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GENESYS™

# Interaction SIP Bulk Caller

## Technical Reference

### Abstract

This document shows how to install, configure, and use Interaction SIP Bulk Caller, a utility that enables you to test your server by making or receiving (in bulk) large numbers of calls. It also simulates station registrations for testing.

For the latest version of this document, see the PureConnect Documentation Library at: http://help.genesys.com/cic.

# Table of Contents

# Overview

Interaction SIP Bulk Caller is a utility that enables you to test your CIC server by making or receiving (in bulk) large numbers of calls. It also simulates station registrations for testing.

Though it is normally used with a CIC server, Interaction SIP Bulk Caller does not require it and can be used with other servers.

# System Requirements

Interaction SIP Bulk Caller requires:

- Windows 7 or higher version
- MSXML 6.0 Parser

Note: Do not install Interaction SIP Bulk Caller on the same machine where SIP Proxy, CIC server, or any CIC server-related component (which uses the SIP Engine) is installed.

# Glossary

**$SBC_ROOT:** The folder where you installed Interaction SIP Bulk Caller.

**Callset:** A callset is a configuration and set of actions to use for each call. The callset XML configuration file describes it. There are two types of callsets:

- Use an outbound callset to place outbound calls to remote machines. Outbound callset is listed in the OutboundConfig section of the main XML configuration file.
- Use an inbound callset to respond to calls received by Interaction SIP Bulk Caller. Inbound callset is listed in the InboundConfig section of the main XML configuration file.

**SIPBulkCaller:** Interaction SIP Bulk Caller works both as an outbound caller and as an inbound receiver. When used to make calls, it is often referred to as SIPBulkCaller. When used to receive calls, it is often referred to as SIPBulkReceiver.

**SIPBulkReceiver:** Interaction SIP Bulk Caller works both as an outbound caller and as an inbound receiver. When used to make calls, it is often referred to as SIPBulkCaller. When used to receive calls, it is often referred to as SIPBulkReceiver.

**Stationset:** A stationset is a set of stations that Interaction SIP Bulk Caller requires to register dynamically to your CIC server. If some or all stations are configured for dynamic registration with CIC Server, use a stationset to simulate registration and authentication of those stations from Interaction SIP Bulk Caller.

# Installing Interaction SIP Bulk Caller

This subject includes:
- Upgrading Interaction SIP Bulk Caller
- Installing Interaction SIP Bulk Caller
- Uninstalling Interaction SIP Bulk Caller

## Upgrading Interaction SIP Bulk Caller

To upgrade Interaction SIP Bulk Caller:
1. Uninstall the old version of Interaction SIP Bulk Caller. For specific instructions, see Uninstalling Interaction SIP Bulk Caller.
2. Install the new version. For specific instructions, see Installing Interaction SIP Bulk Caller.

## Uninstalling Interaction SIP Bulk Caller

To uninstall Interaction SIP Bulk Caller:
1. In case you need them in the future, back up the files:

- SIPBulkCallerConfig.xml
- SampleCallSet.xml
- SampleStationSet.xml
- DefaultInboundCallSet.xml files

Uninstalling Interaction SIP Bulk Caller deletes those files. However, it does not delete your audio files and callset files.

2. Stop the Interaction SIP Bulk Caller service in Windows.

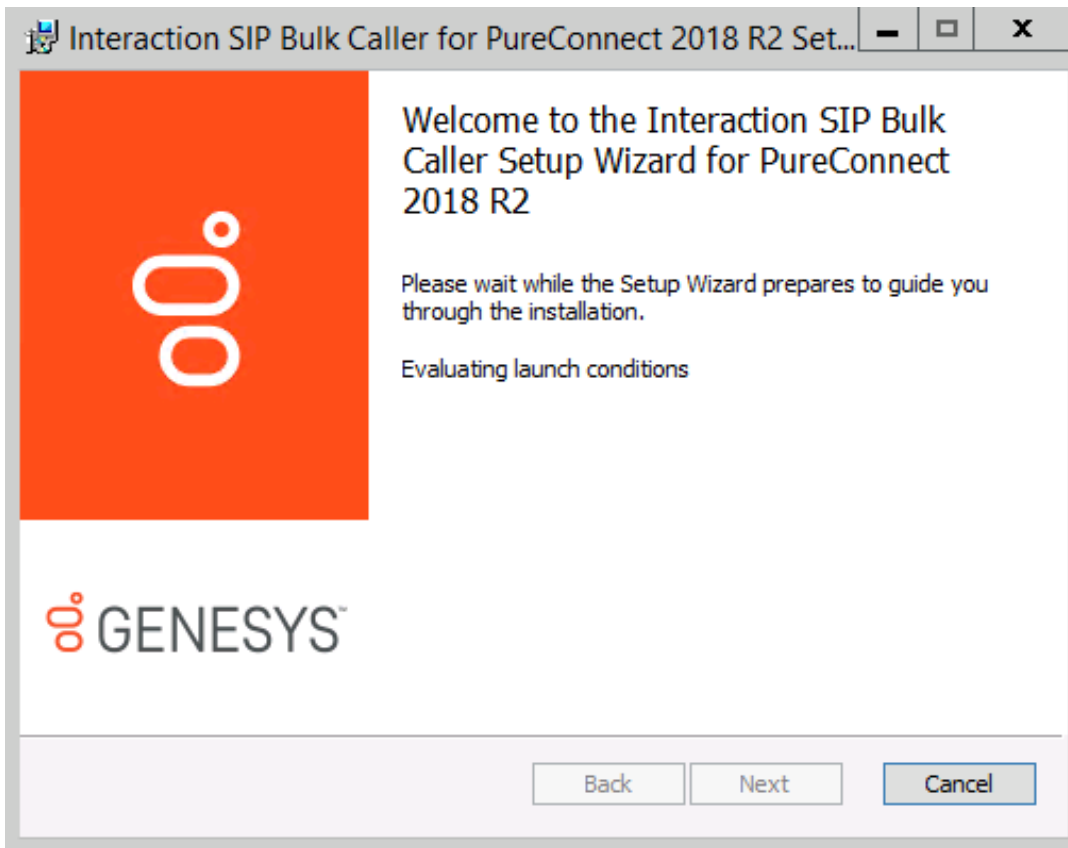   The name of the service is *Interaction SIP Bulk Caller*.

3. On the Windows Control Panel:
   a. Start Add/Remove Programs.
   b. Select Interaction SIP Bulk Caller.
   c. Click Remove.

Windows uninstalls Interaction SIP Bulk Caller from your system.

# Installing Interaction SIP Bulk Caller software

To install the software:

1. Download the setup file (InteractionSIPBulkCaller.msi) from "Utilities and Downloads" on the CIC Product Information site at https://my.inin.com/products/cic/pagees/default.aspx

2. Browse to and double-click the setup file.



3. Work through the first two screens and then click the **Install** button.

4. Once the installation is complete, click the **Finish** button.

The setup program installs Interaction SIP Bulk Caller as a Windows service. The name of the service is *Interaction SIP Bulk Caller*.

The setup program also installs a copy of this technical reference in the following folder:

$SBC_ROOT\UserGuide

The setup program creates two shortcuts that link to this technical reference. The first shortcut is on the Windows Desktop. The second is on the Windows Start menu under Program Files | Genesys | Interaction SIP Bulk Caller | User Guide.

After you have installed the program, you need to apply the license.

# Applying the License

After installing Interaction SIP Bulk Caller, you must apply the Interaction SIP Bulk Caller license you obtained from Genesys. To apply the license:

1. Copy the license file to any local folder.

   Genesys recommends the folder $SBC_ROOT\Resources, where you installed Interaction SIP Bulk Caller (see the section Abbreviations).
2. Start the Interaction SIP Bulk Caller service.
3. Log on to the SIPBulkCaller web interface (http://localhost:8080).

   The default user name is **admin** and **1234** is the default password.
4. Select the **Configuration** tab and then select **License**.



5. Click **Choose File**, then browse to and select the license file.
6. Click **Submit** to update Interaction SIP Bulk Caller's license data.

   Interaction SIP Bulk Caller shows the license details.

**SIP Bulk Caller**

## Configuration

Outbound CallSets

Inbound CallSets

StationSets

Add CallSet/StationSet

SIPLine Config

Audio Config

License

Administration

**License Configuration**

| | |
|---|---|
| License Status | License expires in 7.5 days. |
| License File | C:\Program Files (x86)\Interactive Intelligence\SIP Bulk Caller\Resources\SIPBulkCallerLicense_ININ_20180321081841578.xml |
| Product | I3_FEATURE_INTERNAL_SIP_BULK_CALLER |
| Outbound CallSets | Allowed |
| Inbound CallSets | Allowed |
| StationSets | Allowed |
| Max Outbound CallSets | No Limit |
| Max Active Calls per CallSet | No Limit |
| Max CallRate | No Limit |
| Max G729 Sessions | No Limit |

Add/Update License

**Note:** On the Status tab, you can click **About** to view details of Interaction SIP Bulk Caller version, build date, and local machine information.

# Configuring and Using the Software

Topics covering this subject:

- Adding a Callset or Stationset
- Editing, Reloading, or Deleting a Callset or Stationset
- Edit SIP Line Configuration
- Edit Audio Configuration
- Start or Stop Callset or Stationset
- Generating Security Certificates for TLS
- Location of Log Files
- Troubleshooting

# Adding a Callset or Stationset

For inbound Callset, outbound callset, and stationset, you must create the XML configuration file. For more information, see:

- Inbound Callset
- Outbound Callset
- Stationset

> Note: Default files for inbound callset, outbound callset, and stationset are included with the installation. You can use the inbound callset file as is. You must modify the outbound callset and stationset files for your specific site characteristics.

To add a callset or stationset:

1. Create the required configuration file:

- For an inbound callset, create the inbound callset configuration file. Put it and all audio files in the Interaction SIP Bulk Caller installation resources folder.

- For an outbound callset, create the outbound callset configuration file. Put it and all audio files in the Interaction SIP Bulk Caller installation resources folder.

- For a stationset, create the stationset configuration file. Put it in the Interaction SIP Bulk Caller installation resources folder.

2. Start Interaction SIP Bulk Caller, then:
   b. Display the **Configuration** page.
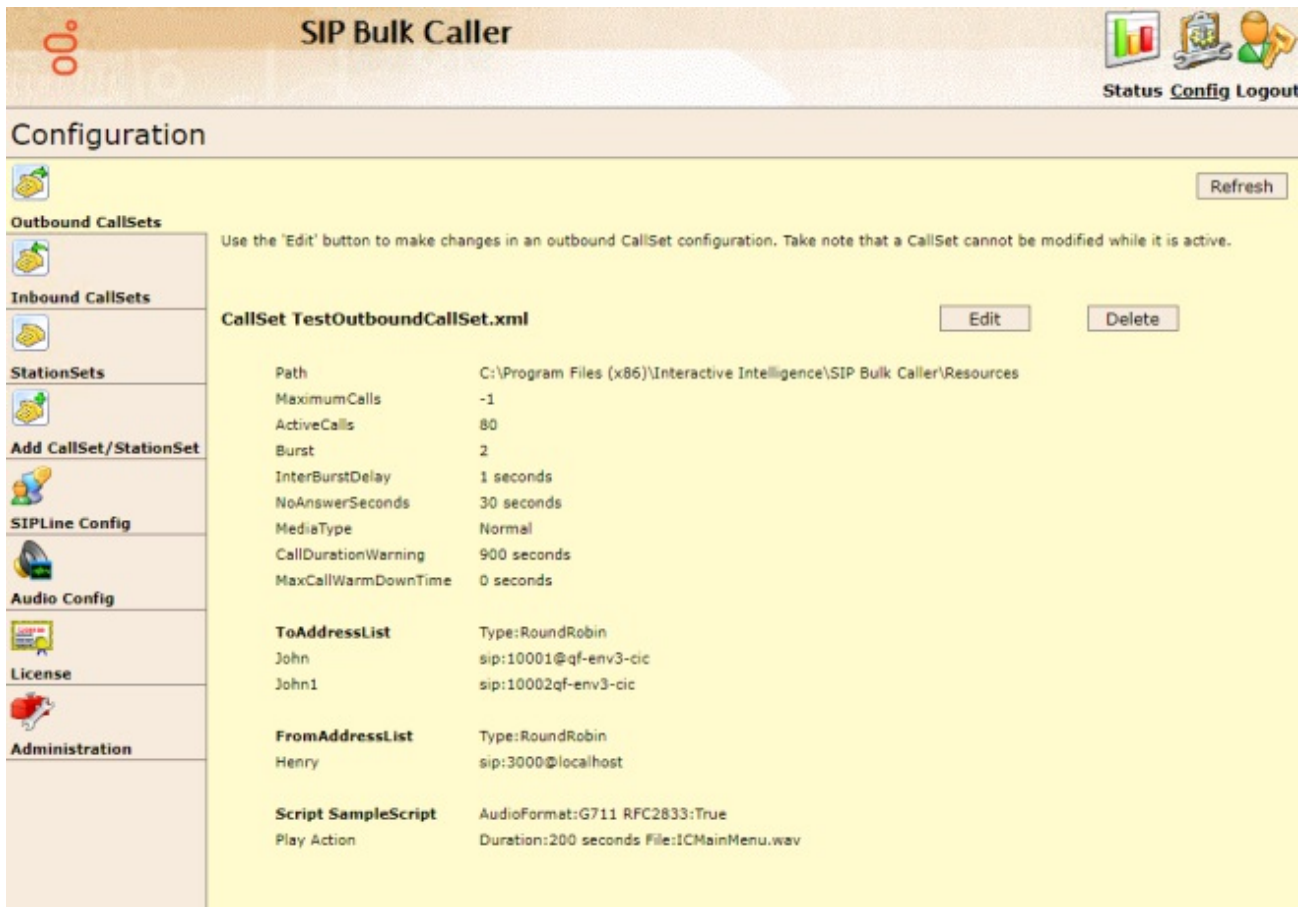   c. Click **Add CallSet/StationSet**.



3. Select **Inbound CallSet**, **Outbound CallSet**, or **StationSet**, then input or browse to the name of the callset or stationset.
4. Click **Submit**.

Interaction SIP Bulk Caller adds your new callset or stationset.

# Editing, Reloading, or Deleting a Callset or Stationset

To edit, reload, or delete a callset or stationset:

1. Display the **Configuration** page, then click the tab for the desired type of callset or stationset.



Each tab displays configuration information, such as the name and path of the configuration file. It also has Edit, Delete, and Refresh buttons.

2. To make changes in the configuration file for the callset or stationset:

   a. Click **Edit**.

   Interaction SIP Bulk Caller displays the XML configuration in a text editing box.

   b. Make any desired changes in the configuration.

   c. Click **Submit**.

   If an error occurs, then Interaction SIP Bulk Caller displays an error message.

3. To delete the callset or stationset, click **Delete**.

   Interaction SIP Bulk Caller unloads the callset or stationset. However, it does not delete the configuration file for the callset or stationset.

# Edit SIP Line Configuration

The SIP Line Configuration page shows current SIP line configuration. You may change protocol, port number and other SIP line-specific elements. Since Interaction SIP Bulk Caller has only one SIP line, you do not need to define the name of the SIP line.

**SIP Bulk Caller**

## Configuration

Refresh

**Outbound CallSets**

**Inbound CallSets**

**StationSets**

**Add CallSet/StationSet**

**SIPLine Config**

**Audio Config**

**License**

**Administration**

**SIP Line Configuration**

Make changes to the SIP Line configuration. Stop all CallSets/StationSets prior to making changes in SIP line configuration.

| | |
|---|---|
| Network Interface | Ethernet (172.26.23.182) ▼ |
| IP Family | IPv4 ▼ |
| Domain Name | inin.com |
| Protocol | UDP ▼ |
| Port Number | 5060 |
| SIP QoS DSCP Value | 18 (24, 011000) CS3 ▼ |
| TCP Connect Timer (ms) | 3500 |
| UDP T1 Timer (ms) | 500 |
| UDP T2 Timer (ms) | 4000 |
| Max Packet Retry | 10 |
| Max Invite Retry | 6 |
| Fax Protocol | T.38 ▼ |
| Enable PRACK | NO ▼ |

Root Certificate File, Line Certificate File and Line PrivateKey File are required when TLS protocol is selected. Obtain certificate files from appropriate authority. Browse to your certificate files below.

**Root Certificate File**          Choose File  No file chosen
**Currently Selected File**

**Line Certificate File**          Choose File  No file chosen
**Currently Selected File**

**Line PrivateKey File**          Choose File  No file chosen
**Currently Selected File**

Submit     Reset

The table explains each element of the SIP Line Configuration page.

| Element | Explanation |
|---|---|
| NetworkInterface | The network connection name on your local machine. The default is Local Area Connection. If you have more than one network interface, or if the name of the network interface is different from the default, you can change the name here. Unless the correct name is set, the SIP engine will not start. |
| IP Family | The family of protocols used. For more information, see http://wiki.wireshark.org/InternetProtocolFamily. |
| DomainName | The domain name for unique identification. |
| Protocol | The protocol to use for SIP communication. The default protocol is UDP. Available protocols are UDP, TCP, and TLS. |
| PortNumber | The port number to bind the SIP line. The default is 5060. |
| SIP QoS DSCP Value | The default DSCP value is 24 (0x18). Select the value from the drop-down list. At the time of installation, Interaction SIP Bulk Caller sets a value of 0 for registry HKEY_LOCAL_MACHINE\ SYSTEM\ CurrentControlSet\ Services\ Tcpip\ Parameters\ DisableUserTOSSetting. This enables Interaction SIP Bulk Caller to send the QoS byte with a SIP message. Any changes to the registry value require reboot of the machine. |
| TCP Connect Timer | This is the timer setting for the TCP protocol. In most cases, the default works fine. |
| UDP T1Timer | This is a timer setting for the UDP protocol. In most cases, the default works fine. |
| UDP T2Timer | This is a timer setting for the UDP protocol. In most cases, the default works fine. |
| Max Packet Retry | This is the maximum number of times Interaction SIP Bulk Caller will retry sending a packet. In most cases, the default works fine. |
| Max Invite Retry | This is the maximum number of times Interaction SIP Bulk Caller will retry sending an invitation. In most cases, the default works fine. |
| Fax Protocol | The fax protocol to use. |
| Enable PRACK | Whether or not to use PRACK as defined in RFC 3262: "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)." |

The following three elements apply only to the TLS (Transport Layer Security) protocol and only work if you select TLS as the SIP protocol. For more information about using TLS, see the section Generating Security Certificates for TLS.

| Element | Explanation |
|---|---|
| RootCertificateFile | The path for the Authority certificate file. |
| CertificateFile | The path for the Certificate file. This certificate must be signed by the Authority Certificate. |
| PrivateKeyFile | The path for the private key file. The certificate file and private key file are generated by the Authority certificate file. |

# Edit Audio Configuration

The Audio Configuration page displays the current audio configuration.



The table explains each element of the Audio Configuration page.

| Element | Explanation |
|---|---|
| RTP Network Interface | The network interface for audio data. The default is Local Area Connection. If you have a dual NIC machine, then you may set up different network interfaces for SIP data and audio data. Set the network interface for SIP data on the SIP Line Configuration page. |
| Play Audio | The default is Yes. If you set this to No, then Interaction SIP Bulk Caller plays no audio, it does not activate the audio source, and it ignores script configuration. This setting is useful in testing of SIP-only messages. |
| Audio Protocol | There are options to play either RTP or SRTP for the audio. In most situations, SRTP works well with the TLS SIP protocol. |
| Enable RTP QoS Tagging | If set to Yes, then RTP packets add Quality of Service (QoS) tagging. |
| RTP QoS DSCP Value | This is the DSCP value for RTP Quality of Service (QoS). The default value is 46 (0x2E). Select the value from the drop-down list. |
| Enable RTCP QoS Tagging | If set to Yes, then RTCP packets will add Quality of Service (QoS) tagging. |
| RTCP QoS DSCP Value | This is the DSCP value for RTCP Quality of Service (QoS). The default value is 46 (0x2E). Select the value from the drop-down list. |
| Audio Engine Selection Algorithm | The default algorithm is FewestGraphs, which works well in most situations. If the system has more than four processor cores and Windows Server 2008, then set the algorithm to LoadWeightedStochastic and set HLKM\ System\ CurrentControlSet\ Services\ Ndis\ Parameters\ MaxNumRssCpus (DWORD) to the number of CPUs you have. This balances the load across all CPUs so your system can handle more calls. |
| Enable RTP Diagnostic Capture | If set to Yes, then Interaction SIP Bulk Caller records the receiving audio stream for each call in HPAA Capture format. It can be used for troubleshooting. |
| RTP Diagnostic Capture Directory | The root directory below which to store diagnostic capture data. Beneath the root directory, Interaction SIP Bulk Caller creates a separate directory for each callset and adds the default directory path to this text box. |

# Start or Stop Callset or Stationset

The Statistics page lets you start or stop callsets and Stationsets. It shows all outbound callsets, inbound callsets, and stationsets loaded into Interaction SIP Bulk Caller. It also displays callset statisticssuch as calls placed, received, and answered.

Once you start a CallSet, you can save of copy of the general or detailed statistics by clicking the Export button at the bottom of the page. The statistics are saved in a CSV file in the Resources folder. When the file is saved, a unique identifier will be appended to the filename. This allows you to export the statistics file multiple times without overwriting previous export files.



To start a callset:
- Click **Start** in the callset's row.

> Note: If your application also receives calls, you must start the appropriate inbound callset. If you do not start the inbound callset, Interaction SIP Bulk Caller will reject inbound calls.

After you start at least one callset, the Statistics page displays buttons to stop active callsets. There are two ways to stop a callset:

## SIP Bulk Caller

### Status

Statistics

About

Refresh statistics automatically every [60 sec ▼] [Refresh]

Click CallSet/StationSet name to view detailed statistics. Once CallSet is started, it shows WarmStop and HardStop button. WarmStop button stops callset, but let currently active calls to finish. HardStop button stops callset and also immediately stops currently active calls.

**Outbound CallSets**

| Name | Placed | Answered | Active | Disconnected | | |
|------|--------|----------|--------|--------------|---|---|
| TestOutboundCallSet.xml | 7126 | 0 | 0 | 0 | HardStop | WarmStop |

**Inbound CallSets**

| Name | Received | Answered | Active | Disconnected | | |
|------|----------|----------|--------|--------------|---|---|
| DefaultInboundCallSet.xml | 0 | 0 | 0 | 0 | Start | |

**StationSets**

| Name | Total | Registered | Requested | Accepted | | |
|------|-------|-----------|-----------|----------|---|---|
| SampleStationSet.xml | 11 | 0 | 0 | 0 | Start | |
| TestStationSet.xml | 1000 | 0 | 2000 | 2000 | Start | |

**MWI Statistics Details**

| Type | Count | Description |
|------|-------|-------------|
| Current MWI Notifications | 0 | The number of MWI notifications received, since the last file write. |
| Aggregrate MWI Notifications | 0 | The number of MWI notifications received, since the SIP Bulk Caller service was started. |
| MWI On SIP Stations | 0 | The number of SIP stations that have the MWI On status. |
| MWI Off SIP Stations | 0 | The number of SIP stations that have the MWI Off status. |
| Total SIP Stations | 0 | The total number of MWI notifications received by the SIP Bulk Caller from the SIP stations. |

Export statistics [Export]

- **HardStop** stops placing and receiving new calls. In addition, it immediately stops all currently placed or received calls.
- **WarmStop** stops placing and receiving new calls. Unlike HardStop, it allows currently active calls to finish.

To view detailed statisticsabout a callset:
1. Click the link for the callset.

   Interaction SIP Bulk Caller displays detailed statistics for the callset.

# SIP Bulk Caller

Status Config Logout

## Status

Refreshing statistics automatically every 60 sec.   [Refresh]

**Statistics**

**About**

Back

### Statistics Details of Outbound CallSet TestOutboundCallSet.xml

|  | Current | Aggregate | Description |
|---|---|---|---|
| Placed | 656 | 7950 | Calls placed by the SIP Bulk Caller. Placed = Answered + UnansweredTotal. |
| Answered | 0 | 0 | Calls answered by a remote party.<br>Answered = Active + LocalDisconnected + RemoteDisconnected. |
| Active | 0 | 0 | Calls answered and currently playing. |
| LocalDisconnected | 0 | 0 | Calls answered and disconnected by the SIP Bulk Caller. |
| RemoteDisconnected | 0 | 0 | Calls answered and disconnected by a remote party. |
| AverageAnswerTime (ms) | 0 | 0 | Average time to receive SIP answer. |
| MaxAnswerTime (ms) | 0 | 0 | Maximum time to receive SIP answer. |
| AverageAnswerTime100 (ms) | 0 | 0 | Average time to receive SIP answer for the last 100 calls. |
| StddevAnswerTime100 (ms) | 0 | 0 | Standard deviation of SIP answertime for the last 100 calls. |
| AverageAudioTime (ms) | 0 | 0 | Average time to receive first RTP packet from a remote party. |
| MaxAudioTime (ms) | 0 | 0 | Maximum time to receive first RTP packet from a remote party. |
| AverageAudioTime100 (ms) | 0 | 0 | Average time to receive first RTP packet from a remote party for the last 100 calls. |
| StddevAudioTime100 (ms) | 0 | 0 | Standard deviation of audio time for the last 100 calls. |
|  |  |  |  |
| UnansweredTotal | 656 | 7950 | UnansweredTotal = WaitingForAnswer + Unanswered + Rejected. |
| WaitingForAnswer | 40 | 40 | Calls placed but not yet answered by a remote party. |
| Unanswered | 297 | 3952 | Calls unanswered either due to remote party not responding or local errors. |
| Rejected | 319 | 3958 | Calls rejected by a remote party. |
| LocalDisUnanswered | 0 | 38 | Calls unanswered and disconnected by the SIP Bulk Caller. |
| RemoteDisUnanswered | 616 | 7872 | Calls unanswered and disconnected by a remote party. |
|  |  |  |  |
| Redirected | 0 | 0 | Calls redirected to another location by a remote party. |
| BlindTransferred | 0 | 0 | Calls blindly transferred to another location by a remote party. |

Aggregate statistics show aggregate counts since service was started or callset/stationset was modified.

Export statistics   [Export]

2. Click the **Back** link at the top left of the page to return to the main page.

# Generating Security Certificates for TLS

If you select the Transport Layer Security (TLS) protocol in SIP Line configuration, then Interaction SIP Bulk Caller needs security certificate files to operate.

To generate security certificates for TLS:

1.  On the CIC server, navigate to the **\I3\IC\Certificates\Lines** folder.
2.  Make a backup of all current line certificates in the folder.

    This step is very important! Do not skip it. You will need the backup in Step 7.
3.  In Windows, open command window:
    a.  OOn the Windows Start menu, click **Run**.
    b.  In the Open text box, type **cmd** and then click **OK**.
4.  IIn the command window, navigate to the **\I3\IC\Server** folder.
5.  In that folder, create a new line certificate by entering:

    GenSSLCertsU.exe -l sipbulkcalleripaddress
6.  Copy the new line certificate file and Private Key file from the **\I3\IC\Certificates\Lines** folder to the $SBC_ROOT\Resources folder on the Interaction SIP Bulk Caller machine.

    These files are CertificateFile and PrivateKeyFile in Interaction SIP Bulk Caller.
7.  On the CIC Server, delete any new certificates and restore the original line certificate files from your backup.
8.  Copy the **ServerGroupCertificate.cer** file from \I3\IC\Certificates\ServerGroup folder to the $SBC_ROOT\Resources folder on the Interaction SIP Bulk Caller machine.

    This would be RootCertificateFile in Interaction SIP Bulk Caller.
9.  Start Interaction SIP Bulk Caller and use the [SIP Line Configuration page](#) to make any required changes.

# Location of Log Files

Interaction SIP Bulk Caller uses same tracelib to write logs as do other Customer Interaction Center components. Logs are in the %ININ_TRACE_ROOT%\inin_tracing directory.

You need to define the **ININ_TRACE_ROOT** environment variable and provide a directory where you want to write logs. If this environment variable is not defined, then Interaction SIP Bulk Caller writes logs in the **%temp%\inin_tracing** directory. You need IninTraceViewer to view the logs.

# Troubleshooting

| Problem | SIP Line fails to activate. |
|---|---|
| Solution | The Network Interface parameter in SIP line configuration must match the name of the network card on the machine. The default value is Local Area Connection. If your network card has a different name or you have more than one network card on the machine, then provide the exact name and submit the changes.<br><br>Make sure some other application is not using the same port specified in the SIP line configuration. |
| Problem | Callset does not load. |
| Solution | Check the format of the audio file. AudioLib supports 8-bit 8 KHz CCITT u-law, 8-bit 8 KHz CCITT a-law, 8-bit 8 KHZ GSM, and 16-bit linear PCM for audio wav files. If your audio wav file is not in one of these formats, then you must convert it before using it in Interaction SIP Bulk Caller. You can use the Windows Sounds Recorder utility (located in Program \| Accessories \| Entertainment) to convert the file format.<br><br>Element names and attribute names in the XML configuration file must match as expected. The XML parser is case-sensitive. |
| Problem | SIP Bulk Caller rejects a callset, stationset, license, or certificate file. |
| Solution | SIP Bulk Caller accepts alphanumeric, hyphen (-), underscore (_), and period (.) characters for callset, stationset, license and certificate files. It rejects any other file names. |

# Appendix: Configuration Files

This appendix has a section for each of the configuration files you use with Interaction SIP Bulk Caller.

# Outbound Callset Configuration File

Each callset requires a separate XML configuration file. For example, if your callset name is ICServerTest, then the callset XML file name must be ICServerTest.xml. The callset XML configuration file and audio files the callset uses must be either in the $SBC_ROOT\Resources folder if DirPath is not specified, or must be in the folder specified by the DirPath attribute.

This configuration file illustrates all options that are available. You do not need to use all the options.

```
<?xml version="1.0"?>
<CallSetConfig>

<MaximumCalls>-1</MaximumCalls>
<ActiveCalls>100</ActiveCalls>
<Burst>2</Burst>
<InterBurstDelay>2</InterBurstDelay>
<MaxCallWarmDownTime>0</MaxCallWarmDownTime>
<CallDurationWarning>900</CallDurationWarning>
<MediaType>Normal</MediaType>
<NoAnswerSeconds>30</NoAnswerSeconds>
<AutoStart>False</AutoStart>

<ToAddressList Type="Random">
<Address CallerId="John">sip:1111@icserver1</Address>
<Address CallerId="James">sip:2222@icserver1</Address>
<AddressRange CallerId="Nancy" StartNumber="5001" EndNumber="5100" MachineAddr="icserver1"/>
</ToAddressList>

<FromAddressList Type="RoundRobin">
<Address CallerId="Joe Smith">sip:anyext@localmachinename</Address>
</FromAddressList>

<Script Name="SampleScript" AudioFormat="G711" RFC2833Dtmf="True" AbsoluteCallLength="0" CheckAudioFiles="True">
<Action Type="Wait" Duration="15"/>
<Action Type="Dtmf" Digits="34#" InterDigitDelay="50" ToneDuration="100"/>
<Action Type="Wait" Duration="5"/>
<Action Type="SequenceDtmf" StartDigits="1000" EndDigits="2000" InterDigitDelay="50" ToneDuration="100"/>
<Action Type="RandomDtmf" StartDigits="1000" EndDigits="2000" InterDigitDelay="50" ToneDuration="100"/>
<Action Type="Play" FileName="Yourturn.wav"/>
<Action Type="Play" Duration="15" FileName="ICMainMenu.wav"/>
<Action Type="RandomPlay" Duration="30">
<FileName>AudioFile1.wav</FileName>
<FileName>AudioFile2.wav</FileName>
<FileName>AudioFile3.wav</FileName>
<FileName>AudioFile4.wav</FileName>
</Action>
<Action Type="RandomPlay">
<FileName>AudioFile1.wav</FileName>
<FileName>AudioFile2.wav</FileName>
<FileName>AudioFile3.wav</FileName>
<FileName>AudioFile4.wav</FileName>
</Action>
<Action Type="SequencePlay" Duration="30">
<FileName>AudioFile1.wav</FileName>
<FileName>AudioFile2.wav</FileName>
<FileName>AudioFile3.wav</FileName>
<FileName>AudioFile4.wav</FileName>
</Action>
<Action Type="SequencePlay">
<FileName>AudioFile1.wav</FileName>
<FileName>AudioFile2.wav</FileName>
<FileName>AudioFile3.wav</FileName>
<FileName>AudioFile4.wav</FileName>
</Action>
</Script>
</CallSetConfig>
```

## Explanation of Outbound Callset Configuration File

| Element | Explanation |
|---|---|
| MaximumCalls | The callset stops after making the number of calls set in MaximumCalls. If the value of MaximumCalls is -1, then the callset runs continuously until the user stops it. |
| ActiveCalls | The number of calls that can be active simultaneously. If Interaction SIP Bulk Caller reaches this limit, then it stops placing new calls until a call is disconnected. |
| Burst | The total number of new calls placed together as a burst. |
| BurstRange | If you want to set up a variable burst rate, then use the BurstRange element instead of Burst. For example,<br><br><BurstRange Type="Sequence" StartNumber="2" EndNumber="20" IncrStep="1"/><br><br>This code specifies the range of burst and the size of the increment step. If incrstep is set to 2, then burst would be 2,4,6,...,20. If type is Sequence, then the first burst of calls would be 2, the next would be 4, and so on in the cycle. If the type is Random, then the burst could have any value in the range of 2,4,6, … 20. |
| InterBurstDelay | The delay in seconds between two bursts of calls.<br><br>For example, if Burst=6 and InterBurstDelay=2, then Interaction SIP Bulk Caller places six calls every two seconds. You can also use decimal values in increments of 0.5, such as 0.5, 1, 1.5, and 2. For example, if Burst=3 and InterBurstDelay=0.5, then Interaction SIP Bulk Caller places three calls every 0.5 seconds.<br><br>Remember that even if you have a high burst rate, Interaction SIP Bulk Caller stops placing new calls after it reaches the ActiveCalls limit until old calls get disconnected. |
| MaxCallWarmDownTime | This value is in seconds and the default value is 0. Use this parameter to insert a delay when outbound callset reaches its activecall limit and then a call gets disconnected. Before placing new call, Interaction SIP Bulk Caller waits for the number of seconds that this value specifies.<br><br>This is used most often when calls are placed to a PBX or gateway where a delay is required to clear the channel before placing a new call. |

| | |
|---|---|
| CallDurationWarning | This value is in seconds and the default value is 900 seconds (15 minutes). If call duration exceeds this value, then Interaction SIP Bulk Caller stores it as a warning message in its logs. This is used to track hung calls. |
| MediaType | The MediaType values are Normal and Delayed. In a delayed media type, media information is sent in ACK. The default value is Normal. |
| NoAnswerSeconds | This is the number of seconds for the sipengine will wait for a response. If a response is not received in the specified time, then the sipengine sends a disconnect message for the call. The default value is 30 seconds. |
| AutoStart | This is a Boolean (true or false) value. If this is set to true, then the callset starts automatically when the Interaction SIP Bulk Caller service starts. |
| ToAddressList | This is a list of addresses where you want to send SIP calls. Interaction SIP Bulk Caller sends calls to the addresses in either RoundRobin or Random mode based on the Type attribute value. The default is RoundRobin. |
| Address | This contains CallerIds and SIP addresses where Interaction SIP Bulk Caller can send SIP calls. |
| AddressRange | You can use this element with a range of addresses in sequence. For example, SIP addresses might be added as sip:5001@icserver1, sip:5002@icserver1, sip:5003@icserver1, ... upto sip:5100@icserver1. |
| FromAddressList | This is a list of "from" addresses. |
| Script | The Script consists of one or more Actions to execute in the SIP call. |
| AudioFormat | The format of the audio data to play. Acceptable values are G711, G711A, G722, G726, G729AB, G729A, or GSM.<br><br>Note that current Audiolib (HPAA) supports 8-bit 8 KHz CCITT u-law, 8-bit 8 KHz CCITT a-law, 8-bit 8 KHZ GSM, and 16-bit linear PCM for audio wav files. If your audio wav file is not in one of these formats, then manually convert it before using it in Interaction SIP Bulk Caller. |
| RFC2833Dtmf | This is a Boolean (true or false) value. If it is set to true, then Interaction SIP Bulk Caller sends an RFC2833 event for Dtmf tones. If it is set to false, then Interaction SIP Bulk Caller sends inband dtmf. |

| | |
|---|---|
| AbsoluteCallLength | If this value is 0, then Interaction SIP Bulk Caller ignores it. If it is greater than 0, Interaction SIP Bulk Caller cuts off the call after the number of seconds specified in AbsoluteCallLength. This parameter ensures that the length of SIP calls will not exceed the time specified in AbsoluteCallLength even if calls are transferred from station to station. |
| CheckAudioFiles | When you load the callset, Interaction SIP Bulk Caller checks the audio file using the callset. If you set this value to false, then Interaction SIP Bulk Caller does not do this check. |
| SendPTime | Packetization interval in milliseconds of audio sent by the script. Valid values are 10 - 160. For G729A and G729AB, the value is rounded up to the next multiple of 10 ms. This setting has no effect on:<br><br>• The G722, GSM, and G723 AudioFormats.<br>• The packetization interval of RFC 2833 events or T.38 UDPTL streams.<br><br>With or without this setting, SIP Bulk Caller ignores any a=ptime and a=maxtime lines in the SDP it receives. |
| Wait Action | This specifies the number of seconds that a call will send silence RTP packets. |
| Dtmf Action | Each call plays the dtmf digits specified in the Digits attribute. There is a delay of InterDigitDelay milliseconds between two dtmf digit play. You can also set the duration in milliseconds of each tone. |
| SequenceDtmf Action | Each call plays different dtmf digits in sequence from StartDigits to EndDigits. For example, the first call might play Dtmf 1000, the second call might play Dtmf 1001, and so on up to Dtmf 2000. After reaching EndDigits Dtmf, the cycle repeats. |
| RandomDtmf Action | Each call plays different random dtmf digits between StartDigits and EndDigits. For example, a call might play any dtmf digits between 1000 and 2000. |
| Play Action | Each call plays an audio wav file. In the FileName attribute, put only the name of the audio file; do not include the directory path. If the Play action has a Duration attribute, then a call will play audio file up to the duration. |
| RandomPlay Action | Each call randomly selects one file from files listed in this action and plays its full length. If the RandomPlay action has a Duration attribute, then the call will play the audio file until the end of the specified duration. |

| SequencePlay Action | Each call selects an audio file in sequence and plays its full length. The first call plays the first audio file, the second call plays the second audio file, and so on. If the SequencePlay action has a Duration attribute, then calls play the audio file until the end of the specified duration. |
|---|---|
| Fax Action | This lets you specify the outbound calls as faxes. Interaction SIP Bulk Caller supports T.30 and T.38 fax transmission. To send a fax, add this code to the configuration file:<br><br>\<Action Type="Fax" Direction="Send" FileName="FaxSend.tif"/><br><br>Note that a .TIFF file (same as the FileName attribute) must be presentin the resource directory for fax send. In the example, the file FaxSend.tifmust be in the resource directory. TIFF files generated by the CIC Fax Client are not supported attime of releaseby the MediaServer Audio library that SIPBulkCaller uses. |
| Hold Action | Each call can be placed on hold. In the Filename attribute, put only the name of the audio file. If the Hold action has a Duration attribute, then the hold lasts for the specified duration. The Hold makes the SBC send a SIP UPDATE message to the remote user agent (UA). This action simulates pressing a Hold button on an IP phone. If the Filename attribute is empty ( ""), the SBC sends the unspecified IP (0.0.0.0 or ::) address in the SIP UPDATE message depending on the IP family SIP Line config setting. |

# Inbound Callset Configuration File

The inbound callset has fewer parameters than the outbound callset.

```
<?xml version="1.0"?>
<CallSetConfig>

<StationList>
<StationRange StartExt="7000" EndExt="8000"/>
</StationList>
<WaitBeforeAnswer>0</WaitBeforeAnswer>
<CallDurationWarning>900</CallDurationWarning>
<MediaType>Normal</MediaType>
<AutoStart>False</AutoStart>

<!-- //comment
Modify script in defaultinboundcallset according to your requirement
-->
<Script Name="SampleScript" AudioFormat="G711" RFC2833Dtmf="True" AbsoluteCallLength="0"
CheckAudioFiles="True">
<Action Type="Wait" Duration="15"/>
<Action Type="Play" Duration="20" FileName="ICMainMenu.wav"/>
</Script>

</CallSetConfig>
```

## Explanation of Inbound Callset Configuration File

| Element | Explanation |
|---|---|
| StationList | This contains StationRange elements. When a call is received, it contains the station extension. Inbound callset, which contains this specific station extension, answers the call. If none of the inbound callsets contain the station extension received, then DefaultInboundCallSet answers the call. DefaultInboundCallSet does not contain the StationList element, since it answers calls for all station extensions.<br><br>In addition to the StationRange elements, the StationList element can contain the Station Name attribute. See Registering a station based on a SIP name. |
| WaitBeforeAnswer | This is the delay in seconds before an inbound call is answered (picked up). The default value is 0.<br><br>This is different from a wait action. This element delays answering of the inbound call. In wait action, by contrast, a call is answered and connected, but does not play audio until after the duration specified in wait action. |
| WaitBeforeAnswerRange | If you want variable delay before answering the call, then use the WaitBeforeAnswerRange element. For example,<br><br>&lt;WaitBeforeAnswerRange Type="Random" StartNumber="2" EndNumber="30" IncrStep="2" /&gt;<br><br>Here, it creates a range of values between 2 and 30 with increments of 2. Values are 2,4,6....30. Interaction SIP Bulk Caller assigns a random value within the range to each call. |
| CallDurationWarning | This value is in seconds and the default value is 900 seconds (15 minutes). If call duration exceeds this value, then Interaction SIP Bulk Caller stores it as a warning message in its logs. This is used to track hung calls. |
| MediaType | The MediaType value can be Normal or Early. If media type is set to early, then media information is sent in Session Progress (183). In such a case, set the number of seconds in WaitBeforeAnswer. Interaction SIP Bulk Caller sends an answer after the number of seconds in WaitBeforeAnswer. |
| AutoStart | If this value is set to true, then the callset is automatically starts when the Interaction SIP Bulk Caller service starts. |
| Script | This is a set of actions executed when an inbound call is received. It is the same as the outbound callset. Refer to Outbound CallSet for more details. |
| Fax Action | This lets you specify inbound calls as faxes. Interaction SIP Bulk Caller supports T.30 and T.38 fax transmission. To receive faxes, add this code to the configuration file:<br><br>&lt;Action Type="Fax" Direction="Receive" ReceiveDirectory="d:\Temp\"/&gt;<br><br>In the code line's ReceiveDirectory parameter, replace "d:\Temp\" in the example code line with the wanted directory path.<br><br>Note that a call set can contain only one fax action. Combining a fax action withother actions (e.g., play, wait, etc.) or another fax action is not allowed. |

| Hold Action | Each call can be placed on hold. In the Filename attribute, put only the name of the audio file. If the Hold action has a Duration attribute, then the hold lasts for the specified duration. The Hold makes the SBC send a SIP UPDATE message to the remote user agent (UA). This action simulates pressing a Hold button on an IP phone. If the Filename attribute is empty ( ""), the SBC sends the unspecified IP (0.0.0.0 or ::) address in the SIP UPDATE message depending on the IP family SIP Line config setting. |
|---|---|
| SendPTime | Packetization interval in milliseconds of audio sent by the script. Valid values are 10 - 160. For G729A and G729AB, the value is rounded up to the next multiple of 10 ms. This etting has no effect on:<br><br>• The G722, GSM, and G723 AudioFormats.<br>• The packetization interval of RFC 2833 events or T.38 UDPTL streams.<br><br>With or without this setting, SIP Bulk Caller ignores any a=ptime and a=maxptime lines in the SDP it receives. |

**DefaultInboundCallSet**: DefaultInboundCallSet is installed with Interaction SIP Bulk Caller. As its name suggests, it is the default callset for inbound answering. When you have only DefaultInboundCallSet listed, then it answers calls received for all stations. If you have more than one inbound callset, then Interaction SIP Bulk Caller finds an inbound callset matching the station extension received in the inbound call. If none of the inbound callsets match the station extension, then it is answered by DefaultInboundCallSet.

# StationSet Configuration File

StationSet is used to simulate station registration from Interaction SIP Bulk Caller. It sends registration and authentication requests to the CIC server. The SIP station on the CIC server is dynamically registered from Interaction SIP Bulk Caller (sipbulkreceiver). Each stationset requires a separate XML configuration file.

The name of the XML file must be *stationsetname.xml*. SampleStationSet.xml is included as part of the installation. If you wish, you may simply edit that file.

You can use a stationset alone to test registration functionality of the CIC server. You might also need to configure and use a stationset if you have stations on CIC server configured for dynamic registration and you want to receive calls. In this scenario, configure InboundCallSet to answer the call and configure StationSet for station registration.

```xml
<?xml version="1.0"?>
<StationSetConfig>

<Burst>20</Burst>
<InterBurstDelay>1</InterBurstDelay>
<ExpirationInterval>3600</ExpirationInterval>
<Username></Username>
<Password></Password>

<ICServerList>
<ICServer>ICServerIPAddr</ICServer>
</ICServerList>

<StationList>
<StationRange StartExt="7000" EndExt="7100"/>
<StationRange StartExt="9500" EndExt="9700"/>
</StationList>

</StationSetConfig>
```

## Explanation of StationSet Configuration File

| Element | Explanation |
| --- | --- |
| Burst | Total number of stations to send registration request simultaneously. |
| InterBurstDelay | Delay in seconds between two bursts of registrations. You can also use decimal values in increments of 0.5, such as 0.5, 1, 1.5, and 2. |
| ExpirationInterval | Time in seconds when registration expires for an SIP station. Interaction SIP Bulk Caller re-registers an SIP station before the SIP station registration expires. Set expirationinterval long enough to register all stations before registration of the first station expires. For example, if you have 5000 stations to register and the rate is 20 stations per second, then expirationinterval must be higher than 250 (5000/20 = 250) seconds in order to finish registration of all stations before the first station registration expires. |
| Username and Password | Set username and password if an SIP Station on CIC server requires authentication at time of registration. It must be same for all SIP stations in the stationset. |
| ICServerList | Provide the name or IP address of the CIC server where you want your station to register. If you have a switchover pair, then add the primary and backup server names in the list. |
| StationList | This contains a list of StationRange elements.<br><br>In addition to the StationRange elements, the StationList element can contain the Station Name attribute. See Registering a station based on a SIP name. |
| StationRange | This has StartExt and EndExt attributes. Interaction SIP Bulk Caller creates a list of station ranges from StartExt to EndExt. |

# Main Xml Configuration File

The main XML configuration file is in the <u>$SBC_ROOT\Resources</u> directory. Its name must be SIPBulkCallerConfig.xml. You make most of the changes via the Web GUI. The elements that require manual changes are explained in the table below.

```xml
<?xml version="1.0"?>
<SIPBulkCallerConfig version="1.0">
<ApplicationConfig>
<SIPBulkCallerName></SIPBulkCallerName>
<HttpPort>8080</HttpPort>
<TelnetPort>0</TelnetPort>
<MaxThreadsInPool>100</MaxThreadsInPool>
<PlayAudio>True</PlayAudio>
<RtpNetworkInterface>Local Area Connection</RtpNetworkInterface>
<UseSrtp>False</UseSrtp>
<RtpQosTagging>False</RtpQosTagging>
<RtpQosDscpValue>46</RtpQosDscpValue>
<RtcpQosTagging>False</RtcpQosTagging>
<RtcpQosDscpValue>46</RtcpQosDscpValue>
<AudioEngineSelectionAlgorithm>FewestGraphs</AudioEngineSelectionAlgorithm>
<RtpDiagCaptureEnabled>False</RtpDiagCaptureEnabled>
<RtpDiagCaptureDirectory></RtpDiagCaptureDirectory>
</ApplicationConfig>

<SIPLineConfig>
<DomainName>inin.com</DomainName>
<Protocol>UDP</Protocol>
<NetworkInterface>Local Area Connection</NetworkInterface>
<PortNumber>5060</PortNumber>
<SIPQosByte>96</SIPQosByte>
<T1Timer>500</T1Timer>
<T2Timer>4000</T2Timer>
<TCPConnectTimer>3500</TCPConnectTimer>
<MaxPacketRetry>10</MaxPacketRetry>
<MaxInviteRetry>6</MaxInviteRetry>
<RootCertificateFile></RootCertificateFile>
<CertificateFile></CertificateFile>
<PrivateKeyFile></PrivateKeyFile>
</SIPLineConfig>


<!-- //comment
IMP: Do not use space in callsetname.
-->
<OutboundConfig>
<CallSet Name="SampleCallSet"/>
<CallSet Name="ICServer1Test" DirPath="D:\MyCallSetFiles"/>
</OutboundConfig>

<!-- //comment
IMP: Do not remove or rename DefaultInboundCallSet
-->
<InboundConfig>
<CallSet Name="DefaultInboundCallSet"/>
<CallSet Name="User1CallSet" DirPath="D:\MyCallSetFiles"/>
</InboundConfig>

<StationConfig>
<StationSet Name="SampleStationSet"/>
</StationConfig>

</SIPBulkCallerConfig>
```

**Explanation of Main Xml Configuration File**

| Element | Explanation |
|---|---|
| SIPBulkCallerName | Optional. Set this value if you want to give a customized name to Interaction SIP Bulk Caller. |
| HttpPort | The Web interface of Interaction SIP Bulk Caller starts at port number 8080. The Web address for the application is http://localhost:8080. To start the Web interface on some other port, change port number in HttpPort. |
| TelnetPort | The port on which Interaction SIP Bulk Caller would start the Telnet server. The default port value is 0 (zero) so by default, Interaction SIP Bulk Caller does not start the Telnet server. When this is set to 23, you can Telnet to Interaction SIP Bulk Caller. The Telnet server gives command-line access to Interaction SIP Bulk Caller. It lets you start or stop callsets and even add or delete callsets. <br><br> The Telnet server is useful for programmatic access or batch script access to Interaction SIP Bulk Caller. You can also use it for scheduling the start and stop time of your callsets, which gives you better control of your simulation. |
| MaxThreadsinPool | Total number of threads spawned for handling calls. You might not need to change its default value. |

# Registering a station based on a SIP name

In addition to registering station extensions using the StartExt and EndExt attributes, you can now register individual stations with the SIP name by using the Station Name attribute inside the Station List tag.

# Example

```
<StationList>

<StationRange StartExt="7000" EndExt="7010"/>

<Station Name="test_name"/>

</StationList>
```

This specifies an extension range as well as a name of "test_name". This new tag can be used in incoming call sets (to selectively answer calls based on incoming name) as well as station sets (for registration).

# sip bulk caller tr

# Change Log

| Date | Changes |
|---|---|
| 28-July-2010 | Initial release of this document. |
| 18-August-2010 | In section on installation, added setup file location on the Support Site. |
| 01-September-2010 | Updated figures to show user interface changes. Made minor changes in procedures and text, such as changing the name of the Windows Service. |
| 26-October-2012 | Added information to the appendix about fax support in callsets; licensing page changes; browsing to configuration files. |
| 08-April-2013 | Added information about T.30 fax support. Updated copyrights and made other minor tweaks. |
| 03-April-2014 | Updated information about SIP line configuration, audio configuration, and other sections. Updated copyrights. |
| 20-September-2014 | Updated documentation to reflect changes required in the transition from version 4.0 SU# to CIC 2015 R1, such as updates to product version numbers, system requirements, installation procedures, references to Interactive Intelligence Product Information site URLs, and copyright and trademark information. |
| 07-October-2014 | Updated the SIP Line Config page figure and accompanying table in the section "Edit SIP Line Configuration" to include IPv6 and PRACK, along with other minor changes. Added coverage of the new Hold action to the config tables for inbound and outbound callsets. |
| 13-October-2014 | Added row about file name requirements to table in Troubleshooting. Added coverage of the new script attribute SendPTime to the sections on Outbound and Inbound callsets. |
| 12-June-2015 | Updated cover page to reflect new color scheme and logo. Updated copyright and trademark information. |
| 09-October-2015 | <ul><li>Updated MSXML Parser version from 4.0 to 6.0 in System Requirements.</li><li>Added information about a new feature that allows you to export callset statistics from the Status > Statistics page.</li><li>Updated screen shots</li></ul> |
| 04-February-2016 | Updated documentation to reflect 2016 R2 Release |
| 06-April-2016 | <ul><li>Updated documentation to reflect 2016 R3 Release</li><li>Updated copyright and trademark information.</li><li>Added information about the new Station Name attribute, which allows you to register a station with a SIP name.</li></ul> |
| 05-February-2018 | Conversion to HTML. |
| 21-March-2018 | Rebranded screen captures and terminoloogy. |

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