Genesys Voice Platform

MediaManager Section

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- DTMFSinglePacket
- rtpdejitterdelay
- RtpDTMFPayloadType
- RtpLocalCodecType
- rtprecvaudiobuffersize

DTMFSinglePacket

Default Value: true
Valid Values:
Changes Take Effect: After restart

Specifies whether to send a single RFC2833 packet for every DTMF digit recognized at the Dialogic side. If this is set to false, then PSTNC tries to send 8 packets corresponding to 1280 ticks

rtpdejitterdelay

Default Value: 0
Valid Values: rtpdejitterdelay must be an integer that is greater than or equal to 0 and less than or equal to 10000.
Changes Take Effect: At start/restart

Specifies the total duration (in milliseconds) of RTP packets to buffer for the inter-arrival dejittering purpose. This will translate to an initial delay before the packets are dispatched internally for further processing. 0 disables the inter-arrival jitter removal functionality.

RtpDTMFPayloadType

Default Value: 101
Valid Values:
Changes Take Effect: After restart

The payload/encoding type of DTMF packets 96, 97, 98, 99, 100, etc

RtpLocalCodecType

Default Value:
Valid Values:
Changes Take Effect: After restart

Local Codec number to be used in RTP values are 0 - Mu law, 8 - A law

rtprecvaudiobuffersize

Default Value: 0
Valid Values: rtprecvaudiobuffersize must be an integer that is greater than or equal to 0 and less than or equal to the maximum integer as defined by the Genesys Administrator Help.
Changes Take Effect: At start/restart

Specifies the buffer size used for the RTP packet reordering feature and audio packets. This optional feature provides support for receiving RTP packets out of order and reordering them before further processing. If enabled, the suggested normal value is 1000. Note that if rtpdejitterdelay is non-zero, and the resulting jitter buffer size to accommodate the delay is greater than the size defined by this configuration, the size will be set to the greater value specified by the rtpdejitterdelay. Setting the buffer size to 0 and rtpdejitterdelay to 0 disables the RTP packet reordering feature for audio. Note that if the selected buffer size is non-zero but too small (say less than 200), the packet may not fit and be dispatched immediately without the re-ordering.