to create a switch, how to associate an application with a switch, and so on). Only the most important configuration will be dealt with. It is assumed that the reader has Genesys knowledge and is familiar with deploying a basic Genesys environment. If this is not the case, please refer to the respective Genesys documentation as listed in the 'References' section.

4.2.1 SIP Server Switch

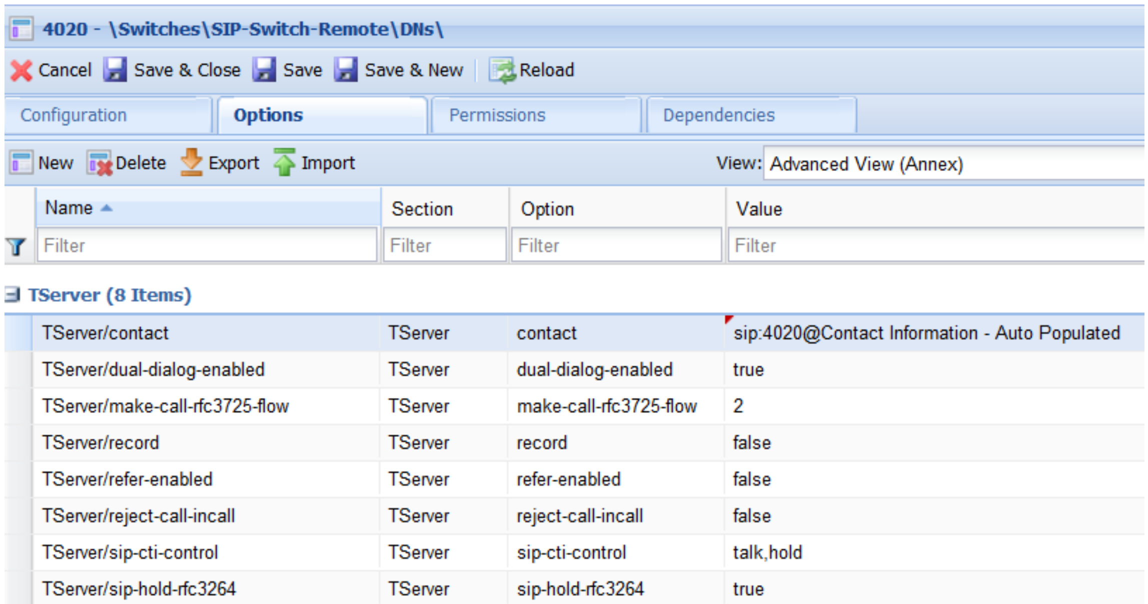
In this section, the SIP Server Switch resources are configured. It is assumed that:

- A SIP Server switching office exists, and is associated with a SIP Server Switch.
- For the purposes of call routing (using URS), the SIP extensions (DNs) are associated with a Place, which in turn is assigned to Place Groups.

Step	Description
1	In the navigation bar, select **Switches**. On the SIP Server Switch/[DNs] tab, create the necessary Extensions, Communication DN, Routing Points, Trunks, Trunk Groups, and Voice over IP Service DNs. 

2

Extension DNs are defined for SIP endpoints (phones) that register with SIP Server. Make sure that the following options are set in the <Extension DN>/[Options] tab/TServer section (extension 05511500100 is used as an example):



Name	Section	Option	Value
TServer/contact	TServer	contact	sip:4020@Contact Information - Auto Populated
TServer/dual-dialog-enabled	TServer	dual-dialog-enabled	true
TServer/make-call-rfc3725-flow	TServer	make-call-rfc3725-flow	2
TServer/record	TServer	record	false
TServer/refer-enabled	TServer	refer-enabled	false
TServer/reject-call-incall	TServer	reject-call-incall	false
TServer/sip-cti-control	TServer	sip-cti-control	talk,hold
TServer/sip-hold-rfc3264	TServer	sip-hold-rfc3264	true

contact

Contains the contact URI, specifying the device's IP address. It is aautomatically populated after a device successfully registers with SIP Server.

dual-dialog-enabled

Set the option to true for endpoints that accept more than one SIP dialog and provide remote CTI control by the NOTIFY message. Set the option to false for endpoints that can only accept one active SIP dialog, or cannot provide remote CTI control by the NOTIFY message.

make-call-rfc3725-flow

Controls which SIP call flow to choose when a call is initiated by a TmakeCall request. The specified value is equal to the call flow number as described in RFC 3725. Only flow 1 and flow 2 are currently supported.

sip-cti-control = talk,hold

Specifies the behaviour of a DN that represents either of the following types of SIP endpoints:

- SIP endpoint built on the Genesys SIP Endpoint SDK 8.0, using proprietary SIP extensions. For this endpoint, you can configure this option with the values both beep and dtmf. For more information, see; see the *SIP Endpoint SDK 8.0 API Reference*.
- SIP endpoint which supports the BroadSoft SIP Extension Event Package. For this endpoint, you can configure this option with the values talk and hold.

sip-hold-rfc3264

Specifies which implementation of hold media SDP is used by SIP Server for hold operations.

- true = RFC3264-compliant implementation.
- false = RFC2543-compliant implementation

authenticate-requests

Determines if incoming SIP requests (REGISTER or INVITE) are treated with an authentication procedure.

password

Specifies the password for the SIP endpoint registration with the local registrar. If it is present, registration attempts are challenged and the password is verified. If it is not present, the registration is not challenged. The authentication procedure can also be applied to INVITE requests.

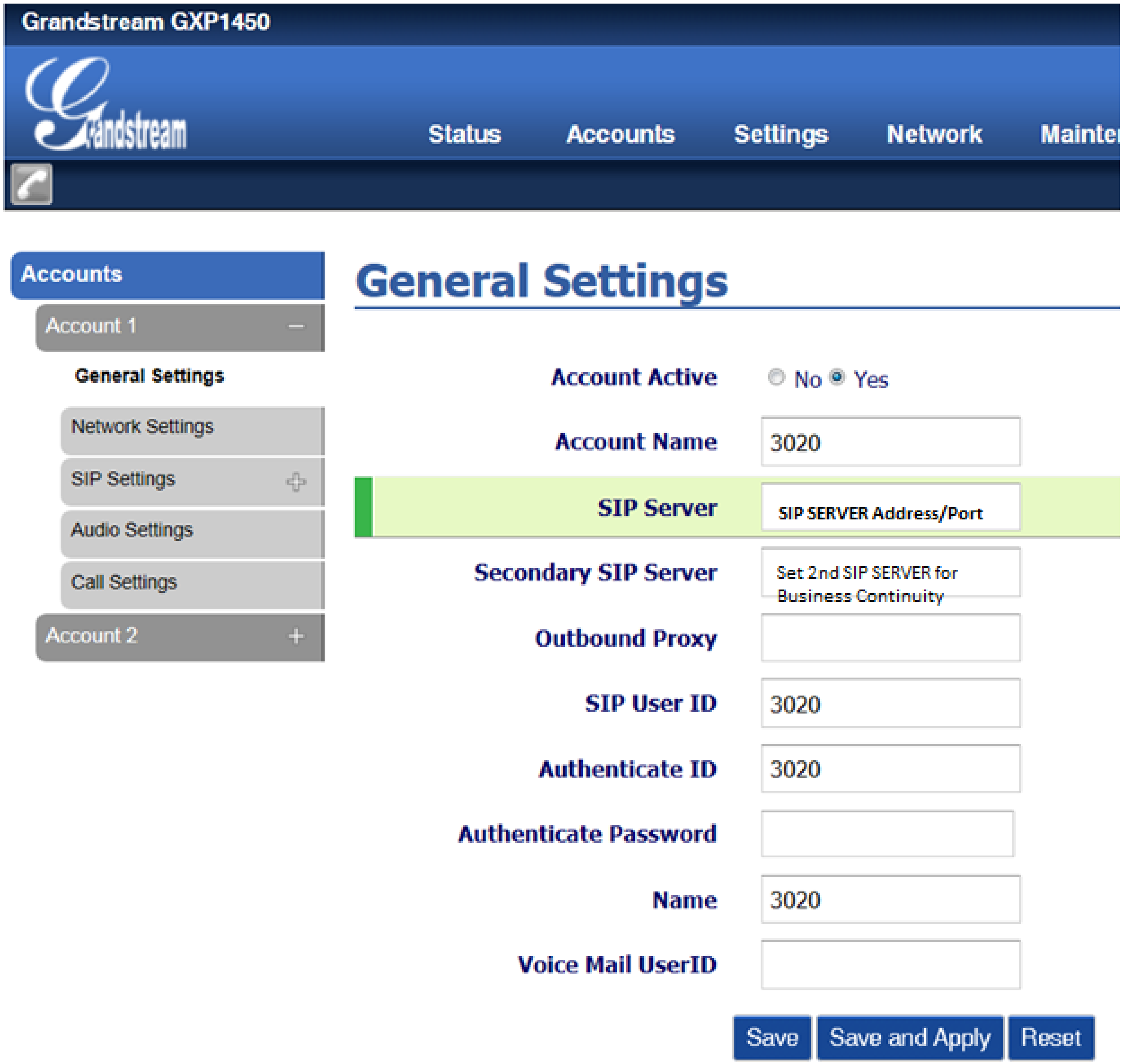
4.2.2 SIP Server Application

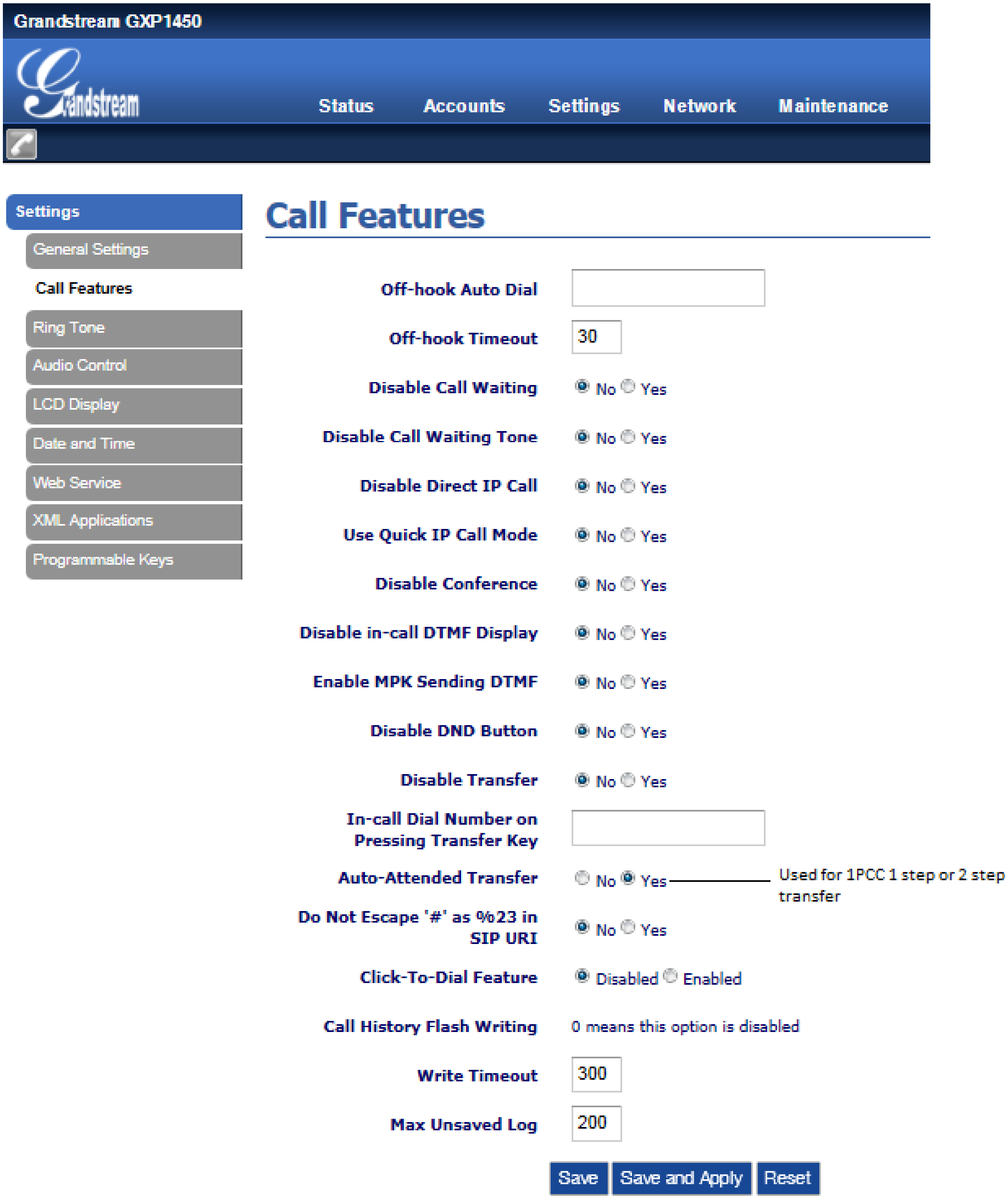
In this section, the SIP Server application is configured. It is assumed that SIP Server is:

- Installed and configured as per the Genesys SIP Server Deployment Guide.
- Associated with the SIP switch configured in section 4.2.1.
- Optional connection to Message Server (to ensure that component log information reaches the Log database and can be viewed in the Solution Control Interface).
- SIP Server must have Full Control permission for the DN objects to update a configuration object. By default, it does NOT have this permission. You need to grant **Full Control permission** for the **System account** for the all DNs on the corresponding switch.

Step	Description
1	In the navigation bar, select the SIP Server application and navigate to the [Options] tab/TServer section. Make sure the following options are set: **internal-registrar-domains = <SIP Server IP address>** **internal-registrar-enabled = true** SIP Server's internal registrar is enabled. **internal-registrar-persistent = true** Enables SIP Server to update the DN attribute contact in the configuration database. When an endpoint registers, SIP Server takes the contact information from the REGISTER request and updates or creates a key called contact in the Annex tab of the corresponding DN. **msml-support = true** SIP Server engages the Media Server via the MSML service for all media services operations such as treatments, music, video, greetings and conferences. **sip-port = <any valid and unique TCP/IP port>** SIP Server listening port for incoming SIP requests. The same port number is used for both TCP and UDP transports. **sip-enable-moh = <true or false>** SIP Set this option to true to enable music-on-hold for any party engaged with this device in the call. **sip-address = <IP address of the SIP Server>** Specifies an IP address of the SIP Server interface. This option must be set when deploying SIP Server on a host with multiple network interfaces.

4.3 Grandstream GXP Phone Configuration Section




Grandstream GXP1450
Status
Accounts
Settings
Network
Maintenance
Settings
General Settings
Call Features
Ring Tone
Audio Control
LCD Display
Date and Time
Web Service
XML Applications
Programmable Keys
Call Features
Off-hook Auto Dial
Off-hook Timeout 30
Disable Call Waiting No Yes
Disable Call Waiting Tone No Yes
Disable Direct IP Call No Yes
Use Quick IP Call Mode No Yes
Disable Conference No Yes
Disable in-call DTMF Display No Yes
Enable MPK Sending DTMF No Yes
Disable DND Button No Yes
Disable Transfer No Yes
In-call Dial Number on Pressing Transfer Key
Auto-Attended Transfer No Yes
Used for 1PCC 1 step or 2 step transfer
Do Not Escape '#' as %23 in SIP URI No Yes
Click-To-Dial Feature Disabled Enabled
Call History Flash Writing 0 means this option is disabled
Write Timeout 300
Max Unsaved Log 200
Save
Save and Apply
Reset

5 SIP Endpoint - Vendor Information.

Grandstream Networks, Inc. (http://www.grandstream.com) is an award-winning designer and ISO 9001 certified manufacturer of next generation IP voice & video products for broadband networks. Grandstream's products deliver superb sound and picture quality, rich telephony features, full compliance with industry standards, and broad interoperability with most service providers and 3rd party SIP-based VOIP products. Grandstream is consistently recognized in the VoIP industry for their innovation, affordability and superior value in their products.

5.1 Comparision Table

							
Model	GXP110x	GXP116x	GXP140x	GXP1450	GXP2130	GXP2140	GXP2160
LCD Screen (pixel)	No	128 x 40		180 x 60	320 x 240 (TFT color LCD)	480 x 272 (TFT color LCD)	
Lines (SIP Account)	1		2		3	4	6
Programmable Keys	4	3 (XML)			4 (XML)	5 (XML)	
Speed Dial/BLF	No				8	No	24
Voicemail LED	Yes, Access Key to Voicemail Box						
HD Audio	Handset with HD Audio, supports codec G.722 (wideband)	No	Handset with HD Audio, supports codec G.722 (wideband)				
Speaker	No	Yes	Yes, with AEC	Yes, G.722 (Wideband) HD Audio, Full Duplex with Advanced Echo Cancellation (AEC)			
Headset Jack	No	RJ9 supporting EHS with Plantronics Headset	RJ9	2.5mm / RJ9	RJ9 supporting EHS with Plantronics Headset		
Backlit LCD	N/A	No		Yes			
Extension Module	No					Yes, up to 4 Modules	No
Provisioning	HTTP, HTTPS, TFTP, TR-069, XML						
Network (10/100Mbps) Ports	1 RJ45	Switch mode 2 x RJ45			Switch mode 2 x RJ45 (10/100/1000Mbps)		
Integrated PoE	Yes-GXP1105 No-GXP1100	Yes-GXP1165 No-GXP1160	Yes-GXP1405 No-GXP1400	Yes			
Security	SIP/TLS, SRTP, AES-256, 802.1x						
Operating System	Linux Based						
Phonebook	No	500 contacts			2,000 contacts		
Conference	3				4	5	
Bluetooth	No					Yes	
USB, SD	No					USB (No SD)	

6 Appendices

6.1 References

References		
Document Name	**Version**	**Date Published**
SIP Server Deployment Guide	81fr_dep-sip_05-2013_v8.1.101.05	2013-05
Genesys Media Server Deployment Guide	81gvp_dep-gms_12-2012_v8.1.601.00	2012-12

6.2 Glossary & Acronyms

Glossary & Acronyms	
Term	**Definition**
ACD	Automatic Call Distribution Queueing Device
CTI	Computer Telephony Integration
DNIS	Dialed Number Identification Service
DTMF	Dual Tone Multie Frequency
ExtDN	Customer DN external to the contact center
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
IRD	Genesys Interaction Routing Designer Application
ISCC	Genesys Inter Server Call Control Functionality
ISDN	Integrated Services Digital Network
LAN	Local Area Network
MCP	Genesys Media Control Platform
MGW	Media Gateway
PSTN	Public System Telephone Network
RM	Genesys Resource Manager
RP	Genesys Routing Point Device
RTP	Real-Time Transport Protocol
SBC	Session Border Controller
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SMI	Genesys Supported Media Interface Guide