Microsoft Skype for Business Deployment Guide

Calling using Back-to-Back User Agent

12/17/2025

Calling using Back-to-Back User Agent

Starting with release 8.5.001.63, T-Server for Skype for Business can dial a new destination using the Back-to-Back User Agent (B2BUA) method, where call switching and control is performed by Genesys components. This feature allows the provisioning of a configured Caller ID and the correct reporting of the origination party of ISCC calls. It also permits calls to be made to TEL URI destinations, Response Groups, and destinations with forwarded calls, none of which can be dialed directly by T-Server.

The use of B2BUA may adversely affect performance. Therefore, Genesys recommends that its configuration be limited to destinations matching a given dial plan or even to the level of individual calls.

Configuring Calling using B2BUA

To enable the B2BUA calling method, configure a DN of type Voice over IP Service with the **service-type** option set to `dialplan`, and assign this DN in the **calling-method-dialplan** option of the T-Server application. In the dialplan DN, configure dial-plan rules if required. If the destination matches the dial plan and **calling-method** = `b2b`, the destination will be called using the B2BUA method. To customize Caller ID information that is displayed on a destination party phone, use the **cpn** configuration option on the Application and/or DN levels.

The destination of a call dialed using B2BUA will be reported as the requested URI until the call is answered. At this point, the destination URI will be replaced with the actual URI of the answering party. This means, for example, that if the destination URI is a TEL URI of a Skype for Business user, then the TEL URI party will be replaced with a SIP URI party when the Skype for Business user answers. Further examples can be found in the call flow tables below.

Configuration Options

Application-level options:

- calling-method-dialplan
- cpn

DN-level options:

- dial-plan-rule-<n>
- calling-method

Dial Plan Rule Examples

The dial-plan rules use the following metacharacters:

- { } (braces)—the start and end of the variable area of the pattern
- D—any single digit
- S—any single case-insensitive character
- # (pound)—any number of digits
- * (asterisk)—any number of any characters

Dial plan pattern	Description	Examples
+{DDDDDDDD}	Matches 8 digits with a '+' prefix	+12345678
+{69DDDDDD}, +69{DDDDDD}	Matches 8 digits with a '+69' prefix	+69123456
+{DD812DDD}	Matches 8 digits with a '+' prefix and 812 in positions 4-6	+078129876; +008121234
{DD812#}	Matches a number with 812 in positions 3-5	
sip:{SSSS}@domain.com	Matches any SIP URI that belongs to domain.com with a user part containing exactly 4 characters	sip:andy@domain.com
sip:{SSSS}@{*}	Matches any SIP URI that contains exactly 4 characters in the user part	sip:andy@domain.com; sip:mike@domain01.uk.com
sip:{*}@domain02.uk.com	Matches the SIP URI of any user that belongs to domain02.uk.com	sip:123@domain02.uk.com; sip:alice321@domain02.uk.com;sip:987bob@domain02.uk.com; sip:michael@domain02.uk.com
sip:{#}@domain02.	Matches any SIP URI that belongs to domain02.uk.com and contains only digits in the user part	sip:1234567@domain02.uk.com; sip:987@domain02.uk.com

Dial plan pattern	Description	Examples
uk.com		
tel:{#}	Matches any TEl URI that contains digits only	tel:012345; tel:987
tel:+{#}	Matches any TEl URI that with a '+' prefix followed by digits only	tel:+012345; tel:+987
sip:SSS{SSSS}@domain.com	Matches any SIP URI that belongs to domain.com and has a user part containing 'SSS' and 4 additional characters	sip:SSSabcd@domain.com; sip:sssdcba@domain.com;
sip:SSS{DDDD}@domain.com	Matches any SIP URI that belongs to domain.com and has a user part containing 'SSS' followed by 4 digits	sip:SSS1234@domain.com; sip:sss4321@domain.com;

AttributeExtensions

Key: **calling-method**
Values: `b2b, dialout`
Description: To provide B2BUA ability for the following requests:

- TMakeCall
- TRouteCall
- TSingleStepConference
- TSingleStepTransfer
- TInitiateConference
- TInitiateTransfer

Providing an Extension key with value `b2b` in one of the above T-Library requests ensures that the call is made using B2BUA regardless of any dial plan. In addition, EventRinging generated by T-Server for a destination that is called by the B2BUA method will contain the same key-value pair in AttributeExtensions.

Providing the Extension key with value `dialout` in one of the above T-Library requests ensures that the call is dialed out directly regardless of any dial plan.

Key: **CPNDigits**
Value: A valid SIP URI
Description: To provide caller ID information to the destination of the B2BUA call in the following requests:

- TMakeCall
- TRouteCall
- TSingleStepConference
- TSingleStepTransfer
- TInitiateConference
- TInitiateTransfer

Providing this Extension key in any of the above T-Library requests in a B2BUA call overrides any value configured in the T-Server option cpn.

Event Flow Diagrams for Typical Destinations

Event flow in scenario "User A makes a call to User B’s Tel URI"

A	B SIP URI
Make Call to B’s Tel URI	
EventDialing	
EventNetworkReached	
	Answer Call
	EventRinging
Established	Established

PARTY A	PARTY B
Make Call to B's Tel URI (TMakeCall)	

PARTY A	PARTY B
EventDialing ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B Tel URI** OtherDNRole **Destination** CallState **Ok**	
EventNetworkReached ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B Tel URI** OtherDNRole **Destination** CallState **Ok**	
	Answer Call
	EventRinging ConnID **1** ThisDN **B SIP URI** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Forwarded**
EventEstablished ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B SIP URI** OtherDNRole **Destination** CallState **Ok**	**EventEstablished** ConnID **1** ThisDN **B SIP URI** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Ok**

Event flow in scenario "User B has forwarded calls to User C, User A makes a call to User B"

A	B	C
Make Call to B		
EventDialing		
	EventRinging	
		Answer Call
	EventReleased	EventRinging
EventEstablished		EventEstablished

PARTY A	PARTY B	PARTY C
Make Call to B's Tel URI (TMakeCall)		
EventDialing ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B** OtherDNRole **Destination** CallState **Ok**		
	EventRinging ConnID **1** ThisDN **B** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Ok**	
		Answer Call
	EventReleased ConnID **1** ThisDN **B** ThisDNRole **Destination**	**EventRinging** ConnID **1** ThisDN **C** ThisDNRole **Destination**

PARTY A	PARTY B	PARTY C
	ThirdPartyDN **C** ThirdPartyDNRole **DeletedBy** CallState **Forwarded**	OtherDN **A** OtherDNRole **Origination** CallState **Forwarded**
EventEstablished ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **C** OtherDNRole **Destination** CallState **Ok**		**EventEstablished** ConnID **1** ThisDN **C** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Ok**

Event flow in scenario "User A makes a call to Response Group B, the call delivered to Response Group Member C"

A	B	C
Make Call to B		
EventDialing		
	EventRinging	
EventEstablished	EventEstablished	
		SfB Client Accept Call
		EventRinging
EventPartyAdded	EventPartyAdded	
		EventEstablished
	EventReleased	
EventPartyDeleted		EventPartyDeleted

PARTY A	PARTY B	PARTY C
Make Call to B (TMakeCall)		

PARTY A	PARTY B	PARTY C
EventDialing ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B** OtherDNRole **Destination** CallState **Ok**		
	EventRinging ConnID **1** ThisDN **B** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Ok**	
EventEstablished ConnID **1** ThisDN **A** ThisDNRole **Origination** OtherDN **B** OtherDNRole **Destination** CallState **Ok**	**EventEstablished** ConnID **1** ThisDN **B** ThisDNRole **Destination** OtherDN **A** OtherDNRole **Origination** CallState **Ok**	
		SfB Client Accept Call
		EventRinging ConnID **1** ThisDN **B** ThisDNRole **ConfMember** CallState **Ok**
EventPartyAdded ConnID **1** ThisDN **A** ThisDNRole **ConfMember** OtherDN **C**	**EventPartyAdded** ConnID **1** ThisDN **B** ThisDNRole **ConfMember** OtherDN **C**	

PARTY A	PARTY B	PARTY C
OtherDNRole **NewParty** ThirdPartyDN **B** ThirdPartyDNRole **AddedBy** CallState **Conferenced**	OtherDNRole **NewParty** ThirdPartyDN **B** ThirdPartyDNRole **AddedBy** CallState **Conferenced**	
		EventEstablished ConnID **1** ThisDN **C** ThisDNRole **ConfMember** CallState **Conferenced**
	EventReleased ConnID **1** ThisDN **B** ThisDNRole **ConfMember** CallState **Ok**	
EventPartyDeleted ConnID **1** ThisDN **A** ThisDNRole **ConfMember** OtherDN **B** OtherDNRole **Deleted** ThirdPartyDN **B** ThirdPartyDNRole **DeletedBy** CallState **Ok**		**EventPartyDeleted** ConnID **1** ThisDN **C** ThisDNRole **ConfMember** OtherDN **B** OtherDNRole **Deleted** ThirdPartyDN **B** ThirdPartyDNRole **DeletedBy** CallState **Ok**

Feature Limitations

- The B2BUA method cannot be applied to direct 1pcc calls to agents. That is, if an agent has forwarding or simultaneous ringing configured, an incoming direct call to that agent cannot be answered and will be immediately cleared.
- The B2BUA method cannot be applied to 1pcc Single-Step Conference that is performed from a Skype for Business Client.

- The B2BUA method cannot be applied to an established originator leg during a TMakeCall request.
- The B2BUA method cannot be applied to any Call Supervision or Supervisor Assistance scenarios.
- This feature does not affect Remote Treatments and Remote Recording functionalities.
- If an agent using Workspace Desktop in suppression mode receives a forwarded call, there is no toast for this call to the agent. Therefore, Genesys does not recommend that you use B2BUA calls in an environment where agents are using Workspace Desktop in suppression mode.

Response Groups Limitations

- If CPN Digits are used for a B2BUA call to a Response Group that contains DNs handled by Genesys components, the call cannot be answered by a Response Group member agent.
- If CPN Digits are used for a B2BUA call to a Response Group that contains Skype for Business users not handled by Genesys components, the CPN Digits are displayed in the ringing toast, but the CPN Digits are replaced in the Skype for Business conversation window with the conference service portal name after the call is answered.
- The B2BUA feature is available only for Response Groups where either all the users are monitored by Genesys, or all the users are not monitored by Genesys. The Response Group members cannot be a mixture of users monitored by Genesys components and users not monitored by Genesys components.