Composer Help

Transfer Block

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Transfer Block

Contents

Use the Transfer block to transfer the call to another destination. By default, blind transfer is enabled, and it has no outports. However, if you enable bridging, the block will have one or more outports. In case of user input blocks (Menu, Input, Record, Transfer), Composer adds a global variable of type "Block" to the variables list. You can conveniently use this variable for accessing the user input value. The Transfer block has the following properties:

Transfer Block Exception Events

The Transfer block has the following exception events as described in Exception Event Descriptions:

- connection.disconect.hangup
- connection.disconnect.transfer (supported by default)
- error (supported by default)
- error.connection.baddestination (supported by default)
- error.connection.noauthorization
- error.connection.noresource
- error.connection.noroute
- error.connection
- error.unsupported.transfer.blind
- error.unsupported.transfer.consultation
- error.unsupported.uri

Name Property

Please find this property's details under Common Properties.

Block Notes Property

Can be used for both callflow and workflow blocks to add comments.

Exceptions Property

Find this property's details under Common Properties.

Language Property

The language set by this property overrides any language set by the Set Language block, the Project preferences, or the incoming call parameters. The property takes effect only for the duration of this block, and the language setting reverts back to its previous state after the block is done. In the case of the Transfer block, this property affects the language of grammars used for ASR input:

1. Click under Value to display a down arrow.
2. Click the down arrow and select English - United States (en-US) or the variable that contains the language.

Condition Property

Find this property's details under Common Properties for Callflow Blocks.

Logging Details Property

Find this property's details under Common Properties for Callflow Blocks.

Log Level Property

Find this property's details under Common Properties for Callflow Blocks.

Enable Status Property

Find this property's details under Common Properties for Callflow Blocks.

Output Result Property

You must use the Output Result property to assign the collected data to a user-defined variable for further processing. Note: This property is mandatory. You must select a variable for the output result even if you do not plan on using the variable. If this is not done, a validation error will be generated in the Problems view.

1. Select the Output Result row in the block's property table.
2. In the Value field, click the down arrow and select a variable.

For more information, see Upgrading Projects/Diagrams.

Transfer Audio Property

The optional Transfer Audio property plays a prompt to the end user while the number is being dialed out. You provide the URI of the audio source to play while the transfer attempt is in progress (before the other end answers). If the callee answers, the interpreter terminates playback of the recorded audio immediately. If the end of the audio file is reached and the callee has not yet answered, the interpreter plays the audio tones from the far end of the call (ringing, busy). If the resource cannot be fetched, the error is ignored and the transfer continues. To provide a Transfer Audio value:

1. Select the Transfer Audio row in the block's property table.
2. In the Value field, type a Transfer Audio value URI (HTTP or RTSP) specifying the location of the audio file to play.

Aai Property

Use the optional Application-to-Application Information (the Aai property) for the data that is to be transferred from the current application to another application. Use this option to transfer the call to a number that initiates another voice application. To assign a value to the Aai property:

1. Select the Aai row in the block's property table.
2. In the Value field, select a value from the drop-down list.

Values are the Voice Application Variables described under the Variables Property.

Authorization Code Property

GVP supports dialing of an authorization code as part of an outbound call on a two-leg transfer. Use free form text to specify the authorization code in the application.

Connect Timeout Property

Use the Connect Timeout property for the connection timeout value. The default is 15 seconds. For information on what happens if a timeout occurs, select Help > Contents and see the GVP 8.1Voice XML 2.1 Reference Help. Specifically see Standard VoiceXML > Variables > Transfer, attribute connecttimeout.   To provide a timeout value:

1. Select the Connect Timeout row in the block's property table.
2. In the Value field, type a timeout value, in seconds.

Connect When Property

This property controls when the call is connected to the end point. To assign a value:

1. Select the Connect When row in the block's property table.
2. In the Value field, select answered or immediate from the drop-down list.

Destination Property

The Destination property contains the destination phone number. The destination number can be one of the following:

- A Virtual Route point number on which the IRD Strategy is loaded
- Extension number of an Agent
- External number

The value must be specified in one of the formats below:

- sip:[user@]host[:port]
- tel:phonenumber e.g., tel:+358-555-1234567

For information on this property, select Help > Contents and see the GVP 8.1Voice XML 2.1 Reference Help. Specifically see Standard VoiceXML > Variables > Transfer, attribute dest. To assign a value to the Destination property:

1. Select the Destination row in the block's property table.
2. In the Value field, select a value from the drop-down list.

Values are the Voice Application Variables described under the Variables Property.

Max Call Duration Property

Use the Max Call Duration property for the maximum call duration. The default is 3600 seconds. (This is not supported for Consultation Transfer Type.) Note: If this is set to 0 (zero), an infinite value is supplied, and there is no upper limit to the call duration. To provide a value for the maximum call duration:

1. Select the Max Call Duration row in the block's property table.
2. In the Value field, type a value for the maximum call duration.

Transfer Type Property

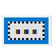
Specifies the type of the Transfer, which determines whether or not the caller's session with the VoiceXML interpeter resumes after the call initiated by the transfer ends. Note: Composer also supports AT&T blind transfers with the following options: Out of Band Courtesy, Out of Band Consult, and Out of Band Conference. For more information on these options, start with the GVP 8.1 Voice XML Reference Help (Help > Contents). Search for ATTOOBCOURTESY, ATTOOBCONSULT, and ATTOOBCONFERENCE (Transfer topic). Also see the Genesys Voice Platform 8.1 Deployment Guide. To assign a value to the Transfer Type property:

1. Select the Transfer Type row in the block's property table.
2. In the Value field, select one of the following from the drop-down list:

Blind is the default setting. The platform redirects the caller to the agent without remaining in the connection, and it does not monitor the outcome. Once the caller is handed off to the network, the caller's session with the VoiceXML application cannot be resumed. The VoiceXML interpeter throws a connection.disconnect.transfer immediately, regardless of whether the transfer was successful or not. **Bridge** causes the platform add the agent to the connection. Document interpretation suspends until the transferred call terminates. The platform remains in the connection for the duration of the transferred call; listening during transfer is controlled by any included <grammar>s.If the caller disconnects by going onhook or if the network disconnects the caller, the platform throws a connection.disconnect.hangup event. If the agent disconnects, then transfer outcome is set to near_end_disconnect and the original caller resumes her session with the VoiceXML application. **Consultation** causes the consultation transfer to be similar to a blind transfer except that the outcome of the transfer call setup is known and the caller is not dropped as a result of an unsuccessful transfer attempt. When performing a consultation transfer, the platform monitors the progress of the transfer until the connection is established between caller and agent. If the connection cannot be established (e.g. no answer, line busy, etc.), the session remains active and returns control to the application. As in the case of a blind transfer, if the connection is established, the interpreter disconnects from the session, connection.disconnect.transfer is thrown, and document interpretation continues normally. Any connection between the caller and the agent remains in place regardless of document execution. Note: The selected transfer type will work only if the platform is provisioned to support that type of transfer.

Variables Property

This is the list of variables that can be optionally sent by the application as part of the Transfer Request to the far end. It corresponds to the signalvars extension attribute of the NGI VXML Interpreter. Check the NGI VXML Reference Guide for more details. All variables that are selected (checked) will be sent as part of the signalvars . The name of the variable will be used as the key name and the actual value will be the corresponding value. Note: Refer to the GVP Documentation for details on the signalvars attribute. The variable name must match the name of the key that will be sent as signalvars . To declare session variables for the application or subcallflow:

1. Click the Variables row in the block's property table.
2. Click the  button to open the Variables dialog box.
3. Select individual variables, or click Select all or Deselect all.

4. Click OK.

Method Property

The Method property specifies the type of SIP transfer method that the Media Control Platform (MCP) uses. To assign a value to the Method property:

1. Select the Method row in the block's property table.
2. In the Value field, select one of the following from the drop-down list (descriptions below):

Bridge

A Bridge method indicates that the Media Control Platform (MCP) bridges the media path.

1. The platform sends an INVITE request to the callee, and a dialog is established between the callee and the platform.
2. The transfer fails if a non-2xx final response is received for the INVITE request.

This is a two-leg transfer (in other words, it occupies two channels on the platform). The platform stays in the signaling path and is responsible for bridging the two call legs.

Hkf (Hookflash)

A Hookflash method indicates a transfer using DTMF digits (RFC 2833).

1. The Media Control Platform (MCP) sends DTMF digits on the media channel. The platform leaves it to the media gateway or switch to perform the transfer on the network.
2. Configurable options enable you to specify whether the call will be disconnected by the platform or by the remote end. Otherwise, the call is disconnected after a configured timeout.

This is a one-leg transfer (in other words, it occupies only one channel on the platform).

Refer

A Refer method indicates that the transfer is based on a SIP REFER message (RFC 3515).

1. The platform sends a REFER request to the caller, with the callee (as specified in the VoiceXML application) in the Refer-To: header.
2. The transfer fails if a non-2xx final response is received for the REFER.

This is a one-leg transfer (in other words, it occupies only one channel on the platform).

Referjoin

A Referjoin method indicates a consultative REFER transfer (RFC 3891).

1. The platform sends an INVITE request to the callee, and a dialog is established between the callee and the platform.
2. The platform also sends a REFER request to the caller, with the callee's information in the Replaces header.
3. The platform considers the transfer to be successful if it receives a BYE from the caller after a 2xx response for the REFER.
4. The transfer fails if a non-2xx final response is received for the INVITE request or for the REFER request.

This is a two-leg, or join-style, transfer (in other words, it occupies two channels on the platform).

Mediaredirect

A Mediaredirect method indicates a media redirection transfer. The Media Control Platform (MCP) uses SIP to handle call control between the caller and the callee, and the RTP media channel is connected directly between the caller and callee.

1. The platform sends an INVITE request to the callee without SDP.
2. If the transfer is proceeding, the callee responds with a 200 OK that includes an SDP offer.
3. The platform forwards the SDP offer in a re-INVITE request to the caller.
4. The caller responds with a 200 OK that includes the SDP answer.
5. The platform forwards the SDP answer to the callee in an ACK response.
6. The transfer fails if a non-2xx final response is received for the initial INVITE request.

This is a two-leg transfer (in other words, it occupies two channels on the platform). attcourtesy attconsult attconference attoobcourtesy attoobconsult attoobconference For information on these methods, consult the section on how the Media Control Platform works in the Genesys Voice Platform 8.1 Deployment Guide.

Disconnect on Answering Machine Property

This property indicates whether or not the FAX / Answering machine has to be detected. To assign a value:

1. Select the Disconnect on Answering Machine row in the block's property table.
2. In the Value field, select true or false from the drop-down list.

Do CPA Analysis Property

This property indicates whether or not the platform is enabled to detect who/what answered the call. To assign a value:

1. Select the Do CPA Analysis row in the block's property table.

2. In the Value field, select true or false from the drop-down list.

Get Shadow Variables Property

Shadow variables (optional) provide a way to retrieve further information regarding the value of an input item. They can provide platform-related information about the interaction/input. For example, for speech recognition, this may be the confidence level the platform receives from the ASR engine about how closely the engine could match the user utterance to specified grammar. By setting this property to true, it will expose the block’s shadow variable within the callflow. When enabled, the shadow variable will be included in the list of available variables. (For example, the Log block’s Logging Details will show Transfer1$.) A shadow variable is referenced as blockname$.shadowVariable, where blockname is the value of the input item's name attribute, and shadowVariable is the name of a specific shadow variable, for example: Transfer1$.duration. To assign a value:

1. Select the Get Shadow Variables row in the block's property table.
2. In the Value field, select true or false from the drop-down list.

Transfer Results Property

There are several types of transfer results supported for applications. When you select a transfer result, a corresponding outport node is added to the block to allow specific actions to be taken for that condition. Please note that a default outport is always present. The default path is executed if none of the selected transfer results are set. The available transfer results are:

- far_end_disconnect (selected by default)
- noanswer (selected by default)
- busy (selected by default)
- near_end_disconnect

Note: Consultation Transfer supports only noanswer, busy, and near_end_disconnect transfer results. To select transfer results:

1. Click the Transfer Results row in the block's property table.
2. Click the  button to open the Transfer Results dialog box.
3. Select items from the list of available CPA results, or click Select all or Deselect all as needed, then click OK.

Input Grammar Dtmf Property

Use the Input Grammar Dtmf (Dual Tone Multi-Frequency) property to specify the DTMF Grammar for the Transfer block, which accepts DTMF signals or speech input from callers. The DTMF Grammar is processed and handled by GVP. In the case of external grammars, this specifies the actual path of the grammar file / resource for DTMF Grammars. This is only valid when the Grammar Type is externalGrammar and Input Mode is dtmf or hybrid. To assign a value to the Input Grammar Dtmf property:

1. Select the Input Grammar Dtmf row in the block's property table.
2. In the Value field, select a value from the drop-down list.

Values are the Voice Application Variables described under the Variables Property. Section 2.3.7.2.1, of the Voice Extensible Markup Language (VoiceXML) Version 2.0 specification (http://www.w3.org/TR/voicexml20/#dml2.3.7.2.1), contains the following information on listening for user input during a transfer (interrupting a transfer): Platforms may optionally support listening for caller commands to terminate the transfer by specifying one or more grammars inside the <transfer> element. The <transfer> element is modal in that no grammar defined outside its scope is active. The platform will monitor during playing of prompts and during the entire length of the transfer connecting and talking phases:

- DTMF input from the caller matching an included DTMF grammar
- an utterance from the caller matching an included speech grammar

A successful match will terminate the transfer (the connection to the callee); document interpretation continues normally. An unsuccessful match is ignored. If no grammars are specified, the platform will not listen to input from the caller. The platform does not monitor in-band signals or voice input from the callee.

Input Grammar Voice Property

Use the Input Grammar Voice property to specify the Voice Grammar for the Input block, which accepts DTMF or speech input from callers. If you are writing hybrid applications that allow both DTMF and Speech input, specify both the DTMF and Voice grammars. The Voice grammar is sent to the ASR Engine for processing, whereas the DTMF grammar is processed by GVP. As a result, you need two separate grammars for Voice and DTMF in the case of hybrid applications that allow both Voice and DTMF inputs. In the case of external grammars, this specifies the actual path of the grammar file / resource for ASR Grammars.. This is only valid when Grammar Type is externalGrammar and Input Mode is voice or hybrid. To assign a value to the Input Grammar Voice property:

1. Select the Input Grammar Voice row in the block's property table.
2. In the Value field, select a value from the drop-down list.

Values are the Voice Application Variables described under the Variables Property.

Input Mode Property

To assign a value to the Input Mode property:

1. Select the Input Mode row in the block's property table.
2. In the Value field, select one of the following from the drop-down list (descriptions below):

DTMF

The DTMF format indicates the menu option mode of input will be via the telephone keypad.

Voice

The Voice format indicates the menu option mode of input will be a voice phrase.

Hybrid

The Hybrid menu mode will handle both DTMF and Voice inputs, that is via telephone keypad and voice phrase.