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## Composer Help

ICM Route Request Block

# ICM Route Request Block

## Contents

- [1 ICM Route Request Block](#)
  - [1.1 Name Property](#)
  - [1.2 Block Notes Property](#)
  - [1.3 Exceptions Property](#)
  - [1.4 Interaction Data Property](#)
  - [1.5 Output Result Property](#)
  - [1.6 Aai Property](#)
  - [1.7 Transfer Audio Property](#)
  - [1.8 Connect Timeout Property](#)
  - [1.9 Connect When Property](#)
  - [1.10 Destination Property](#)
  - [1.11 Max Call Duration Property](#)
  - [1.12 Transfer Type Property](#)
  - [1.13 Method Property](#)
  - [1.14 Do CPA Analysis Property](#)
  - [1.15 Get Shadow Variables Property](#)
  - [1.16 Transfer Result Property](#)
  - [1.17 Input Grammar Dtmf Property](#)
  - [1.18 Input Grammar Voice Property](#)
  - [1.19 Input Mode Property](#)
  - [1.20 Condition Property](#)
  - [1.21 Logging Details Property](#)
  - [1.22 Log Level Property](#)
  - [1.23 Enable Status Property](#)

ICM refers to a Cisco product called Intelligent Contact Management, which provides intelligent routing and Computer Telephony Integration (CTI). You can use the GVP ICM Adapter in VoiceXML applications when invoking services, responding to requests, and sharing data. Use the ICM Route Request block to transfer a call to ICM. Note: This block functions in the same way as the existing [Route Request block](#). Composer uses the VXML <transfer> tag to implement the ICM Route Request functionality.

- For information on ICM Support and variables, see the figure in topic [Project Properties dialog box](#).

The ICM Route Request block has the following properties:

### Name Property

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](#). The following events are supported:

- connection.disconnect.hangup
- connection.disconnect.transfer
- error
- error.connection.noauthorization
- error.connection.baddestination
- error.connection.noresource
- error.connection.noroute
- error.connection
- error.unsupported.transfer.blind
- error.unsupported.transfer.consultation
- error.unsupported.uri
- error.com.genesyslab.composer.unsupported

[Custom events](#) are also supported.

## Interaction Data Property

To select session variables:

1. Click the Interaction Data row in the block's property table.
2. Click the  button to open the Interaction Data dialog box.
3. Select individual global **variables**, or click Select all or Deselect all.
4. Click OK.

## 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.

## 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.

## 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.

## Connect Timeout Property

Use the Connect Timeout property for the connection timeout value. The default is 30 seconds. 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 whether the connection is made after the call is picked up, or immediately. Select one of the following:

- Immediate
- Answered

## 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 in the Entry block.

## Max Call Duration Property

Use the Max Call Duration property for the maximum call duration. Default value is 0. This property is not supported for Consultation and Blind transfer types. 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

The Transfer Type property specifies the type of transfer required. 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

This 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 interpreter throws a `connection.disconnect.transfer` immediately, regardless of whether the transfer was successful or not.

### Bridge

The platform adds 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

The consultation transfer is 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.

## Method Property

The Method property specifies the type of route request required. 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:

## 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*.

## Do CPA Analysis Property

Triggers whether the platform will detect who or what answered the call. Select one of the following:

- True
- False (default, no detection)

## Get Shadow Variables Property

Shadow variables provide a way to retrieve further information regarding the value of an input item. 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 `RouteRequest1$`.) A shadow variable is referenced as `blockname$.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: `RouteRequest1$.duration`. To assign a value to the Get Shadow Variables property:


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 Result Property

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.

For each item selected, an output node is added to allow specific actions to be taken for that condition.

## Input Grammar Dtmf Property

Use the Input Grammar Dtmf property to specify the DTMF Grammar for the Input Block. 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.

## Input Grammar Voice Property

Use the Input Grammar Voice property to specify the Voice Grammar for the Input block. 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:

### 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.

## Condition Property

Find this property's details under [Common Properties for Callflow Blocks](#) or [Common Properties for Workflow Blocks](#).

## Logging Details Property

Find this property's details under [Common Properties for Callflow Blocks](#) or [Common Properties for Workflow Blocks](#).

## Log Level Property

Find this property's details under [Common Properties for Callflow Blocks](#) or [Common Properties for Workflow Blocks](#).

## Enable Status Property

Find this property's details under [Common Properties for Callflow Blocks](#) or [Common Properties for Workflow Blocks](#).