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Dial()

Synopsis

Attempts to establish a new outgoing connection on a channel, and then link it to the existing input channel.

Description

- Dial(*type*/*identifier*,*timeout*,*options*,*URL*) — dial one channel
- Dial(*type1*/*identifier1***&***type2*/*identifier2***&***type3*/*identifier3...*,*timeout*,*options*,*URL*) — dial multiple channels

Attempts to "dial out" on all the specified channels (each specified by a *type* and *identifier*) simultaneously. The first channel that answers "wins", and all the other outgoing channels are hung up. The originating channel that triggered this Dial command is then Answered, if necessary, and the two channels are connected together ("bridged") allowing a conversation to take place between them. When the channel that triggered the Dial command hangs up, the Dial command exits.

Parameters

- *type* specifes the channel type. It should be one of the registered channel types, such as "Zap", "SIP", "IAX2", and so on.
- *identifier* specifies the "phone number" to dial on that channel. The format of the "phone number" depends on the channel, and may contain additonal parameters (e.g. a distinctive ring parameter) specific to the channel module in question; the Dial command simply passes *identifier* to the channel module to process in whatever way is appropriate. See the documentation for the individual channel modules to learn about the correct format for specifying the *identifier* for the Dial command, and the options available to you when doing so.
- If you wish to specify more than one channel for the Dial command to try — remembering that it will dial out on all of them simultaneously — separate them with the **&** symbol. The channels can be different *types*. See Examples, below.
- The *timeout* parameter is optional. If not specifed, the Dial command will wait indefinitely, exiting only when the originating channel hangs up, or all the dialed channels return a busy or error condition. Otherwise it specifies a maximum time, in seconds, that the Dial command is to wait for a channel to answer.
- The *options* parameter, which is optional, is a string containging zero or more of the following flags and parameters:
  - **t**: Allow the *called* user to transfer the call
  - **T**: Allow the *calling* user to transfer the call
  - **r**: Generate a ringing tone for the calling party, passing no audio from the called channel(s) until one answers. Use with care and don't insert this by default into all your dial statements as you are killing call progress information for the user. Really, you almost certainly do not want to use this. Asterisk will generate ring tones automatically where it is appropriate to do so. "r" makes it go the next step and additionally generate ring tones where it is probably not appropriate to do so.
  - **R**: Indicate ringing to the calling party when the called party indicates ringing, pass no audio until answered. This is available only if you are using kapejod's bristuff.
  - **m**: Provide Music on Hold to the calling party until the called channel answers. This is mutually exclusive with option 'r', obviously. Use m(class) to specify a class for the music on hold.
  - **n**: (Asterisk 1.1 and later) July 2005 bug 752 was included in CVS (Asterisk 1.1) and enhances the privacy manager considerably. As part of this patch, the 'n' flag to Dial got changed to be used as part of the privacy features, instead of being the 'dont jump to +101' flag. That flag is now 'j'.
  - **j**: Asterisk 1.1 and later: Don't jump to +101
  - **M(***x***)**: Executes the macro (x) upon connect of the call
  - **h**: Allow the callee to hang up by dialing *
  - **H**: Allow the caller to hang up by dialing *
  - **C**: Reset the CDR (Call Detail Record) for this call. This is like using the NoCDR command
  - **P(***x***)**: Use the PrivacyManager, using *x* as the database (*x* is optional)
  - **g**: When the called party hangs up, exit to execute more commands in the current context.
  - **A(***x***)**: Play an announcement (*x*.gsm) to the called party.
  - **S(***n***)**: Hangup the call *n* seconds AFTER called party picks up.
  - **D(***digits***)**: After the called party answers, send *digits* as a DTMF stream, then connect the call to the originating channel.
  - **L(**x[:y][:z]**)**: Limit the call to 'x' ms, warning when 'y' ms are left, repeated every 'z' ms) Only 'x' is required, 'y' and 'z' are optional. The following special variables are optional for limit calls: (pasted from app_dial.c)
    - **LIMIT_PLAYAUDIO_CALLER** - yes|no (default yes) - Play sounds to the caller.
    - **LIMIT_PLAYAUDIO_CALLEE** - yes|no - Play sounds to the callee.
    - **LIMIT_TIMEOUT_FILE** - File to play when time is up.
    - **LIMIT_CONNECT_FILE** - File to play when call begins.
    - **LIMIT_WARNING_FILE** - File to play as warning if 'y' is defined. If **LIMIT_WARNING_FILE** is not defined, then special sound macro to auto-say the time left is used ("You have [XX minutes] YY seconds").
  - **f**: forces callerid to be set as the extension of the line making/redirecting the outgoing call. For example, some PSTNs don't allow callerids from other extensions than the ones that are assigned to you.
  - **w**: Allow the *called* user to start recording after pressing *1 or what defined in features.conf (Asterisk > v1.0.x)
  - **W**: Allow the *calling* user to start recording after pressing *1 or what defined in features.conf (Asterisk > v1.0.x)
- The optional *URL* parameter will also be sent to the called party upon successful connection, if the channel technology supports the sending of URLs in this way.

New: Introduced in/for Asterisk 1.1, see bug 2905

You can now add args to the macro by using a '^' char

```
Dial(Zap/1|60|M(mymacro^cat^dog^bark))
```

Also, the macro can set the MACRO_RESULT variable to do the following:

```
ABORT - Hangup both legs of the call
BUSY
CONTINUE - Hangup the called party and continue on in the dialplan from where you called Dial
GOTO:<context>^<exten>^<priority> - Transfer the call.
```

Example usage of the above patch

```
screen-record: Please record your name press pound when finished.
screen-from: You have a call from
screen-accept: Press 1 to accept this call or any other key to reject.

exten => 890,1,Wait(0.2)
exten => 890,2,Playback(screen-record)
exten => 890,3,SetVar(SCREEN_FILE=/tmp/${CALLERIDNUM}-${EPOCH})
exten => 890,4,Record(${SCREEN_FILE}.gsm|6|25)
exten => 890,5,Dial(SIP/16|60|gM(screen^${SCREEN_FILE}))
```

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```
exten => 890,6,Voicemail(10@default)

[macro-screen]
exten => s,1,Wait(0.2)
exten => s,2,Playback(screen-from)
exten => s,3,Playback(${ARG1})
exten => s,4,Read(ACCEPT|screen-accept|1)
exten => s,5,GotoIf($[${ACCEPT} = 1 ] ?7:6)
exten => s,6,SetVar(MACRO_RESULT=CONTINUE)
exten => s,7,System(/bin/rm ${ARG1})
```

Notes:

- Do not put spaces between the arguments to the Dial command, it will not work.
- When options *t* or *T* are applied, Asterisk will remain in the media path, even if *canreinvite=yes* (a SIP channel option) has been specified.

Return codes

If all the called channels are busy, Dial will exit with a return code of 0 and will continue in the current context at priority n+101, if it exists, where n is the priority of the Dial command. This allows you to set up different behavior for a busy response as opposed to a no-answer response.

If Dial was unable to place the call for some other reason, such as timing out before any channel answered successfully, then Dial will exit with a return code of 0 and execute the next higher priority (n+1) command.

If the *g* option is specified, and the called party hangs up before the calling party, then Dial exits with a return code of 0 to continue execution at priority n+1.

Dial returns -1 if the originating channel hangs up, or if the call is bridged and either of the parties in the bridge terminate the call.

Example by dsfr (Chris Hozian) - Can be used for home answering machine with call screen. Provides CallerID to handset even if you don't pay for CallerID. Must associate CID name with number in Asterisk database in order to work.

```
[from-iax-fwd]

exten => night-mode,1,WaitForRing(30)
exten => night-mode,n,Goto(s|night-mode-start)

exten => ${FWD-HOME-NUMBER},1,Set(INVALID-PRIVACY-TRIES=0) ; used for determining number of invalid tries used during privacymanager.
exten => ${FWD-HOME-NUMBER},n,Set(TIMEOUT(response)=20)
exten => ${FWD-HOME-NUMBER},n,Set(TIMEOUT(digit)=7)
exten => ${FWD-HOME-NUMBER},n,SetMusicOnHold(default)
exten => ${FWD-HOME-NUMBER},n,GotoIfTime(20:00-7:59|*|*|*?night-mode|1)
exten => ${FWD-HOME-NUMBER},n,Answer
exten => ${FWD-HOME-NUMBER},n,Wait(2)
exten => ${FWD-HOME-NUMBER},n,Zapateller
exten => ${FWD-HOME-NUMBER},n,Goto(skip-night-mode)
exten => ${FWD-HOME-NUMBER},n(night-mode-start),Answer
exten => ${FWD-HOME-NUMBER},n,Wait(2)
exten => ${FWD-HOME-NUMBER},n(skip-night-mode),Playback(dsfr-hozian-residence) ; welcome the caller.
exten => ${FWD-HOME-NUMBER},n,Playback(dsfr-greeting) ; welcome the caller.
exten => ${FWD-HOME-NUMBER},n,NoOp(Pre Privacy Manager Number ${CALLERID})
exten => ${FWD-HOME-NUMBER},n(privacy-manager),PrivacyManager
exten => ${FWD-HOME-NUMBER},s+101,GotoIf($[${INVALID-PRIVACY-TRIES} = ${MAX-PRIVACY-TRIES}] ?i|1)
exten => ${FWD-HOME-NUMBER},n,Playback(dsfr-privacy-invalid) ; explain that # is not valid and that 10 digits must be entered.
exten => ${FWD-HOME-NUMBER},n,Set(INVALID-PRIVACY-TRIES=$[${INVALID-PRIVACY-TRIES} + 1])
exten => ${FWD-HOME-NUMBER},n,Goto(privacy-manager)
exten => ${FWD-HOME-NUMBER},privacy-manager+1,NoOp(Post Privacy Manager Number ${CALLERID})
exten => ${FWD-HOME-NUMBER},n,LookupCIDName
exten => ${FWD-HOME-NUMBER},n,NoOp(Post LookupCIDName ${CALLERID})
exten => ${FWD-HOME-NUMBER},n,Playback(pls-rcrd-name-at-tone)
exten => ${FWD-HOME-NUMBER},n,Playback(and-prs-pound-whn-finished)
exten => ${FWD-HOME-NUMBER},n,Set(SCREEN-FILE=/var/lib/asterisk/tmp/${TIMESTAMP}-${CALLERIDNUM})
exten => ${FWD-HOME-NUMBER},n,Record(${SCREEN-FILE}.gsm|${SILENCE-SECONDS}|${SCREEN-FILE-SECONDS})
exten => ${FWD-HOME-NUMBER},n,Playback(this-call-may-be-monitored-or-recorded)
exten => ${FWD-HOME-NUMBER},n,Playback(pls-hold-while-try)
exten =>
${FWD-HOME-NUMBER},n,Dial(${PHONE-1}|${INCOMING-RING-SECONDS}|gmM(from-iax-fwd-screen^${SCREEN-FILE}^${UNIQUEID}^${CALLERIDNUM}))
exten => ${FWD-HOME-NUMBER},n,GotoIf($[${${UNIQUEID}} = 1] ?hangup) ; checks whether bypass voicemail option is set in macro-from-iax-fwd-screen.
exten => ${FWD-HOME-NUMBER},n(from-iax-fwd-ivr),System(/bin/rm ${SCREEN-FILE}.gsm)
exten => ${FWD-HOME-NUMBER},n,Goto(from-iax-fwd-ivr|s|1)
exten => ${FWD-HOME-NUMBER},n(hangup),System(/bin/rm ${SCREEN-FILE}.gsm)
exten => ${FWD-HOME-NUMBER},n,Hangup
exten => i,1,System(/bin/rm ${SCREEN-FILE}.gsm)
exten => i,n,Playback(call-terminated)
exten => i,n,Playback(goodbye)
exten => i,n,Hangup
exten => t,1,System(/bin/rm ${SCREEN-FILE}.gsm)
exten => t,n,Playback(call-terminated)
exten => t,n,Playback(goodbye)
exten => t,n,Hangup
exten => h,1,System(/bin/rm ${SCREEN-FILE}.gsm)

[macro-from-iax-fwd-screen]
exten => s,1,Set(MACRO_RESULT=GOTO:from-iax-fwd^${FWD-HOME-NUMBER}^from-iax-fwd-ivr)
exten => s,n,Set(TIMEOUT(response)=15)
exten => s,n,Playback(call)
exten => s,n,Playback(from)
exten => s,n(repeat-screen),Playback(${ARG1})
exten => s,n(repeat-options),Read(ACCEPT-CALL|dsfr-macro-from-iax-fwd-options|1)
exten => s,n,GotoIf($["${ACCEPT-CALL}" = ""] ?t|1)
exten => s,n,GotoIf($[${ACCEPT-CALL} = 1] ?call-accepted)
exten => s,n,GotoIf($[${ACCEPT-CALL} = 5] ?repeat-screen)
exten => s,n,GotoIf($[${ACCEPT-CALL} = 9] ?call-declined:repeat-options)
exten => s,n(call-declined),Playback(dsfr-screen-declined) ; declined - caller will be sent to ivr.
exten => s,n,GotoIf($[${MACRO_RESULT} = GOTO:from-iax-fwd^${FWD-HOME-NUMBER}^from-iax-fwd-ivr] ?skip-monitor)
exten => s,n(call-accepted),Playback(dsfr-screen-accepted) ; accepted - call will be bridged momentarily.
exten => s,n,Set(MACRO_RESULT=)
exten => s,n,Set(${ARG2}=1|g) ; if set to 1 then bypass voicemail.
exten => s,n,Playback(this-call-may-be-monitored-or-recorded)
exten => s,n,Monitor(wav49|${TIMESTAMP}-${ARG3}|mb)
exten => s,n(skip-monitor),NoOp
exten => t,1,Playback(connection-timed-out)
exten => t,n,Goto(s|call-declined)

[from-iax-fwd-ivr]
exten => s,1,Set(INVALID-IVR-TRIES=0) ; used for determining number of invalid tries used during ivr.
exten => s,n,Set(RETRIES-FWD-WORK=0) ; used for determining number of retry attempts when calling fwd home.
exten => s,n,Set(RETRIES-WEATHER-SERVICE=0) ; used for determing number of retry attempts when checking weather service.
exten => s,n,Set(RETRIES-VOICEMAIL=0) ; used for determing number of retry attempts when checking voicemail.
exten => s,n,Set(TIMEOUT(response)=20)
exten => s,n,Set(TIMEOUT(digit)=7)
```

```
exten => s/1000,n,Background(dsfr-personalized-greeting-test1)
exten => s/0123456789,s,Background(dsfr-personalized-greeting-test2)
exten => s/123456789,s,Background(dsfr-personalized-greeting-test2)
exten => s/0987654321,s,Background(dsfr-personalized-greeting-test3)
exten => s/987654321,s,Background(dsfr-personalized-greeting-test3)
exten => s,s,NoOp(This caller does not have a personalized greeting.)
exten => s,n(ivr-options),Background(dsfr-from-iax-fwd-ivr) ; provide various ivr options excluding voicemail and admin-auth extensions.
exten => s,n,WaitExten
exten => 1,1,Voicemail(${PERSONAL-1-VMBOX}|u)
exten => 2,1,Voicemail(${PERSONAL-2-VMBOX}|u)
exten => 3,1,Playback(dsfr-ivr-fwd-info) ; explain that fwd work phone is experimental, will ring for 90 seconds, then come back to ivr.
exten => 3,n,SetCallerId(${FWD-HOME-CID})
exten => 3,n,Monitor(wav49|${TIMESTAMP}-${CALLERIDNUM}-FWD-WORK|mb)
exten => 3,n,Dial(IAX2/to-iax-fwd/${FWD-WORK-NUMBER}|${OUTGOING-RING-SECONDS}|rg)
exten => 3,n,Macro(from-iax-fwd-ivr-retries|RETRIES-FWD-WORK)
exten => 4,1,Playback(dsfr-weather-service) ; explain that there will be a pause while the Huntsville, AL weather information is downloaded.
exten => 4,n,System(/usr/bin/curl -s ftp://weather.noaa.gov/data/forecasts/city/al/huntsville.txt > /var/lib/asterisk/tmp/weather.txt)
exten => 4,n,System(/usr/bin/text2wave /var/lib/asterisk/tmp/weather.txt -F 8000 -o /var/lib/asterisk/tmp/weather.wav)
exten => 4,n,Playback(/var/lib/asterisk/tmp/weather)
exten => 4,n,Macro(from-iax-fwd-ivr-retries|RETRIES-WEATHER-SERVICE)
exten => 9,1,Playback(dsfr-emergency-phone-info) ; explain that emergency phone will ring for 90 seconds then come back to ivr.
exten => 9,n,Monitor(wav49|${TIMESTAMP}-${CALLERIDNUM}-EMERGENCY|mb)
exten => 9,n,Dial(${PHONE-1}${EMERGENCY-RING-TONE}|${EMERGENCY-RING-SECONDS}|r)
exten => 9,n,Goto(s|ivr-options)
exten => 8500,1,VoicemailMain
exten => 8500,n,Macro(from-iax-fwd-ivr-retries|RETRIES-VOICEMAIL)
exten => i,1,GotoIf($[${INVALID-IVR-TRIES} = ${MAX-IVR-INVALID-TRIES}] ?from-iax-fwd|i|1)
exten => i,n,Playback(invalid)
exten => i,n,Set(INVALID-IVR-TRIES=$[${INVALID-IVR-TRIES} + 1])
exten => i,n,Goto(s|ivr-options)
exten => t,1,Goto(from-iax-fwd|t|1)
exten => h,1,Goto(from-iax-fwd|h|1)

[macro-from-iax-fwd-ivr-retries]
exten => s,1,GotoIf($[${${ARG1}} = ${MAX-IVR-RETRIES}] ?from-iax-fwd|i|1)
exten => s,n,Set(${ARG1}=$[${${ARG1}} + 1])
exten => s,n,Goto(from-iax-fwd-ivr|s|ivr-options)
```

If you are using kapejod's bristuff, you will have a n + 201 priority as well. Dial goes to this priority if no one is logged in on the called extension. I.e. giving an unavailable message instead of a busy message when no phone is connected to the extension.

Example:

```
exten => 4000,1,Dial(SIP/${EXTEN},15)
exten => 4000,2,Voicemail(u${EXTEN})
exten => 4000,102,Voicemail(b${EXTEN})
exten => 4000,202,Voicemail(u${EXTEN})
```

Return values in channel variables

- For PRI connections over a ZAP channel, as well as IAX2 channels, the hangup result code will be found the HANGUPCAUSE variable.
- Note that causecode 0 is effectively a notice that the causecode has not been set. Causecode 16 is not an error, but simply a notice that the call went through and was terminated normally.
- For all channels, the DIALSTATUS variable contains the result of the call, which may be used for dial plan logic
- The variables DIALEDTIME and ANSWEREDTIME contains timing for billing (as in CDR records)

Examples

```
exten => 1265,1,Dial(Phone/phone0,15)
exten => s,3,Dial(SIP/oej,20)
exten => _908.,1,Dial(Modem/ttyI0:${EXTEN:1})
exten => 233,1,Dial(SIP/4029&SIP/4027&Zap/4&IAX/jaz,15,tTr)
exten => 500,1,Dial(Zap/2r2,20,crh)
exten => 20,1,Dial(Zap/3/5551234)
```

Version comments

- Option **A** and **S**: added to CVS after release 0.7.2
- Option **D**: in CVS since May 2004
- Option **f**: added to CVS in June/July 2004
- Variables DIALSTATUS, ANSWEREDTIME and DIALEDTIME: added to CVS head in june 2004

See also

- Set variable ALERT_INFO to change ring cadence on Cisco 79xx phones. See also MySQL custom ringtones
- Phones running the SCCP (skinny) firmware have some support for pushing XML pages. If you want to test it, set the variable VXML_URL to point to a Cisco XML file on a web server.
- For distinctive ringing on Cisco Ata, see Asterisk phone cisco ATA18x
- For a Least Cost (and Failsave) Dial Command see Application LCDial

Asterisk | Configuration | The Dialplan - extensions.conf | Dialplan Commands

Created by: oej last modification: Thursday 25 of August, 2005 [15:39:22 UTC] by dsfr