Genesys Application Note

Grandstream GXP1625/GCC1700 SIP Phones With Genesys SIP Server

Version 1.0

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Each product has its own documentation for online viewing at the Genesys Technical Support website or on the Documentation Library DVD, which is available from Genesys upon request. For more information, contact your sales representative.

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1 Summary

This document details the configuration to successfully integrate Genesys SIP Server with Grandstream GXP1625 and GCC1700 SIP phones.

The supported phone firmware is 1.0.2.27 and above.

Testing was carried out using GXP1625 SIP phones, however the GCC1700 is also certified as both phones use the same SIP/Feature Stack.

2 SIP End Point Features

2.1 Feature Chart

Feature Name	
General features (1PCC)	**Supported**
Agent Login from the Phone	Yes
Auto-Answer	Yes
Alternate Ringtones	No
Caller ID	Yes
Call Forward	Yes
Do Not Disturb	Yes
DNS-based redundancy (SIP Proxy, SIP Cluster)	Yes
DTMF tones generation	Yes
IPv6 support	Yes
Multiple calls on one extension	Yes
Message Waiting Indicator	Not Tested
SIP authentication	Yes
TLS/SRTP	Yes
Call Control with 1PCC	**Supported**
Basic calling	Yes
Conference	Yes
Hold / Retrieve	Yes
Transfer	
1-step	Yes
2-step semi-attended	Yes
2-step consultation	Yes
Call Control with 3PCC	**Supported**
Answer Incoming Call	Yes
Conference	Yes
Hold/Retrieve	Yes
Make Outgoing Call	Yes
Remote Auto-Answer	No
Transfer	
1-step	Yes
2-step semi-attended	Yes
2-step consultation	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 phone keypad.

3pcc: Third Party Call Control is a method to handle calls using T-Library desktop connected to the 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. This functionality is based on subscription packages described in the *SIP Access Side Extensions Interface* document by BroadSoft.

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

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

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

Call Forward: Phone can forward the 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 in case if current SIP Server becomes unavailable. This functionality is required to deploy a phone with Genesys SIP Proxy and SIP Cluster. 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 message.

IPv6 support: Phone can support 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 HTTP Digest algorithm (RFC3261 and RFC2617).

TLS/SRTP: Phone supports SIP secure environment using 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:

- **1-step**: Call transfer using REFER.
- **2-step semi-attended**: Completing the transfer when one party is on hold and the other party is ringing using REFER with Replaces.
- **2-step consultation**: 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:

- **1-step**: Phone supports single-step transfer initiated by SIP Server using REFER or re-INVITE.
- **2-step semi-attended**: Phone supports completion of two-step transfer initiated by SIP Server when one party is on hold and the other party is ringing.
- **2-step consultation**: 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 Support: Phone is certified with Genesys Voice Mail solution, optionally with advanced features to support group Voice Mail Boxes, work with multiple Voice Mail Boxes 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.

3.1 Genesys Components

Genesys Components		
Component	**Version**	**Release Type**
SIP Server	8.1.101.93	HF
Media Server	8.1.700.61	HF
Universal Routing Server	8.1.400.12	GA
Stat Server	8.1.200.46	GA

3.2 Grandstream IP Phones

3rd Party Hardware Components		
Component	**Version**	**Release Type**
Grandstream GXP1625/GCC1700	1.0.2.27	Official

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

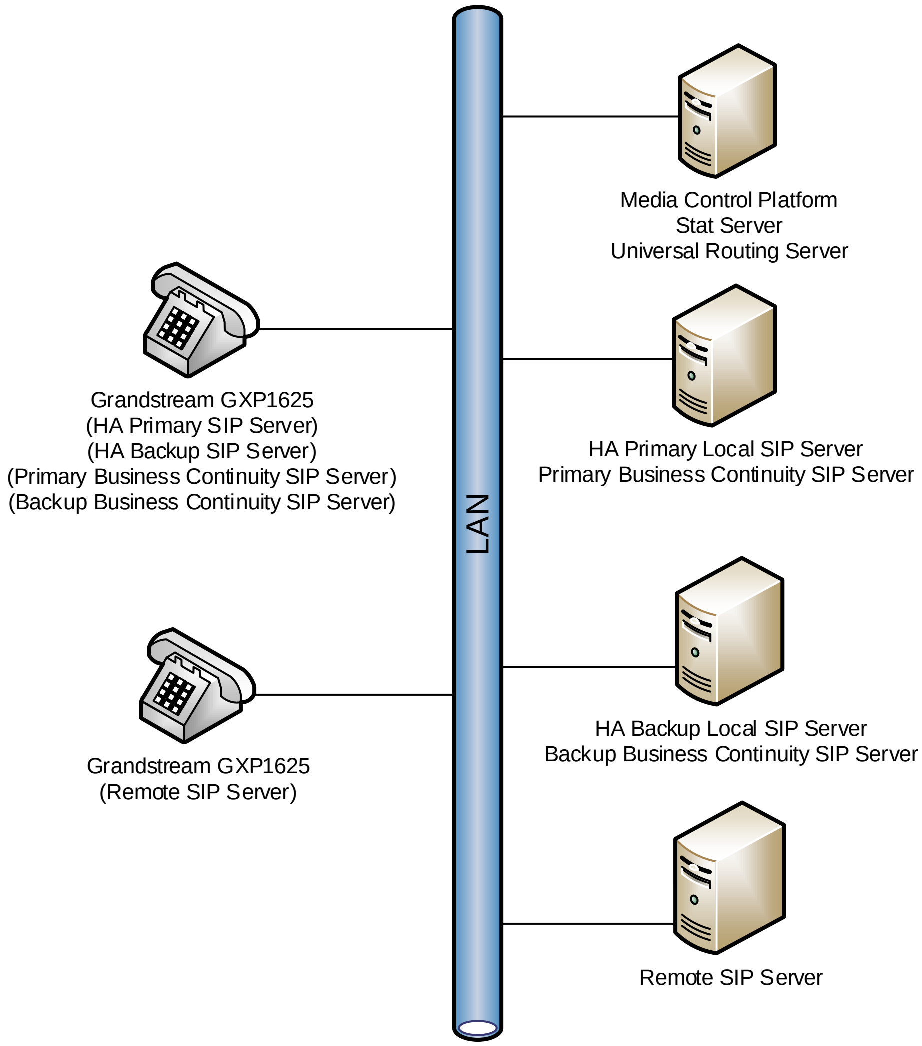
3rd Party Product Issues		
Description	**Product**	**Version**
None found		

4 Integration and Configuration Section

This section describes the various components involved with integrating the Grandstream GXP1625/GCC1700 SIP 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

The Genesys SIP 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 to a switch, etc.). 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.

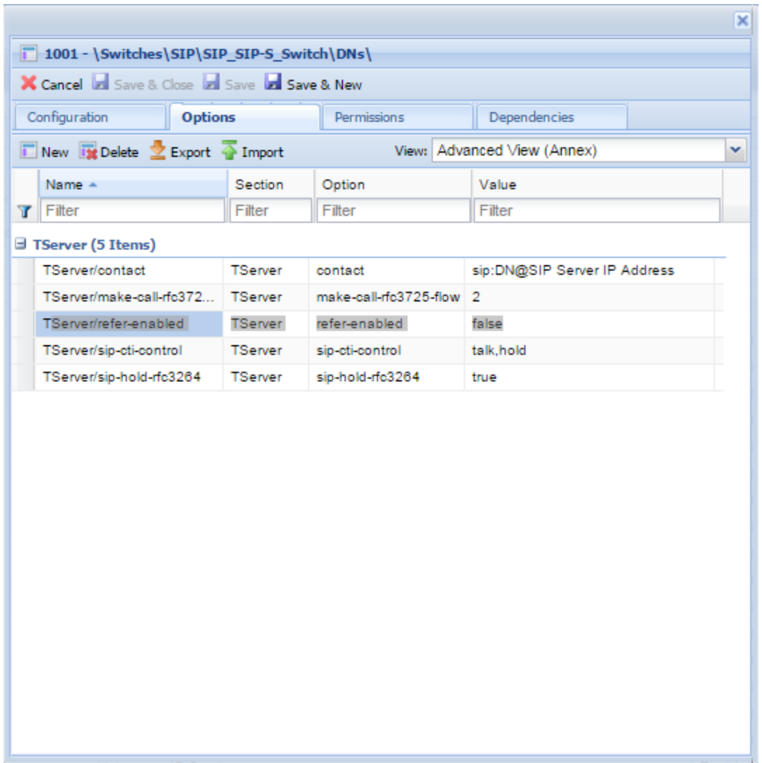
4.2.1 SIP Server Switch

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

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1</td><td>In the navigation bar, select "Switches". Select the 'SIP Server Switch' / [DNs] tab – create the necessary Extensions, Communication DN, Route Points, Trunks, Trunk Groups and Voice over IP Service DNs.

</td></tr>
</table>

<table>
<tr>
<td>2</td>
<td>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 4020 is used as an example):


refer-enabled = false
Set to false as the GXP1625/GCC1700 phones are currently incompatible with 3PCC Make Call using SIP REFER method.</td>
</tr>
</table>

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 in order 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.).

4.2.3 Media Control Platform (MCP) Application

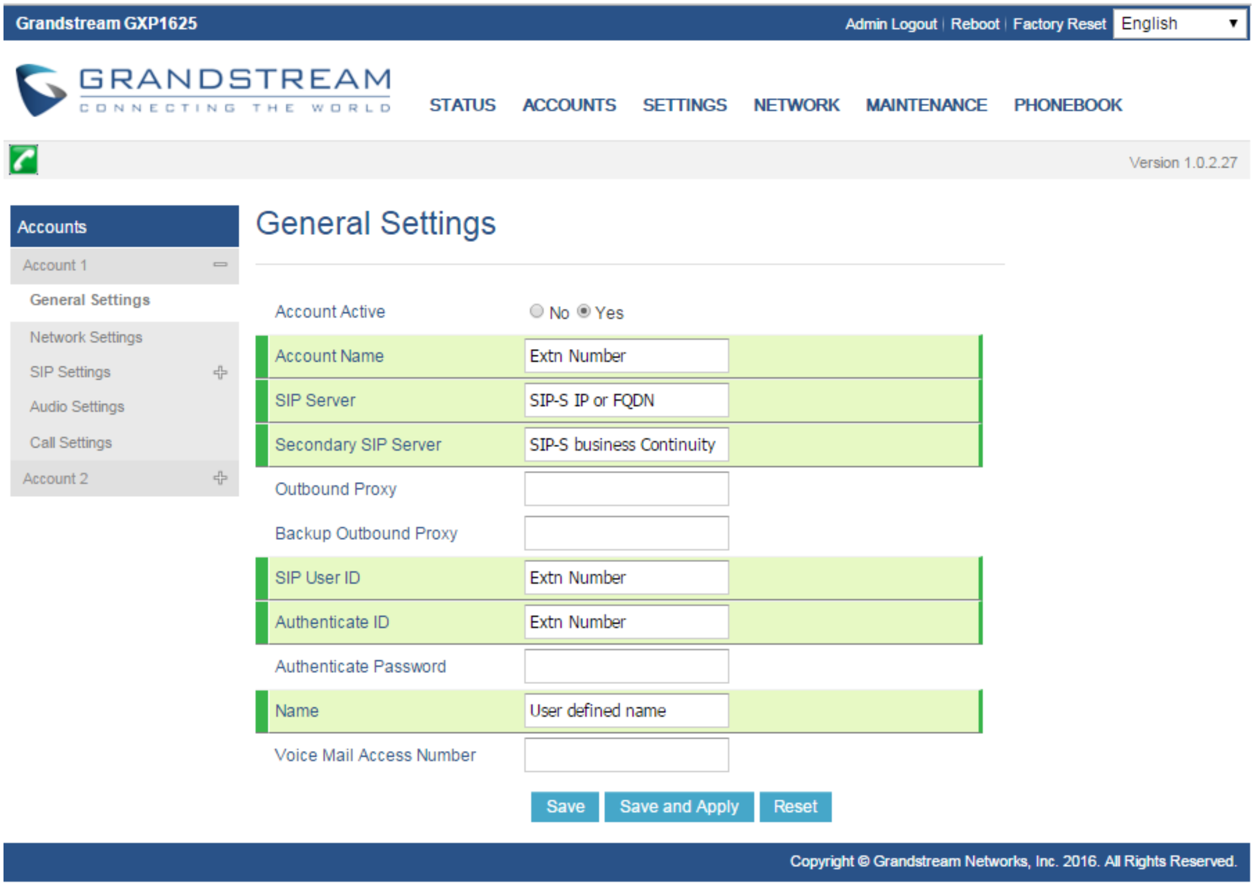
In this section, the MCP application is configured. It is assumed that MCP is:

- Installed and configured as per the Genesys Media Control Platform Deployment Guide.

Step	Description
1	If SRTP functionality is required, select the MCP application and navigate to the [Options] tab / mpc section - make sure the following options are set. srtp-mode = offer srtp.cryptomethods = AES_CM_128_HMAC_SHA1_80

4.3 Grandstream GXP Phone Configuration Section

4.3.1 SIP Server Connectivity

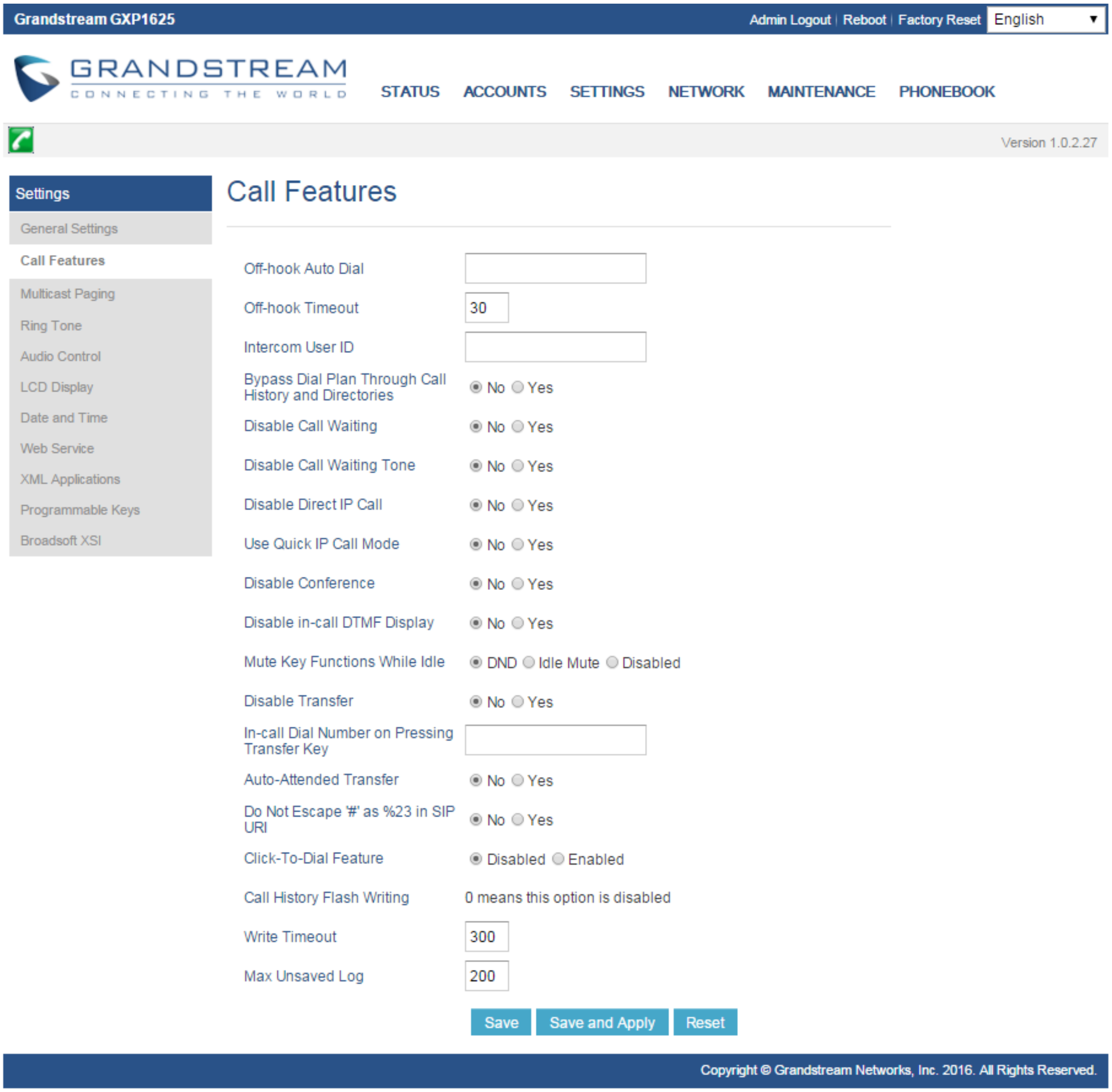


Note:

SIP Server field - Connectivity to primary SIP Server

Secondary SIP Server field - Connectivity to 2nd SIP Server in Business Continuity pair

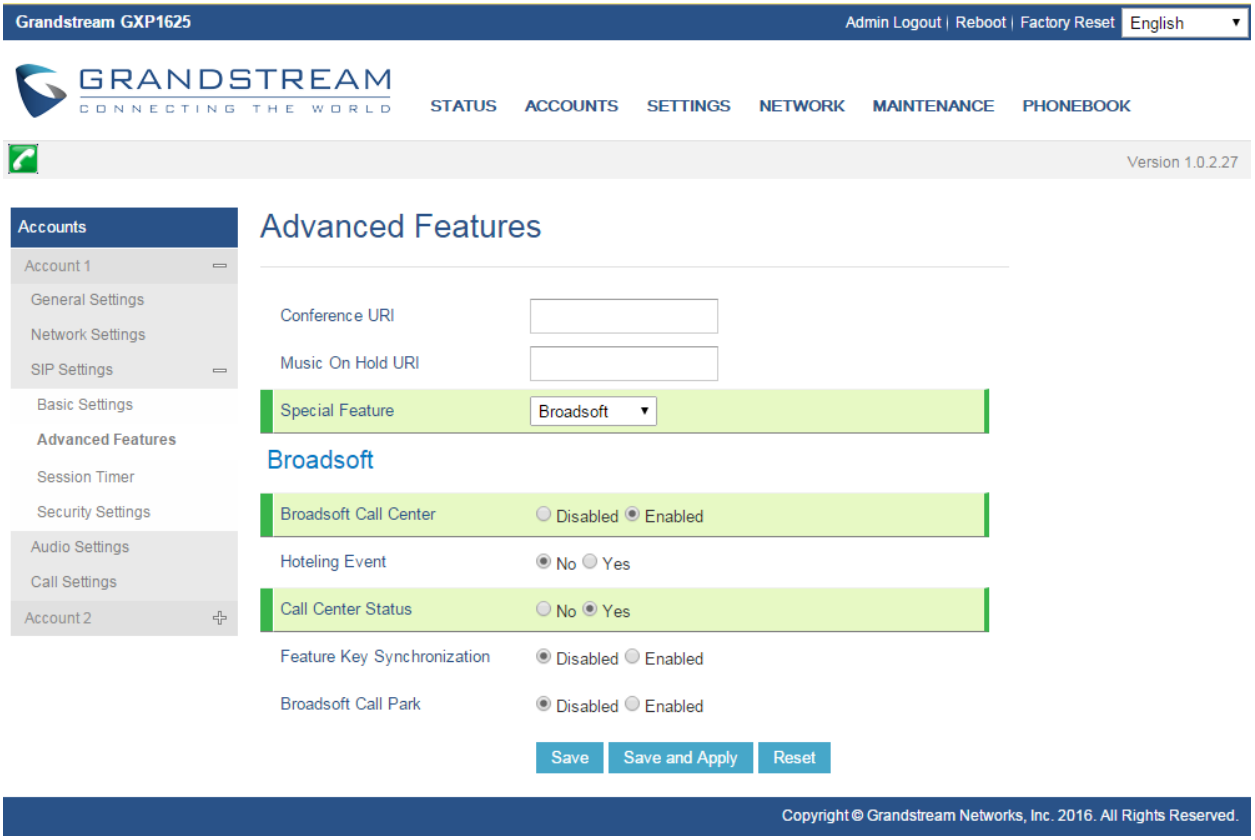
4.3.2 3PCC Functionality



Note:

Disable Call Waiting = No - When using 3PCC functionality

4.3.3 Broadsoft Agent Synchronisation



5 SIP Endpoint - Vendor Information

Company Summary from http://www.grandstream.com/company/about-grandstream

Grandstream Networks, Inc. has been connecting the world since 2002 with SIP Unified Communications solutions that allow businesses to be more productive than ever before. Our award-winning solutions serve the small and medium business and enterprises markets and have been recognized throughout the world for their quality, reliability and innovation. Grandstream solutions lower communication costs, increase security protection and enhance productivity. Our open standard SIP-based products offer broad interoperability throughout the industry, along with unrivaled features, flexibility and price competitiveness.

Grandstream products are available through our established global distribution channels. We are a private corporation headquartered in Boston, MA USA with regional locations in Los Angeles, CA, Dallas, TX, China, Venezuela, Morocco, Malaysia, Spain, Ukraine, Israel and Colombia.

Grandstream was named the 2016 Global Enterprise IP Endpoints Company of the Year by renowned market research firm, Frost & Sullivan.

Product summary from http://www.grandstream.com/our-products

The Grandstream IP voice & video products offer the best price-performance point in the industry. All of our products and solutions are designed to fully leverage the benefits of VoIP broadband networks. Each portfolio is based on SIP standard and is feature rich – supporting both traditional and advanced features - support a broad range of voice codecs, and are easy to manage through web-based GUI interfaces.

6 Appendices

6.1 Revision History

Revision History		
Version	**Date Published**	**Reason for Revision**
0.1	2016-01-20	Initial draft for review.
1.0	2016-01-22	Document approved

6.2 References

References		
Document Name	**Version**	**Date Published**
SIP Server Deployment Guide	81fr_dep-sip_12-2015_v8.1.101.30	2015-12
Genesys Media Server Deployment Guide	81gvp_dep-gms_07-2013_v8.1.701.00	2013-07

6.3 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 Gate Way
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