



ACD and DIALER OVERVIEW



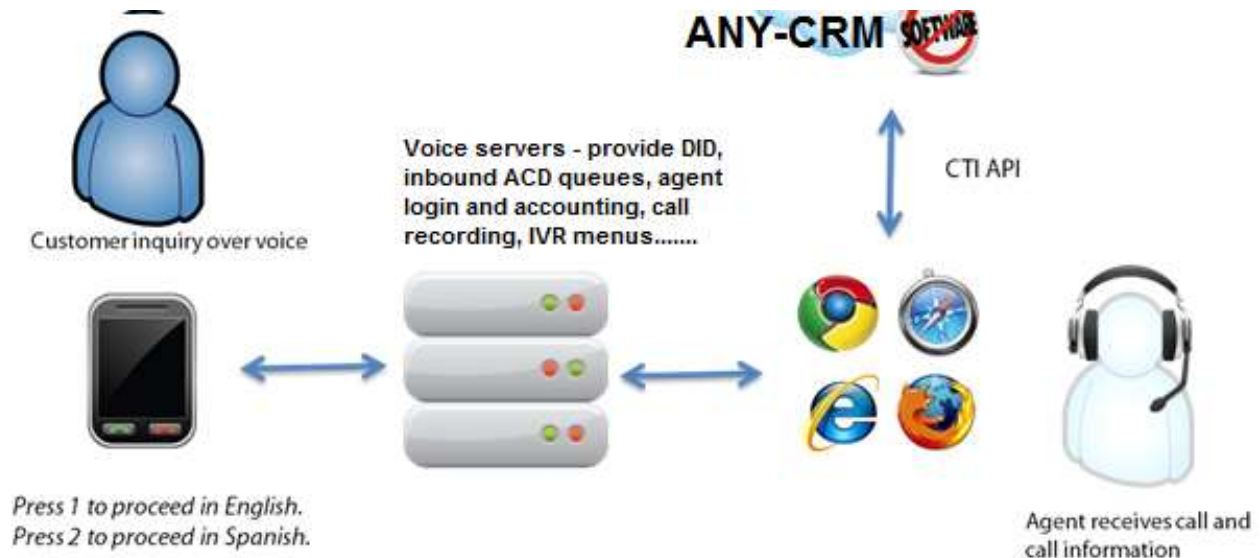
INBOUND CALLS

ACD = Automated Call Distribution = call queuing

ACDs route incoming calls to an agent depending upon the queue strategy {least active agent, next agent, best qualified agent...} using static configuration and data derived from the caller {caller ID, IVR input from caller, speech input from caller... etc.}.

The queue strategies can be set up within an algorithm for each agent group and will depend upon the requirements for that group. The groups can consist of agents within a call center and/or remote and virtual call center agents.

Computer Telephony Integration is often used to allow the inbound call to be routed to the appropriate agent directly or via ACD queue and have the agent screen populated by the corresponding data for that call. The agent screen pop is often part of the call center CRM. A few agent groups are still using manual data screens rather than have the full integration - this is less efficient and often wastes time on calls while the agent has to ask and enter client details for the call.....



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There are multiple choices for distributing incoming calls from a queue

- Linear Call Distribution – Calls are distributed in order, starting at the beginning each time
- Circular/Rotary Call Distribution – Calls are distributed in order, starting with the next in order
- Uniform Call Distribution – Calls are distributed uniformly, starting with the person who has handled the fewest calls
- Simultaneous Call Distribution – Calls are presented to all available extensions simultaneously
- Weighted Call Distribution – Calls are distributed according to a configurable weighting, such as differing skill sets within customer service representatives

Agents login to become queue members so that the ACDs can route inbound calls to them when available. Agents can set their status to pause so that the ACD will not send inbound calls to them {if they are on a break, lunch, meeting, training etc..}. This method of agent time keeping can also be used for agents that are NOT configured to receive calls from the inbound queues, but provides a method of time reporting and can be configured so that the agent has to login in order to make outbound calls. Agents that are not members of an ACD queue will often have directly dialled inbound extensions or inbound via AA {automatic attendant} IVR which allows the caller to enter the agents extension number.

Wallboards and or display screens are usually essential to show which agents are logged-in and which agents are available for calls from the ACD queues. Obviously, if agents fail to logout their time keeping data will be incorrect and the statistics used meaningless.

Agents can be members of several queues at the same time and can be configured with priority and skill levels to optimise the inbound calls to the agent.

{ draw and discussed a typical call flow and example ACD and time reporting agent groups }

The ACD queue has many configurable variables ;

In order to configure an ACD queue, the business/call center operation manager has to provide the details and any SLAs required, such as percentage of calls expected to be answered within 30 seconds, expected call duration, caller in-queue announcements {welcome messages, 'you are caller number #..' , 'please stay on the line..' , advertisements and announcements etc...}, how long before call is routed to voicemail, how long an agent should be given for 'wrap-up time' after each call, how many calls should the agent be allow to reject before notification, what route calls should take during holidays and out of hours for the business.....

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Typical Asterisk queue configuration example ;

```
[general]
;
; Global settings for call queues
;
; Persistent Members
;   Store each dynamic member in each queue in the astdb and/or MySQL so that
;   when asterisk is restarted, each member will be automatically
;   read into their recorded queues. Default is 'no'.
;
persistentmembers = yes
;
; AutoFill Behavior
;   The old behavior of the queue (autofill=no) is to have a serial type behavior
;   in that the queue will make all waiting callers wait in the queue
;   even if there is more than one available member ready to take
;   calls until the head caller is connected with the member they
;   were trying to get to. The next waiting caller in line then
;   becomes the head caller, and they are then connected with the
;   next available member and all available members and waiting callers
;   waits while this happens. The new behavior, enabled by setting
;   autofill=yes makes sure that when the waiting callers are connecting
;   with available members in a parallel fashion until there are
;   no more available members or no more waiting callers. This is
;   probably more along the lines of how a queue should work and
;   in most cases, you will want to enable this behavior. If you
;   do not specify or comment out this option, it will default to yes.
;
;autofill = no
;
; Monitor Type
;   By setting monitor-type = MixMonitor, when specifying monitor-format
;   to enable recording of queue member conversations, app_queue will
;   now use the new MixMonitor application instead of Monitor so
;   the concept of "joining/mixing" the in/out files now goes away
;   when this is enabled. You can set the default type for all queues
;   here, and then also change monitor-type for individual queues within
;   queue by using the same configuration parameter within a queue
;   configuration block. If you do not specify or comment out this option,
;   it will default to the old 'Monitor' behavior to keep backward
;   compatibility.
;
monitor-type = MixMonitor
;
; UpdateCDR behavior.
;   This option is implemented to mimic chan_agents behavior of populating
;   CDR dstchannel field of a call with an agent name, which you can set
;   at the login time with AddQueueMember membername parameter.
;
; updatecdr = no
;
; Note that a timeout to fail out of a queue may be passed as part of
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```



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```
; an application call from extensions.conf:
; Queue(queueename,[options],[optionalurl],[announceoverride],[timeout])
; example: Queue(dave,t,,,45)

; shared_lastcall will make the lastcall and calls received be the same in
; members logged in more than one queue. This is useful to make the queue
; respect the wrapuptime of another queue for a shared member.
; The default value is no.
;
;shared_lastcall=no
;
; Negative_penalty_invalid will treat members with a negative penalty as logged off
;
;negative_penalty_invalid = no
;
; log_membername_as_agent will cause app_queue to log the membername rather than
; the interface for the ADDMEMBER and REMOVEMEMBER events when a state_interface
; is set. The default value (no) maintains backward compatibility.
;
;log_membername_as_agent = no
;
;[markq]
;
; A sample call queue
;
; Musicclass sets which music applies for this particular call queue.
; The only class which can override this one is if the MOH class is set
; directly on the channel using Set(CHANNEL(musicclass)=whatever) in the
; dialplan.
;
;musicclass = default
;
; An announcement may be specified which is played for the member as
; soon as they answer a call, typically to indicate to them which queue
; this call should be answered as, so that agents or members who are
; listening to more than one queue can differentiated how they should
; engage the customer
;
;announce = queue-markq
;
; A strategy may be specified. Valid strategies include:
;
; ringall - ring all available channels until one answers (default)
; leastrcent - ring interface which was least recently hung up by this queue
; fewestcalls - ring the one with fewest completed calls from this queue
; random - ring random interface
; rrmemory - round robin with memory, remember where we left off last ring pass
; rrorordered - same as rrmemory, except the queue member order from config file
;               is preserved
; linear - rings interfaces in the order specified in this configuration file.
;         If you use dynamic members, the members will be rung in the order in
;         which they were added
; wrandom - rings random interface, but uses the member's penalty as a weight
;         when calculating their metric. So a member with penalty 0 will have
;         a metric somewhere between 0 and 1000, and a member with penalty 1 will
```

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```
; have a metric between 0 and 2000, and a member with penalty 2 will
have
; a metric between 0 and 3000. Please note, if using this strategy, the member
; penalty is not the same as when using other queue strategies. It is ONLY used
; as a weight for calculating metric.
;
;strategy = ringall
;
; Second settings for service level (default 0)
; Used for service level statistics (calls answered within service level time frame)
; There are 2 Service Level Performance Calculations SL and SL2
; SL = (total calls answered within threshold / total calls answered) * 100
; SL2 = ((calls answered within threshold + calls abandoned within threshold) / (answered
calls + abandoned calls)) * 100
;servicelevel = 60
;
; A context may be specified, in which if the user types a SINGLE
; digit extension while they are in the queue, they will be taken out
; of the queue and sent to that extension in this context.
;
;context = qoutcon
;
; A limit can be set to disregard penalty settings when the queue has
; too few members. No penalty will be weighed in if there are only X
; or fewer queue members. (default 0)
;
;penaltymemberslimit = 5
;
; -----QUEUE TIMING OPTIONS-----
; A Queue has two different "timeout" values associated with it. One is the
; timeout parameter configured in queues.conf. This timeout specifies the
; amount of time to try ringing a member's phone before considering the
; member to be unavailable. The other timeout value is the second argument
; to the Queue() application. This timeout represents the absolute amount
; of time to allow a caller to stay in the queue before the caller is
; removed from the queue. In certain situations, these two timeout values
; may clash. For instance, if the timeout in queues.conf is set to 5 seconds,
; the retry value in queues.conf is set to 4, and the second argument to Queue()
; is 10, then the following may occur:
;
; A caller places a call to a queue.
; The queue selects a member and attempts to ring that member.
; The member's phone is rung for 5 seconds and he does not answer.
; The retry time of 4 seconds occurs.
; The queue selects a second member to call.
;
; How long does that second member's phone ring? Does it ring for 5 seconds
; since the timeout set in app_queue is 5 seconds? Does it ring for 1 second since
; the caller has been in the queue for 9 seconds and is supposed to be removed after
; being in the queue for 10 seconds? This is configurable with the timeoutpriority
; option. By setting the timeoutpriority to "conf" then you are saying that you would
; rather use the time specified in the configuration file even if it means having the
; caller stay in the queue longer than the time specified in the application argument.
; For the scenario described above, timeoutpriority=conf would result in the second
; member's phone ringing for 5 seconds. By specifying "app" as the value for
; timeoutpriority, you are saying that the timeout specified as the argument to the
```

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```
; Queue application is more important. In the scenario above, timeoutpriority=app
; would result in the second member's phone ringing for 1 second.
;
; There are a few exceptions to the priority rules. For instance, if timeoutpriority=app
; and the configuration file timeout is set to 0, but the application argument timeout is
; non-zero, then the timeoutpriority is ignored and the application argument is used as
; the timeout. Furthermore, if no application argument timeout is specified, then the
; timeoutpriority option is ignored and the configuration file timeout is always used
; when calling queue members.
;
; In timeoutpriority=conf mode however timeout specified in config file will take higher
; priority than timeout in application arguments, so if config file has timeout 0, each
; queue member will be called indefinitely and application timeout will be checked only
; after this call attempt. This is useful for having queue members with custom timeouts
; specified within Dial application of Local channel, and allows handling NO ANSWER which
; would otherwise be interrupted by queue destroying child channel on timeout.
;
; The default value for timeoutpriority is "app" since this was how previous versions of
; Asterisk behaved.
;
;timeout = 15
;retry = 5
;timeoutpriority = app|conf
;
; -----END QUEUE TIMING OPTIONS-----
; Weight of queue - when compared to other queues, higher weights get
; first shot at available channels when the same channel is included in
; more than one queue.
;
;weight=0
;
; After a successful call, how long to wait before sending a potentially
; free member another call (default is 0, or no delay)
;
;wrapuptime=15
;
; Autofill will follow queue strategy but push multiple calls through
; at same time until there are no more waiting callers or no more
; available members. The per-queue setting of autofill allows you
; to override the default setting on an individual queue level.
;
;autofill=yes
;
; Autopause will pause a queue member if they fail to answer a call
; no: Member will not be paused
; yes: Member will be paused only in the queue where the timeout took place
; all: Member will be paused in all queues he/she is a member
;autopause=yes
;
; Autopausedelay delay autopause for autopausdelay seconds from the
; last call if a member has not taken a call the delay has no effect.
;autopausedelay=60
;
; Autopausebusy controls whether or not a queue member is set as paused
; automatically upon the member device reporting busy. The autopausdelay
; option applies. Defaults to 'no'.
```

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```
;autopausebusy=no
;
; Autopauseunavail controls whether or not a queue member is set as paused
; automatically upon the member device reporting congestion. The autopauseonly
; option applies. Defaults to 'no'.
;autopauseunavail=no
;
; Maximum number of people waiting in the queue (0 for unlimited)
;
;maxlen = 0
;
; Note: for below queue channel options (setinterfacevar, setqueueentryvar,
; setqueuevar) if the caller channel is a local channel and optimizations
; is enabled then after optimization has occurred only the queue member
; channel will contain the variables.
;
; If set to yes, just prior to the caller being bridged with a queue member
; the following variables will be set on the caller and queue member channels:
; MEMBERINTERFACE is the interface name (eg. Agent/1234)
; MEMBERNAME is the member name (eg. Joe Soap)
; MEMBERCALLS is the number of calls that interface has taken,
; MEMBERLASTCALL is the last time the member took a call.
; MEMBERPENALTY is the penalty of the member
; MEMBERDYNAMIC indicates if a member is dynamic or not
; MEMBERREALTIME indicates if a member is realtime or not
;
;setinterfacevar=no
;
; If set to yes, just prior to the caller being bridged with a queue member
; the following variables will be set on the caller and queue member channels:
; QEHOLDTIME callers hold time
; QEORIGINALPOS original position of the caller in the queue
;
;setqueueentryvar=no
;
; If set to yes, the following variables will be set
; just prior to the caller being bridged with a queue member (set on the
; caller and queue member channels) and just prior to the caller
; leaving the queue
; QUEUENAME name of the queue
; QUEUEMAX maximum number of calls allowed
; QUEUESTRATEGY the strategy of the queue;
; QUEUECALLS number of calls currently in the queue
; QUEUEHOLDTIME current average hold time
; QUEUECOMPLETED number of completed calls for the queue
; QUEUEABANDONED number of abandoned calls
; QUEUESRVLEVEL queue service level
; QUEUESRVLEVELPERF current service level performance
;
;setqueuevar=no

; if set, run this macro when connected to the queue member
; you can override this macro by setting the macro option on
; the queue application
;
;membermacro=macro_name[,arg1[,...][,argN]]
```

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```
; if set, run this gosub when connected to the queue member
; you can override this gosub by setting the gosub option on
; the queue application
;
;membergosub=gosub_context_name[,arg1[,...][,argN]]

; How often to announce queue position and/or estimated
; holdtime to caller (0=off)
; Note that this value is ignored if the caller's queue
; position has changed (see min-announce-frequency)
;
;announce-frequency = 90
;
; The absolute minimum time between the start of each
; queue position and/or estimated holdtime announcement
; This is useful for avoiding constant announcements
; when the caller's queue position is changing frequently
; (see announce-frequency)
;
;min-announce-frequency = 15
;
; How often to make any periodic announcement (see periodic-announce)
;
;periodic-announce-frequency=60
;
; Should the periodic announcements be played in a random order? Default is no.
;
;random-periodic-announce=no
;
; If set to yes, the periodic announcement frequency will be timed from the end
; of each announcement rather than from the start of each announcement. This
; defaults to off.
;
;relative-periodic-announce=yes
;
; Should we include estimated hold time in position announcements?
; Either yes, no, or only once.
; Hold time will be announced as the estimated time.
;
;announce-holdtime = yes|no|once
;
; Queue position announce?
; Valid values are "yes," "no," "limit," or "more." If set to "no," then the caller's
; position will
; never be announced. If "yes," then the caller's position in the queue will be announced
; to the caller. If set to "more," then if the number of callers is more than the number
; specified by the announce-position-limit option, then the caller will hear that there
; are more than that many callers waiting (i.e. if a caller number 6 is in a queue with
; the
; announce-position-limit set to 5, then that caller will hear that there are more than 5
; callers waiting). If set to "limit," then only callers within the limit specified by
; announce-position-limit
; will have their position announced.
;
;announce-position = yes
```

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```
;
; If enabled, play announcements to the first user waiting in the Queue. This may mean
; that announcements are played when an agent attempts to connect to the waiting user,
; which may delay the time before the agent and the user can communicate. Disabled by
; default.
;
; announce-to-first-user = no
;
; If you have specified "limit" or "more" for the announce-position option, then the
following
; value is what is used to determine what announcement to play to waiting callers. If you
have
; set the announce-position option to anything else, then this will have no bearing on
queue operation
;
; announce-position-limit = 5
;
; What's the rounding time for the seconds?
; If this is non-zero, then we announce the seconds as well as the minutes
; rounded to this value.
; Valid values are 0, 5, 10, 15, 20, and 30.
;
; announce-round-seconds = 10
;
; Only announce the caller's position if it has improved since the last announcement.
; The default value is no.
;
; announce-position-only-up = yes
;
; Use these sound files in making position/holdtime announcements. The
; defaults are as listed below -- change only if you need to.
;
; Keep in mind that you may also prevent a sound from being played if you
; explicitly set a sound to be an empty string. For example, if you want to
; prevent the queue from playing queue-thankyou, you may set the sound using
; the following line:
;
; queue-thankyou=
;
;           ;           ("You are now first in line.")
;queue-youarenext = queue-youarenext
;           ;           ("There are")
;queue-thereare= queue-thereare
;           ;           ("calls waiting.")
;queue-callswaiting = queue-callswaiting
;           ;           ("The current est. holdtime is")
;queue-holdtime = queue-holdtime
;           ;           ("minute.")
;queue-minute = queue-minute
;           ;           ("minutes.")
;queue-minutes = queue-minutes
;           ;           ("seconds.")
;queue-seconds = queue-seconds
;           ;           ("Thank you for your patience.")
;queue-thankyou = queue-thankyou
;           ;           ("Hold time")
```

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```
;queue-reporthold = queue-reporthold
;           ;           ("All reps busy / wait for next")
;periodic-announce = queue-periodic-announce
;
; A set of periodic announcements can be defined by separating
; periodic announcements to reproduce by commas. For example:
;periodic-announce = queue-periodic-announce,your-call-is-important,please-wait
;
; The announcements will be played in the order in which they are defined. After
; playing the last announcement, the announcements begin again from the beginning.
;
; Calls may be recorded using Asterisk's monitor/MixMonitor resource
; This can be enabled from within the Queue application, starting recording
; when the call is actually picked up; thus, only successful calls are
; recorded, and you are not recording while people are listening to MOH.
; To enable monitoring, simply specify "monitor-format"; it will be disabled
; otherwise.
;
; You can specify the monitor filename with by calling
;   Set(MONITOR_FILENAME=foo)
; Otherwise it will use MONITOR_FILENAME=${UNIQUEID}
;
; Pick any one valid extension for monitor format recording. If you leave
; monitor-format commented out, it will not record calls.
;
; monitor-format = gsm|wav|wav49
;
; Monitor Type
;   By setting monitor-type = MixMonitor, when specifying monitor-format
;   to enable recording of queue member conversations, app_queue will
;   now use the new MixMonitor application instead of Monitor so
;   the concept of "joining/mixing" the in/out files now goes away
;   when this is enabled. If you do not specify or comment out this option,
;   it will default to the old 'Monitor' behavior to keep backward
;   compatibility.
;
; ----- Queue Empty Options -----
;
; Asterisk has provided the "joinempty" and "leavewhenempty" options for a while
; with tenuous definitions of what they actually mean. The "joinempty" option controls
; whether a caller may join a queuedepending on several factors of member availability.
; Similarly, then leavewhenempty option controls whether a caller may remain in a queue
; he has already joined. Both options take a comma-separated list of factors which
; contribute towards whether a caller may join/remain in the queue. The list of
; factors which contribute to these option is as follows:
;
; paused: a member is not considered available if he is paused
; penalty: a member is not considered available if his penalty is less than
QUEUE_MAX_PENALTY
; inuse: a member is not considered available if he is currently on a call
; ringing: a member is not considered available if his phone is currently ringing
; unavailable: This applies mainly to Agent channels. If the agent is a member of the
queue
;           but has not logged in, then do not consider the member to be available
; invalid: Do not consider a member to be available if he has an "invalid" device state.
```

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```
; This generally is caused by an error condition in the member's channel driver.
; unknown: Do not consider a member to be available if we are unable to determine the
member's
; current device state.
; wrapup: A member is not considered available if he is currently in his wrapuptime after
; taking a call.
;
; For the "joinempty" option, when a caller attempts to enter a queue, the members of
that
; queue are examined. If all members are deemed to be unavailable due to any of the
conditions
; listed for the "joinempty" option, then the caller will be unable to enter the queue.
For the
; "leavewhenempty" option, the state of the members of the queue are checked periodically
during
; the caller's stay in the queue. If all of the members are unavailable due to any of the
above
; conditions, then the caller will be removed from the queue.
;
; Some examples:
;
;joinempty = paused,inuse,invalid
;
; A caller will not be able to enter a queue if at least one member cannot be found
; who is not paused, on the phone, or who has an invalid device state.
;
;leavewhenempty = inuse,ringing
;
; A caller will be removed from the queue if at least one member cannot be found
; who is not on the phone, or whose phone is not ringing.
;
; For the sake of backwards-compatibility, the joinempty and leavewhenempty
; options also accept the strings "yes" "no" "strict" and "loose". The following
; serves as a translation for these values:
;
; yes - (empty) for joinempty; penalty,paused,invalid for leavewhenempty
; no - penalty,paused,invalid for joinempty; (empty) for leavewhenempty
; strict - penalty,paused,invalid,unavailable
; loose - penalty,invalid
;
; If you wish to report the caller's hold time to the member before they are
; connected to the caller, set this to yes.
;
; reportholdtime = no
;
; If you want the queue to avoid sending calls to members whose devices are
; known to be 'in use' (via the channel driver supporting that device state)
; uncomment this option. This can be controlled on a per member basis by
; setting 'ringinuse' on that member. This can be done in the member definition,
; in the 'ringinuse' field on a realtime member, via the QUEUE_MEMBER dialplan
; function, or with CLI/AMI. By default, the per member value will be the same
; as the queue's ringinuse value if it isn't set on the member deliberately.
; (Note: only the SIP channel driver currently is able to report 'in use'.)
; ringinuse = no
;
```

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```
; If you wish to have a delay before the member is connected to the caller (or
; before the member hears any announcement messages), set this to the number of
; seconds to delay.
;
; memberdelay = 0
;
; If timeoutrestart is set to yes, then the timeout for an agent to answer is
; reset if a BUSY or CONGESTION is received. This can be useful if agents
; are able to cancel a call with reject or similar.
;
; timeoutrestart = no
;
; If you wish to implement a rule defined in queuerules.conf (see
; configs/queuerules.conf.sample from the asterisk source directory for
; more information about penalty rules) by default, you may specify this
; by setting defaultrule to the rule's name
;
; defaultrule = myrule
;
; Each member of this call queue is listed on a separate line in
; the form technology/dialstring. "member" means a normal member of a
; queue. An optional penalty may be specified after a comma, such that
; entries with higher penalties are considered last. An optional member
; name may also be specified after a second comma, which is used in log
; messages as a "friendly name". Multiple interfaces may share a single
; member name. An optional state interface may be specified after a third
; comma. This interface will be the one for which app_queue receives device
; state notifications, even though the first interface specified is the one
; that is actually called.
;
; A hint can also be used in place of the state interface using the format
; hint:<extension>@<context>. If no context is specified then 'default' will
; be used.
;
; It is important to ensure that channel drivers used for members are loaded
; before app_queue.so itself or they may be marked invalid until reload. This
; can be accomplished by explicitly listing them in modules.conf before
; app_queue.so. Additionally, if you use Local channels as queue members, you
; must also preload pbx_config.so and chan_local.so (or pbx_ael.so, pbx_lua.so,
; or pbx_realtime.so, depending on how your dialplan is configured).
;
;                                     syntax:                               member
interface, [,penalty][,membername][,state_interface][,ringinuse][,wrapuptime] =>
;
;member => DAHDI/1
;member => DAHDI/2,10
;member => DAHDI/3,10,Bob Johnson
;member => Local/1001@agents,0,May Flowers,Agent:1001
```

{ draw & discuss the agent login and status configuration }

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ACD and DIALER OVERVIEW

OUTBOUND CALLS

Configuration will control the dialing rights for each user. Discuss the dialing rights context groups and how they work to include less priority groups.

- Understand what a dialplan is
 - extensions.conf syntax
 - Contexts, Extensions, Priorities
 - 'n' priority
 - Priority labels
 - "same" designator
 - build some basic extensions
-
- The Asterisk dialplan is **unlike** the dialplan in traditional PBX systems
 - In a **traditional PBX**, the dialplan is a **simple table** that maps extension numbers to phones
 - The **Asterisk dialplan** is much more like a small **scripting language**
 - This is what makes Asterisk so **powerful** and **flexible**
 - The dialplan is the **heart** of any Asterisk system
 - It provides the **logic** of the system
 - It consists of a **list of instructions**

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ACD and DIALER OVERVIEW

ALL configuration files have the filename extension = conf and take the form ;

```
[context]

{something} => name, priority, application|options

{something} = {something}

; remark

{something} => Gotof({something} = {something})

{something} => Answer

Include => {another-context}
```

Dialplan configuration files can include logic programming steps and routine that will be familiar to anyone with any C, PHP, perl or shell scripting experience. Configuration files can also include other configuration files and contexts can include other contexts much like scripting a program.

Each configuration file is loaded into memory so changes can be made to an individual function or protocol without disruption and loaded without effecting any other function or protocol.

Compare an HTML 'hello world' with an Asterisk dialplan 'hello world' ;

```
Example /var/www/html/helloworld.html
<html>
  <head>
    <title>Hello World Demo</title>
  </head>
  <body>
    <h1>Hello World!</h1>
  </body>
</html>
! print Hello World on the browser screen
```

```
Example /etc/asterisk/extensions.conf
[some-context]
exten => 100,1,Answer()
exten => 100,n,Wait(1)
exten => 100,n,Playback(hello-world) ! play 'Hello World' to the caller to #100
exten => 100,n,Hangup()
```

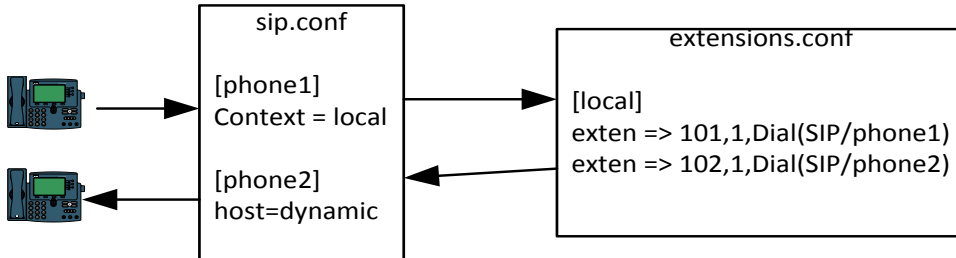
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ACD and DIALER OVERVIEW

Simple 2 x SIP telephone example showing relationship between the channel module (in this case SIP) and the dialplan (extensions.conf) ;



sip.conf configuration

```
[device-name]
type=friend
host=dynamic
context=local
secret=MyT4u
```

sip.conf example above defines the SIP device as 'device-name'

where type = friend {matches inbound calls on username or IP address}

user {matches inbound calls on username only}

peer {matches inbound calls on IP address only}

host = xxx.xxx.xxx.xxx {defined ipv4 or ipv6 address}

hostname

dynamic {endpoint needs to register with Asterisk}

context = {inbound calls are sent to this context in the dialplan first}

secret = password to match from the endpoint

{insert example sip.conf here from local system & discuss}

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ACD and DIALER OVERVIEW

Automated outbound calls = auto-dialers, robo-dialers and predictive dialers

Auto, preview or 'robo-dialers' automatically dial outbound calls from a list of contact numbers. When a call is answered the called party is either connected to an announcement or IVR system, direct to an agent or to an agent queue. Calls are dialed based upon how many agents are available and/or how many channels available outbound.

Predictive dialing obviously requires much more computer power and configuration to that of auto-dialing because the predictive dialler has to consider all of the delays and statistics to make the outbound call when it is predicted to be an agent available.

When dialing outbound, there are several sources of delay;

Outbound call dialing and connection - average is 11 seconds from dial to answer

Not all of outbound calls are answered - for example, if 1 out of 3 dials are answered, a predictive dialer might dial 3 lines every time an agent becomes available.

A predictive dialer may be configured to start dialing 80 seconds before an agent 'should' be available and continue to modify the timing as the calls are made and statistics gathered for a particular campaign

Dialing one number at a time, only when an agent is available, typically keeps agents utilized for 40 minutes per hour (33% idle time). Predictive dialing can increase utilization to 57 minutes per hour (5% idle time).

Predictive dialers may be standalone hardware devices, cloud-based, or they may be integrated in software with call center or contact center platforms or Asterisk clusters. A cloud-based predictive dialer provides the functionalities of a traditional predictive dialer without requiring installation on the local device and reduces the dependency on the hardware.

Discuss how Asterisk uses 'outgoing spool' + how AGI and PHP can place outbound calls

VERY IMPORTANT - REGULATIONS ARE NOW VERY STRICT

For example, in the UK, Ofcom requires that predictive dialers abandon **fewer than 3% of answered calls** on a daily basis. Ofcom also requires that if an agent is **not available within 2 seconds** the call is considered "abandoned" and an automated message is played. The automated message must identify the company making the call, the purpose of the call, a free phone or basic rate phone number to call back on and must not contain any form of marketing. A phone call to the return number must not be treated by the company as an opportunity to market, but to be removed from the calling list. In the UK "abandoned" **calls must not be called back within 72 hours** unless there is a dedicated agent available.

In the United States, if someone answers but **no agent is available within 2 seconds** of the person's greeting, Federal Communications Commission (FCC) regulations consider the call "abandoned" and require the dialer to play a recorded message. The FCC requires that predictive **dialers abandon less than 3% of answered calls.**

In 1991 the Telephone Consumer Protection Act prohibited the use of an "automatic telephone dialing system" to contact "any telephone number assigned to a mobile telephone service" without "express prior consent" from the party being called. Dialing "accidents" are not an effective legal defense and **penalties run from \$500 to \$1,500 per violating call.**

In Canada, the **maximum abandon rate is 5%**, and calls cannot be made to numbers registered with the National Do Not Call Registry, emergency or health care providers.^[2] However, not-for-profit organizations such as survey companies are exempt from the National Do Not Call Registry.

So it should be very clear - it is essential to get the outbound dialer configuration correct or risk very large fines

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ACD and DIALER OVERVIEW

Discuss Predictive Dialer Algorithm considerations

- Average rings before called party answers
- % of calls answered
- Talk time average
- Number of available agents in time period
- Desired abandoned rate
- Agent wrap-up time

Asterisk Dialer Screenshots ;

Create New Campaign

Campaign Name
Acme-survey-group-1

Import CSV File
Choose file campaign1-test.csv

Greeting Audio
Choose file No file chosen

Message Audio - advert or announcement
Choose file No file chosen

The basic dialer configuration will take a list of contacts from a CSV file and any required custom audio announcements.

The campaign can then be scheduled or started immediately

In this example we have a campaign with at least 14 available agents, for a survey that has an average talk time of 50 seconds. The only announcements to the called party are if there are no agents available to take the answered call immediately – informing the called party that the agent will be on the line within a few seconds..... Agent wrap-up time is only 5 seconds and the quantity of calls in the CSV are chosen to match the expected work load for a single agent shift and assuming there are only 2 agents on a break at any one time.

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ACD and DIALER OVERVIEW

In this example the dialer will start the campaign outbound dialing immediately, send the answered calls to the queue with 12 calls simultaneously within the defined time period (1 minute default).... If calls fail they will be retried twice after 5 minutes then not called again

- Deliver Message - Announcement Only
- Dial Only - Direct to Queue
- Dial Only - Direct to Agent

Maximum Concurrent Calls

Maximum Retries **Retry Time**

At any time during or after the campaign, the call data can be examined and custom reports generated which combine the CDRs with the agent status and dialer stats. Alarms can be set to highlight SLAs on wallboards and to supervisors via SMS and email etc...

Campaign Data										Download Data	
ID	Name	LastName	Telephone	Tries	CallStatus	Queue	Account	SIP Cause	Deliver	Edit	Delete
1	kcc	another test	16465701658	1	answered	sales	kccvoip		no	Edit	Delete
2	keith	kcctest	011441207668118	1	answered	sales	kccvoip		10/20/2018	Edit	Delete

More complex configuration is required for predictive dialing as the stats from agent call rates, abandoned calls, average talk times, wrap-up times and announcement timings are used to continually change the call rates and optimize the agents time.

BLENDED QUEUES are ACD queues that are configured for both inbound and outbound calling.

{discuss the options and how it is configured}

It is also possible to have chat, SMS, email, video and voice calls {discuss requirements}

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