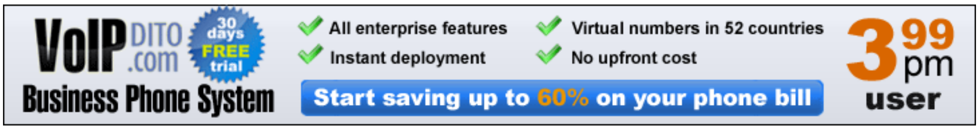

View Discussion (0) History

Asterisk config chan_dahdi.conf

Configuration file for Chan_DAHDi lookups in Asterisk

See also

Asterisk config files

```
!Sample chan_dahdi.conf
^
;
;
; DAHDI telephony
;
; Configuration file
;
; You need to restart Asterisk to re-configure the DAHDI channel
; CLI> reload chan_dahdi.so
;               will reload the configuration file,
;               but not all configuration options are
;               re-configured during a reload (signalling, as well as
;               PRI and SS7-related settings cannot be changed on a
;               reload.
;
; This file documents many configuration variables.  Normally unless you
; know what a variable means or that it should be changed, there's no
; reason to unrem lines.
;
; remmed-out examples below (those lines that begin with a ';' but no
; space afterwards) typically show a value that is not the defauult value,
; but would make sense under cetain circumstances. The default values
; are usually sane. Thus you should typically not touch them unless you
; know what they mean or you know you should change them.

~np~[trunkgroups]~/np~
;
; Trunk groups are used for NFAS or GR-303 connections.
;
; Group: Defines a trunk group.
;        trunkgroup => <trunkgroup>,<dchannel>[,<backup1>...]
;
;        trunkgroup  is the numerical trunk group to create
;        dchannel    is the DAHDI channel which will have the
;                    d-channel for the trunk.
;        backup1     is an optional list of backup d-channels.
;
;trunkgroup => 1,24,48
;trunkgroup => 1,24
;
; Spanmap: Associates a span with a trunk group
;        spanmap => <dahdispan>,<trunkgroup>[,<logicalspan>]
;
;        dahdispan     is the DAHDI span number to associate
;        trunkgroup  is the trunkgroup (specified above) for the mapping
;        logicalspan is the logical span number within the trunk group to use.
;                    if unspecified, no logical span number is used.
;
;spanmap => 1,1,1
;spanmap => 2,1,2
;spanmap => 3,1,3
;spanmap => 4,1,4

~np~[channels]~/np~
;
; Default language
;
;language=en
;
; Context for calls. Defaults to 'default'
;
;context=incoming
;
```

```
; Switchtype:  Only used for PRI.
;
; national:       National ISDN 2 (default)
; dms100:         Nortel DMS100
; 4ess:           AT&T 4ESS
; 5ess:           Lucent 5ESS
; euroisdn:       EuroISDN (common in Europe)
; ni1:            Old National ISDN 1
; qsig:           Q.SIG
;
;switchtype=euroisdn
;
; Some switches (AT&T especially) require network specific facility IE
; supported values are currently 'none', 'sdn', 'megacom', 'tollfreemegacom', 'accunet'
;
; nsf cannot be changed on a reload.
;
;nsf=none
;
; PRI Dialplan: The ISDN-level Type Of Number (TON) or numbering plan, used for
; the dialed number.  For most installations, leaving this as 'unknown' (the
; default) works in the most cases.  In some very unusual circumstances, you
; may need to set this to 'dynamic' or 'redundant'.  Note that if you set one
; of the others, you will be unable to dial another class of numbers.  For
; example, if you set 'national', you will be unable to dial local or
; international numbers.
;
; PRI Local Dialplan:  Only RARELY used for PRI (sets the calling number's
; numbering plan).  In North America, the typical use is sending the 10 digit
; callerID number and setting the prilocaldialplan to 'national' (the default).
; Only VERY rarely will you need to change this.
;
; Neither pridialplan nor prilocaldialplan can be changed on reload.
;
; unknown:        Unknown
; private:        Private ISDN
; local:          Local ISDN
; national:       National ISDN
; international:  International ISDN
; dynamic:        Dynamically selects the appropriate dialplan
; redundant:      Same as dynamic, except that the underlying number is not
;                 changed (not common)
;
;pridialplan=unknown
;prilocaldialplan=national
;
; pridialplan may be also set at dialtime, by prefixing the dialled number with
; one of the following letters:
; U - Unknown
; I - International
; N - National
; L - Local (Net Specific)
; S - Subscriber
; V - Abbreviated
; R - Reserved (should probably never be used but is included for completeness)
;
; Additionally, you may also set the following NPI bits (also by prefixing the
; dialled string with one of the following letters):
; u - Unknown
; e - E.163/E.164 (ISDN/telephony)
; x - X.121 (Data)
; f - F.69 (Telex)
; n - National
; p - Private
; r - Reserved (should probably never be used but is included for completeness)
;
; You may also set the prilocaldialplan in the same way, but by prefixing the
; Caller*ID Number, rather than the dialled number.  Please note that telcos
; which require this kind of additional manipulation of the TON/NPI are *rare*.
; Most telco PRIs will work fine simply by setting pridialplan to unknown or
; dynamic.
;
;
; PRI caller ID prefixes based on the given TON/NPI (dialplan)
; This is especially needed for EuroISDN E1-PRIs
;
; None of the prefix settings can be changed on reload.
;
; sample 1 for Germany
;internationalprefix = 00
;nationalprefix = 0
;localprefix = 0711
;privateprefix = 07115678
;unknownprefix =
;
; sample 2 for Germany
;internationalprefix = +
;nationalprefix = +49
```

```
;localprefix = +49711
;privateprefix = +497115678
;unknownprefix =
;
; PRI resetinterval: sets the time in seconds between restart of unused
; B channels; defaults to 'never'.
;
;resetinterval = 3600
;
; Overlap dialing mode (sending overlap digits)
; Cannot be changed on a reload.
;
;overlapdial=yes
;
; Allow inband audio (progress) when a call is RELEASEd by the far end of a PRI
;
;inbanddisconnect=yes
;
; PRI Out of band indications.
; Enable this to report Busy and Congestion on a PRI using out-of-band
; notification. Inband indication, as used by Asterisk doesn't seem to work
; with all telcos.
;
; outofband:      Signal Busy/Congestion out of band with RELEASE/DISCONNECT
; inband:         Signal Busy/Congestion using in-band tones (default)
;
; priindication cannot be changed on a reload.
;
;priindication = outofband
;
; If you need to override the existing channels selection routine and force all
; PRI channels to be marked as exclusively selected, set this to yes.
;
; priexclusive cannot be changed on a reload.
;
;priexclusive = yes
;
; ISDN Timers
; All of the ISDN timers and counters that are used are configurable.  Specify
; the timer name, and its value (in ms for timers).
; K:    Layer 2 max number of outstanding unacknowledged I frames (default 7)
; N200: Layer 2 max number of retransmissions of a frame (default 3)
; T200: Layer 2 max time before retransmission of a frame (default 1000 ms)
; T203: Layer 2 max time without frames being exchanged (default 10000 ms)
; T305: Wait for DISCONNECT acknowledge (default 30000 ms)
; T308: Wait for RELEASE acknowledge (default 4000 ms)
; T309: Maintain active calls on Layer 2 disconnection (default -1,
;       Asterisk clears calls)
;       EuroISDN: 6000 to 12000 ms, according to (N200 + 1) x T200 + 2s
;       May vary in other ISDN standards (Q.931 1993 : 90000 ms)
; T313: Wait for CONNECT acknowledge, CPE side only (default 3000 ms)
;
;pritimer => t200,1000
;pritimer => t313,4000
;
; To enable transmission of facility-based ISDN supplementary services (such
; as caller name from CPE over facility), enable this option.
; Cannot be changed on a reload.
;
;facilityenable = yes
;
; pritimer cannot be changed on a reload.
;
; Signalling method. The default is "auto". Valid values:
; auto:           Use the current value from DAHDI.
; bri_cpe         ISDN BRI in TE mode and Point to Point
; bri_nt           ISDN BRI in NT mode and Point to Point
; bri_cpe_ptmp ISDN BRI in TE mode and Point to multi Point
; em:             E & M
; em_e1:          E & M E1
; em_w:           E & M Wink
; featd:          Feature Group D (The fake, Adtran style, DTMF)
; featdmf:        Feature Group D (The real thing, MF (domestic, US))
; featdmf_ta:     Feature Group D (The real thing, MF (domestic, US)) through
;                 a Tandem Access point
; featb:          Feature Group B (MF (domestic, US))
; fgccama         Feature Group C-CAMA (DP DNIS, MF ANI)
; fgccamamf       Feature Group C-CAMA MF (MF DNIS, MF ANI)
; fxs_ls:         FXS (Loop Start)
; fxs_gs:         FXS (Ground Start)
; fxs_ks:         FXS (Kewl Start)
; fxo_ls:         FXO (Loop Start)
; fxo_gs:         FXO (Ground Start)
; fxo_ks:         FXO (Kewl Start)
; pri_cpe:        PRI signalling, CPE side
; pri_net:        PRI signalling, Network side
; gr303fxoks_net: GR-303 Signalling, FXO Loopstart, Network side
; gr303fxsks_cpe: GR-303 Signalling, FXS Loopstart, CPE side
```

```
The following SF signaling does not appear to be supported in /etc/dahdi/system.conf, dahdi responds with an error of u

; sf:             SF (Inband Tone) Signalling
; sf_w:           SF Wink
; sf_featd:       SF Feature Group D (The fake, Adtran style, DTMF)
; sf_featdmf:     SF Feature Group D (The real thing, MF (domestic, US))
; sf_featb:       SF Feature Group B (MF (domestic, US))

; e911:           E911 (MF) style signalling
; ss7:            Signalling System 7
;
; The following are used for Radio interfaces:
; fxs_rx:         Receive audio/COR on an FXS kewlstart interface (FXO at the
;                 channel bank)
; fxs_tx:         Transmit audio/PTT on an FXS loopstart interface (FXO at the
;                 channel bank)
; fxo_rx:         Receive audio/COR on an FXO loopstart interface (FXS at the
;                 channel bank)
; fxo_tx:         Transmit audio/PTT on an FXO groundstart interface (FXS at
;                 the channel bank)
; em_rx:          Receive audio/COR on an E&M interface (1-way)
; em_tx:          Transmit audio/PTT on an E&M interface (1-way)
; em_txrx:        Receive audio/COR AND Transmit audio/PTT on an E&M interface
;                 (2-way)
; em_rxtx:        Same as em_txrx (for our dyslexic friends)
; sf_rx:          Receive audio/COR on an SF interface (1-way)
; sf_tx:          Transmit audio/PTT on an SF interface (1-way)
; sf_txrx:        Receive audio/COR AND Transmit audio/PTT on an SF interface
;                 (2-way)
; sf_rxtx:        Same as sf_txrx (for our dyslexic friends)
; ss7:            Signalling System 7
;
; signalling of a channel can not be changed on a reload.
;
;signalling=fxo_ls
;
; If you have an outbound signalling format that is different from format
; specified above (but compatible), you can specify outbound signalling format,
; (see below). The 'signalling' format specified will be the inbound signalling
; format. If you only specify 'signalling', then it will be the format for
; both inbound and outbound.
;
; outsignalling can only be one of:
;   em, em_e1, em_w, sf, sf_w, sf_featd, sf_featdmf, sf_featb, featd,
;   featdmf, featdmf_ta, e911, fgccama, fgccamamf
;
; outsignalling cannot be changed on a reload.
;
;signalling=featdmf
;
;outsignalling=featb
;
; For Feature Group D Tandem access, to set the default CIC and OZZ use these
; parameters (Will not be updated on reload):
;
;defaultozz=0000
;defaultcic=303
;
; A variety of timing parameters can be specified as well
; The default values for those are "-1", which is to use the
; compile-time defaults of the DAHDI kernel modules. The timing
; parameters, (with the standard default from DAHDI):
;
;    prewink:     Pre-wink time (default 50ms)
;    preflash:    Pre-flash time (default 50ms)
;    wink:        Wink time (default 150ms)
;    flash:       Flash time (default 750ms)
;    start:       Start time (default 1500ms)
;    rxwink:      Receiver wink time (default 300ms)
;    rxflash:     Receiver flashtime (default 1250ms)
;    debounce:    Debounce timing (default 600ms)
;
; None of them will update on a reload.
;
; How long generated tones (DTMF and MF) will be played on the channel
; (in milliseconds).
;
; This is a global, rather than a per-channel setting. It will not be
; updated on a reload.
;
;toneduration=100
;
; Whether or not to do distinctive ring detection on FXO lines:
;
;usedistinctiveringdetection=yes
;
; enable dring detection after caller ID for those countries like Australia
```

```
; where the ring cadence is changed *after* the caller ID spill:
;
;distinctiveringaftercid=yes
;
; Whether or not to use caller ID:
;
usecallerid=yes
;
; Hide the name part and leave just the number part of the caller ID
; string. Only applies to PRI channels.
;hidecalleridname=yes
;
; Type of caller ID signalling in use
;     bell     = bell202 as used in US (default)
;     v23      = v23 as used in the UK
;     v23_jp   = v23 as used in Japan
;     dtmf     = DTMF as used in Denmark, Sweden and Netherlands
;     smdi     = Use SMDI for caller ID.  Requires SMDI to be enabled (usesmdi).
;
;cidsignalling=v23
;
; What signals the start of caller ID
;     ring        = a ring signals the start (default)
;     polarity    = polarity reversal signals the start
;     polarity_IN = polarity reversal signals the start, for India,
;                   for dtmf dialtone detection; using DTMF.
;                   (see doc/India-CID.txt)
;
;cidstart=polarity
;
; Whether or not to hide outgoing caller ID (Override with *67 or *82)
; (If your dialplan doesn't catch it)
;
;hidecallerid=yes
;
; The following option enables receiving MWI on FXO lines.  The default
; value is no.  When this is enabled, and MWI notification indicates on or off,
; the script specified by the mwimonitornotify option is executed.  Also, an
; internal Asterisk MWI event will be generated so that any other part of
; Asterisk that cares about MWI state changes will get notified, just as if
; the state change came from app_voicemail. The energy level that must be seen
; before starting the MWI detection process can be set with 'mwilevel'.
;
;mwimonitor=no
;mwilevel=512
;
; This option is used in conjunction with mwimonitor.  This will get executed
; when incoming MWI state changes.  The script is passed 2 arguments.  The
; first is the corresponding mailbox, and the second is 1 or 0, indicating if
; there are messages waiting or not.
;
;mwimonitornotify=/usr/local/bin/dahdinotify.sh
;
; Whether or not to enable call waiting on internal extensions
; With this set to 'yes', busy extensions will hear the call-waiting
; tone, and can use hook-flash to switch between callers. The Dial()
; app will not return the "BUSY" result for extensions.
;
;callwaiting=yes
;
; Whether or not restrict outgoing caller ID (will be sent as ANI only, not
; available for the user)
; Mostly use with FXS ports
;
;restrictcid=no
;
; Whether or not use the caller ID presentation for the outgoing call that the
; calling switch is sending.
; See README.callingpres. FIXME: file no longer exists.
;
;usecallingpres=yes
;
; Some countries (UK) have ring tones with different ring tones (ring-ring),
; which means the caller ID needs to be set later on, and not just after
; the first ring, as per the default (1).
;
;sendcalleridafter = 2
;
;
; Support caller ID on Call Waiting
;
;callwaitingcallerid=yes
;
; Support three-way calling
;
;threewaycalling=yes
;
; For FXS ports (either direct analog or over T1/E1):
```

```
;   Support flash-hook call transfer (requires three way calling)
;   Also enables call parking (overrides the 'canpark' parameter)
;
; For digital ports using ISDN PRI protocols:
;   Support switch-side transfer (called 2BCT, RLT or other names)
;   This setting must be enabled on both ports involved, and the
;   'facilityenable' setting must also be enabled to allow sending
;   the transfer to the ISDN switch, since it sent in a FACILITY
;   message.
;
;transfer=yes
;
; Allow call parking
; ('canpark=no' is overridden by 'transfer=yes')
;
;canpark=yes
;
; Support call forward variable
;
;cancallforward=yes
;
; Whether or not to support Call Return (*69, if your dialplan doesn't
; catch this first)
;
;callreturn=yes
;
; Stutter dialtone support: If a mailbox is specified without a voicemail
; context, then when voicemail is received in a mailbox in the default
; voicemail context in voicemail.conf, taking the phone off hook will cause a
; stutter dialtone instead of a normal one.
;
; If a mailbox is specified *with* a voicemail context, the same will result
; if voicemail received in mailbox in the specified voicemail context.
;
; for default voicemail context, the example below is fine:
;
;mailbox=1234
;
; for any other voicemail context, the following will produce the stutter tone:
;
;mailbox=1234@context
;
; Enable echo cancellation
; Use either "yes", "no", or a power of two from 32 to 256 if you wish to
; actually set the number of taps of cancellation.
;
; Note that when setting the number of taps, the number 256 does not translate
; to 256 ms of echo cancellation.  echocancel=256 means 256 / 8 = 32 ms.
;
; Note that if any of your DAHDI cards have hardware echo cancellers,
; then this setting only turns them on and off; numeric settings will

; be treated as "yes". There are no special settings required for
; hardware echo cancellers; when present and enabled in their kernel
; modules, they take precedence over the software echo canceller compiled
; into DAHDI automatically.
;
;
;echocancel=yes
;
; Some DAHDI echo cancellers (software and hardware) support adjustable
; parameters; these parameters can be supplied as additional options to
; the 'echocancel' setting. Note that Asterisk does not attempt to
; validate the parameters or their values, so if you supply an invalid
; parameter you will not know the specific reason it failed without
; checking the kernel message log for the error(s) put there by DAHDI.
;
;echocancel=128,param1=32,param2=0,param3=14
;
; Generally, it is not necessary (and in fact undesirable) to echo cancel when
; the circuit path is entirely TDM.  You may, however, change this behavior
; by enabling the echo canceller during pure TDM bridging below.
;
;echocancelwhenbridged=yes
;
; In some cases, the echo canceller doesn't train quickly enough and there
; is echo at the beginning of the call.  Enabling echo training will cause
; DAHDI to briefly mute the channel, send an impulse, and use the impulse
; response to pre-train the echo canceller so it can start out with a much
; closer idea of the actual echo.  Value may be "yes", "no", or a number of
; milliseconds to delay before training (default = 400)
;
; WARNING:  In some cases this option can make echo worse!  If you are
; trying to debug an echo problem, it is worth checking to see if your echo
; is better with the option set to yes or no.  Use whatever setting gives
; the best results.
;
; Note that these parameters do not apply to hardware echo cancellers.
```

```
;
;echotraining=yes
;echotraining=800
;
; If you are having trouble with DTMF detection, you can relax the DTMF
; detection parameters.  Relaxing them may make the DTMF detector more likely
; to have "talkoff" where DTMF is detected when it shouldn't be.
;
;relaxdtmf=yes
;
; You may also set the default receive and transmit gains (in dB)
;
; Gain Settings: increasing / decreasing the volume level on a channel.
;                The values are in db (decibells). A positive number
;                increases the volume level on a channel, and a
;                negavive value decreases volume level.
;
;                There are several independent gain settings:
;   rxgain: gain for the rx (receive - into Asterisk) channel. Default: 0.0
;   txgain: gain for the tx (transmit - out of Asterisk Asterisk) channel.
;           Default: 0.0
;   cid_rxgain: set the gain just for the caller ID sounds Asterisk
;               emits. Default: 5.0 .

;rxgain=2.0
;txgain=3.0
;
; Logical groups can be assigned to allow outgoing roll-over.  Groups range
; from 0 to 63, and multiple groups can be specified. By default the
; channel is not a member of any group.
;
; Note that an explicit empty value for 'group' is invalid, and will not
; override a previous non-empty one. The same applies to callgroup and
; pickupgroup as well.
;
;group=1
;
; Ring groups (a.k.a. call groups) and pickup groups.  If a phone is ringing
; and it is a member of a group which is one of your pickup groups, then
; you can answer it by picking up and dialing *8#.  For simple offices, just
; make these both the same.  Groups range from 0 to 63.
;
;callgroup=1
;pickupgroup=1

; Channel variable to be set for all calls from this channel
;setvar=CHANNEL=42
;setvar=ATTENDED_TRANSFER_COMPLETE_SOUND=beep   ; This channel variable will
                                                ; cause the given audio file to
                                                ; be played upon completion of
                                                ; an attended transfer.

;
; Specify whether the channel should be answered immediately or if the simple
; switch should provide dialtone, read digits, etc.
; Note: If immediate=yes the dialplan execution will always start at extension
; 's' priority 1 regardless of the dialed number!
;
;immediate=yes
;
; Specify whether flash-hook transfers to 'busy' channels should complete or
; return to the caller performing the transfer (default is yes).
;
;transfertobusy=no
;
; caller ID can be set to "asreceived" or a specific number if you want to
; override it.  Note that "asreceived" only applies to trunk interfaces.
; fullname sets just the
;
; fullname: sets just the name part.
; cid_number: sets just the number part:
;
;callerid = 123456
;
;callerid = My Name <2564286000>
; Which can also be written as:
;cid_number = 2564286000
;fullname = My Name
;
;callerid = asreceived
;
; should we use the caller ID from incoming call on DAHDI transfer?
;
;useincomingcalleridondahditransfer = yes
;
; AMA flags affects the recording of Call Detail Records.  If specified
; it may be 'default', 'omit', 'billing', or 'documentation'.
;
```

```
;amaflags=default
;
; Channels may be associated with an account code to ease
; billing
;
;accountcode=lss0101
;
; ADSI (Analog Display Services Interface) can be enabled on a per-channel
; basis if you have (or may have) ADSI compatible CPE equipment
;
;adsi=yes
;
; SMDI (Simplified Message Desk Interface) can be enabled on a per-channel
; basis if you would like that channel to behave like an SMDI message desk.
; The SMDI port specified should have already been defined in smdi.conf.  The
; default port is /dev/ttyS0.
;
;usesmdi=yes
;smdiport=/dev/ttyS0
;
; On trunk interfaces (FXS) and E&M interfaces (E&M, Wink, Feature Group D
; etc, it can be useful to perform busy detection either in an effort to
; detect hangup or for detecting busies.  This enables listening for
; the beep-beep busy pattern.
;
;busydetect=yes
;
; If busydetect is enabled, it is also possible to specify how many busy tones
; to wait for before hanging up.  The default is 3, but it might be
; safer to set to 6 or even 8.  Mind that the higher the number, the more
; time that will be needed to hangup a channel, but lowers the probability
; that you will get random hangups.
;
;busycount=6
;
; If busydetect is enabled, it is also possible to specify the cadence of your
; busy signal.  In many countries, it is 500msec on, 500msec off.  Without
; busypattern specified, we'll accept any regular sound-silence pattern that
; repeats <busycount> times as a busy signal.  If you specify busypattern,
; then we'll further check the length of the sound (tone) and silence, which
; will further reduce the chance of a false positive.
;
;busypattern=500,500
;
; NOTE: In make menuselect, you'll find further options to tweak the busy
; detector.  If your country has a busy tone with the same length tone and
; silence (as many countries do), consider enabling the
; BUSYDETECT_COMPARE_TONE_AND_SILENCE option.
;
; To further detect which hangup tone your telco provider is sending, it is
; useful to use the ztmonitor utility to record the audio that main/dsp.c
; is receiving after the caller hangs up.
;
; Use a polarity reversal to mark when a outgoing call is answered by the
; remote party.
;
;answeronpolarityswitch=yes
;
; In some countries, a polarity reversal is used to signal the disconnect of a
; phone line.  If the hanguponpolarityswitch option is selected, the call will
; be considered "hung up" on a polarity reversal.
;
;hanguponpolarityswitch=yes
;
; polarityonanswerdelay: minimal time period (ms) between the answer
;                        polarity switch and hangup polarity switch.
;                        (default: 600ms)
;
; On trunk interfaces (FXS) it can be useful to attempt to follow the progress
; of a call through RINGING, BUSY, and ANSWERING.   If turned on, call
; progress attempts to determine answer, busy, and ringing on phone lines.
; This feature is HIGHLY EXPERIMENTAL and can easily detect false answers,
; so don't count on it being very accurate.
;
; Few zones are supported at the time of this writing, but may be selected
; with "progzone".
;
; progzone also affects the pattern used for buzydetect (unless
; busypattern is set explicitly). The possible values are:
;   us (default)
;   ca (alias for 'us')
;   cr (Costa Rica)
;   br (Brazil, alias for 'cr')
;   uk
;
; This feature can also easily detect false hangups. The symptoms of this is
; being disconnected in the middle of a call for no reason.
;
```

```
;callprogress=yes
;progzone=uk
;
; Set the tonezone. Equivalent of the defaultzone settings in
; /etc/dahdi.conf . This sets the tone zone by number.
; Note that you'd still need to load tonezones (loadzone in dahdi.conf).
; The default is -1: not to set anything.
;tonezone = 0 ; 0 is US
;
; FXO (FXS signalled) devices must have a timeout to determine if there was a
; hangup before the line was answered.  This value can be tweaked to shorten
; how long it takes before DAHDI considers a non-ringing line to have hungup.
;
; ringtimeout will not update on a reload.
;
;ringtimeout=8000
;
; For FXO (FXS signalled) devices, whether to use pulse dial instead of DTMF
; Pulse digits from phones (FXS devices, FXO signalling) are always
; detected.
;
;pulsedial=yes
;
; For fax detection, uncomment one of the following lines.  The default is *OFF*
;
;faxdetect=both
;faxdetect=incoming
;faxdetect=outgoing
;faxdetect=no
;
; This option specifies a preference for which music on hold class this channel
; should listen to when put on hold if the music class has not been set on the
; channel with Set(CHANNEL(musicclass)=whatever) in the dialplan, and the peer
; channel putting this one on hold did not suggest a music class.
;
; If this option is set to "passthrough", then the hold message will always be
; passed through as signalling instead of generating hold music locally. This
; setting is only valid when used on a channel that uses digital signalling.
;
;mohinterpret=default
;
; This option specifies which music on hold class to suggest to the peer channel
; when this channel places the peer on hold.
;
;mohsuggest=default
;
; PRI channels can have an idle extension and a minunused number.  So long as
; at least "minunused" channels are idle, chan_dahdi will try to call "idledial"
; on them, and then dump them into the PBX in the "idleext" extension (which
; is of the form exten@context).  When channels are needed the "idle" calls
; are disconnected (so long as there are at least "minidle" calls still
; running, of course) to make more channels available.  The primary use of
; this is to create a dynamic service, where idle channels are bundled through
; multilink PPP, thus more efficiently utilizing combined voice/data services
; than conventional fixed mappings/muxings.
;
; Those settings cannot be changed on reload.
;
;idledial=6999
;idleext=6999@dialout
;minunused=2
;minidle=1
;
; Configure jitter buffers in DAHDI (each one is 20ms, default is 4)
; This is set globally, rather than per-channel.
;
;jitterbuffers=4
;
;------------------------------ JITTER BUFFER CONFIGURATION --------------------------
; jbenable = yes              ; Enables the use of a jitterbuffer on the receiving side of a
                              ; DAHDI channel. Defaults to "no". An enabled jitterbuffer will
                              ; be used only if the sending side can create and the receiving
                              ; side can not accept jitter. The DAHDI channel can't accept jitter,
                              ; thus an enabled jitterbuffer on the receive DAHDI side will always
                              ; be used if the sending side can create jitter.

; jbmaxsize = 200             ; Max length of the jitterbuffer in milliseconds.

; jbresyncthreshold = 1000    ; Jump in the frame timestamps over which the jitterbuffer is
                              ; resynchronized. Useful to improve the quality of the voice, with
                              ; big jumps in/broken timestamps, usually sent from exotic devices
                              ; and programs. Defaults to 1000.

; jbimpl = fixed              ; Jitterbuffer implementation, used on the receiving side of a DAHDI
                              ; channel. Two implementations are currently available - "fixed"
                              ; (with size always equals to jbmax-size) and "adaptive" (with
                              ; variable size, actually the new jb of IAX2). Defaults to fixed.
```

```
; jblog = no                    ; Enables jitterbuffer frame logging. Defaults to "no".
;-----------------------------------------------------------------------------------
;
; You can define your own custom ring cadences here.  You can define up to 8
; pairs.  If the silence is negative, it indicates where the caller ID spill is
; to be placed.  Also, if you define any custom cadences, the default cadences
; will be turned off.
;
; This setting is global, rather than per-channel. It will not update on
; a reload.
;
; Syntax is:  cadence=ring,silence[,ring,silence[...]]
;
; These are the default cadences:
;
;cadence=125,125,2000,-4000
;cadence=250,250,500,1000,250,250,500,-4000
;cadence=125,125,125,125,125,-4000
;cadence=1000,500,2500,-5000
;
; Each channel consists of the channel number or range.  It inherits the
; parameters that were specified above its declaration.
;
; For GR-303, CRV's are created like channels except they must start with the
; trunk group followed by a colon, e.g.:
;
; crv => 1:1
; crv => 2:1-2,5-8
;
;
;callerid="Green Phone"<(256) 428-6121>
;channel => 1
;callerid="Black Phone"<(256) 428-6122>
;channel => 2
;callerid="CallerID Phone" <(630) 372-1564>
;channel => 3
;callerid="Pac Tel Phone" <(256) 428-6124>
;channel => 4
;callerid="Uniden Dead" <(256) 428-6125>
;channel => 5
;callerid="Cortelco 2500" <(256) 428-6126>
;channel => 6
;callerid="Main TA 750" <(256) 428-6127>
;channel => 44
;
; For example, maybe we have some other channels which start out in a
; different context and use E & M signalling instead.
;
;context=remote
;sigalling=em
;channel => 15
;channel => 16

;signalling=em_w
;
; All those in group 0 I'll use for outgoing calls
;
; Strip most significant digit (9) before sending
;
;stripmsd=1
;callerid=asreceived
;group=0
;signalling=fxs_ls
;channel => 45

;signalling=fxo_ls
;group=1
;callerid="Joe Schmoe" <(256) 428-6131>
;channel => 25
;callerid="Megan May" <(256) 428-6132>
;channel => 26
;callerid="Suzy Queue" <(256) 428-6233>
;channel => 27
;callerid="Larry Moe" <(256) 428-6234>
;channel => 28
;
; Sample PRI (CPE) config:  Specify the switchtype, the signalling as either
; pri_cpe or pri_net for CPE or Network termination, and generally you will
; want to create a single "group" for all channels of the PRI.
;
; switchtype cannot be changed on a reload.
;
; switchtype = national
; signalling = pri_cpe
; group = 2
; channel => 1-23

;
```

```
;  Used for distinctive ring support for x100p.
;  You can see the dringX patterns is to set any one of the dringXcontext fields
;  and they will be printed on the console when an inbound call comes in.
;
;  dringXrange is used to change the acceptable ranges for "tone offsets".  Defaults to 10.
;  Note: a range of 0 is NOT what you might expect - it instead forces it to the default.
;  A range of -1 will force it to always match.
;  Anything lower than -1 would presumably cause it to never match.
;
;dring1=95,0,0
;dring1context=internal1
;dring1range=10
;dring2=325,95,0
;dring2context=internal2
;dring2range=10
; If no pattern is matched here is where we go.
;context=default
;channel => 1

; ---------------- Options for use with signalling=ss7 -----------------
; None of them can be changed by a reload.
;
; Variant of SS7 signalling:
; Options are itu and ansi
;ss7type = itu

; SS7 Called Nature of Address Indicator
;
; unknown:        Unknown
; subscriber:     Subscriber
; national:       National
; international:  International
; dynamic:        Dynamically selects the appropriate dialplan
;
;ss7_called_nai=dynamic
;
; SS7 Calling Nature of Address Indicator
;
; unknown:        Unknown
; subscriber:     Subscriber
; national:       National
; international:  International
; dynamic:        Dynamically selects the appropriate dialplan
;
;ss7_calling_nai=dynamic
;
;
; sample 1 for Germany
;ss7_internationalprefix = 00
;ss7_nationalprefix = 0
;ss7_subscriberprefix =
;ss7_unknownprefix =
;

; This option is used to disable automatic sending of ACM when the call is started
; in the dialplan.  If you do use this option, you will need to use the Proceeding()
; application in the dialplan to send ACM.
;ss7_explictacm=yes

; All settings apply to linkset 1
;linkset = 1

; Point code of the linkset.  For ITU, this is the decimal number
; format of the point code.  For ANSI, this can either be in decimal
; number format or in the xxx-xxx-xxx format
;pointcode = 1

; Point code of node adjacent to this signalling link (Possibly the STP between you and
; your destination).  Point code format follows the same rules as above.
;adjpointcode = 2

; Default point code that you would like to assign to outgoing messages (in case of
; routing through STPs, or using A links).  Point code format follows the same rules
; as above.
;defaultdpc = 3

; Begin CIC (Circuit indication codes) count with this number
;cicbeginswith = 1

; What the MTP3 network indicator bits should be set to.  Choices are
; national, national_spare, international, international_spare
;networkindicator=international

; First signalling channel
;sigchan = 48

; Additional signalling channel for this linkset (So you can have a linkset
```

```
; with two signalling links in it).  It seems like a silly way to do it, but
; for linksets with multiple signalling links, you add an additional sigchan
; line for every additional signalling link on the linkset.
;sigchan = 96

; Channels to associate with CICs on this linkset
;channel = 25-47
;
; For more information on setting up SS7, see the README file in libss7 or
; the doc/ss7.txt file in the Asterisk source tree.
; ----------------- SS7 Options ----------------------------------------

; Configuration Sections
; ~~~~~~~~~~~~~~~~~~~~~~
; You can also configure channels in a separate dahdi.conf section. In
; this case the keyword 'channel' is not used. Instead the keyword
; 'dahdichan' is used (as in users.conf) - configuration is only processed
; in a section where the keyword dahdichan is used. It will only be
; processed in the end of the section. Thus the following section:
;
;[phones]
;echocancel = 64
;dahdichan = 1-8
;group = 1
;
; Is somewhat equivalent to the following snippet in the section
; [channels]:
;
;echocancel = 64
;group = 1
;channel => 1-8
;
; When starting a new section almost all of the configuration values are
; copied from their values at the end of the section [channels] in
; dahdi.conf and [general] in users.conf - one section's configuration
; does not affect another one's.
;
; Instead of letting common configuration values "slide through" you can
; use configuration templates to easily keep the common part in one
; place and override where needed.
;
;[phones](!)
;echocancel = yes
;group = 0,4
;callgroup = 3
;pickupgroup = 3
;threewaycalling = yes
;transfer = yes
;context = phones
;faxdetect = incoming
;
;[phone-1](phones)
;dahdichan = 1
;callerid = My Name <501>
;mailbox = 501@mailboxes
;
;
;[fax](phones)
;dahdichan = 2
;faxdetect = no
;context = fax
;
;[phone-3](phones)
;dahdichan = 3
;pickupgroup = 3,4
;
group=1
context=inside-users
signalling=fxo_ks
callerid="Joe the Plomer" <123>
mailbox="123"
callwaiting=yes
threewaycalling=yes
transfer=yes
channel =>25
;
group=4
context=from_pstn
signalling=fxs_ks
callerid=asreceived
callwaiting=o
channel => 28
;
;configuration for PRI
group=5
context=from_pri_provider
echocancel=yes
echocancelwhenbridged=no
```

```
echotraining=yes
switchtype=5ess
signaling=pri_cpe
channel => 1-23
;
;
;
;
```

^

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