Genesys Application Note

CounterPath Bria SIP Phones With Genesys SIP Server

Document version 1.8

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Table of Contents

1 Summary

CounterPath Bria phones are recommended as SIP "soft phones" to be integrated and used with the Genesys SIP solution. All voice features, from simple calls to voicemail integration, have been successfully validated during extensive testing. This application note details the supported features, and includes reference configuration examples.

The latest tested version of CounterPath Bria is 6.1.

Note: CounterPath Bria 6.1 was tested only with IPv4. IPv6 is not supported at this time. Only audio was tested with CounterPath Bria 6.1.

For a complete list of supported and validated versions of CounterPath Bria and required Genesys components, see Software and Hardware Versions Validated.

For a complete list of 3rd party hardware/software supported by Genesys, see the Genesys Supported Media Interfaces Guide (SMI).

2 SIP Endpoint Features

2.1 Feature Chart

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

* See section 6 for known limitations.
** Video is supported with Bria version 5 only. Video is not tested with Bria version 6.1.

2.2 Feature Chart Glossary

2.2.1 General Features Supported By Phone

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.

Agent State Control from the Phone: This feature enables an agent to perform agent-related operations from the phone: login/logout, change of the state to ready/not ready/ACW, reason code for not ready state. Available for phones which support BroadSoft's Application Server Feature Event Package and Hoteling Event Package.
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.

Shared Call Appearance (SCA): This feature enables a group of SIP phones to receive inbound calls directed to a single destination (shared line); that way, any phone from this group can answer the call, barge-in to the active call, or retrieve the call placed on hold. The shared line has sub-lines called appearances.

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 Using Phone (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 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 Using Desktop Client (3pcc)

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

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

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

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

Remote Auto-Answer: Phone can answer a call automatically based on Call-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.

DTMF tone generation: Phone can generate DTMF tones through RTP when the tone generation was requested by SIP Server through the Genesys T-Library interface.

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 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 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 Validated

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

3.1 Genesys Components

Genesys Components		
Component	**Version**	**Notes**
SIP Server	8.1.1	Call switching and control is performed by Genesys SIP Server. SIP Server communicates via SIP with SIP Endpoints. **Important:** If using Bria 6.1+, SIP Server version 8.1.103.95 (or later) is required.
Genesys Media Server	8.1.7, 8.5.1	Used to handle media interactions such as call treatments (ring back, busy tones and music on hold); also used as MCU. For video calls support, use Media Server version 8.5.1.
Genesys SIP Feature Server	8.1.2	Used as a SIP Voicemail server.

3.2 CounterPath Bria Softphones

3rd Party Hardware Components		
Model	**Version**	**Notes**
CounterPath Bria 6	6.1.0.3 103770	v6.1.0.3 103770 or later are supported
CounterPath Bria 5	5.1.0 89374	v5.1.0 89374 or later are supported
CounterPath Bria 4	4.1.1 74246	v4.1.1 74246 or later are supported

For a complete list of 3rd party hardware/software supported by Genesys, see the Genesys Supported Media Interfaces Guide (SMI).

4 Features Configuration in Genesys Configuration Environment

This section describes how to configure features represented in the Feature Chart in the Genesys configuration environment.

Features can be configured in the SIP Server Switch on a DN object with type Extension (or ACD Position) representing SIP Endpoint devices.

Note: It is assumed that the reader has Genesys knowledge and is familiar with deploying a basic Genesys environment.

Features Configuration In Genesys Configuration Environment	
General Features Supported By Phone (1pcc)	
Feature	**Key Actions and Procedures**
Auto-Answer	No configuration is required.
Caller ID	No configuration is required.
Call Forward	No configuration is required.
Do Not Disturb	No configuration is required.
DTMF tones generation	No configuration is required.
Multiple calls on one extension	No configuration is required.
Message Waiting Indicator	Configure a voice mail box for an Extension. In the TServer section of the DN object, configure: **gvm_mailbox=<voice mail box number>** For example: gvm_mailbox=12002, where 12002 is a mailbox number.
SIP authentication	1. Specify SIP requests (REGISTER, INVITE) which are sent by the phone to be authenticated by SIP Server. In the TServer section of the DN object, configure: **authenticate-requests=register,invite** 2. If required, configure the password used for authentication of incoming REGISTER or INVITE messages to SIP Server. In the TServer section of the DN object, configure: **password=<Any alphanumerical string>** **Note:** A string must match the phone setting in Softphone -> Account Settings -> Add -> SIP Account -> User Details -> Password.

TLS/SRTP	See the Transport Layer Security for SIP Traffic chapter in the Genesys 8.1 SIP Server Deployment Guide for details.
Call Control Using Phone (1pcc)	
Feature	**Key Actions and Procedures**
Basic calling (incoming and outgoing calls)	See the Make Outgoing Call feature.
Conference	No configuration is required.
Hold/Retrieve	No configuration is required.
Unattended transfer	No configuration is required.
Attended transfer	No configuration is required.
Call Control Using Desktop Client (3pcc)	
Feature	**Key Actions and Procedures**
Answer Incoming Call	Enable SIP Server to send the SIP NOTIFY (event talk) message when desktop client requests to answer the incoming call. In the TServer section of the DN object, configure: **sip-cti-control=talk** **Note:** The "talk" value affects the Retrieve feature. See Hold/Retrieve feature for information about setting the **sip-cti-control** option.
Conference	Deploy Genesys Media Server with MCU capabilities. See the *SIP Server Deployment Guide* for details.
Hold/Retrieve	Enable SIP Server to send the SIP NOTIFY (event hold) message when desktop client requests to hold the call, and the SIP NOTIFY (event talk) message when desktop client requests to retrieve the call. In the TServer section of the DN object, configure: **sip-cti-control=talk,hold**
Make Outgoing Call	1. Create a DN object with type Extension or ACD Position in the Genesys configuration environment under Switch object and DNs folders. This object represents the SIP phone. 2. To activate required features described in this Table, configure options in the DN object > TServer section. 3. Configure a phone to make basic calls (incoming, outgoing) with SIP Server. 4. Restart the phone. 5. After successful SIP registration the phone is ready for making outgoing calls and receiving incoming calls. 6. Run your desktop client to make a test call.

<table>
<tr><td>Remote Auto-Answer
(based on the SIP header)</td><td>If required, specify the value that SIP Server will add into the Call-Info header of the INVITE message, which it sends to the SIP Endpoint. In the TServer section of the DN object, configure:
    auto-answer-after=0
where 0 indicates that the phone answers the call immediately.</td></tr>
<tr><td>Unattended transfer
(Genesys Single-Step Transfer)</td><td>No configuration is required.</td></tr>
<tr><td>Semi-attended transfer
(Genesys Blind Transfer)</td><td>Enable completion of transfer when the destination is in alerting state. In the TServer section of the DN object (transfer target DN), configure:
    blind-transfer-enabled=true

Note: This option must be set on the DN object that represents a transfer destination party.</td></tr>
<tr><td>Attended transfer
(Genesys Two-Step Transfer)</td><td>1. Enable dual dialog to be supported on a DN for an attended transfer operation requested from a desktop client. In the TServer section of the DN object, configure:
    dual-dialog-enabled=true

2. Specify the call flow to process a make call/initiate consultative call operation initiated from a desktop client. In the TServer section of the DN object, configure:
    make-call-rfc3725-flow=2

Note: A value of 1 or 2 is sufficient for the phone.

3. Specify the SIP INVITE method to be sent to an Endpoint when a simple call or consultation call is initiated from a desktop client. In the TServer section of the DN object, configure:
    refer-enabled=false

4. If required, specify the SIP REFER method to be sent to an Endpoint when a simple call or consultation call is initiated from a desktop client. In the TServer section of the DN object, configure:
    refer-enabled=true

Warning! The REFER method will work properly if you set the following option on the DN object: sip-cti-control=talk.</td></tr>
<tr><td colspan="2">Video Support</td></tr>
<tr><td>Feature</td><td>Key Actions and Procedures</td></tr>
<tr><td>Basic video calls</td><td>No configuration is required.</td></tr>
<tr><td>Push video</td><td>1. Create the gcti::video device. Under a configured Switch object -> DNs folder, create a new DN object by setting the following properties:
    Number: Enter gcti::video.
    Type: Select Trunk.</td></tr>
</table>

	2. Specify the name of the default video file that will be played to a caller. In the TServer section of the SIP Server Application object, configure: **default-video-file=<name of video file>**
Video call on Hold/Retrieve	No configuration is required.
Video call Transfer	No configuration is required.
Video conference	1. Deploy Genesys Media Server with MCU capabilities. Turn on support of video codecs on Media Server. 2. In the TServer section of the SIP Server Application object, configure: **info-pass-through=*** See the *SIP Server Deployment Guide* for details.
Genesys Business Continuity	Enable call forwarding of incoming calls to the SIP Server peer where an agent is logged in. In the TServer section of the DN object, configure: **dr-forward=no-agent**

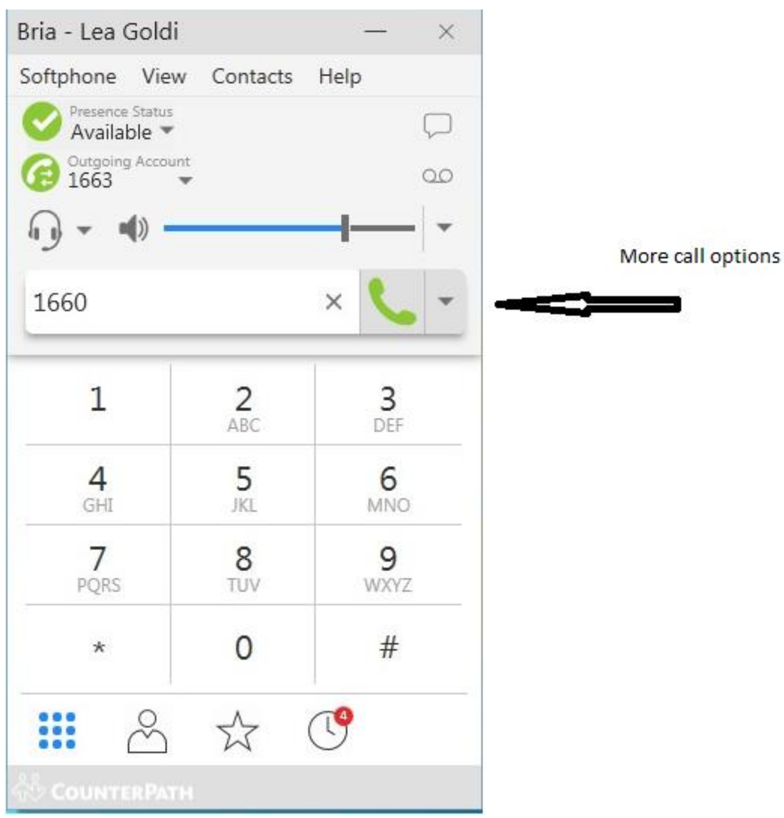
Example of the DN .cfg file:

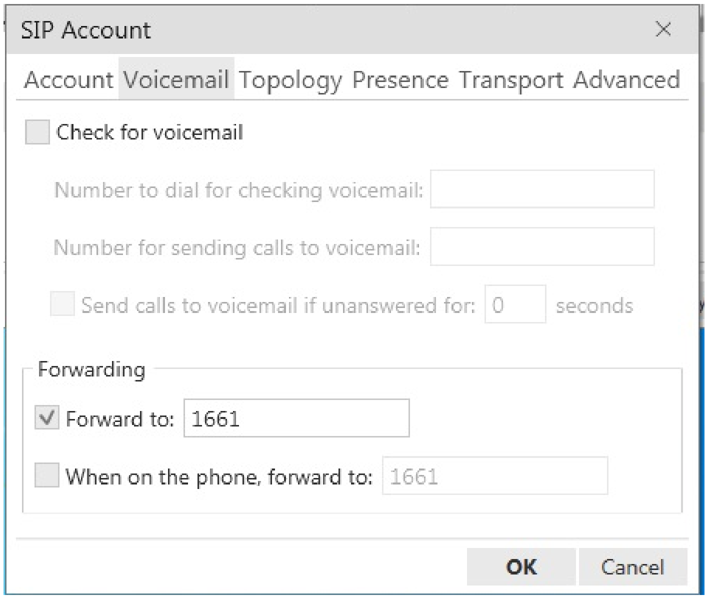
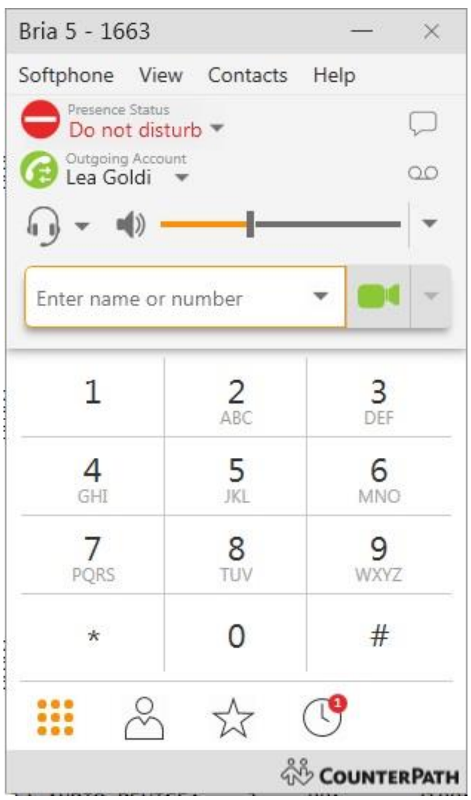
```
[TServer]
contact=sip:1663@172.21.82.239:46200;rinstance=4b9062683af50dfd
dual-dialog-enabled=true
make-call-rfc3725-flow=1
refer-enabled=false
sip-cti-control=talk,hold
```

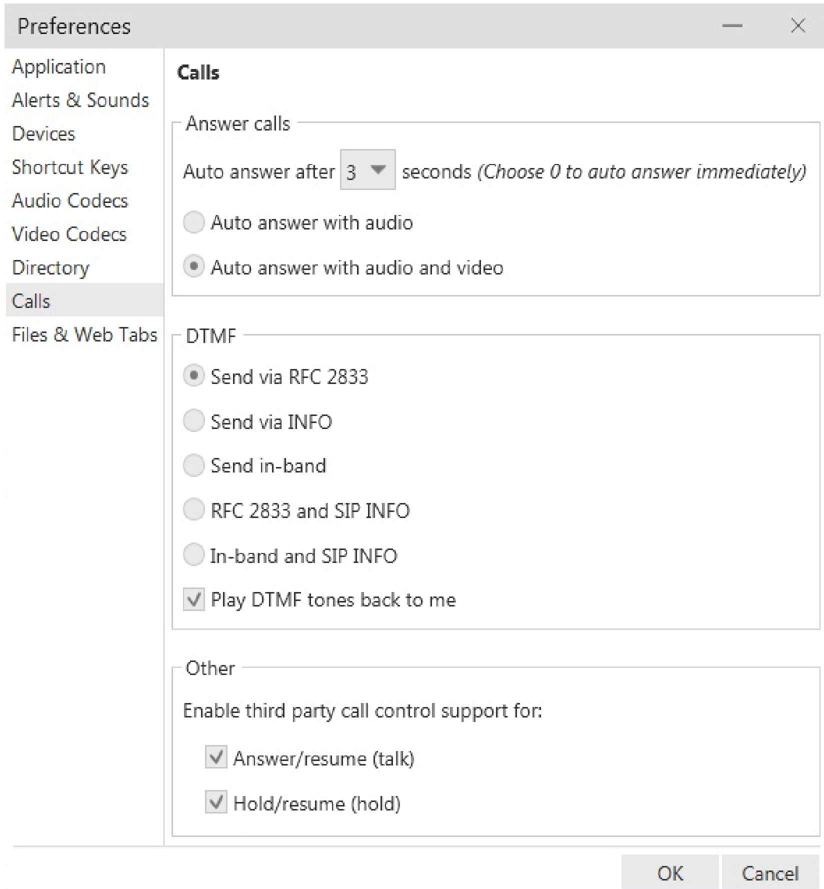
5 CounterPath Bria Phone Configuration

This section describes how to configure features represented in the Feature Chart in the CounterPath Bria using the softphone GUI.

The following table displays screenshots of the softphone GUI of Bria 5.

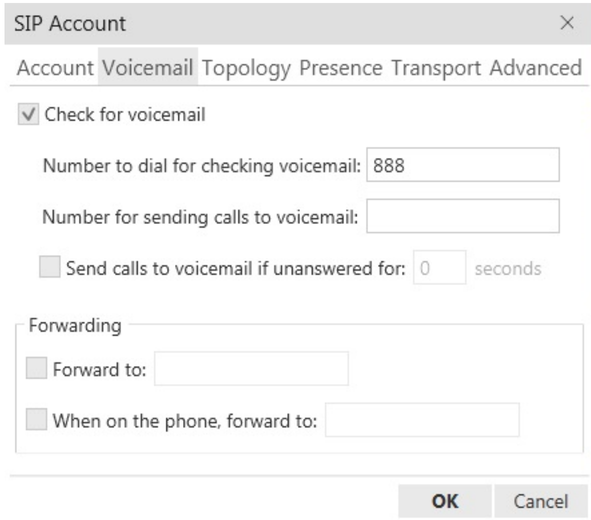
Bria Phone Configuration	
General Features Supported By Phone	
Feature	**Key Actions and Procedures**
Auto-Answer	Turn on and off auto-answer of all SIP calls by selecting **More call options -> Auto Answer**. **Note:** Auto-answer is initially configured to auto-answer after 1 ring and to send only audio. To change these settings, go to **Softphone -> Preferences -> Calls -> Answer call**. 
Caller ID	No configuration is required.

Call Forward	Enable call forwarding by selecting the **Forward to** check box, and specify the forward destination using the Account Settings window: **Softphone -> Account Settings -> SIP Account -> Voicemail -> Forwarding -> Forward To**. 
Do Not Disturb	Select **Do not Disturb** from the drop-down list. 

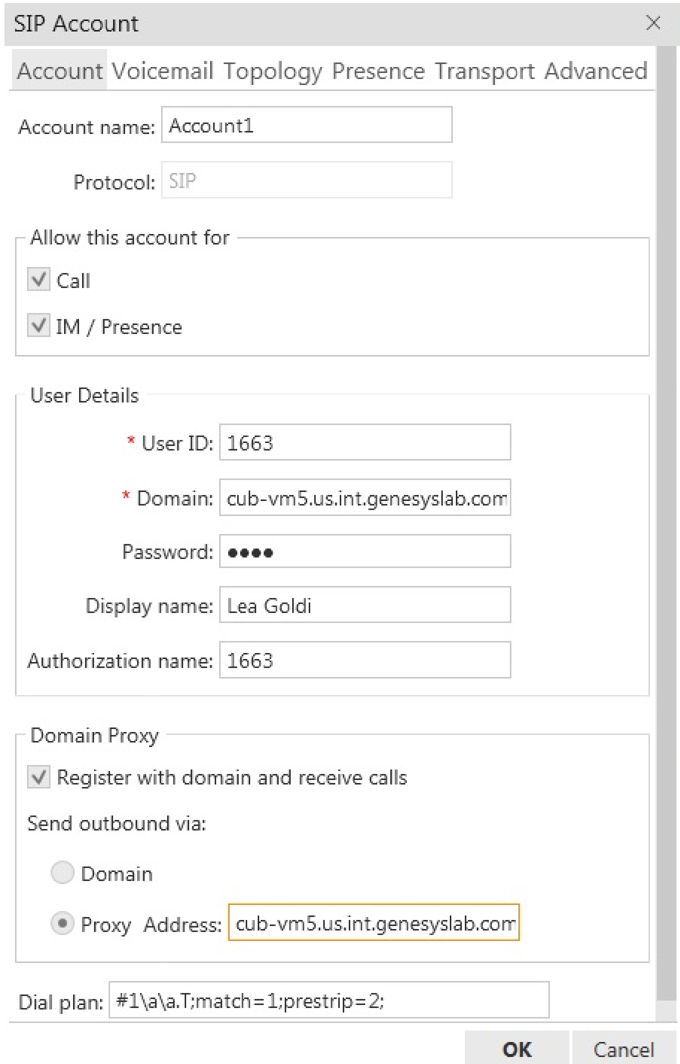
<table>
<tr>
<td>DTMF tones generation</td>
<td>Specify the method for a DTMF tone generation. Go to Softphone -> Preferences -> Calls -> DTMF.


</td>
</tr>
<tr>
<td>Multiple calls on one extension</td>
<td>No configuration is required.

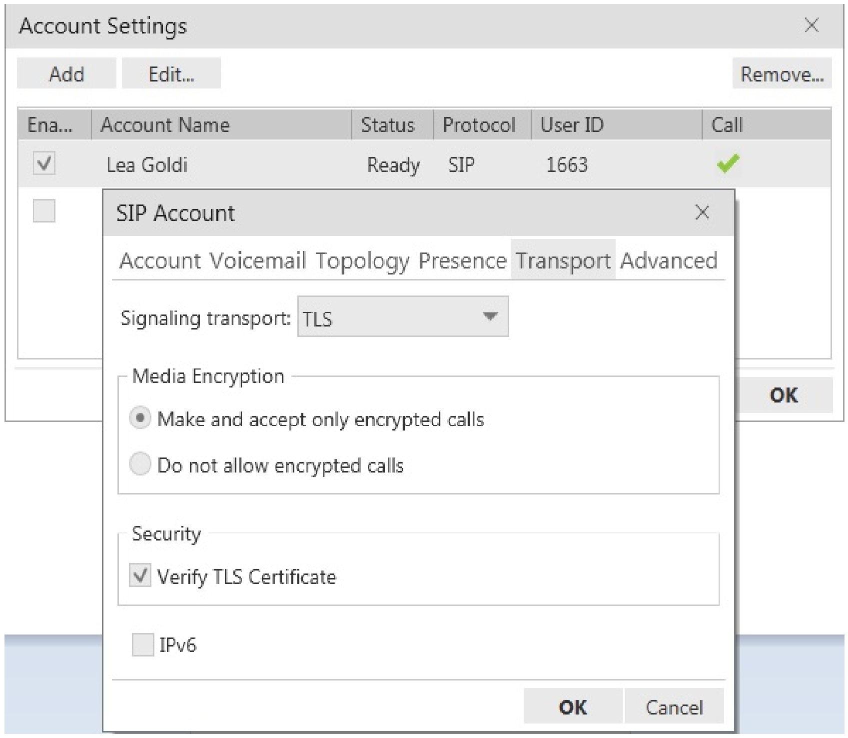
Note: There is no limit to the number of calls you can make using Bria. Genesys recommends having no more than two concurrent calls.</td>
</tr>
</table>

Message Waiting Indicator	1. Using the Account Settings window (Softphone -> Account Settings -> Voicemail), select **Check for voicemail**. 2. In the **Number to dial for checking voicemail** field, enter the number to dial a voicemail system. 

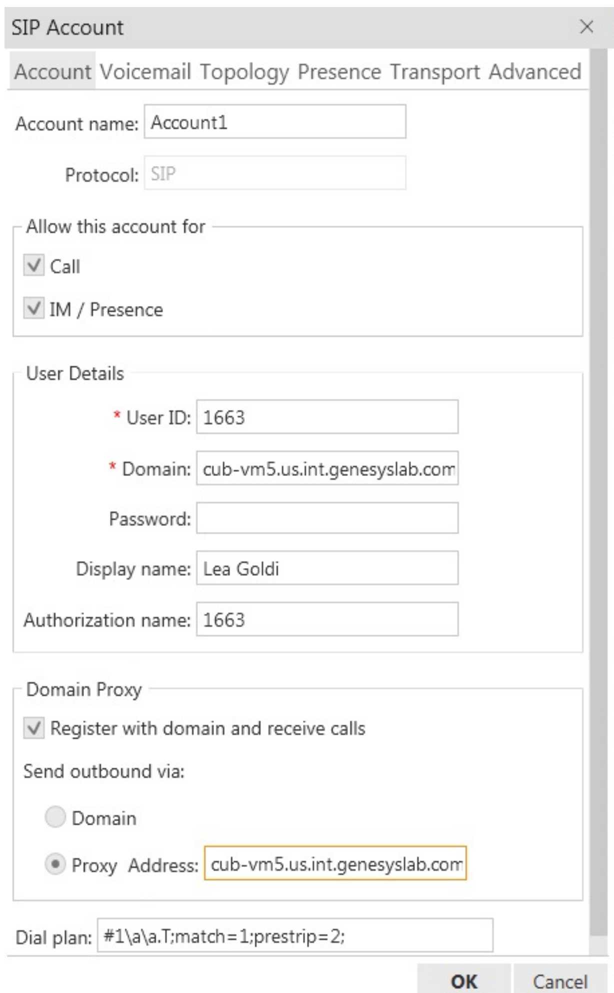
<table>
<tr>
<td>SIP authentication</td>
<td>Using the Account Settings window (Softphone -> Account Settings -> Add -> SIP Account -> User Details), specify the credentials (Password and Authorization name) for SIP authentication.
Note: Authorization name is used as a username in the Authorization of SIP REGISTER and INVITE messages.



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

<table>
<tr>
<td>TLS/SRTP</td>
<td>
1. Configure the SIP Account with the address and secure port of SIP Server.
2. On the SIP Account tab (Softphone -> Account Settings -> Edit account -> Transport), specify the Signaling transport as TLS.
3. On the SIP Account tab, select Make and accept only encrypted calls.


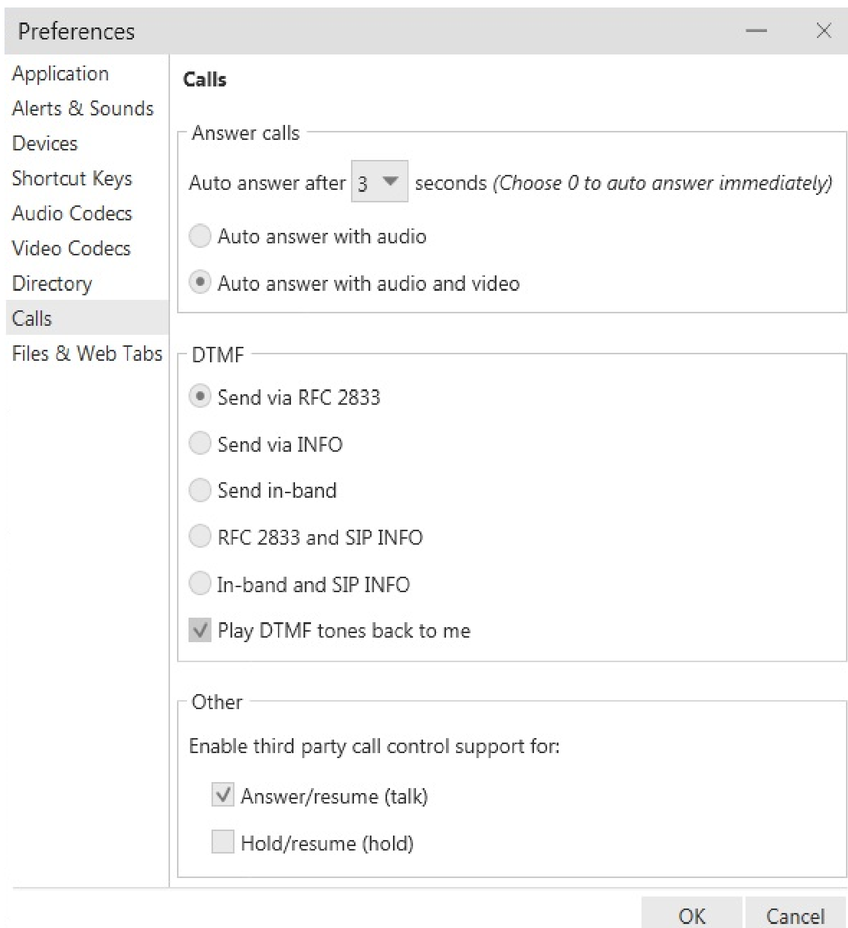
</td>
</tr>
<tr>
<td colspan="2">Call Control Using Phone</td>
</tr>
</table>

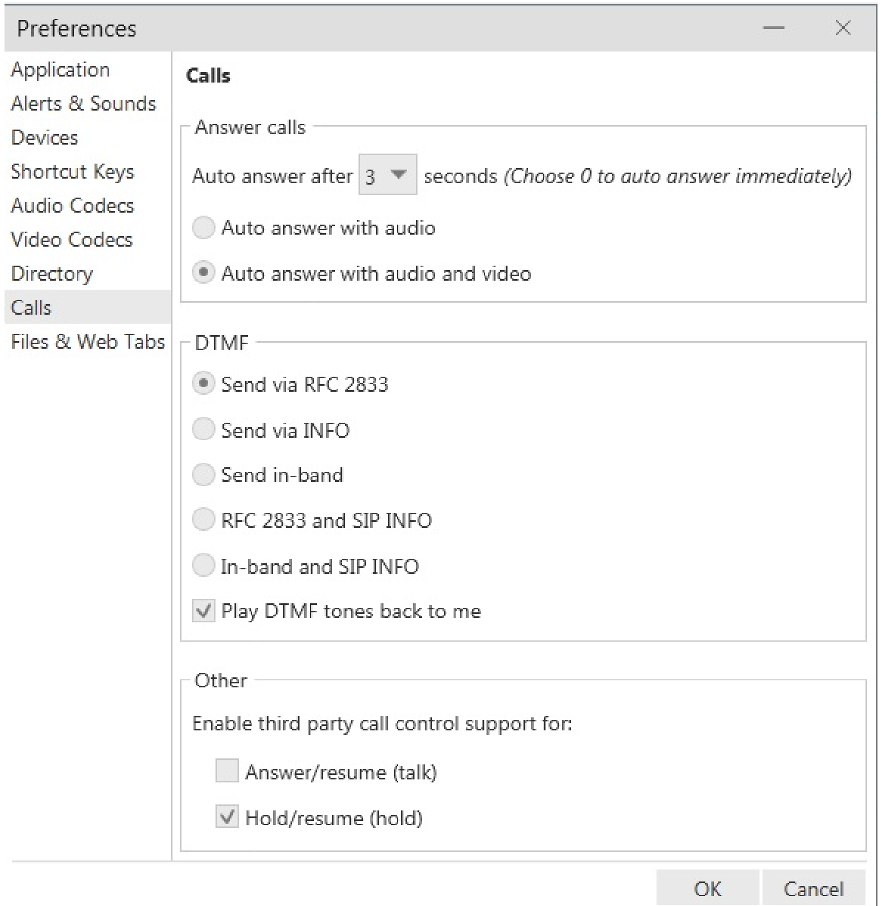
<table>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Basic calling (Incoming and outgoing calls)</td><td>
1. Using the Account Settings window (Softphone -> Account Settings -> Add -> SIP Account -> User Details), specify the User ID (the address that will be registered on the SIP Registrar) and Domain (SIP Registrar).
Note: User ID must equal the DN number in the Genesys configuration environment.
2. If required, specify the Display name in the User Details section.
3. In the Allow this account for section, select the Call check box.
4. In the Domain Proxy section, select the Register with domain and receive calls check box.
5. In the Proxy > Address, specify the address of SIP Proxy. In general cases, it is the SIP Server address.
6. Click OK.


</td></tr>
<tr><td>Conference</td><td>No configuration is required.</td></tr>
<tr><td>Hold/Retrieve</td><td>No configuration is required.</td></tr>
</table>

<table>
<tr><td>Unattended transfer</td><td>No configuration is required.

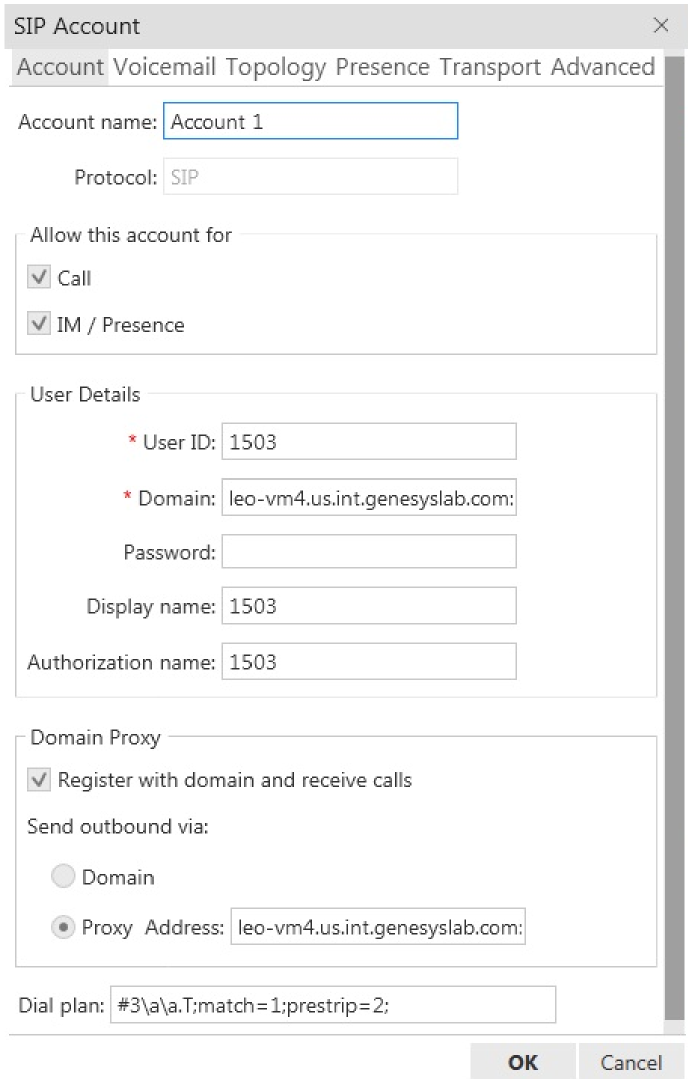
Using the phone, press Transfer this call, enter the number, and then press Transfer now.</td></tr>
<tr><td>Attended (consultative) transfer</td><td>No configuration is required.

Using the phone, press Transfer this call, enter the number, and then press Call first and the destination will be ringing. After the destination answers, then you can see Transfer now. Click it. The call will be transferred.</td></tr>
<tr><th colspan="2">Call Control Using Desktop Client</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Answer Incoming Call</td><td>Enable Third-Party Control in Bria Preferences: go to Softphone -> Preferences -> Calls -> Other, and select Answer/resume (talk).

</td></tr>
<tr><td>Conference</td><td>No configuration is required.</td></tr>
<tr><td>Hold/Retrieve</td><td>Enable Third-Party Control in Bria Preferences: go to Softphone -> Preferences -> Calls -> Other, and select Hold/resume (hold).</td></tr>
</table>

	
Make Outgoing Call	See the [Basic calling (incoming and outgoing calls)] feature.
Remote Auto-Answer	Configuration of the Remote Auto-Answer (based on the SIP Call-Info header) functionality is not available in the phone’s GUI.
Unattended transfer	No configuration is required.
Semi-attended transfer (Genesys Blind Transfer)	No configuration is required.
Attended (consultative) transfer	No configuration is required.

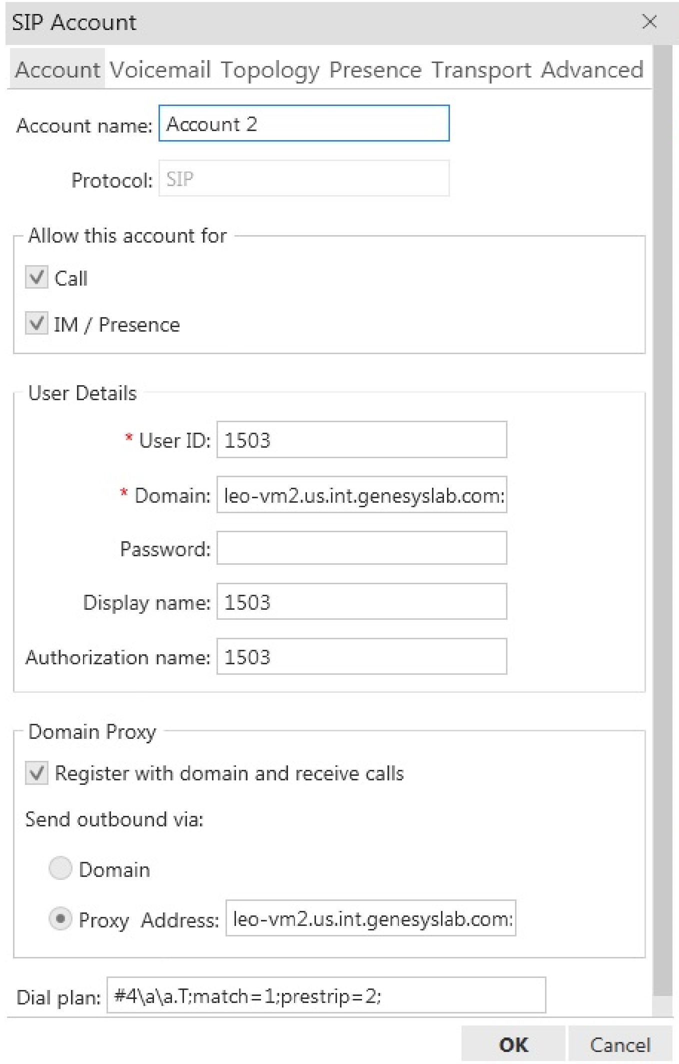
<table>
<tr>
<td>Genesys Business Continuity</td>
<td>

1. Create **Account 1**. See the Basic calling (Incoming and outgoing calls) feature for reference.
2. In the **Domain Proxy Address**, specify the IP address of the peer1 SIP Server.

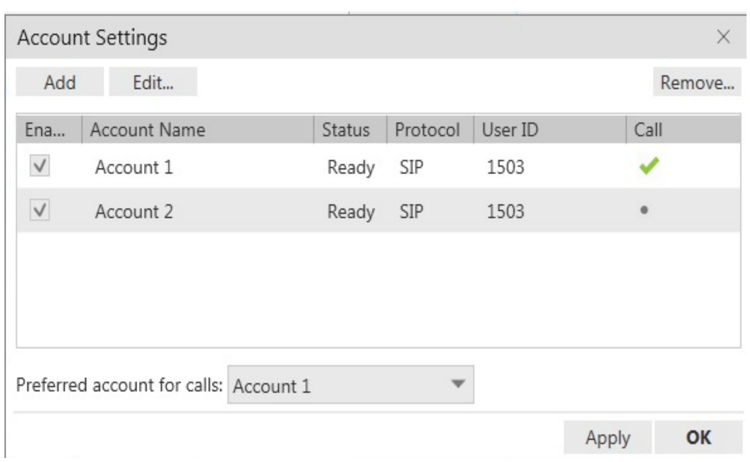


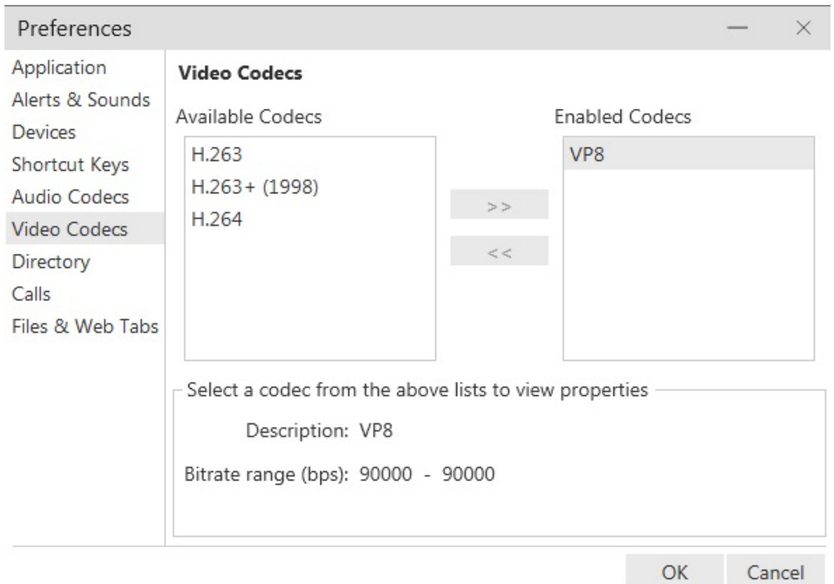
3. Create **Account 2** using the same User ID as in Step 1. See the [Basic calling (Incoming and outgoing calls)]() feature for reference.
4. In the **Domain Proxy Address**, specify the IP address of the peer2 SIP Server.

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



5. Set up the preferred account for calls. Go to **Softphone -> Account Settings -> Preferred accounts for calls**.



<table>
<tr><th colspan="2">Video Support</th></tr>
<tr><th>Feature</th><th>Key Actions and Procedures</th></tr>
<tr><td>Basic video calls</td><td>Enable video codecs via the softphone GUI. Go to Softphone -> Preferences -> Video Codecs.


Video Codec VP8 was used to verify all Video features.</td></tr>
<tr><td>Push video</td><td>See the Basic video calls feature.</td></tr>
<tr><td>Video call on Hold /Retrieve</td><td>See the Basic video calls feature.</td></tr>
<tr><td>Video call Transfer</td><td>See the Basic video calls feature.</td></tr>
<tr><td>Video conference</td><td>See the Basic video calls feature.</td></tr>
</table>

6 Known Issues and Limitations

Issues and Limitations Identified with Genesys Products

When SIP Server is operating with CounterPath Bria phones:

- A three-way conference on the phone is not supported in a Call Center deployment. Call participants can talk to each other, but such a call is not reported as a conference.
- Video was not tested for CounterPath Bria 6.1.

Issues and Limitations Identified with Third-Party Products

- A consultation call initiated by a desktop client using the REFER method will not be successful if the hold operation is configured to be done using the NOTIFY (hold) method.
- Bria is able to recognize that SIP Server is not responsive only when SIP Server does not respond to a SIP REGISTER request.
- Configuration of the Remote Auto-Answer (based on the SIP header) functionality is not available in the phone’s GUI. It was tested on version 5.1.0.3 Build 90054, which is based on 5.1.0 GA Base code. Only immediate auto-answer is supported.