Genesys Application Note

Tenacity accessaphone ipTTY SIP Soft-Phone 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 the Tenacity accessaphone ipTTY SIP soft-phone.

Tools readily available on the web, can be used to create announcement sound files, that are formatted for baudot tones. This enables TTY feedback to be given to a caller, when an inbound call is queued on a Genesys Routing Point.

The supported phone software is 2.0.3.3 and later.

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	No
DNS-based redundancy (SIP Proxy, SIP Cluster)	No
DTMF tones generation	Yes
IPv6 support	No
Multiple calls on one extension	No
Message Waiting Indicator	No
SIP authentication	Yes
TLS/SRTP	No
Call Control with 1PCC	**Supported**
Basic calling	Yes
Conference	No
Hold / Retrieve	No
Transfer	
1-step	Yes
2-step semi-attended	No
2-step consultation	No
Call Control with 3PCC	**Supported**
Answer Incoming Call	No
Conference	No
Hold/Retrieve	Yes *
Make Outgoing Call	No
Remote Auto-Answer	No
Transfer	
1-step	Yes
2-step semi-attended	No
2-step consultation	No
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	No
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

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

3.1 Genesys Components

Genesys Components		
Component	**Version**	**Release Type**
SIP Server	8.1.101.93	HF
Media Server	8.1.700.61	HF

3.2 Non Genesys Components

3rd Party Hardware Components		
Component	**Version**	**Release Type**
Tenacity accessaphone ipTTY	2.0.3.3	Official

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

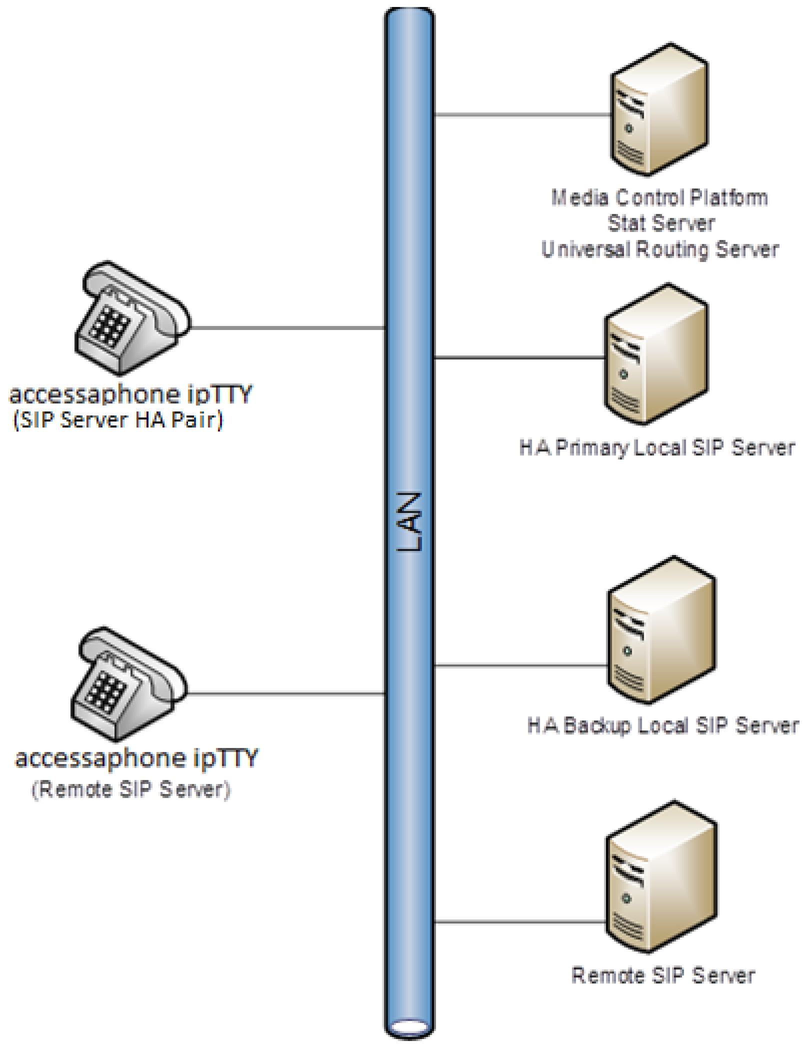
3rd Party Product Issues		
Description	**Product**	**Version**
The ipTTY phone stops unexpectedly in scenarios that involve multiple dialogs. See below: 1) SIP Server DN option dual-dialog-enabled = false ipTTY phone crashes when 3PCC consultation or conference initiated **Resolution** – Ensure any ipTTY phones registered to SIP Server, have option dual-dialog-enabled = true 2) ipTTY phone in a call with another phone (ipTTY or standard SIP phone) Other phone places call on hold, which triggers MOH. The ipTTY phone receives INVITE to start MOH which causes it to crash **Resolution** – Ensure any ipTTY phones registered to SIP Server, have option sip-enable-moh = false **Note:** Logs have been sent accessaphone for assessment	Accessaphone ipTTY Soft-Phone	2.0.3.3
ipTTY phone will accept SIP request to hold call, however the voice path is not disabled *Note: As this phone is aimed at chat sessions then Hold call functionality is probably not required*	Accessaphone ipTTY Soft-Phone	2.0.3.3
ipTTY soft phone only supports G711 ULAW codec	Accessaphone ipTTY Soft-Phone	2.0.3.3
It is not possible to control the ipTTY phone using 3PCC via NOTIFY request handling	Accessaphone ipTTY Soft-Phone	2.0.3.3

4 Integration and Configuration Section

This section describes the various components involved with integrating the Tenacity accessaphone ipTTY SIP Soft 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 Tenacity accessaphone ipTTY SIP Phones

ipTTY is a SIP soft phone that supports the conversion of Baudot Tones sent via IP to text. In other words, ipTTY facilitates a TTY/TDD conversation over IP without the need for modems or analog lines. ipTTY connects directly via IP to PBX and VoIP systems that support SIP. This type of communication is necessary for individuals with hearing impairments.

4.1.2 Genesys SIP Server

The Genesys SIP interface between the Tenacity 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

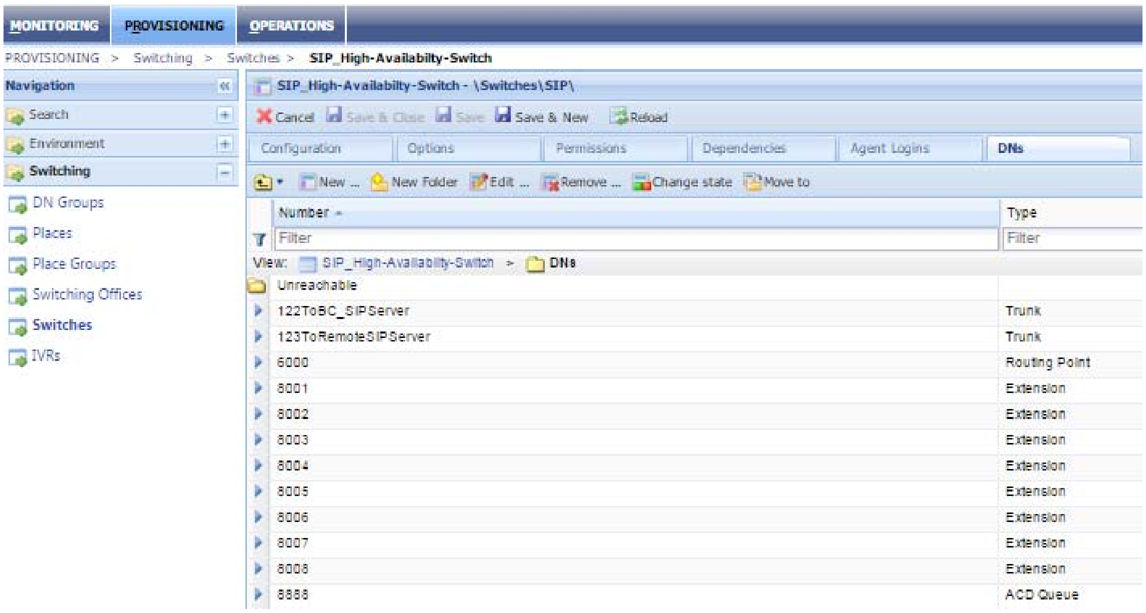
This section covers the configuration of DN objects in Genesys Configuration Server and phone configuration screenshots.

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

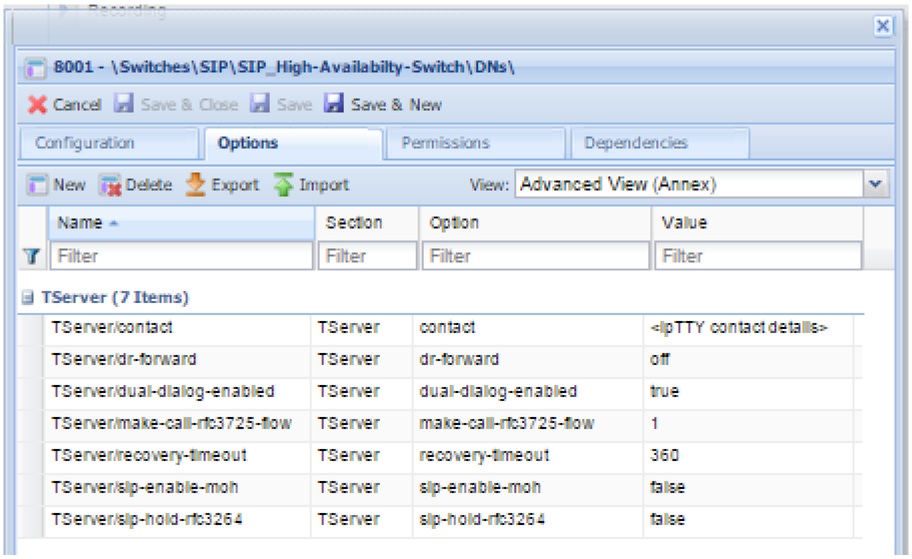
In this section the SIP Server Switch resources are configured. It is assumed that:

- 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”.

Step	Description
1	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. 

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 4020 is used as an example):



Ensure the following options are set for all accessaphone ipTTY phones:
sip-enable-moh = false
dual-dialog-enabled = true

4.2.2 SIP Server Application

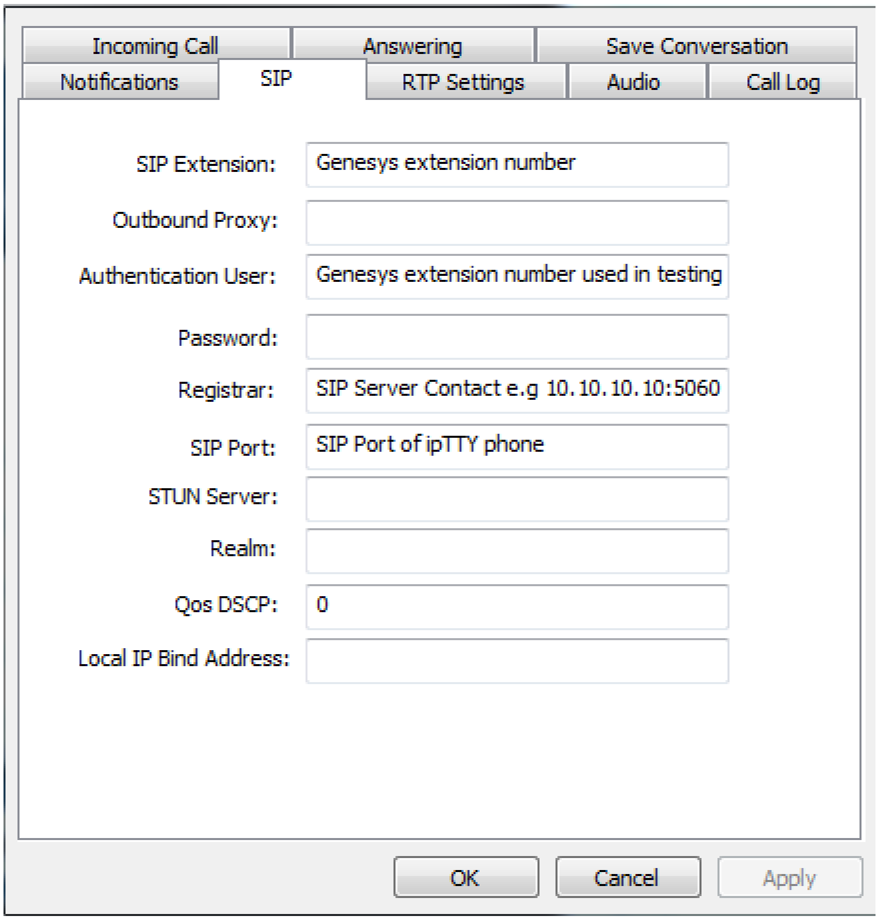
In this section the SIP Server application is configured. It is assumed that SIP Server is:

- 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. 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 = 5060 (SIP Server listening port for incoming SIP requests. The same port number is used for both TCP and UDP transports).

4.3 Tenacity accessaphone ipTTY Configuration Section

4.3.1 SIP Server Connectivity



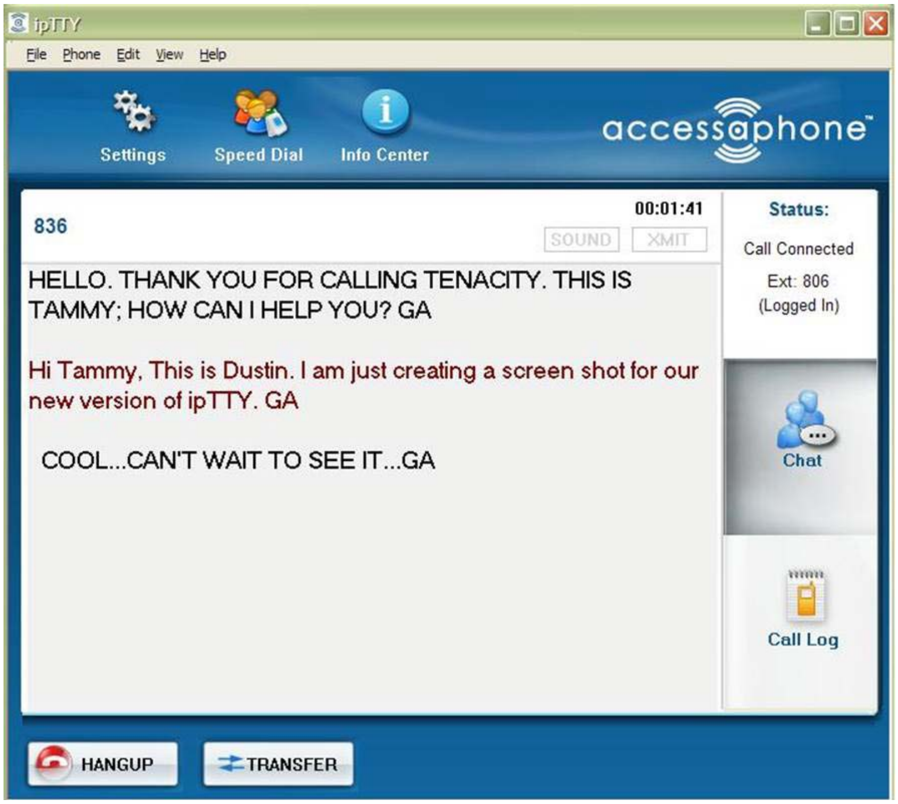
5 SIP Endpoint - Vendor Information.

Company Name: Tenacity Operating
Company Website: http://www.accessaphone.com/iptty/

What is ipTTY?

No analog lines or bulky desktop solutions.

ipTTY (sometimes referred to as TTY over IP or TTY over VoIP) is engineered to allow TTY communications using existing telephony infrastructure. The program communicates through IP and uses standard session initiation protocol (SIP). ipTTY enables virtually every user on your telephony network to communicate with customer TTY machines or the Text Relay Service (TRS), without the need for expensive analog lines or FXS gateways. Additionally ipTTY supports audio for hearing and voice carry-over and real-time-text via RFC 4103; the future in real time text communication.



Packed with features

Create, answer and transfer calls with ease.
Can be configured to act like an answering machine.
Speed Dials for quick calling
Dial from recent calls list
Conversations can be saved
Accessible interface (edit font size and navigate via Tab key)
Supports Hearing and Voice Carry Over
Supports Flashers and Wireless Alerters
Supports Real Time Text (RFC 4103)

6 Appendices

6.1 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

6.2 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 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
SIP-S	Genesys SIP Server Application
SMI	Genesys Supported Media Interface Guide