



# Administration Guide

27 Apr 2024



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# Administration Guide

## Description and Architecture

IVR Module for Queue Manager Enterprise allows you to quickly create and manage IVR services, independently and in real time, without the intervention of technicians or professional services.

Through the web interface, you can design interactive menus, add or modify audio messages, and configure how to route the calls.

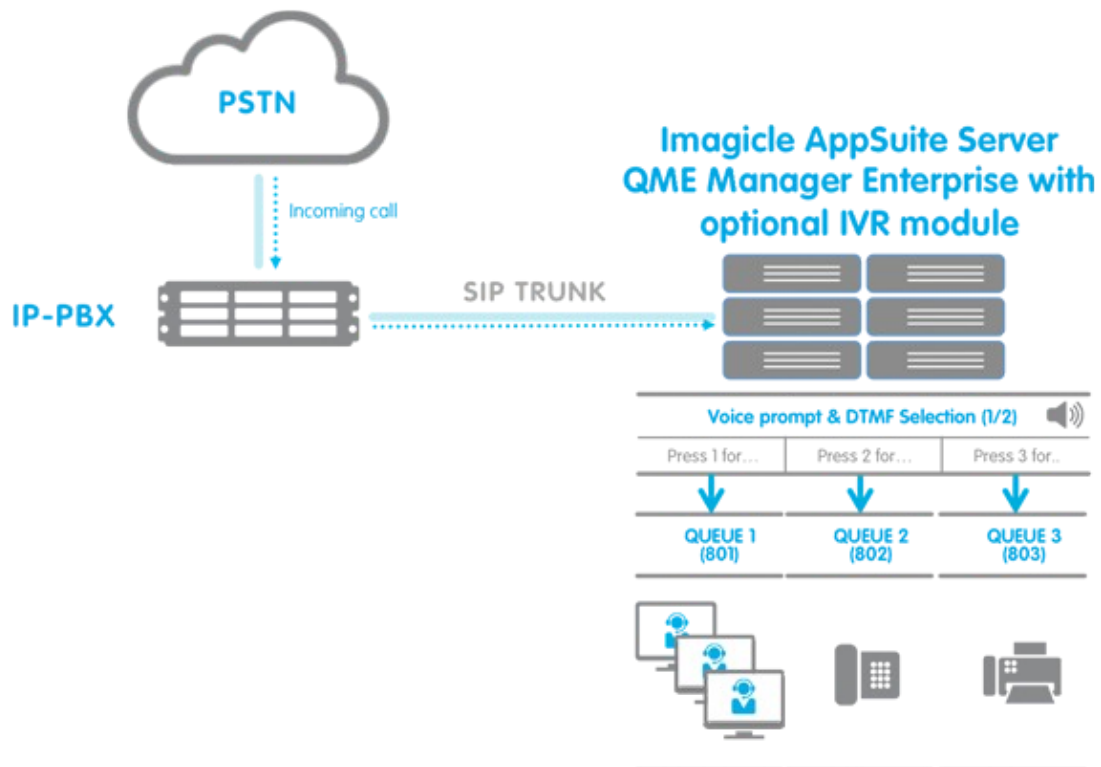
Since the IVR module can run standard VoiceXML scripts, it is possible to handle complex scenarios on a project basis, such as Caller ID, DB integration and more.

## How it works

The IVR module leverages Queue Manager Enterprise service and architecture. The PBX sends incoming calls from the PSTN to the QME SIP trunk. You must assign a unique number (i.e. the Pilot number) to each IVR service you need.

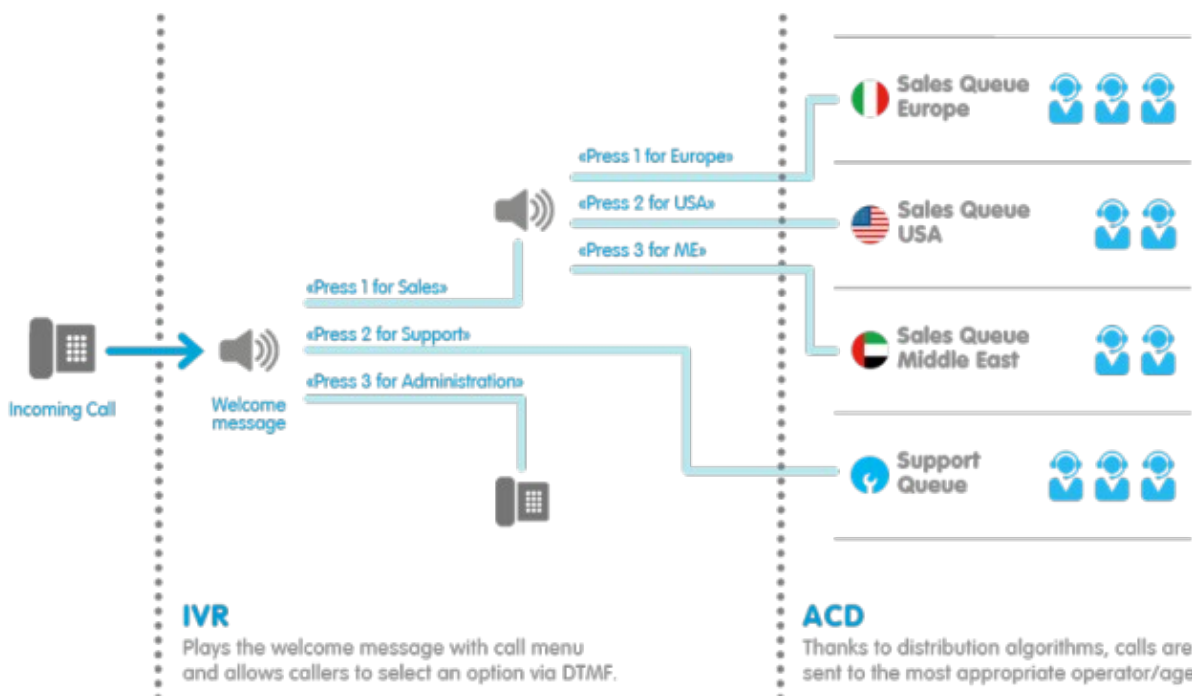
The IVR will answer the call and manage it according to the behaviors you defined through the web interface.

Typically, behaviors play audio messages, wait for DTMF tones and transfer or drop the calls. The destination for the calls include other IVR services, IP phones extensions, fax numbers, third party applications, or QME queues.



While the call is being transferred, the caller hears a ringback tone. IVR services can handle invalid or busy destinations and a definite number of retries.

IVR services send the call to their destination performing a blind transfer. Their control over retries is limited. If you need to play a welcome message and dispatch the call to one or more agents, using different distribution algorithm, prefer a QME queue to an IVR service.



## Configuration Task List

**Warning:** you must install and configure the Application Suite before being able to configure the single applications. Please go through the AppSuite Deployment, Main Configuration, and User Management sections before reading on.

Since the IVR module shares most of the configuration with Queue Manage Enterprise, we strongly recommend that you configure that application before the IVR module.

You can easily create a new IVR service with the following steps:

- Configure the PBX
- Check that the QME windows service is running
- Create and configure the service with the service Wizard or from a template
- Edit service settings
- Change the Date/Time schedule for the service
- Configure the voice prompts
- Activate the license

If you just want to test the application, you can create a special test service called "Echo service", which waits for DTMF selection and pronounces the pressed keys.

If you want to create a production service, please make sure you recorded the IVR audio prompts in WAV, MP3, AAC or mp3 format.

## PBX Configuration

The IVR Module for QME leverages standard SIP communication with the PBX. Since the IVR Module services are run by the QME service, you must configure the PBX as described in Queue Manager Enterprise system guide.

Please make sure CuCM is correctly configured for QME before reading on.

The only difference with QME concerns DTMF tones. IVR Module needs to receive the DTMF tones from the PBX using the RFC2833 standard (i.e. tones are sent out of band in the RTP stream), while QME queues do not negotiate DTMF tones.

### Cisco Gateway Configuration

Here is the configuration needed to let CuCM correctly negotiate DTMF tones with IVR Module.

If the gateway is controlled via **h323**, add the **dtmf-relay rtp-nte h245-alphanumeric** command to the dial-peer routing the calls to CallManager. E.g.

```
dial-peer voice 1002 voip
destination-pattern 6...
voice-class codec 1
session target ipv4:<CuCM IP Address>
dtmf-relay rtp-nte h245-alphanumeric
no vad
```

On a gateway with **SIP** configuration, add the **dtmf-relay rtp-nte** command. E.g.

```
dial-peer voice 1002 voip
destination-pattern 6...
voice-class codec 1
session target ipv4:<CuCM IP Address>
session protocol sipv2
dtmf-relay rtp-nte
no vad
```

If the gateway is controlled by **MGCP**, you must configure the "Type of DTMF relay" to NTE-CA on the gateway configuration.

## Product Specific Configuration Layout



Global ISDN Switch Type	4ESS
Switchback Timing*	Graceful
Switchback uptime-delay (min)	10
Switchback schedule (hh:mm)	12:00
Modem Passthrough*	Enable
Cisco Fax Relay*	Disable
T38 Fax Relay*	Enable
RTP Package Capability*	Enable
MT Package Capability*	Disable
RES Package Capability*	Disable
PRE Package Capability*	Enable
SST Package Capability*	Enable
RTP Unreachable OnOff*	Enable
RTP Unreachable timeout (ms)*	1000
RTCP Report Interval (secs)*	0
Simple SDP*	Enable
Type Of DTMF Relay*	NTE-CA

## Statistics and Reports

Imagicle Auto Attendant administrators and IVR Managers can analyze the details of managed calls, performing some statistics on the available data.

Auto Attendant stores in the database a Call Detail Record (CDR) for each call managed by an IVR service. Such CDRs include following information:

- Date-Time of call arrival
- Duration of the call, before being transferred or dropped
- IVR Service: name of the service that managed the call
- Entry Behaviour: the first behaviour that was triggered when the call arrived
- Type: type of behaviour
- Calling number
- Caller Name: available if Speedy Enterprise is licensed and the number can be found in a Speedy public directory
- Routed To: number the call was routed to
- Outcome: classification of the call based on how it was terminated (see below).
- Refused Reason: why the call was refused: Resource Exhausted or No Behaviour Defined for the pilot number.
- Concurrent calls
- Transfer Attempts
- Last Transfer Duration
- Last Transfer Destination
- Node ID: in HA installations, the cluster node that served the call
- Record Id: needed for diagnostics only. Useful to locate the call in log files

Each CDR includes the **outcome**, that is the final treatment applied to the call. Possible values are:

- Refused: the IVR could not answer the call, for example because there were no available channels. See the Refused Reason field.
- Abandoned: the call has been disconnected by the caller, while being served by the IVR service.
- Served and Completed: the call has been answered by the service, then dropped because of the selected behaviour.
- Served and Transferred: the call has been answered by the service, then transferred to a destination number.

## Running the Reports

The Reports can be accessed through the web interface by selecting relevant option on Auto Attendant menu.

The following reports are available:

- Call Volume by Service - Show number of calls and wait duration per service and per outcome
- Call Volume by Service and Behaviour - Show number of calls and wait duration per service, per behaviour and per outcome
- Trends - Show the number of calls per outcome and time trend
- Busy Channels analysis - Show busy channels statistics including average and max channel occupation over the day
- Call List - Show call detail records
- Interactive Call Detail Analysis - Search call details in an interactive grid

Each report can be filtered by clicking on "Report options" to restrict the results by specific parameters (e.g. call arrival time, service, behaviour...). Once filters are applied, you can "Run report" and display results in the browser.

Once generated, report can be exported in Word, Excel or PDF format. Scheduling feature is available, too.

## Interactive Call Detail Analysis

**Interactive Call Detail Analysis** allows you to dynamically group and filter the call records. Here is how to do it.

- First of all, select a time range for the calls you want to analyze
- To filter the records, use the drop down lists in the column headers
- To group data by a specific column, just drag it to the bar at the top of the grid
- A checkbox allows to enable the extended version, showing all the fields

You can group by one or more column. When data are grouped, the total number of calls and the average values are calculated per group.

## **Filtering data**

The table header include filtering capabilities for the data. Different column filters can be combined to extract the interesting portion of data. For example, setting a filter on the IVR Service column and on the Outcome column, extracts all CDRs related to that queue, with selected classification.

Changing any of the parameters updates the report contents. CDR are reported in the order they have been generated by Auto Attendant (descending order).

Once generated, each report can be exported in Word, Excel or PDF format.

## **Scheduling Reports**

Scheduled reports are executed periodically and sent via e-mail.

To schedule a report, setup the filtering options and run it from the web interface. Once you are satisfied with the result, press the "schedule" button. You'll be asked when the report must be automatically executed, and the recipients' e-mail address check or edit these settings later, through the Scheduled Reports web page.

## License Activation

To run IVR Module, you need two separate licenses. IVR Module for QME requires a valid Imagicle Queue Manager Enterprise license activated on the same server. Besides, IVR Module for QME has its own separate license, which can be purchased separately from Imagicle Queue Manager Enterprise.

### How the license works

The license for IVR Module for QME just unlocks the ability to create IVR services in QME. This feature does not affect the number of concurrent calls (channels) which can be managed by QME. In other words, the channels are shared between the two applications.

As a consequence:

- The number of IVR services you can create is unlimited, just as the number of QME queues.
- You cannot buy and activate the IVR module without buying a consistent QME license. For example, if you need to manage two concurrent incoming calls to a single IVR service, you need to buy two QME channels plus the IVR module license

The IVR service uses a QME channel when a call is connected to the service, and it is playing audio or waiting for DTMF tones. As soon as the call is dropped or successfully transferred to the destination, the channel is free and can handle a new IVR call or QME call.

### License Overflow

When a new incoming call requires IVR treatment but no channel is available, IVR will apply the overflow behaviour configured for the called IVR service. Moreover, an Imagicle Monitoring Service (MAM) event will be triggered to notify system administrator.

### Evaluation

The IVR Module for QME runs for 30 days after the installations, with full features, allowing you to test and evaluate the product. After 30 days, you must activate the license or the IVR services will be disabled.

When IVR Module for QME license is missing or expired, you will still be able to configure the product (creating IVR services, behaviours, schedules, etc...), but no IVR treatment will be applied to incoming calls. Whenever a new call arrives, a MAM event will be triggered to notify system administrator.

### How to activate the license

The license can be activated by purchasing the product from Imagicle. You can enable the license in the license page as described in the Application Suite common configuration section of this guide.

To activate the license, follow the standard procedure you can find in the General configuration section.

### How to check the license status

As for the other applications, you can check the license status from the web interface, accessing the License page through the Admin menu.

## High Availability

**Warning:** High availability for Imagicle Applications is provided through cluster configuration. Please check the general **High Availability** section of the Imagicle Application Suite administration guide before reading on.

### How it works

The IVR Module supports High availability natively, provided QME redundancy is configured on the PBX. Please see "HA Configuration and Call Survivability" in QME Administration Guide.

### Licensing

You need a different IVR Module license for each node of the cluster.

## Product Configuration

For examples and details on how to create IVR services, please read the Advanced Queuing and Auto Attendant User Guide.

This section focuses on some technical details, which the Administrators should know to understand the application behavior better.

### User permissions

To create or modify an IVR service, you must be an Administrator (Complete Management). This will give you full control over the application.

You can set user permissions from the Admin > User Management page.

### Adding a new IVR service

You can add a new IVR service from the Services web page, by clicking on "Add a new IVR service". The service can be created via a Wizard or starting from a template.

The IVR Service name and the Pilot number must be unique for the whole system. Since Advanced Queuing and Auto Attendant are sharing same SIP Trunk, the number and the name must be also different from ACD queue names and numbers.

Usually you define one IVR service (with one pilot number) for each site in your organization, or for each PSTN public number.

For complex scenarios, you can define more than one IVR service and cascade them, transferring the call from the first service to the other services pilot numbers.

### Behaviors and the Main Behavior

Behaviors define how the call is treated after being answered. Behaviors can play audio files, wait for DTMF tones, drop or transfer the calls, and so on.

The first behavior you add to the service becomes the Main Behavior that will be executed when the call is answered.

The following behaviors are available:

### Menu

Through the Menu behavior, you can easily create menus and submenus with DTMF selections. You design the call flow through the web interface, specifying what happens when the user presses a digit on the phone, how many times to repeat the intro message on timeout and so on.

Each option (i.e. DTMF) can lead to a sub menu. The timeout and invalid selection options are inherited by the sub menus, but can be overridden if needed.

### Transfer to

All AutoAttendant call transfers are blind type. They fail instantly if the destination is invalid or cannot be reached. In this case, the caller hears no ringback.

If the destination can be reached, the caller hears the ringback tone from the Calling Platform.

For each transfer you can define a message to be played before initiating the transfer, the number of retries, and messages to be played in case of failure.

After the retries are exceeded, you can choose to drop the call or trigger an existing behavior.

If the Calling Platform in use allows it, the call can be transferred to internal numbers/SIP URI, to a third-party answering groups or ACD, to a voice mail system or to any PSTN number.

## Play message

This behaviour allows to simply play a voice prompts, previously created in MP3 or WAV format and uploaded from web portal.

## Dial in

Dial-in behavior allows caller party to enter the digit of the internal number/extension to be connected to. You can define the minimum and maximum number of digits to be collected, a timeout, and if the caller must press # to terminate digits collection.

## Conversational IVR

This behaviour allows to trigger a VoiceBOT script created in Conversational AI (relevant license required). Please remember that interacting with Cloud-based Imagicle ConvAI application might require to setup a new web server URL, as explained [here](#).

## Audio prompts

You can easily add and audio prompt by clicking on the textbox close to the message label. It is a best practice to add the transcript of the text pronounced in the audio file.

Audio files can be recorded in .MP3 or Wav formats, and they will be automatically converted to PCM 8Kz 16-bit mono.

To check the audio prompt quality after conversion, save the behavior, then click on the play icon, then on the download icon. Play the file locally on your PC.

All the audio messages are optional.

To delete the message, click on the trash icon.

**Note:** please download and backup any audio file before removing it from an existing IVR service, because it will be deleted from the server.

## Scheduling behaviors

The timetable allows you to change the behavior associated to a pilot number based on the date and time when the call arrives to the IVR.

For example, you could define a "day" behavior and a "night" behavior. Select the Night behavior as default, pressing the "Outside of defined schedules, execute the following behavior" button on the bottom of the timetable.

Then add a new scheduled behavior for the working hours, selecting the "day" behavior.

The holiday and special events allows you to override the normal schedule, executing different behaviors.

## IVR Managers

When you edit an existing IVR Service as administrator, you can assign one or more **IVR Managers** from "Permissions" tab.

Each IVR Manager associated to an IVR Service has the following capabilities:

- Change overflow treatment
- Edit Auto Attendant behaviors and relevant voice prompts
- Amend weekly timetable, by choosing opening/closing time
- Configure holidays
- Add mode IVR Managers
- Run Auto Attendant reports, except "Busy Channels"

## Modifying an existing IVR service



You can safely edit any an existing IVR service without the need to restart the service. If you make changes, calls currently connected to the IVR service will retain the old behavior. The changes apply to new calls.

### **Other IVR service settings**

When you edit an existing IVR service, you can define an additional behavior, which is triggered when the maximum number of concurrent calls is exceeded ("overflow"). To understand how Auto Attendant license relates to Advanced Queuing licensed channels, please read the "License Activation" page in this guide.

You can also define a default timeout for DTMF selection.

### **Data Retention**

From IVR "Settings" tab, it is possible to specify maximum lifetime (in days) of historical data. That is, older data will be periodically deleted. Set it to zero (0) to disable automatic deletion. Please, notice that this setting may impact on the database size.

### **Starting the service**

Since Advanced Queuing and Auto Attendant share the same service, you can start and stop the service from Advanced Queuing's Manage Service page. Stopping Advanced Queuing service stops all the queues, as well as all IVR services.

## Troubleshooting

### How to use the troubleshooting guide

This page describes basic troubleshooting techniques and most frequent issues you may face during the application setup and usage.

The first part describes the basic tests to be made after you completed the configuration task list. Those test can reveal issues in the configuration and can help you to identify them.

The second part is a list of common issues and their causes. Look for the symptom and follow the tips. To know how to configure the product, please refer to the relevant pages in this guide.

Please understand that the problem may be related to complex PBX and network configurations, and that is not possible to list all them all. This guide must be considered as a tool to guess the origin of the issue.

### 1. Post installation tests

Create a new Echo service from the Add new service page. Make sure the Auto Attendant Module can receive DTMF tones placing a test call from the PSTN with your mobile phone. Press each button on the keypad once, and the IVR service will tell you if it received the key digits, or if it received the same digit twice.

### 2. Common issues

#### No DTMF digits are received or they are received twice

Please make sure your PBX does not send OOB (out of band) DTMF. Imagicle Auto Attendant only supports RFC2833

#### When a call comes in from the PSTN into the Auto attendant, and the call is transferred back from Auto Attendant to a mobile number, the Imagicle music on hold is heard without the actual transfer happening

- Check that a Calling Search Space to reach an outbound route pattern is set on both the following parameters of the Advanced Queuing SIP trunk:

Rerouting Calling Search Space	ALL	▼
Out-Of-Dialog Refer Calling Search Space	ALL	▼

#### When a call comes into an Auto Attendant service, and an option is selected to transfer to a queue, a closing message is heard

- Note that both Auto Attendant and Advanced Queuing have their own weekly timetable. Auto Attendant can be open, and maybe the destination queue might be closed due to end of shift.

### 3. Reading Wireshark captures

If you capture the network communication between Imagicle server and your PBX with Wireshark or with another packet sniffer, this is what you should see in the SIP call flow.

When a new call arrives, the PBX sends an INVITE to the Auto Attendant module. Auto Attendant replies with TRYING, RINGING and OK. "OK" signaling has SDP headers which include codec and DTMF capabilities.

After the OK, the PBX should send ACK SDP and begin to send the RTP stream encoded with one of the supported codecs (G.711, G.729). Auto Attendant sends RTP stream too, if audio prompts are configured in the behavior.

If the call must be transferred, Auto Attendant sends a REFER with the new destination. The PBX answers "202 accepted" and sends at least two NOTIFY messages to tell Auto Attendant that the destination is ringing. The NOTIFY messages should contain TRYING, RINGING.

When the transferred call is answered, the PBX sends a BYE, followed by a NOTIFY (containing OK) to inform that the transfer is accomplished.

If the transfer fails, one of the NOTIFY messages includes the error code (e.g. 404 NOT FOUND).

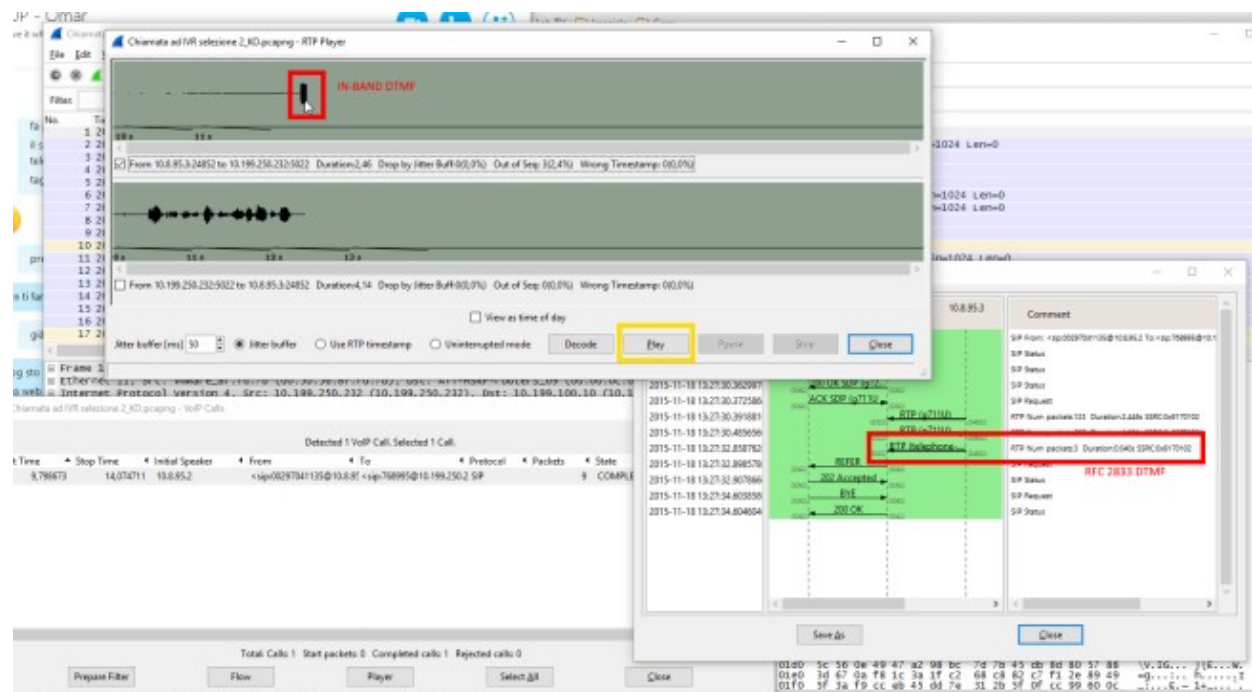
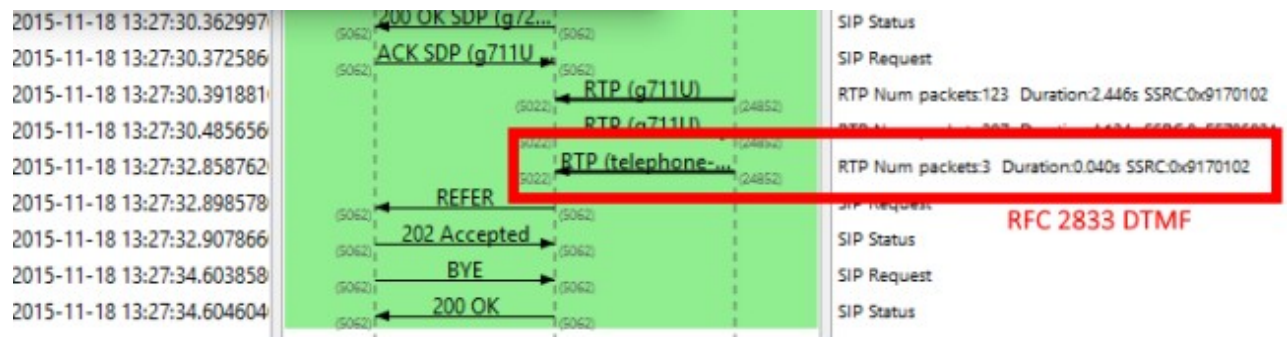
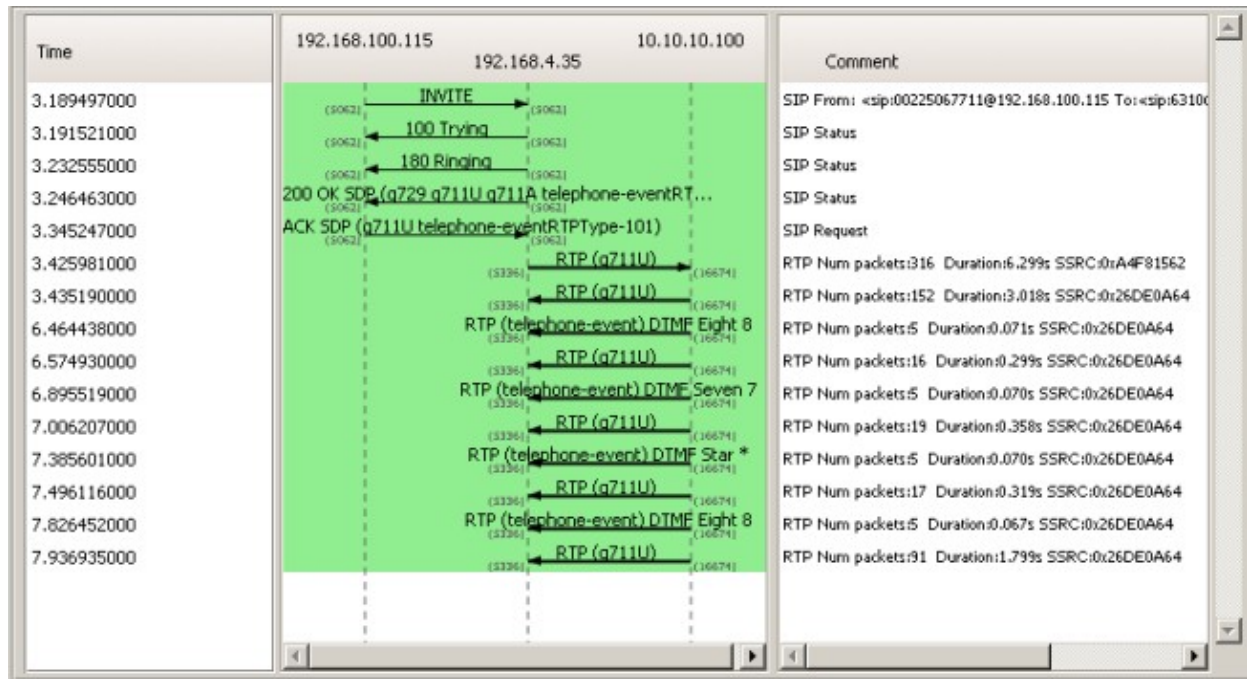
Expression..	Clear	Apply	Save	Time	192.168.150.242	192.168.150.127	Comment
8.150.242	192.168.150.127	SIP		2016-02-11 15:51:44.109959	INVITE		SIP From: <sip:202211779103@192.168.150.242> To: <sip:8127
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:44.111741	100 Trying		SIP Status
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:44.149134	180 Ringing		SIP Status
8.150.127	192.168.150.242	SIP/SDP		2016-02-11 15:51:44.262344	200 OK SDP (g72		SIP Status
8.150.127	192.168.150.242	SIP/SDP		2016-02-11 15:51:44.632306	RTP (g723)		RTP Num packets:7 Duration:0.120s SSRC:0xCT355000
8.150.242	192.168.150.127	SIP/SDP		2016-02-11 15:51:44.762100	200 OK SDP (g72		SIP Status
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:44.772466	RTP (g723)		RTP Num packets:210 Duration:4.170s SSRC:0xCT330000
8.150.242	192.168.150.127	SIP		2016-02-11 15:51:44.847919	ACK SDP (g729)		SIP Request
8.150.242	192.168.150.127	SIP/sipf		2016-02-11 15:51:44.866062	RTP (g723)		RTP Num packets:1305 Duration:25.074s SSRC:0xCAA6521
8.150.242	192.168.150.127	SIP/sipf		2016-02-11 15:51:48.953491	REFER		SIP Request
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:48.964482	202 Accepted		SIP Status
8.150.242	192.168.150.127	SIP/sipf		2016-02-11 15:51:48.964482	NOTIFY		SIP Request
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:48.970440	200 OK		SIP Status
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:49.145532	NOTIFY		SIP Request
8.150.127	192.168.150.242	SIP		2016-02-11 15:51:49.155970	200 OK		SIP Status
8.100.115	192.168.150.127	SIP		2016-02-11 15:52:10.941531	BYE		SIP Request
8.150.127	192.168.100.115	SIP		2016-02-11 15:52:10.952059	NOTIFY		SIP Request
				2016-02-11 15:52:10.930502	200 OK		SIP Status
				2016-02-11 15:52:10.935041	200 OK		SIP Status

## Checking DTMF tones

As per RFC2833 standard, the tones are sent out of band in the in RTP stream. The Payload type is 101.

In the capture, you should see something like:

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
```



#### **4. How to get the log files**

The application log files are shared with Advanced Queuing. You need to get the log files only if requested by Imagicle Support Team. To retrieve log files, please read [here](#).