

GENESYS

This PDF is generated from authoritative online content, and is provided for convenience only. This PDF cannot be used for legal purposes. For authoritative understanding of what is and is not supported, always use the online content. To copy code samples, always use the online content.

Integration Reference Manual

Asterisk as a Media Server

Asterisk as a Media Server

Contents

- 1 Asterisk as a Media Server
 - 1.1 Configuring Asterisk
 - 1.2 Configuring DN Objects

In order for Asterisk to work as a media server integrated with SIP Server, you must enhanced the Asterisk dialing plan with several Genesys macros and global variables as described in this section.

Configuring Asterisk

Dialing Plan Global Variables

You must add the following list of global variables to the [globals] section of the Asterisk dialing plan.

```
SIP_PREFIX=.*sip:.*@.*:[0-9]+.*
DIG_PRMT_REGEX=silence/1?[0-9]
FIND_CLT_REGEX=${SIP_PREFIX}play=[ ]*(music/collect).*
FIND_PLY_REGEX=${SIP_PREFIX}play=[ ]*([^>\;]*)[>\;].*
FIND_REP_REGEX=${SIP_PREFIX}repeat=[ ]*([^>\;]*)[>\;].*
FIND_REC_REGEX=${SIP_PREFIX}record=[ ]*([^>\;]*)[>\;].*
FIND_COF_REGEX=.*sip:conf=(.*)@.*:[0-9]+.*
DEFAULT_FILE_TO_PLAY= /var/lib/asterisk/moh/fpm-calm-river
```

Variable DEFAULT_FILE_TO_PLAY points to the default music file that is played for the Genesys treatments. In the example, above it refers to the voice file, which comes with Asterisk (if Asterisk is installed in the standard directory). You can change this reference to any other file in the actual deployment.

Dialing Plan Macro to Perform Genesys Treatments

You must add this treatment to the Asterisk dialing plan to perform Genesys treatments.

```
[macro-treatment]
;
; ${ARG1} - SIP_HEADER(To)
;
; IF treatment == CollectDigits
;
exten => s, 1, Answer
exten => s, 2, Set(collect=$["${ARG1}":"${FIND_PLY_REGEX}"])
exten => s, 3, GotoIf($[$["${collect}"="music/collect"] | $["${collect}"="music/
silence"]] ? 15 : 20)
exten => s, 15, macro(get-digits,${collect})
exten => s, 16, Goto(s,99)
;
; ELSE IF treatment == record
;
exten => s, 20, Set(rec_file=$["${ARG1}":"${FIND_REC_REGEX}"])
exten => s, 21, Set(ply_file=$["${ARG1}":"${FIND_PLY_REGEX}"])
exten => s, 22, GotoIf($[${LEN(${rec_file})} != 0] ? 30 : 40)
;
; Recording Treatment
exten => s, 30, GotoIf($[${LEN(${ply file})} != 0] ? 32 : 31)
```

```
exten => s, 31, Playback(${ply file}) ;
exten => s, 32, Record(genesys-rec-${rec file}.wav) ;can't detect|report dtmf
exten \Rightarrow s, 33, Goto(s,98)
; ELSE
; Play treatment
exten \Rightarrow s, 40, GotoIf(\{\{\{LEN(\{\{ply file\}\}\}\} = 0\}\}? 41 : 43)
exten => s, 41, Set(ply_file=${DEFAULT_FILE_T0_PLAY})
exten \Rightarrow s, 42, Goto(s,44)
exten => s, 43, Set(ply_count=$["${ARG1}":"${FIND REP REGEX}"])
exten \Rightarrow s, 44, GotoIf(\{[\{\{LEN(\{\{ply count\})\} = 0\} \mid \{[\{ply count]\} = \{[\{ply count]\} = \{[\{ply count]\} \}]\}\}
50:60)
; Playback forever
exten => s, 50, Playback(${ply_file})
exten => s, 51, GotoIf($[${PLAYBACKSTATUS}=FAILED] ? 52 : 50) ;Goto(s, 50)
exten \Rightarrow s, 52, Goto(s, 99)
; Counted playback
; here probably possible to use background()
exten => s, 60, Playback(${ply_file}) ; Playback
exten => s, 61, Set(ply count=$[${ply count} - 1])
exten \Rightarrow s, 62, GotoIf(\{\{\{\{\{p\}\}\}\}\}\} count\} > 0] & \{\{\{p\}\}\}\} \{\{p\}\}\} \{\{p\}\}
exten => s, 98, Hangup
exten => s, 99, NoOp(end-withot-hagup)
```

Dialing Plan Macro to Collect DTMF Digits

You must add this treatment to the Asterisk dialing plan to collect DTMF digits. Replace <COLLECT-MESSAGE-PLACEHOLDER> in the macros below with the name of the file to play to announce digit collection.

```
[macro-get-digits]
exten => s,1, GotoIf($[$[${ARG1}=music/collect] | $[${ARG1}=music/silence]] ? 2 : 3)
exten => s,2, Set(ARG1=silence/2)
exten => s,3,Read(dncdigits,<COLLECT-MESSAGE-PLACEHOLDER>,1,s)
exten => s,4,SendText(Signal=${dncdigits})
exten => s,5, Goto(macro-get-digits,s,3)
```

Dialing Plan Macro to Create a Conference

You must add this treatment to the Asterisk dialing plan to organize a conference using the Asterisk MeetMe application.

```
[macro-conf]
exten => s, 1, Set(conf_id=$["${ARG1}":"${FIND_COF_REGEX}"])
exten => s, 2, NoOp(${ARG1})
exten => s, 3, GotoIf($[${LEN(${conf_id})} != 0] ? 4 : 20)
exten => s, 4, Set(rec_file=$["${ARG1}":"${FIND_REC_REGEX}"])
exten => s, 5, GotoIf($[${LEN(${rec_file})} != 0] ? 6 : 8)
exten => s, 6, MeetMe(${conf_id},drq)
exten => s, 7, Goto(s,20)
exten => s, 8, MeetMe(${conf_id},dq)
exten => s, 20, NoOp()
```

Integrating Genesys Macros into the Dialing Plan

The Asterisk dialing plan all macros provided above. This section suggests one possible way to do that. Add the following macro in the dialing plan:

```
[moh_conf_treatment]
include => macro-treatment
exten => annc, 1, macro(treatment,${SIP_HEADER(To)})
exten => _co[n]f=., 1, macro(conf,${SIP_HEADER(To)})
```

You must include this macro into the context used to process agent calls. If there is no special context created for this purpose, you must include macro into the default dialing plan context.

```
[default]
include => moh_conf_treatment
```

Media Files

Media files used for the Genesys treatments should be placed into the standard Asterisk sounds directory. The default location of this directory is:

/var/lib/asterisk/sounds

Call recordings created by Asterisk are also stored in this directory. There are two types of recordings, which can be activated by SIP Server:

- Regular (proxy mode)
- Emergency

By default, names of the recordings made in regular mode are prefixed with genesys-rec. Names of the emergency recordings start with the meetme-conf-rec prefix. In both cases, the name prefix is followed by a conference ID.

Configuring DN Objects

SIP Server utilizes media services through the DNs of type Voice over IP Service configured under the Switch object. The Voice over IP Service DNs have a service-type configuration option, which defines the kind of service this DN can provide. SIP Server selects an appropriate DN when the client application requests a media service.

When you use Asterisk as a media server for SIP Server, you should configure the Voice over IP Service DNs with the following service-type values in the SIP Server Switch object:

- mcu
- treatment
- recorder
- music

For information about configuring DNs for different types of services, see the "SIP Device



Configuration" topic of the SIP Server Deployment Guide.