FAQ and Solutions

19 Apr 2024

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FAQ and Solutions

How to periodically offer Voice Mail call transfer during queue waiting

This article details how to configure Imagicle Advanced Queuing to offer caller party the possibility to leave a message in a voice mailbox, instead of waiting a long time to be served by an operator.

Imagicle provides a simple script, attached to this KB, to enable VoiceMail transfer feature. Before actually loading the script, it should be edited by configuring the following parameters:

- `<var name="dialog.target" expr="'nnnnn'" />`

where *nnnnn* should be replaced with Voice Mail number to transfer the call.

- `<property name="timeout" value="10s" />`

where *10s* is the timeout while waiting for a DTMF digit. Once timeout is expired, the call returns to the queue.

- `<choice dtmf="1" next="#TransferTo" />`

1 is the expected DTMF digit to trigger call transfer to Voice Mail number. Any other digit is ignored.

- `<audio expr="session.AudioPath + 'transferVM.wav'">`
- `<audio expr="session.AudioPath + 'two.wav'">`
- `<audio expr="session.AudioPath + 'three.wav'">`

â These are the voice messages played during VM transfer transaction. You can create them in WAV format, following the suggested content in below table:

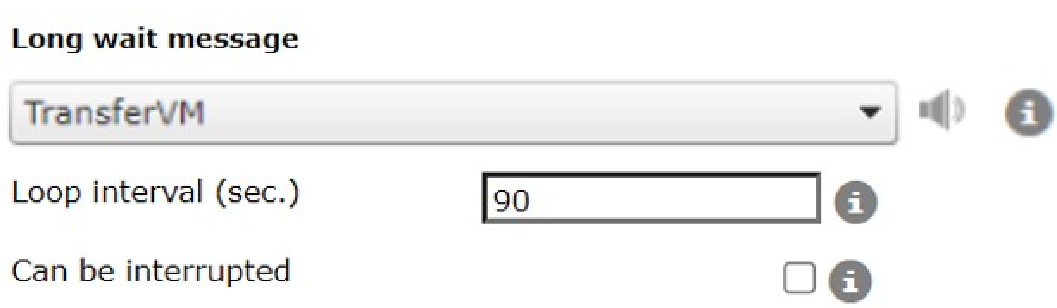
File name	Content
transferVM.wav	Intro message: "Please press 1 if you want to leave a message or just hold the line to avoid losing the acquired queue priority"
two.wav	Please wait, while I connect your call.
three.wav	I'm sorry. I can't connect you with the voice mail service

Once audio files have been created, please copy them in the following UCS folder:

```
C:\Program Files (x86)\StonevoiceAS\Apps\QME\Data\AudioFiles\User\Shared
```

Now you can include the amended "transferVM.vxml" script into the queue of your choice, by uploading it from the following web menu: Advanced Queuing â Double-click on existing queue entry â Voice Messages tab â Manage messages.. Please save configuration and wait one minute; then refresh the Voice Messages web page.

Now you can select new VM script from "Long wait message" pull-down menu and you can configure "Loop interval (sec.)" for the desired frequency between each VM transfer request.



Finally, you need to modify the following setting file, to enable DTMF digits recognition within the script:

`C:\Program Files (x86)\StonevoiceAS\Apps\QME\Settings\`**`QME.Opal.Config.xml`**

Please add the following line, between <Configuration> statements, like below sample:

```
<preference key="voip.exclude.additionalcodecs.whendtmf.notrequired" value="" />
```

```
<?xml version="1.0" encoding="utf-8"?>
<configuration version="1.0.1.0">
  <preference key="voip.exclude.additionalcodecs.whendtmf.notrequired" value="" />
</configuration>
```

Save XML file and restart Imagicle Advanced Queuing service from Advanced Queuing â Manage Service menu option.

QME lookup not working correctly and VIP calls not managed as they should because of Speedy used with different departments

Applies to

All Application Suite with QME and Speedy used with different departments configured.

Description

When the call arrives to QME, during contacts lookup (Speedy), QME does not show the contact NAME in Attendant Console or phone display.

Because of this we have two possible issues:

- Error in lookup (number without name)
- Issue with "VIP Calls"

Cause

The cause of these problems is that the user QME uses to search contacts in Speedy is anonymous and, when anonymous try to search contacts in Speedy, it can't see department contacts.

Solution

In order to solve this issue we can let QME "impersonate" a different user (existing in user management page) that has permissions to see department contacts.
To achieve this:

1) Edit "QME.Engine.config.xml" in <StonevoiceAS>\Apps\QME\Settings\
2) Add this preference key:
<preference key="speedy.callerlookup.identity.asuser" value="**<user>**" />
Where <user> is an existing Application Suite user, for example:

<configuration version="1.0.1.0">
    <preference key="speedy.callerlookup.identity.asuser" value="**test**" />
</configuration>

3) Save the file and restart QME service form the web interface.

Enable detailed VoIP debug (SIP trace) for QME

Applies to:

Imagicle Application Suite - Queue Manager Enterprise

Solution:

To enable low level SIP tracing for Queue Manager Enterprise, follow these steps:

1. Edit (with notepad) text file:

   <StonevoiceAS>\Apps\QME\Settings**QME.Opal.config**.xml
2. Add the following line

   ```
   <preference key="logging.level" value="4" />
   ```

   inside the <configuration> tag, so that the file will look like this:

   ```
   <configuration>

   ...

   <preference key="logging.level" value="4" />

   ...

   </configuration>
   ```
3. Save and close the file
4. Restart "Imagicle Queue Manager Enterprise" Windows Service

To disable tracing, just remove the line at step 2.
When the issue has occured, just download logs from administration support web page, choosing application "Queue Manager Enterprise" and disable tracing.

Queue Manager Enteprise does not transfer calls to agents

Applies from Application Suite 2014.12.1
to version Application Suite 2014.12.1

Applies to:

IAS Winter 2015 (2014.12.1) for Cisco UCM

Description:

QME answers incoming calls correctly, but can't distribute the calls to the agents, so the call remains in queue forever and no agent phone rings.

Cause:

This is related to a bug that affects only IAS Winter 2015 build 1 (2014.12.1).
QME tries to send INVITE for consultation calls to PBX to port 5060 instead of 5062 (QME dedicated).
But Sip trunk security profile is configured for port 5062 and not for port 5060, so CUCM doesn't accept it.
This bug is not observed if the Imagicle Fax Server is configured, because in that case there is also a sip trunk security profile working on port 5060, so QME answers calls from 5062 and send invite to CUCM's 5060 port.

Solution:

This is related to a bug that affects only IAS Winter 2015 build 1 (2014.12.1).

QME tries to send INVITE for consultation calls to PBX to port 5060 instead of 5062 (QME dedicated).

But Sip trunk security profile is configured for port 5062 and not for port 5060, so CUCM doesn't accept it.
This bug is not observed if the Imagicle Fax Server is configured, because in that case there is also a sip trunk security profile working on port 5060, so QME answers calls from 5062 and send invite to CUCM's 5060 port.

Workaround: in CuCM, edit the SIP Trunk Security Profile usedby to SIP Trunk for QME, set "Incoming Port" to 5060 instead of 5062.

How to remove the ringback tone for calls landing to QME/IVR SIP trunk

Applies to:

- Queue Manager Enterprise (CISCO UC)
- IVR for QME (CISCO UC)

Description:

The CUCM will play a ringback tone for incoming calls to ACD. To remove this ringback tone, a SIP Normalization Script must be used. The script will process any SIP messages coming from QME and will discard any 180 ringing.

How-to:

1. Log in into CUCM
2. Create SIP Normalization script
    1. Go to Device -> device settings -> SIP normalization script
    2. Click AddNew
    3. Click Import File and upload the attached file
    4. Enter a name, such as "DiscardIncoming180Ringing"
    5. Click Save
3. Go to Device -> Trunk
4. Find and edit SIP Trunk for QME
5. Within "SIP Information" section, find "Normalization Script"
6. Choose the script "DiscardIncoming180Ringing"
7. Save and Reset trunk

Phone Control: configurable timeout for TAPI Library and line monitoring initialization

to version Application Suite 201x (any version)

Applies to

IAS Winter 2019 or newer

Description

Bug symptoms:

- While autenticanting with Attendant Console, user receives a message of incorrect credentials (username and password) even if these are correct.
- At lines TAPI monitoring start, something goes wrong and log file **Var\Log\ApplicationSuite.Phone.ControlService\ApplicationSuite.log.txt** ends with the following line:

```
INFO  { 10} [ApplicationSuite] [SvTapiManager] Initializing main TAPI object.
```

Cause

Bug root cause is related to the fact that if TAPI monitoring freezes, Phone Control remains locked in a not-ready state and Console, having exceeded its unanswered timeouts from server, returns an error message. In addition to that, if TAPI initialization is unlocked after a certain time, agent is wrongly shown as connected event if he is not.

The only work-around consists in turning off services and restart Phone Control.

Solution

Two configuration parameters have been added:

- Phone Control TAPI Library initialization timeout
- Phone Control line monitoring initialization timeout

TAPI Library initialization timeout parameter configuration

A new configurable parameter has been introduced to allow setting of the Phone Control TAPI library initialization timeout. This parameter is configured by default using the key tapi.initializationTimeout.minutes in the configuration file \ Apps \ ApplicationSuite \ Settings \ ApplicationSuite.Phone.Control.schema.xml

```
<!-- TAPI Manager Initialization Timeout expressed in minutes ( if 0 or less, timeout will be set to max value) -->

<preference key="tapi.initializationTimeout.minutes" defvalue="10" type="System.Int32" attributes="" />
```

Parameter override can be handled, as usual, by adding to the corresponding settings file ApplicationSuite.Phone.Control.config.xml a type key as follows:

```
<preference key="tapi.initializationTimeout.minutes" value="1" />
```

Parameter is in minutes and its default value is 10.

If the value is set to 0 (or any negative number), the system will use the maximum available value (24 days, 20 hours, 31 minutes and 23 seconds).

If TAPI monitoring starting does not end successfully within the indicated timeout, Phone Control generates an error. In this case it is necessary to switch off the service.

Line monitoring initialization Timeout parameter configuration

In addition to the above, a further configurable parameter has been introduced to set the timeout for the Attendant Server to monitor a TAPI line. This parameter is configured by default with the **Attendant.Core.ConsoleServer.PhoneControl.ConnectionTimeout.Seconds** key in the configuration file **\Apps\Attendant\Settings\Attendant.schema.xml**

```
<!-- PhoneControl: TAPI line initialization expressed in seconds ( if 0 or less, timeout will be set to max value) -

<preference key="Attendant.Core.ConsoleServer.PhoneControl.ConnectionTimeout.Seconds" defvalue="17" type="System.Int
```

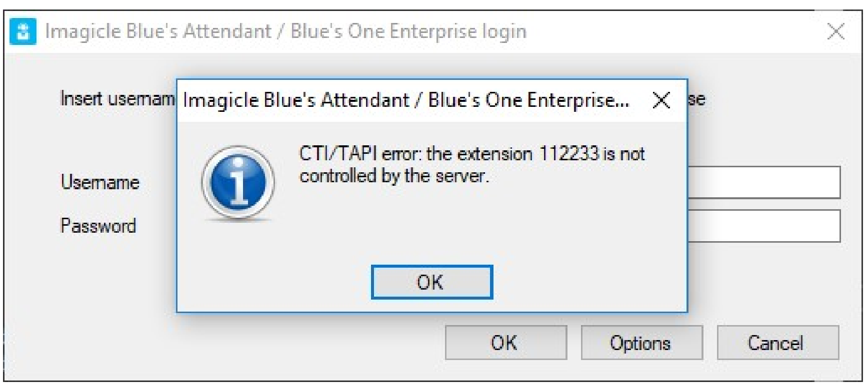
The override can be executed, as usual, by adding to the corresponding settings file **Attendant.config.xml** a key of the type

```
<preference key="Attendant.Core.ConsoleServer.PhoneControl.ConnectionTimeout.Seconds" value="10" />
```

The parameter is in second and its default value is 17. This key must always be set to a value minor than client-side timeout, configurable by the AuthenticationTimeout key in the **Attendant.Client.schema.xml** file (default value 20 seconds).

If the value is set to 0 (or any negative number), the system will use the maximum available value, (24 days, 20 hours, 31 minutes and 23 seconds).

If TAPI monitoring starting does not end successfully within the indicated timeout, Attendant Server generates an error and, after receiving an "Init Complete" message from the client, returns an empty list of controlled lines, which results into the following error message from the client (Error message shown by Attendant Console in case of failure in checking TAPI line within the configured timeout)



Troubeshooting

TAPI Library Initialization Timeout

Log file **Var\Log\ApplicationSuite.Phone.ControlService\ApplicationSuite.log.tx**t, when TAPI library is successfully loaded, shows the following logging (timeout is set to 1 minute in the example below: Phone control log with successful TAPI library inizialization. Timeout set to 1 minute.)

```
1024 11:14:08.631 INFO  { 10} [ApplicationSuite] [SvTapiManager] Initializing main TAPI object.
1024 11:14:08.638 INFO  { 10} [ApplicationSuite] [TapiManager] Tapi manager initialization timeout set: Timeout {00:01:00}
1024 11:14:08.644 INFO  { 12} [ApplicationSuite] [TapiManager] Tapi Initialization thread started
1024 11:14:08.690 INFO  { 12} [ApplicationSuite] [TapiManager] Tapi Initialization thread completed
1024 11:14:08.691 INFO  { 10} [ApplicationSuite] [TapiManager] Tapi initialization completed successfully
```

Phone control log with timeout TAPI library inizialization (In the example below: Phone control log with timeout unsuccessful TAPI library inizialization. Timeout set to 1 minute)

```
1024 10:46:55.500 INFO  { 16} [ApplicationSuite] [SvTapiManager] Initializing main TAPI object.
1024 10:46:55.500 INFO  { 16} [ApplicationSuite] [TapiManager] Tapi manager initialization timeout set: Timeout {00:01:00}
1024 10:46:55.500 INFO  { 18} [ApplicationSuite] [TapiManager] Tapi Initialization thread started
1024 10:47:55.500 ERROR { 16} [ApplicationSuite] [TapiManager] Tapi initialization has not completed within timeout, aborting thread: Timeout {00:01:00}, Thread id {18}
1024 10:47:55.500 ERROR { 18} [ApplicationSuite] [TapiManager] Tapi initialization has failed within timeout: Timeout {00:01:00}, {
Exception Type {System.Threading.ThreadAbortException}
Message {Thread was being aborted.}
StackTrace {
   at System.Threading.Thread.SleepInternal(Int32 millisecondsTimeout)
   at System.Threading.Thread.Sleep(Int32 millisecondsTimeout)
   at System.Threading.Thread.Sleep(TimeSpan timeout)
   at ApplicationSuite.Phone.Control.TAPI.ATAPI.TapiManager.<>c__DisplayClass15_0.<Initialize>b__0() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\Applicatior
}
1024 10:47:55.516 ERROR { 16} [ApplicationSuite] [SvTapiMonitor] Unable to initialize Tapi: {
Exception Type {System.TimeoutException}
Message {Tapi initialization timed out.}
StackTrace {
   at ApplicationSuite.Phone.Control.TAPI.ATAPI.TapiManager.Initialize() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\ApplicationSuite\Component\Assembly\App
   at ApplicationSuite.Phone.Control.SvTapiManager..ctor(ILoggerFactory loggerFactory, ITapiManagerFactory tapiManagerFactory) in E:\Sviluppo\GIT\ApplicationSuite\StoneC
   at ApplicationSuite.Phone.Control.SvTapiMonitor..ctor(ILoggerFactory loggerFactory, ITapiManagerFactory tapiManagerFactory, Type lineFactoryType, Object clientContext
}
```

In case a timeout occurs, the following logs are produced

- in **Var\Log\Attendant.Core.ConsoleServerService\Attendant.log.txt (** (Attendant Server Logâ Application Suite in case of timeout unsuccessful TAPI library inizialization):

```
1024 12:03:03.353 ERROR { 11} [ApplicationSuite] [SvClientRemoted`1] An error occurred during remote execution: {
Exception Type {System.TimeoutException}
Message {Tapi initialization timed out.}
StackTrace {
   at ApplicationSuite.Base.Library.SvResilientClientRemoted`1.<>c__DisplayClass7_0.<.ctor>b__0() in E:\Sviluppo\GIT
   at ApplicationSuite.Base.Library.SvClientRemoted`1.Reconnect() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Sourc
}
```

- in **Var\Log\Attendant.Core.ConsoleServerService\Attendant.log.txt (**Attendant Server Log â Attendant in case of timeout unsuccessful TAPI library inizialization):

```
1024 12:02:03.322 ERROR {  8} [Attendant] [ClientManager:(192.168.150.69:11317|lucao|Unknown)] GetClientData - DoLoginProcedure Error: System.TimeoutException: Tapi initialization timed out.
   at ApplicationSuite.Base.Library.SvResilientClientRemoted`1.<>c__DisplayClass7_0.<.ctor>b__0() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\ApplicationSuite\Component\Assembly\
   at ApplicationSuite.Base.Library.SvClientRemoted`1.Reconnect() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\ApplicationSuite\Component\Assembly\ApplicationSuite.Base.Library\Re
   at ApplicationSuite.Base.Library.SvResilientClientRemoted`1.Reconnect() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\ApplicationSuite\Component\Assembly\ApplicationSuite.Base.L
   at Attendant.Core.ConsoleServer.PhoneControl.SvPhoneControlGuest.Start() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\Attendant\Component\Assembly\Attendant.Core.ConsoleServer\
   at Attendant.Core.ConsoleServer.Private.SvClientHost.SubscribeLine(SvFWUser user) in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\Attendant\Component\Assembly\Attendant.Core.Conso
   at Attendant.Core.ConsoleServer.Private.SvClientHost.DoLoginProcedure(String username, String password, AttendantClientType clientType, Boolean isInBackupmode) in E:\Sviluppo\GIT\Applicati
   at Attendant.Core.ConsoleServer.Clients.Private.ClientManager.UserLogin(String userName, String encriptedPwd, AttendantClientType clientType, enBacErrorCodes& errorNumber) in E:\Sviluppo\G
```

Line Monitoring Inizialization Timeout

Log file **Var\Log\Attendant.Core.ConsoleServerService\Attendant.log.txt**, in case of line monitoring inizialization timeout, shows the following log lines (Attendant Server Log â Attendant in case of timeout unsuccessful line monitoring inizialization):

```
1119 10:27:15.099 INFO  { 38} [Attendant] [ClientManager:(192.168.150.69:1395|lucao|Unknown)] Logon request username=lucao for client type=BacEnt (raw=BAE)
1119 10:27:15.100 INFO  { 38} [Attendant] [SvClientHost] DoLoginProcedure(lucao): Attempting login on client type BacEnt (normale mode).
1119 10:27:15.131 INFO  { 38} [Attendant] [SvClientHost] SubscribeLine(lucao): trying to subscribe to phone control the extension 112233 (Device name='')
1119 10:27:15.133 TRACE { 38} [Attendant] [SvPhoneControlGuest:[Subscription: Address=112233] (conference enabled)] Object created: Instance {[Subscription: Address=112233] (conference enabled)}
1119 10:27:15.135 TRACE { 38} [Attendant] [SvPhoneControlGuest:[Subscription: Address=112233] (conference enabled)] is starting. Line Factory=Attendant.Core.ConsoleServer.PhoneControl.SvPhoneControlConferenceLineFactory
1119 10:27:15.138 INFO  { 42} [Attendant] [SvPhoneControlGuest:[Subscription: Address=112233] (conference enabled)] starting resilient reconnection procedure thread
1119 10:27:30.140 ERROR { 38} [Attendant] [SvPhoneControlGuest:[Subscription: Address=112233] (conference enabled)] Resilient reconnection procedure has not completed within timeout, aborting thread: Timeout {00:00:15}, Thread id {42}
1119 10:27:30.282 ERROR { 42} [Attendant] [SvPhoneControlGuest:[Subscription: Address=112233] (conference enabled)] Resilient reconnection procedure has failed within timeout: Timeout {00:00:15}, {
Exception Type {System.Threading.ThreadAbortException}
Message {Thread was being aborted.}
StackTrace {
   at System.Threading.Thread.SleepInternal(Int32 millisecondsTimeout)
   at System.Threading.Thread.Sleep(Int32 millisecondsTimeout)
   at System.Threading.Thread.Sleep(TimeSpan timeout)
   at Attendant.Core.ConsoleServer.PhoneControl.SvPhoneControlGuest.<>c__DisplayClass15_0.<Reconnect>b__0() in E:\Sviluppo\GIT\ApplicationSuite\StoneCME\Source\Apps\Attendant\Component\Assembly\Attendant.Core.ConsoleServer\PhoneControl\SvPhoneControlGuest.cs:line 189}
}
```

In addition, after receiving the message of Init_complete from the CLient, the following line is produced (Attendant Server Log â Attendant sending to client an empty list of controlled lines):

```
1119 10:27:31.000 INFO  {  8} [Attendant] [ServerConnectionStateMachine] (192.168.150.69:1395|lucao) CurrentConnectionState - <<CLIENT_INIT_COMPLETED>>
1119 10:27:31.001 ERROR {  8} [Attendant] [ClientManager:(192.168.150.69:1395|lucao|BacEnt)] BacMessage_CCLineCallData - non è stata trovata la linea , restituisco al client la struttura CallControlLineCallData vuota
```

The SIP invite coming from the CUCM and directed to the QME is rejected by the operating system; problems using network interface E1000/E1000e

Applies from Application Suite 201x (any version)
to version Application Suite 201x (any version)

Applies to

All Windows Server OS

Description

Symptoms:

QueueManager > QME is running but is not managing any incoming calls.
The SIP invite coming from the CUCM and directed to the QME is rejected by the operating system.
In a wireshark trace you can see that the TCP/IP stack of the operating system is sending back to CUCM an ICMP message "port unreachable".
If the "ping option" is enabled in CUCM Trunk configuration you can see it "Not Service"
Event viewer: > You can see some messages with event id "27" on "e1iexpress" errors

StoneFax > Fatal Error in Incoming/Outgoing
Event viewer: > You can see some messages with event id "27" on "e1iexpress" errors

Cause

Issue with E1000E: Link here

Network adapters: Link here

Solution

Change the network interface card from type E1000e/E1000 to VMXNET3 imagicle

QME Call List report, assign a call to the original called number instead of to the number that has answered

Applies to:

- Queue Manager Enterprise - minimum version 2020.Winter.1

Description:

It is added again the possibility to associate a call to the original destination, instead of assign it to the extension that has answered. Consider for example the case of the destination number is forwarded to another extension. Starting from 2018.Summer.1 release, the call was associated to the target of the forwarding. Now, it's possibile to change this behaviour and associate the call to the forwarded extension, that is the original called.

How-to:

- Access to Imagicle Server/VM using RDP session;
- Go to <StonevoiceAS>\Apps\QME\Settings\ folder;
- Open file QME.Engine.schema.xml, there is a new preference with the defvalue="Actual", that means the call is assigned to the extension that has actually answered to;

```
<preference key="cdr.routetotargetnumber.value" defvalue="Actual"
type="System.string" attributes="enum">

  <element value="Actual"  desc="real answering number"/>

  <element value="Expected"  desc="target called number"/>

 </preference>
```

- Change the preference's defvalue to Expected, in this way the call will be associated to the original called number, regardless of whom has answered;
- Expected value has to be define in the file <StonevoiceAS>\Apps\QME\Settings\ QME.Engine.config.xml, adding this preference:
  `<preference key="cdr.routetotargetnumber.value" value="Expected"/>`
- if the the value is different from Expected, it will be use Actual and an error will be logged.

FAQ

to version Application Suite 201x (any version)

Configurations

Q: Can an agent serve multiple queues?

A: Yes, this is one of the supported and commonly used configurations.

Q: Can two different agents have the same extension number?

A: NO, QME cannot distinguish them and it cannot place indipendent consultation calls to each of them. On CUCM platforms, if the 2 agents directory numbers stay in 2 different partitions, only one of them will be reachable by QME (depending on the CSS assigned to the Imagicle QME SIP trunk).

Q: Can QME transfer calls to an offnet (PSTN) number?

A: Yes. This permits, for instance, to answer an incoming call for the queue from a mobile phone.
You must ensure the Imagicle QME SIP trunk is enabled for such kind of calls. On CUCM platforms this requires to assign a proper CSS to the "Inbound Calls" and "Reroute CSS" sections of the Imagicle QME SIP trunk.

Q: Can I configure the system to prioritize VIP callers in the waiting queues?

A: Yes, it is possible to define a list of calling numbers that enter the waiting queues skipping the queue (they're put ahead of the other regular waiting calls). This requires a Speedy Enteprise license and an advanced configuration, please contact the Imagicle support service to do it.

Particular scenarios and events

Q: What happens if an incoming call hits the QME and there are no queues associated to the called number?

A: If the called number is not associated to any ACD queue or AutoAttendant service, a voice prompt is given to the caller saying the called number (spelt by the voice prompt) is not associated to any valid service.

Q: What happens to an incoming call when there are no more available licensed channels?

A: The queue overflow treatment defined for the specific queue is applied to the incoming call. Therefore, queue by queue, you can decide if disconnecting the call (possibly with a voice prompt) or bouncing it to another phone number (for example a voicemail system).

Advanced Queuing report failure due to out of memory

Affecting:

Imagicle UC Suite rel. 2022.Summer.1.h1 and before

Description:

When you launch an Advanced Queuing report for a large time interval, it might happen you get an "out of memory" warning after a long waiting time.

Cause:

Report needs to retrieve a large amount of data from SQL and eventually the report transaction times-out.

Solution:

Starting from Imagicle 2022.Summer.1.h2 and above. two new parameters have been added in the following setting file:

C:\Program Files (x86)\StonevoiceAS\Apps\QME\Settings**QME.Engine.config.xml**

Within this file, you can add the following statement:

```
<preference key="qme.report.callList.maxCallsCount" value="nnn"/>
```

Where nnn is the maximum number of lines to display within the report. Default value is 30000.

Another important parameter you can tweak is the following:

```
<preference key="qme.report.aggregatedReportsMaxCallDetailsCount" value="nn"/>
```

Where nn is the maximum number of calls to show in Advanced Queuing aggregated reports, when "Show details" is enabled. Default value is 100.

As you notice, both parameters are limiting the amount of data to retrieve from SQL DB, thus avoiding time-out or memory overload.

imagicle

Wrong Caller ID and report data in Advanced Queuing for Webex Calling MT

Applies to:

Imagicle UCX Cloud Suite - all versions

Problems Description:

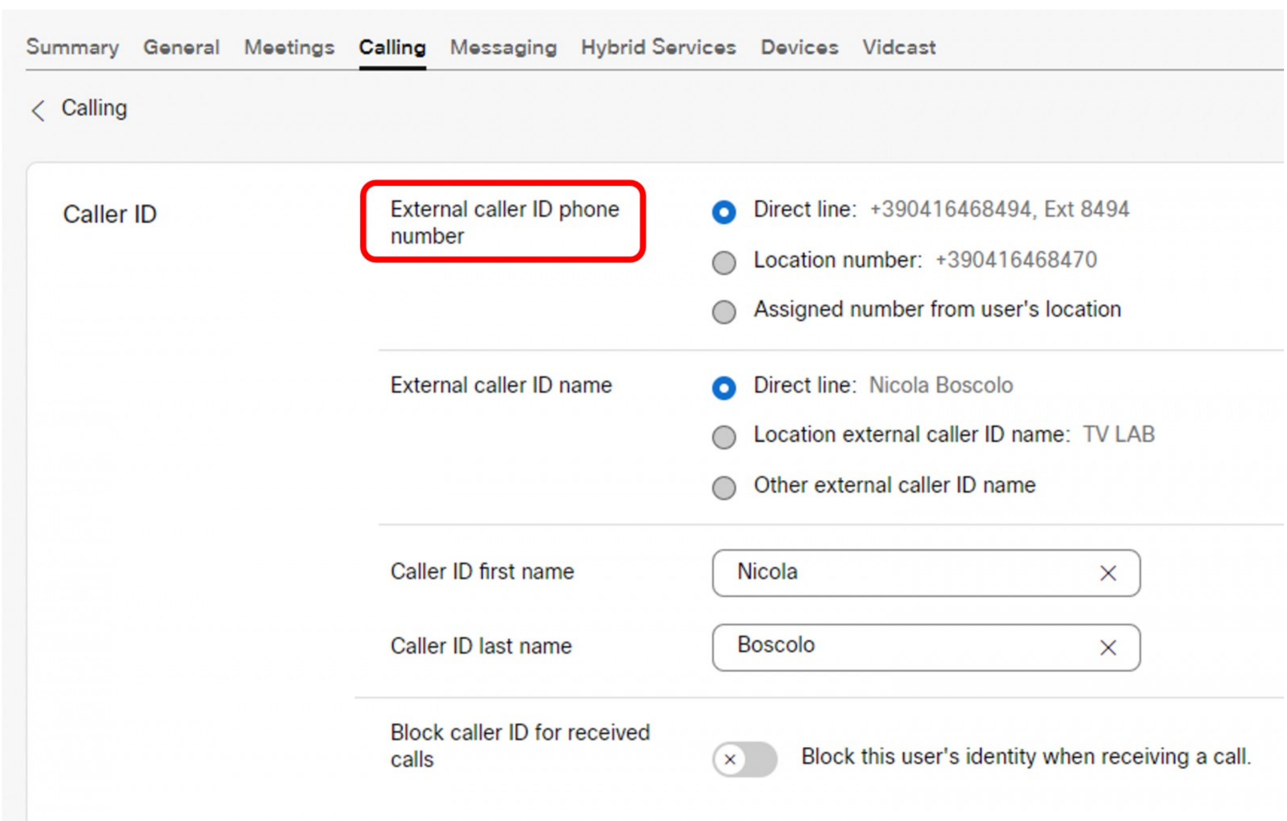
- Caller ID displayed on Imagicle Attendant Console "CURRENT CALLS" panel does not include queue name.
- If the call comes to the queue through a call forward, forwader number/name is not disclosed.
- Advanced Queuing does not track the conversation time.
- In Advanced Queuing reports, the "Served By" column is filled with wrong operator's phone number.

Cause:

There are two Webex Control Hub configurations which impact the consultation transfer from Imagicle queue to relevant operators:

Caller ID external phone number

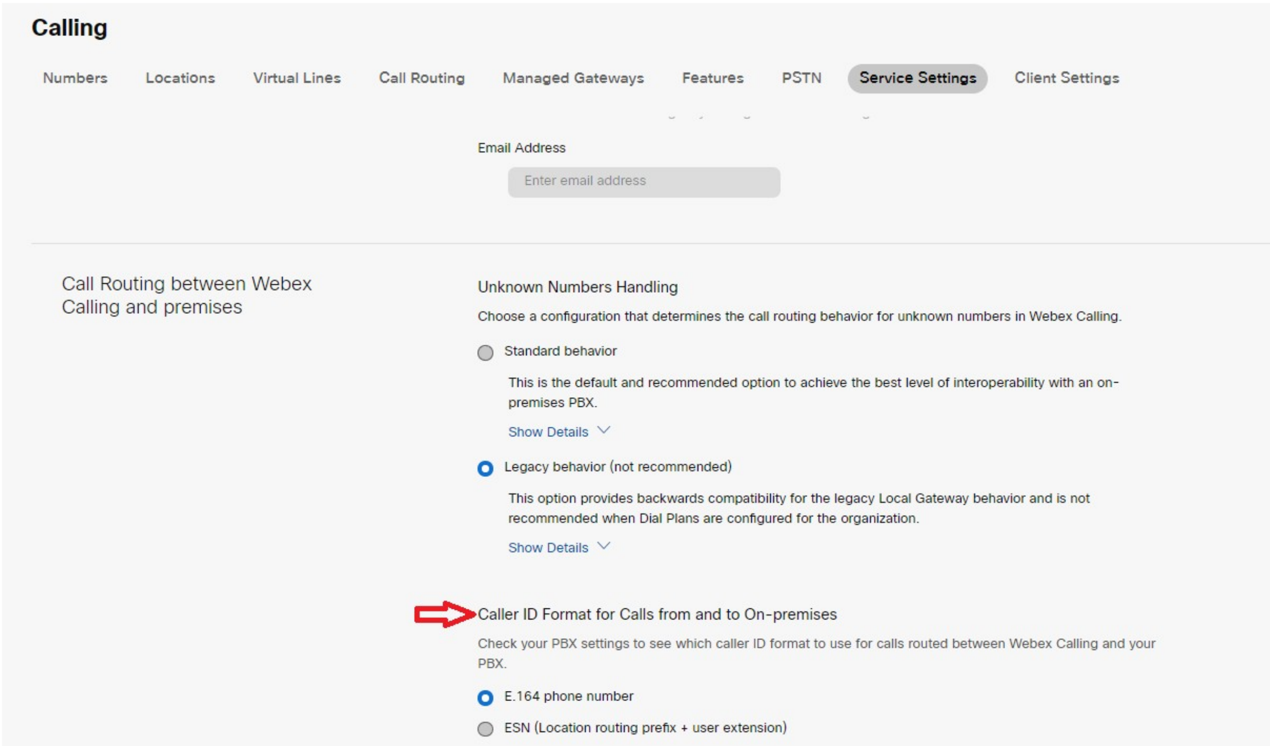
This setting dictates which phone number to take into consideration for Caller ID purposes. Standard setting is like below screenshot:



If customer configures above setting with "Location number" or "Assigned number from user's location", then Advanced Queuing can't correctly correlate the ringback signalling coming back from Webex Cloud with the actual operator's personal phone number, causing wrong statistics data retrieval in selected reports and incomplete Caller ID display in Imagicle Attendant Console.

Caller ID Format for Calls from/to On-premises

This setting dictates which phone number is added in Webex SIP signalling as Caller ID. Standard setting is like below screenshot:



If customer adopts "ESN" as Caller ID format, again Imagicle can't correctly correlate the ringback signalling coming back from Webex Cloud with the actual operator's phone number in +E.164 format, causing wrong statistics data retrieval in selected reports.

Solution:

If customer wants to keep above settings different than standard setup, then Imagicle needs to apply a specific configuration in its Cloud SBC. Please contact Imagicle Support for details.