

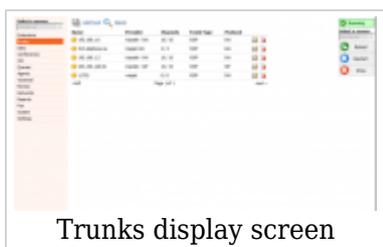
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From Bicom Systems Wiki

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Trunks



Trunks are a transmission line between two systems. This transmission is done using a wide range of PSTN and VOIP technologies. This screen lists all system Trunks with the following details:

- **Name**

This field tell us name of the trunk.
(ex. It can be TrunkName or IP address 192.168.1.6)
(Display)

- **Provider**

Provider template name.

(ex. SIP TRUNK)

- **Channels**

Maximum number of inbound/outbound channels available for the trunk.
e.g. 10/10

- **Trunk Type**

Type of a trunk
(ex. PSTN, VOIP etc.)

- **Protocol**

Protocol used by the trunk
(ex. SIP, IAX etc.)

-  Click this button if you want to edit trunk configuration.
-  Click this button in order to delete a trunk from the system

Search

The search bar filters trunks by name and provider.



Search phrase
(ex. Provide a search phrase here and hit enter to filter the records)
([a-z][0-9])

- **Name**

Should the search filter be applied to trunk names.
(ex. Check the box to search trunk names)
(Check box)

- **Provider**

Should the search filter be applied to provider names.
(ex. Check the box to search provider names)
(Check box)

Add/Edit Trunk

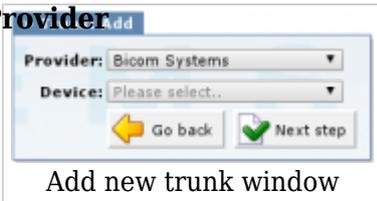
When adding a new trunk, the first step requires 'Provider' and 'Device' selection.

TIP:

Although a new trunk can be created without it, it is preferred that the 'Provider' template be created first under 'Settings: Providers'.

For basic Provider and Trunk HowTo check our HOWTO Create a Trunk

▪ Provider



Select a service provider template.
(ex. BT)
(Select box)

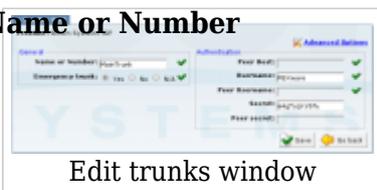
▪ Device

If the providers service requires a device in order to provide the service, this field will become visible.
(ex. None, T100)
(Select box)

VoIP (SIP/IAX)

The second step of the trunk installation and trunk edit command, opens the following options:

▪ Name or Number



Some providers require this field to be equal to the DID number (e.g.55510205); but if connecting two systems, the IP address may be used as well.
(ex. 2554433, myvoiceboxlink)
([a-z] [0-9])

▪ Emergency trunk

Should emergency services (Police, Ambulance etc) be dialed through this trunk.
(ex. Dialing 911 will pass the call through this trunk)
(Option buttons)

- **Peer Host**

The IP of a peer host system sends the calls to.
(ex. 192.168.1.1)
(IP Address)

- **Username**

The username for authenticating with the service provider.
(ex. 2554433)
([0-9])

- **Peer Username**

The peer username for authenticating with the service provider.
(ex. 2554433)
([a-z] [0-9])

- **Secret**

The Secret/Password used for authenticating with the service provider.
(ex. 123456)
([a-z] [0-9])

- **Peer secret**

The peer secret/password used for authenticating with the service provider.
(ex. 123456)
([a-z] [0-9])

Advanced Options

General

These options are used frequently and required for normal trunk operation. Some of these fields are pre-configured with the default values. It is not recommended to change them.

- **User Type**

General options

The user's relationship to the system

Example:

- **user** - The trunk accepts incoming calls only
- **peer** - The trunk makes outgoing calls only
- **friend** - The trunk does both incoming and outgoing calls

(Select box)

▪ **DTMF Mode (Dual Tone Multi-Frequency)**

Trunk DTMF mode. A specific frequency (consisting of two separate tones) to each key so that it can be easily identified by a microprocessor

Example:

- **inband** - inband audio (requires 64 kbit codec - alaw, ulaw)
- **rfc2833** - default
- **info** - SIP INFO messages

(Select box)

▪ **RFC2833 Compensate (1.2)**

Compensate for pre-1.4 Asterisk DTMF transmission from another machine. You must have this turned on or DTMF reception will work improperly.

(ex. Yes, No, N/A)

(Option buttons)

▪ **Context**

Contexts define a scope within PBXware. The trunk context cannot be modified and is the same as the trunk name or number.

(ex. 2554433)

([a-z][0-9])

▪ **Status**

Set trunk status on the system.

(ex. Rather than deleting the trunk, you can disable it on the system level by selecting 'Not Activated' here)

(Select box)

▪ **Qualify**

Timing interval in milliseconds at which a 'ping' is sent to a host in order to find out its status.

(ex. Set this field to 2000, for example. If more time than provided here is needed to reach the host, the host is considered offline)

([0-9])

▪ **VoiceMaster Trunk**

Set whether this trunk leads to the VoiceMaster gateway.

(ex. If you have a VoiceMaster gateway and are creating this trunk to connect it with this system, select 'Yes' here)

(Option buttons)

▪ **Country**

Country where the service provider resides.

(Select USA, for example, if the provider is from the United States)

(Select box)

▪ **National Dialing Code**

National dialing code used at the trunk destination.

(ex. For USA 1. For the United Kingdom and Germany 0)

(Select box)

▪ **International Dialing Code**

International dialing code used at the trunk destination.

(ex. For USA 011. For the United Kingdom and Germany 00)

([0-9])

▪ **E164 Accepted**

Does the trunk support dialing destinations in the E164 format.
(ex. Enabling this option will reformat any dialed number into the following form COUNTRY_CODE + AREA_CODE + DIALED_NUMBER. For example, if you dial 55510205, the system will dial 121255510205)
(Option buttons)

▪ **Pass-thru mode**

Pass the digits dialed without any conversion (E164, National, Area code).

NOTE: When active, 'Leave National Code and 'Local Area Code' will be disabled.
(ex. If this option is disabled, PBXware will convert all dialed numbers to E164 format (COUNTRY_CODE + AREA_CODE + DIALED_NUMBER) and then make a call to converted number. If this option is enabled, PBXware will directly call the DIALED_NUMBER without making any number conversions)
(Option buttons)

▪ **Leave National Code**

In some countries, the national code is stripped automatically. If set to 'Yes', the national code will not be stripped from the dialed number.

NOTE: Before setting this option to 'Yes', go to 'Settings: Servers' and enable this option as well.
(ex. 035123456 will not be striped of 0)
(Option buttons)

▪ **Local Area Code**

Add the local area code to the dialed number, if required by the service provider. (By default, the local area code is stripped when dialing).
(ex. The user dials 55510205, the local area code is 212. If the call goes through this trunk, PBXware will dial 21210205)
([0-9])

▪ **Prefix**

Value added to all dialed numbers going over the trunk.
(ex. Prefix 5 + Dialed number 123 = System dials 5123)
([0-9])

▪ **Outbound Caller ID**

If the Caller ID is not set by the UAD, the value provided here will be used instead for all outgoing calls.
(ex. 55599999)
([0-9])

▪ Allow ES Caller ID

Should ES (Enhanced Services) caller id be allowed over this trunk.
(ex. Any extension can set a custom caller id for each system trunk. With this option enabled, that caller id will be used instead of the trunk outbound caller ID).
([0-9])

▪ Test number

This is the number that the trunk will try to call when Trunk Monitoring is turned on. If the answer is not: ANSWER, BUSY, CANCEL, or NOANSWER, this call is considered non-functional and the monitoring system will send a notification email.
(1009)
([0-9])

▪ Send PAI header

The P-Asserted-Identity contains the caller ID information for the call on the INVITE SIP packet. This is the acceptable way to specify privacy information for calls. This field enables you to send preferred PAI header using several variables:

- %CALLERIDNUM%
- %TENANT%
- %EXT%
- %TENANTEXT%

Authentication

▪ Host

Authentication settings

The IP address that the host trunk is connecting to.

Example:

Enter a host IP, 192.168.1.1, for example, or set 'dynamic' if the host is behind the dynamic IP address.

([0-9][a-z])

- **Authname:**

Value assigned to the Digest username= SIP header
(ex. 2222)
([0-9][a-z])

- **Auth**

Authenticate for outbound calls to other realms.
(ex. user:secret@realm)
([0-9][a-z])

- **Register**

Method for registering to the remote server.
(ex. Providers may require a different form of registration to their server. You may choose between 'registration not required', 'register with phone number', and 'register with username').
(Select box)

- **Register suffix**

The service provider may request different registration methods for their services. Select the proper method, as required by the provider.
(ex. 1234567)
([0-9])

- **Insecure**

Typically used to allow incoming calls while having a "friend" type entry defined with a username and password

Example:

- Yes
- No
- very
- port
- invite
- port, invite

(Select box)

- **From User**

What to show when calling TO this peer FROM asterisk.
(ex. 152)
([0-9])

- **From Domain**

From domain data is required by some providers.
(ex. If your provider requires this information, provide the exact value here).
([a-z][0-9])

- **user=phone in URI**

If yes, ";user=phone" is added to uri that contains a valid phone number.
(ex. Yes, No, N/A)
(Option buttons)

- **Outbound Proxy**

Send outbound signaling to this proxy, not directly to the peer.
(ex. outbound.proxy.com)
([0-9][a-z].)

- **Incoming IP addresses (new line seperated)**

You may limit SIP traffic to and from this peer to a certain IP or network.
(ex. 10.1.1.9)
([0-9].)

IAX Specific Authentication Settings

- **Encryption**

Should encryption be used when authenticating with the peer.
(ex. Yes, No, N/A)
([a-z][0-9])

- **Auth Method**

Authentication method required by the provider.
(ex. md5)
([a-z] [0-9])

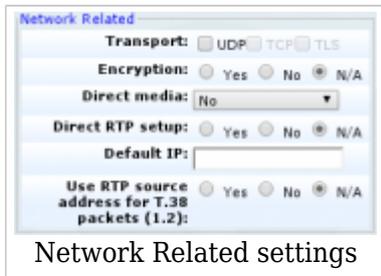
- **RSA key**

RSA authentication key
(ex. If the Auth Method is set to RSA, provide the RSA key here).
([a-z][0-9])

Network Related

These options set important network related values regarding NAT.

- **Transport:**



Type of transfer protocol that will be used on PBXware.

UDP (User Datagram Protocol) - With UDP, computer applications can send messages, in this case referred to as datagrams, to other hosts on an Internet Protocol (IP) network without prior communications to set up special transmission channels or data paths.

TCP (Transmission Control Protocol) - provides reliable, ordered, error-checked delivery of a stream of octets between programs running on computers connected to an intranet or the public Internet.

TLS (Transport Layer Security) - cryptographic protocol that provide communication security over the Internet.[1] They use asymmetric cryptography for authentication of key exchange, symmetric encryption for confidentiality, and message authentication codes for message integrity.

Type: Checkbox

- **Encryption:**

This option enables or disables encryption in PBXware transport.

Options: Yes, No, N/A.

- **Direct media**

Should you allow RTP voice traffic to bypass PBXware.

NOTE: All enhanced services for the extension have to be disabled.

(ex. Some devices do not support this especially if one of them is behind a NAT).

(Options buttons)

- **Direct RTP setup:**

Here you can enable or disable the new experimental direct RTP setup. Setting this value to yes sets up the call directly with media peer-2-peer without re-invites. Will not work for video and cases where the callee sends RTP payloads and fntp headers in the 200 OK that does not match the callers INVITE. This will also fail if directmedia is enabled when the device is actually behind NAT.

Options: Yes, No, N/A

- **Default IP**

The IP address to be used until registration.

(ex. 192.168.1.1)
(IP Address)

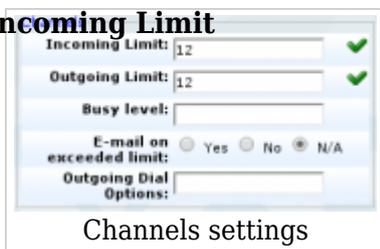
- **Use RTP source address for T.38 packets (1.2)**

Use the source IP address of RTP as the destination IP address for T.38 packets if the nat option is enabled. If a single RTP packet is received, Asterisk will know the external IP address of the remote device. If port forwarding is done at the client side, then T.38 will flow to the remote device.

(ex. Yes, No, N/A)
(Option buttons)

Channels

- **Incoming Limit**



The screenshot shows the 'Channels settings' window. It contains several fields: 'Incoming Limit' with a value of 12 and a green checkmark; 'Outgoing Limit' with a value of 12 and a green checkmark; 'Busy level' which is empty; 'E-mail on exceeded limit' with radio buttons for 'Yes', 'No', and 'N/A' (selected); and 'Outgoing Dial Options' which is empty.

The number of simultaneous incoming calls that the trunk can handle.

(ex. 4 = four simultaneous incoming calls. Any additional calls will get a busy sound).
([0-9])

- **Outgoing Limit**

Number of simultaneous outgoing calls that the trunk can handle.

(ex. 4 = four simultaneous outgoing calls. Any additional calls attempting to use this trunk will be rejected or will be redirected to other trunks depending on what is set in the system/extensions).
([0-9])

- **Busy Level**

Number of concurrent calls until user/peer is busy.

- **E-mail on exceeded limit**

Send an e-mail when the outgoing limit is reached, can be set to Yes, No or N/A.
(Option buttons)

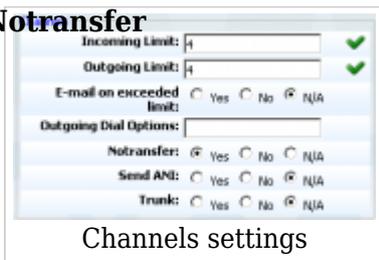
- **Outgoing Dial Options**

Advanced dial options for all outgoing calls.

(ex. trT)
([a-z])

IAX Specific Channels settings

▪ Notransfer



Channels settings

Disable the native IAX transfer.

(ex. Yes, No, N/A)

(Option buttons)

▪ Send ANI

Should ANI ("super" Caller ID) be sent over this trunk.

(ex. Yes, No, N/A)

(Option buttons)

▪ Trunk

Use IAX2 trunking with this host.

(ex. Yes, No, N/A)

(Option buttons)

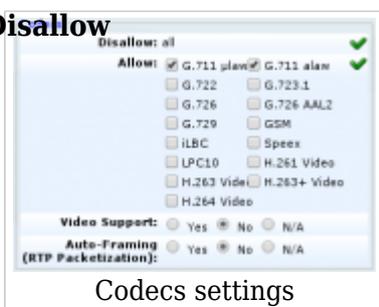
Codecs

Codecs are used to convert analog to digital voice signals and vice versa. These options set preferred codecs used by the extension.

TIP:

If some of the desired codecs cannot be checked, go to 'Settings: Servers: Edit: Default Codecs' and enable them under the 'Remote' group.

▪ Disallow



Codecs settings

The set that the codecs trunk is now allowed to use.

(ex. This field is very unique. In order to work properly, this setting is automatically set to 'Disallow All' and it cannot be modified).

(Read only)

▪ Allow

The set that the codecs extension is allowed to use.

Example:

Only the codecs set under 'Settings: Server' will be available to choose from

- **ITU G.711 ulaw** - 64 Kbps, sample-based, used in US
- **ITU G.711 alaw** - 64 Kbps, sample-based, used in Europe
- **ITU G.723.1** - 5.3/6.3 Kbps, 30ms frame size
- **ITU G.726** - 16/24/32/40 Kbps
- **ITU G.729** - 8 Kbps, 10ms frame size
- **GSM** - 13 Kbps (full rate), 20ms frame size
- **iLBC** - 15Kbps,20ms frame size: 13.3 Kbps, 30ms frame size
- **Speex** - 2.15 to 44.2 Kbps
- **LPC10** - 2.5 Kbps
- **H.261 Video** - Used over ISDN lines with resolution of 352x288
- **H.263 Video** - Low-bit rate encoding solution for video conferencing
- **H.263+ Video** - Extension of H.263 that provides additional features that improve compression over packet switched networks.

(Check box)

▪ Auto-Framing (RTP Packetization)

If auto framing is turned on, system will choose packetization level based on remote ends preferences.

(ex. Yes)

(Option buttons)

PSTN

» Trunks: Edit

Provider: Generic Analog Device: TDM12B

Zapata General

Name: 032445231 ✓

Emergency trunk: Yes No N/A ✓

Channel(s): 2,4 ✓

Group: 10 ✓

FXS Channels

FXS Kewlstart: 2,4

Locality

Country: Bosnia and Herzego ✓

E164: Yes No N/A ✓

International dialing code: 00 ✓

Advanced Options

Save Go back

▪ Name

Trunk name/number.

(ex. 032445231)

([a-z][0-9])

- **Emergency trunk**

Should emergency services (Police, Ambulance, etc) be called over this trunk.
(ex. Select 'Yes' in order to dial emergency services over this trunk).
(Option buttons)

- **Channel(s)**

Which card channels are used.
(ex. If channel 2 and 4 are used on your card, set '2, 4' here. If all four channels are used set '1-4' here).
([0-9], [-,])

- **Group**

Every PSTN trunk has to belong to a group. Selecting any group will enable the trunk.
(ex. With most of the cards, this option is auto detected and set. If that is the case with your card - do not change this field).
(Select box)

- **FXS Kewlstarts**

Signaling protocol for analog circuits that better detects far-end disconnects.
(ex. Select card channels to be monitored with it. For example '1, 4' or '1-4'. These numbers should match the 'Channel(s)' field).
([0-9], [-,])

- **Country**

Destination of the trunk connection.
(ex. If your system is located in the USA, select USA here)
(Select box)

- **E164 Accepted**

Does the trunk support dialing destinations in the E164 format. Enabling this option will reformat any dialed number into the following form
COUNTRY_CODE+AREA_CODE+DIALED_NUMBER.
(ex. If you dial 55510205, the system will dial 121255510205).
(Option buttons)

- **International Dialing Code:**

The international dialing code at the trunk destination.
(ex. For the USA 011, for the United Kingdom and Germany 00 etc.)
([0-9])

Advanced

Zapata

General

▪ Language

Zapata General settings

Default language.

(ex. us)

(Select box)

▪ Context

Contexts define a scope within the PBXware.

(ex. default)

([a-z][0-9])

▪ Status:

Trunk status

Example:

- Active
- Not Activated

(Select box)

▪ Signalling

Signaling method

Example:

- FXS Loopstart
- FXS Groundstart
- FXS Kewlstart
- FXO Loopstart
- FXO Groundstart
- FXO Kewlstart
- PRI CPE side
- PRI Network side

- BRI CPE side
- BRI Network side
- BRI CPE PTMP
- BRI Network PTMP

▪ **Test number**

This is the number which the trunk will try to call when Trunk Monitoring is turned on. If the answer is not: ANSWER, BUSY, CANCEL, or NOANSWER, this call is considered non-functional and the monitoring system will send a notification email.

(ex. 1009)

([0-9])

▪ **Music On Hold**

Select which class of music to use for music on hold. If not specified, the 'default' will be used.

(ex. default)

(Select box)

▪ **Mailbox**

Define a voicemail context.

(ex. 1234, 1234@context)

([a-z][0-9])

▪ **Group Method**

([a-z][0-9])

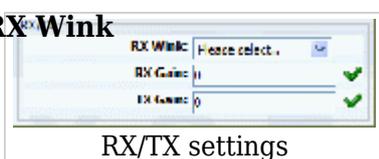
This option specifies how to choose a channel to use in the specified group.

The four possible options are:

- **g**: select the lowest-numbered non-busy DAHDI channel (aka. ascending sequential hunt group).
- **G**: select the highest-numbered non-busy DAHDI channel (aka. descending sequential hunt group).
- **r**: use a round-robin search, starting at the next highest channel than last time (aka. ascending rotary hunt group).
- **R**: use a round-robin search, starting at the next lowest channel than last time (aka. descending rotary hunt group).

RX/TX

▪ **RX Wink**



Set timing parameters

Example:

- Pre-wink (50ms)
- Pre-flash (50ms)
- Wink (150ms)
- Receiver flashtime (250ms)
- Receiver wink (300ms)
- Debounce timing (600ms)

(Select box)

▪ **RX Gain**

Receive signal decibel.

(ex. If incoming sound is low and you cannot hear the other party well, set this option to 2. That should increase incoming sound by 2 decibels).

([0-9])

▪ **TX Gain**

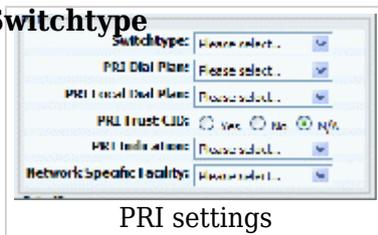
Transmit signal decibel.

(ex. If outgoing sound is low and the other party cannot hear you well, set this option to 2. That should increase outgoing sound by 2 decibels).

([0-9])

PRI

▪ **Switchtype**



Set switch type

Example:

- National ISDN 2
- Nortel DMS100
- AT&T 4ESS
- Lucent 5ESS
- EuroISDN
- Old National ISDN 1

(Select box)

▪ **PRI Dial Plan**

Set the dial plan used by some switches

Example:

- Unknown
- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

(Select box)

▪ **PRI Local Dial Plan**

Set the numbering dial plan for destinations called locally

Example:

- Unknown
- Private ISDN
- Local ISDN
- National ISDN
- International ISDN

(Select box)

▪ **PRI Trust CID**

Trust provided caller id information.

(ex. Yes, No, N/A)

(Option buttons)

▪ **PRI Indication**

How to report 'busy' and 'congestion' on a PRI

Example:

- **outofband** - Signal Busy/Congestion out of band with RELEASE/DISCONNECT
- **inband** - Signal Busy/Congestion using in-band tones

(Select box)

▪ **Network Specific Facility**

If required by switch, select the network specific facility

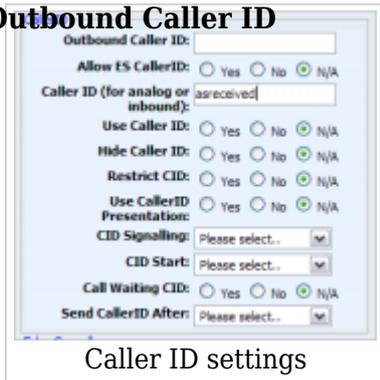
Example:

- none
- sdn
- megacom
- accunet

(Select box)

Caller ID

▪ Outbound Caller ID



The screenshot shows a 'Caller ID settings' dialog box with the following fields and options:

- Outbound Caller ID: [Empty text box]
- Allow ES CallerID: Yes No N/A
- Caller ID (for analog or inbound): [asreceived]
- Use Caller ID: Yes No N/A
- Hide Caller ID: Yes No N/A
- Restrict CID: Yes No N/A
- Use CallerID Presentation: Yes No N/A
- CID Signalling: [Please select...] [OK]
- CID Start: [Please select...] [OK]
- Call Waiting CID: Yes No N/A
- Send CallerID After: [Please select...] [OK]

Caller ID settings

Caller ID set for all outbound calls where the Caller ID is not set or supported by a device.
(ex. john@domain.com)
([0-9])

▪ Allow ES Caller ID

Should the ES (Enhanced Services) Caller ID be allowed over this trunk.
(ex. Any extension can set a custom Caller ID for each system trunk. With this option enabled, that Caller ID will be used instead of the Trunk Outbound Caller ID).
([0-9])

▪ Caller ID (for analog or inbound)

CallerID can be set to 'asreceived' or a specific number if you want to override it.
NOTE: Caller ID can only be transmitted to the public phone network with supported hardware, such as a PRI. It is not possible to set external caller ID on analog lines.
(ex. 'asreceived', 555648788)
([a-z][0-9])

▪ Use Caller ID

Whether or not to use caller ID.
(ex. Yes, No, N/A)
(Option buttons)

▪ Hide Caller ID

Whether or not to hide the outgoing caller ID.
(ex. Yes, No, N/A)
(Option buttons)

▪ Restrict CID

Whether or not to use the caller ID presentation for the outgoing call that the calling switch is sending.

(ex. Yes, No, N/A)
(Option buttons)

▪ **Use CallerID Presentation**

Whether or not to use the caller ID presentation for the outgoing call that the calling switch is sending.

(ex. Yes, No, N/A)
(Option buttons)

▪ **CID Signalling**

Set the type of caller ID signaling

Example:

- **bell** - US
- **v23** - UK
- **dtmf** - Denmark, Sweden and Netherlands

(Select box)

▪ **CID Start**

What signals the start of the Caller ID

Example:

- ring = a ring signals the start
- polarity = polarity reversal signals the start

(Select box)

▪ **Call Waiting CID**

Whether or not to enable call waiting on FXO lines.

(ex. Yes, No, N/A)
(Option buttons)

▪ **Send CallerID After**

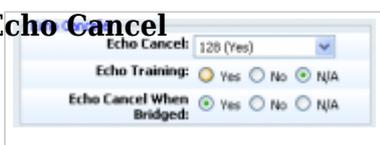
Some countries, like the UK, have different ring tones (ring-ring), which means the caller id needs to be set later on, and not just after the first ring, as per the default.

(ex. Yes)

(Select box)

Echo Canceler

▪ **Echo Cancel**



Echo Canceler settings

Level of enabled echo cancellation.
(ex. 128 (Yes))
(Select box)

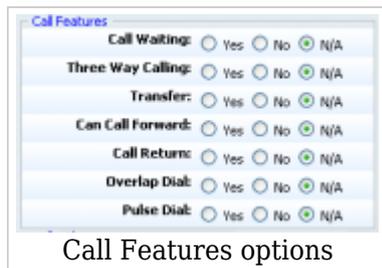
▪ Echo Training

Mute the channel briefly, for 400ms, at the beginning of the conversation, cancelling the echo. (Use this only if 'Echo Cancel' doesn't work as expected).
(ex. Yes, No, N/A)
(Option buttons)

▪ Echo Cancel When Bridged

Enable echo cancellation when bridged. Generally not necessary, and in fact undesirable, to echo cancel when the circuit path is entirely TDM.
(ex. Yes, No, N/A)
(Option buttons)

Call Features



▪ Call Waiting

Whether or not to enable call waiting on FXO lines.
(ex. Yes, No, N/A)
(Option buttons)

▪ Three Way Calling

Support three-way calling. If enabled, the call can be put on hold and one is able to make another call.
(ex. Yes, No, N/A)
(Option buttons)

▪ Transfer

Support call transfer and also enable call parking (overrides the 'canpark' parameter).

Requires 'Three Way Calling' = 'Yes'.
(ex. Yes, No, N/A)
(Option buttons)

▪ **Can Call Forward**

Support call forwarding.
(ex. Yes, No, N/A)
(Option buttons)

▪ **Call Return**

Whether or not to support Call Return '*69'. Dials last caller extension number.
(ex. Yes, No, N/A)
(Option buttons)

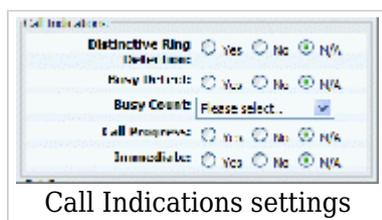
▪ **Overlap Dial**

Enable overlap dialing mode (sends overlap digits).
(ex. Yes, No, N/A)
(Option buttons)

▪ **Pulse Dial**

Use pulse dial instead of DTMF. Used by FXO (FXS signalling) devices.
(ex. Yes, No, N/A)
(Option buttons)

Call Indications



▪ **Distinctive Ring Detection**

Whether or not to do distinctive ring detection on FXO lines.
(ex. Yes, No, N/A)
(Option buttons)

▪ **Busy Detect**

Enable listening for the beep-beep busy pattern.
(ex. Yes, No, N/A)

(Option buttons)

▪ **Busy Count**

How many busy tones to wait before hanging up. Bigger settings lower the probability of random hang ups. 'Busy Detect' has to be enabled.

- 4
- 6
- 8

(Select box)

▪ **Call Progress**

Easily detect false hangups.

(ex. Yes, No, N/A)

(Option buttons)

▪ **Immediate**

Should the channel be answered immediately or should the simple switch provide dialtone, read digits, etc.

(ex. Yes, No, N/A)

(Option buttons)

Call Groups

▪ **Call Group**



A screenshot of a web form titled 'Call Groups'. It contains two input fields: 'Call Group:' and 'Pickup Group:'. Below the fields is a green checkmark icon. The text 'Call Groups' is centered below the form.

Which group is allowed to pick up incoming calls by dialing *8. The default value is empty.

(ex. 1, 1-4)

([0-9])

▪ **Pickup Group:**

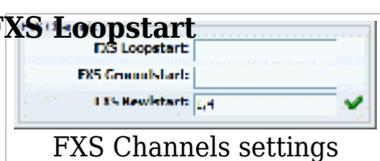
Which groups are allowed to pick up calls by dialing *8. The default value is empty.

(ex. 1, 1-4)

([0-9])

FXS Channels

▪ **FXS Loopstart**



A screenshot of a web form titled 'FXS Channels settings'. It contains three input fields: 'FXS Loopstart:', 'FXS Group/Start:', and 'FXS New/Start:'. The 'FXS New/Start:' field has a green checkmark icon next to it. The text 'FXS Channels settings' is centered below the form.

Signals the far end that it wants the dial tone by shorting the leads.
(ex. default)
([0-9])

▪ **FXS Groundstart**

Signals the far end that it wants the dial tone by grounding one of the leads.
(ex. default)
([0-9])

Locality

▪ **Country**

Country: USA ✓
National dialing code: 0,9 ✓
International dialing code: 00 ✓
E164: Yes No N/A ✓
Pass-thru mode: Yes No N/A ✓
Leave National Code: Yes No N/A ✓
Local Area Code: Yes No N/A ✓
Write dialing code: Yes No N/A ✓
Prefix: _____

Locality settings

Destination of the trunk connection
(ex. USA)
(Select box)

▪ **E164 Accepted**

Does the trunk support dialing destinations in E164 format.
(ex. Enabling this option will reformat any dialed number into the following form COUNTRY_CODE+AREA_CODE+DIALED_NUMBER. For example, if you dial 55510205, the system will dial 121255510205).
(Option buttons)

▪ **Pass-thru Mode**

If this option is enabled, the number which is dialed is passed through trunk without modification.
(ex. Yes, No, N/A)
(Option buttons)

▪ **National Dialing Code**

The national dialing code at the trunk destination.
(ex. For the USA 1. For the United Kingdom and Germany 0 etc.)
([0-9])

▪ **Leave National Code**

In some countries, the national code is stripped automatically. If set to 'Yes', the national code will not be stripped from the dialed number.

NOTE: Before settings this option to 'Yes', go to 'Settings: Servers' and enable this options as well.

(ex. John dials 121255510205. With this option enabled)
([0-9])

▪ **International Dialing Code**

International dialing code at the trunk destination.

(ex. For the USA 011. For the United Kingdom and Germany 00 etc.)
([0-9])

▪ **Local Area Code**

Add the local area code to the dialed number, if required by the service provider. (By default, the local area code is stripped when dialing)

(ex. The user dials 55510205, the local area code is 212. If the call goes through this trunk PBXware will dial 21210205).
([0-9])

▪ **Write dialing code**

Should the National and International prefix be written into the configuration files.

(ex. Enable this option if required by provider).
([0-9])

▪ **Prefix**

Value added to all dialed numbers going over the trunk.

(ex. Prefix 5, Dialed number 123, System dials 5123).
([0-9])

Other Zapata Options

▪ **ADSI (Analog Display Services Interface)**



Enable remote controlling of the screen phone with softkeys. (Only if you have ADSI compatible CPE equipment).

(ex. Yes, No, N/A)
(Option buttons)

▪ **Jitter Buffers**

Configure jitter buffers. Each one is 20ms long.
(ex. 4)
([0-9])

▪ Relax DTMF

If you are having trouble with DTMF detection, you can relax the DTMF detection parameters.
(ex. Yes, No, N/A)
(Option buttons)

▪ Fax Detect

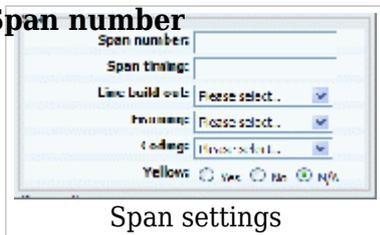
Enable fax detection
Example:

- both
- incoming
- outgoing
- no

(Select box)

Span

▪ Span number



Number of the span.
(ex. 1)
([0-9])

▪ Span timing

How to synchronize the timing devices
Example:

- 0 - do not use this span as sync source
- 1 - use as primary sync source
- 2 - set as secondary and so forth

([a-z])

▪ Line build out

Length of the last leg of the connection and is set to zero if the length is less than 133 feet

Example:

- 0 db (CSU) / 0-133 feet (DSX-1)
- 133-266 feet (DSX-1)
- 266-399 feet (DSX-1)
- 399-533 feet (DSX-1)
- 533-655 feet (DSX-1)
- -7.5db (CSU)
- -15db (CSU)
- -22.5db (CSU)

(Select box)

▪ Framing

How to communicate with the hardware at the other end of the line

Example:

- For T1: Framing is one of d4 or esf.
- For E1: Framing is one of cas or ccs.

(Select box)

▪ Coding

How to encode the communication with the other end of line hardware.

Example:

- For T1: coding is one of ami or b8zs
- For E1: coding is one of ami or hdb3 (E1 may also need crc)

(Select box)

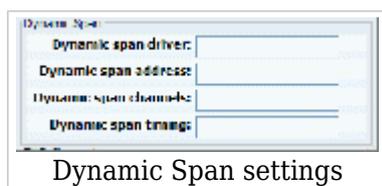
▪ Yellow

Whether the yellow alarm is transmitted when no channels are open.

(ex. Yes, No, N/A)

(Option buttons)

Dynamic Span



▪ Dynamic span driver

The name of the driver (e.g. eth).
([0-9][a-z])

- **Dynamic span address**

Driver specific address (like a MAC for eth).
([0-9][a-z])

- **Dynamic span channels**

Number of channels.
(6)

- **Dynamic span timing**

Sets timing priority, like for a normal span. Use '0' in order to not use this as a timing source, or prioritize them as primary, secondary, etc.
(0)

FXO Channels



- **FXO Loopstart**

Channel(s) are signaled using FXO Loopstart protocol.

- **FXO Groundstart**

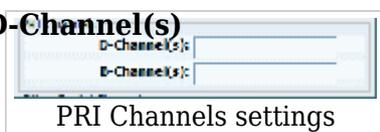
Channel(s) are signaled using FXO Groundstart protocol.

- **FXO Kewlstart**

Channel(s) are signaled using FXO Kewlstart protocol.

PRI Channels

- **D-Channel(s)**



For example, every ISDN BRI card has 1 D- (control) channel.

(ex. 1)
([0-9])

- **B-Channels(s)**

For example, every ISDN BRI card has 2 B- (data) channels.

(ex. 2)
([0-9])

Custom Trunks



Custom trunks are used in rare cases when PBXwares trunk setup features are not providing enough flexibility for some provider configurations. In such cases one will create a custom trunk and then change the dialplan in the *trunks-in.conf* file.

For information on how to modify asterisk dialplan, please check sites like www.voip-info.org

Add Custom Trunk

When you click on *Add Custom Trunk* button, you will get following screen:



It is the same screen when adding or editing the current custom trunk.

- **Trunk**

Name of the trunk which will be visible in PBXware GUI.

(ex. Global Trunk)
([0-9][a-z])

- **Context**

Context by which the system sees this trunk and which will be used when editing the dialplan.

(ex. Set *gl_trunk* for example, and when sending calls to this trunk, you would use this to refer to the given trunk).

([0-9][a-z])

Next -> 5.DID

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