Objects diagram for pjsip.conf and pjsip_wizard.conf (Asterisk certified/13.8-cert1):

Access Control Lists

acl.conf

Access Control List.

res_pjsip_acl.so

Object type: acl

+acl	
+contact_acl	List of IP ACL section names in acl.conf.
+deny	List of Contact ACL section names in acl.conf.
+permit	List of IP addresses to deny access from.
+contact_deny	List of IP addresses to permit access from.
+contact_permit	List of Contact header addresses to deny.
	List of Contact header addresses to permit.

Dialplan logic.

extensions.conf

Configures res_pjsip transport layer interaction.

res_pjsip.so

Object type: transport

+protocol	Protocol to use for SIP traffic.
+bind	IP Address and optional port to bind to for this transport.
+domain	Domain the transport comes from.
+cert_file	Certificate file for endpoint (TLS ONLY).
+priv_key_file	Will read *.crt or .pem file but only uses cert, a .key file must be specified via priv_key_file.
+cipher	Private key file (TLS ONLY).
+method	Preferred cryptography cipher names (TLS ONLY).
+ca_list_file	Method of SSL transport (TLS ONLY).
+ca_list_path	File containing a list of certificates to read (TLS ONLY).
+require_client_cert	Path to directory containing certificates to read (TLS ONLY).
+verify_client	Require client certificate (TLS ONLY).
+verify_server	Require verification of client certificate (TLS ONLY).
+local_net	Require verification of server certificate (TLS ONLY).
+external_media_address	Network to consider local (used for NAT purposes).
+external_signaling_address	External IP address to use in RTP handling.
+external_signaling_port	External address for SIP signaling.
+async_operations	External port for SIP signalling.
+password	Number of simultaneous Asynchronous Operations.
+tos	Password required for transport.
+cos	Enable TOS for the signalling sent over this transport.
+websocket_write_timeout	Enable COS for the signalling sent over this transport.
+allow_reload	The timeout (in milliseconds) to set on WebSocket connections.
	Allow this transport to be reloaded.

Configures core SIP functionality related to SIP endpoints.

res_pjsip.so

Object type: endpoint

+transport	Desired transport configuration.
+context	Dialplan context for inbound sessions.
+auth	Authentication Object(s) associated with the endpoint.
+outbound_auth	Authentication object used for outbound requests.
+aors	AOR(s) to be used with the endpoint.
+disallow	Media codec(s) to disallow.
+allow	Media Codec(s) to allow.
+direct_media	Determines whether media may flow directly between endpoints.
+force_rport	Force use of return port. Help the far-end NAT/firewall.
+rtp_symmetric	Enforce that RTP must be symmetric. Help the far-end NAT/firewall.
+rewrite_contact	Allow Contact header to be rewritten with the source IP address-port. Necessary if endpoint does not know/register public ip:port.
+ice_support	Enable the ICE mechanism to help traverse NAT.
+media_encryption	Determines whether res_pjsip will use and enforce usage of media encryption for this endpoint.
+media_encryption_optimistic	Determines whether encryption should be used if possible but does not terminate the session if not achieved.
+rtp_ipv6	Allow use of IPv6 for RTP traffic.
+aggregate_mwi	Condense MWI notifications into a single NOTIFY.
+mailboxes	NOTIFY the endpoint when state changes for any of the specified mailboxes.
+mwi_from_user	For endpoints that cannot SUBSCRIBE for MWI, This enables unsolicited MWI NOTIFYs to the endpoint.
+allow_subscribe	Username to use in From header for unsolicited MWI NOTIFYs to this endpoint.
+sub_min_expiry	Determines if endpoint is allowed to initiate subscriptions with Asterisk.
+100rel	The minimum allowed expiry time for subscriptions initiated by the endpoint.
+callerid	Allow support for RFC3262 provisional ACK tags.
+callerid_privacy	CallerID information for the endpoint.
+callerid_tag	Default privacy level.
+direct_media_glare_mitigation	Internal id_tag for the endpoint.
+direct_media_method	Mitigation of direct media (re)INVITE glare.
+connected_line_method	Direct Media method type.
+disable_direct_media_on_nat	connected line method type.
+dtmf_mode	Disable direct media session refreshes when NAT obstructs the media session.
+media_address	DTMF mode.
+bind_rtp_to_media_address	IP address used in SDP for media handling.
+identify_by	Bind the RTP session to the media_address. This causes all RTP packets to be sent from the specified address.
+redirect_method	Way(s) for Endpoint to be identified.
+moh_suggest	How redirects received from an endpoint are handled.
+outbound_proxy	Default Music On Hold class.
+send_diversion	Proxy through which to send requests. A full SIP URI must be provided.
+send_pai	Send the Diversion header conveying the diversion information to the called user agent.
+send_rpid	Send the P-Asserted-Identity header.
+rpid_immediate	Send the Remote-Party-ID header.
+timers_min_se	Send connected line updates on unanswered incoming calls immediately.
+timers	Minimum session timers expiration period.
+timers_sess_expires	Session timers for SIP packets.
+trust_id_inbound	Maximum session timer expiration period.
+trust_id_outbound	Accept identification information received from this endpoint.
+use_ptime	Send private identification details to the endpoint.
+use_avpf	Use Endpoint's requested packetization interval.
+force_avp	Determines whether res_pjsip will use and enforce usage of AVPF for this endpoint.
+media_use_received_transport	Determines whether res_pjsip will use and enforce usage of AVP, regardless of the RTP profile in use for this endpoint.
g726_non_standard	Determines whether res_pjsip will use the media transport received in the offer SDP in the corresponding answer SDP.
+inband_progress	Force g.726 to use AAL2 packing order when negotiating g.726 audio.
+call_group	Determines whether chan_pjsip will indicate ringing using inband progress.
+pickup_group	The numeric pickup groups for a channel.
+named_call_group	The numeric pickup groups that a channel can pickup.
+named_pickup_group	The named pickup groups for a channel.
+device_state_busy_at	The named pickup groups that a channel can pickup.
+t38_udptl	The number of in use channels which will cause busy to be returned as device state.
+t38_udptl_ec	Whether T.38 UDPTL support is enabled or not.
+t38_udptl_maxdatagram	T.38 UDPTL error correction method.
+fax_detect	T.38 UDPTL maximum datagram size.
+t38_udptl_nat	Whether CNG tone detection is enabled.
+t38_udptl_ipv6	Whether NAT support is enabled on UDPTL sessions.
+tone_zone	Whether IPv6 is used for UDPTL Sessions.
+language	Set which country's indications to use for channels created for this endpoint.
+one_touch_recording	Set the default language to use for channels created for this endpoint.
+record_on_feature	Determines whether one touch recording is allowed for this endpoint.
+record_off_feature	The feature to enact when one touch recording is turned on.
+rtp_engine	The feature to enact when one touch recording is turned off.
+allow_transfer	Name of the RTP engine to use for channels created for this endpoint.
+user_eq_phone	Determines whether SIP REFER transfers are allowed for this endpoint.
+sdp_owner	Determines whether a user=phone parameter is placed into the request URI if the user is determined to be a phone number.
+sdp_session	String placed as the username portion of an SDP origin (o=) line.
+tos_audio	String used for the SDP session (s=) line.
+tos_video	DSCP TOS bits for audio streams.
+cos_audio	DSCP TOS bits for video streams.
+cos_video	Priority for audio streams.
+from_user	Priority for video streams.
+from_domain	Username to use in From header for requests to this endpoint.
+dtls_verify	Domain to user in From header for requests to this endpoint.
+dtls_rekey	Verify that the provided peer certificate is valid.
+dtls_cert_file	Interval at which to renegotiate the TLS session and rekey the SRTP session.
+dtls_private_key	Path to certificate file to present to peer.
+dtls_cipher	Path to private key for certificate file.
+dtls_ca_file	Cipher to use for DTLS negotiation.
+dtls_ca_path	Path to certificate authority certificate.
+dtls_setup	Path to a directory containing certificate authority certificates.
+dtls_fingerprint	Whether we are willing to accept connections, connect to the other party, or both.
+srtp_tag_32	Type of hash to use for the DTLS fingerprint in the SDP.
+set_var	Determines whether 32 byte tags should be used instead of 80 byte tags.
+message_context	Variable set on a channel involving the endpoint. For multiple channel variables specify multiple 'set_var'(s).
+accountcode	Context to route incoming MESSAGE requests to.
+rtp_keepalive	An accountcode to set automatically on any channels created for this endpoint.
+rtp_timeout	Number of seconds between RTP comfort noise keepalive packets. This setting is useful for ensuring that holes in NATs and firewalls are kept open throughout a call.
+rtp_timeout_hold	Maximum number of seconds without receiving RTP (while off hold) before terminating call.
	Maximum number of seconds without receiving RTP (while on hold) before terminating call.

Required to be defined to have USERNAME and SECRET automatically set on phone provisioning.

Stores inbound or outbound authentication credentials for use by trunks, endpoints, registrations.

res_pjsip.so

Object type: auth

+auth_type	Authentication type.
+nonce_lifetime	Lifetime of a nonce associated with this authentication config.
+md5_cred	MD5 Hash used for authentication.
+username	Username to use for account.
+password	Plain-Text password used for authentication.
+realm	SIP realm for endpoint.

The configuration for a location of an endpoint.

res_pjsip.so

Object type: aor

+contact	Permanent contacts assigned to AoR.
+max_contacts	Maximum number of contacts that can bind to an AoR.
+remove_existing	Determines whether new contacts replace existing ones.
+default_expiration	Default expiration time in seconds for contacts that are dynamically bound to an AoR.
+minimum_expiration	Minimum keep alive time for an AoR.
+maximum_expiration	Maximum time to keep an AoR.
+qualify_frequency	Interval at which to qualify an AoR.
+qualify_timeout	Timeout (in fractional seconds) for qualify.
+authenticate_qualify	Authenticates a qualify request if needed.
+outbound_proxy	Outbound proxy used when sending OPTIONS request.
+mailboxes	Allow subscriptions for the specified mailbox(es). For endpoints that SUBSCRIBE for MWI.
+support_path	Enables Path support for REGISTER requests and Route support for other requests.

Contacts are a way to hide SIP URIs from the dialplan directly. They are also used to make a group of contactable parties when in use with 'AoR' lists.

res_pjsip.so

Object type: contact

+uri	SIP URI to contact peer.
+expiration_time	Time to keep alive a contact.
+qualify_frequency	Interval at which to qualify a contact.
+qualify_timeout	Timeout for qualify.
+outbound_proxy	Outbound proxy used when sending OPTIONS request.
+path	Stored Path vector for use in Route headers on outgoing requests.
+user_agent	User-Agent header from registration.

Identifies endpoints via source IP address.

res_pjsip_endpoint_identifier_ip.so

Object type: identify

+endpoint	Name of Endpoint.
+match	IP addresses or networks to match against.

Allows you to specify an alias for a domain. If the domain on a session is not found to match an AoR then this object is used to see if we have an alias for the AoR to which the endpoint is binding.

res_pjsip.so

Object type: domain_alias

+domain	Domain to be aliased.

The configuration for outbound registration.

res_pjsip_outbound_registration.so

Object type: registration

+transport	Transport used for outbound authentication.
+outbound_auth	Authentication object to be used for outbound registrations.
+outbound_proxy	Outbound Proxy used to send registrations.
+endpoint	Endpoint to use for incoming related calls. When line support is enabled this configured endpoint name is used for incoming calls that are related to the outbound registration.
+server_uri	SIP URI of the server to register against.
+client_uri	Client URI to use when attempting outbound registration.
+contact_user	Contact User to use in request.
+max_retries	Maximum number of registration attempts.
+retry_interval	Interval in seconds between retries if outbound registration is unsuccessful.
+forbidden_retry_interval	Interval used when receiving a 403 Forbidden response.
+fatal_retry_interval	Interval used when receiving a Fatal response. A Fatal response is any permanent failure (non-temporary 4xx, 5xx, 6xx) response received from the registrar. It also set the 'forbidden_retry_interval' takes precedence over this one when a 403 is received. Also, if 'auth_rejection_permanent' equals 'yes' a 401 and 407 become subject to this retry interval.
+auth_rejection_permanent	Determines whether failed authentication challenges are treated as permanent failures.
+expiration	Expiration time for registrations in seconds.
+line	Whether to add a 'line' parameter to the Contact for inbound call matching.
+support_path	Enables Path support for outbound REGISTER requests.

Support used for RFC 4662 Resource Lists, allowing Asterisk to act as a Resource List Server (RLS). Resource lists can contain either message-summary or presence events, can be composed of specific resources that provide the event, or other resource lists. This allows for an endpoint to, through a single subscription, subscribe to the states of multiple resources.

res_pjsip_pubsub.so

Object type: resource_list

+event	The SIP event package that the list resource belong to.
+list_item	The name of a resource to report state on.
+full_state	Indicates if the entire list's state should be sent out.
+notification_batch_interval	Time Asterisk should wait, in milliseconds, before sending notifications.

Options that apply to the SIP stack as well as other system-wide settings.

res_pjsip.so

Object type: system

+timer_t1	Set transaction timer T1 value (milliseconds).
+timer_b	Set transaction timer B value (milliseconds).
+compact_headers	Use the short forms of common SIP header names.
+threadpool_initial_size	Initial number of threads in the res_pjsip threadpool.
+threadpool_auto_increment	The amount by which the number of threads is incremented when necessary.
+threadpool_idle_timeout	Number of seconds before an idle thread should be disposed of.
+threadpool_max_size	Maximum number of threads in the res_pjsip threadpool. A value of 0 indicates no maximum.
+disable_tcp_switch	Disable automatic switching from UDP to TCP transports if outgoing request is too large. See RFC 3261 section 18.1.1.

Provides the mechanisms for sending PUBLISH requests for specific event packages to another SIP User Agent.

res_pjsip_outbound_publish.so

Object type: outbound-publish

+transport	
+expiration	Expiration time for publications in seconds.
+outbound_auth	Authentication object to be used for outbound publishes.
+outbound_proxy	SIP URI of the outbound proxy used to send publishes.
+server_uri	SIP URI of the server and entity to publish to.
+from_uri	SIP URI to use in the From header.
+to_uri	SIP URI to use in the To header.
+event	Event type of the PUBLISH.
+max_auth_attempts	Maximum number of authentication attempts before stopping the publication.

Contains variables for autoprovisioning each user.

res_pjsip_phoneprov_provider.so

Object type: phoneprov

+endpoint	The endpoint from which variables will be retrieved.
+PROFILE	The phoneprov.conf profile to use for this user.
+MAC	The MAC address for this user.
+OTHERVAR	Any other name value pair to be used in templates. Common variables include LINE, LINEKEYS, etc.
+*	Other name/value pairs to be passed through for use in templates.

Inbound publications support. This configures to accept PUBLISH requests from a particular resource. Which events are accepted is constructed dynamically.

res_pjsip_pubsub.so

Object type: inbound-publication

+endpoint	Optional name of an endpoint that is only allowed to publish to this config
+event_*	Event type that links to an endpoint.

Adds support for PUBLISH requests of Asterisk information to other Asterisk servers. This module is intended only for Asterisk to Asterisk exchanges of information. Currently, this includes both mailbox state and device state information.

res_pjsip_publish_asterisk.so

Object type: asterisk-publication

+devicestate_publish	Optional name of a publish item that can be used to publish a request for full device state information.
+mailboxstate_publish	Optional name of a publish item that can be used to publish a request for full mailbox state information.
+device_state	Whether we should permit incoming device state events.
+device_state_filter	Optional regular expression used to filter what devices we accept events for.
+mailbox_state	Whether we should permit incoming mailbox state events.
+mailbox_state_filter	Optional regular expression used to filter what mailboxes we accept events for.

Persists SIP subscriptions so they survive restarts.

res_pjsip_pubsub.so

Object type: subscription_persistence

+packet	Entire SIP SUBSCRIBE packet that created the subscription.
+src_name	The source address of the subscription.
+src_port	The source port of the subscription.
+transport_key	The type of transport the subscription was received on.
+local_name	The local address the subscription was received on.
+local_port	The local port the subscription was received on.
+cseq	The sequence number of the next NOTIFY to be sent.
+tag	The local tag of the dialog for the subscription.
+endpoint	The name of the endpoint that subscribed.
+expires	The time at which the subscription expires.

pjsip_wizard.conf:

Provides config wizard. This allows creation of common PJSIP configuration scenarios without having to specify individual endpoint, aor, auth, identify and registration objects.

res_pjsip_config_wizard.so

Object type: wizard

+transport	The name of a transport to use for this object.
+remote_hosts	List of remote hosts.
+sends_auth	Send outbound authentication to remote hosts.
+accepts_auth	Accept incoming authentication from remote hosts.
+sends_registrations	Send outbound registrations to remote hosts.
+accepts_registrations	Accept inbound registration from remote hosts.
+has_phoneprov	Create a phoneprov object for this endpoint.
+server_uri_pattern	A pattern to use for constructing outbound registration server_uris.
+client_uri_pattern	A pattern to use for constructing outbound registration client_uris.
+contact_pattern	A pattern to use for constructing outbound contact uris.
+has_hint	Create hint and optionally a default application.
+hint_context	The context in which to place hints.
+hint_exten	Extension to map a PJSIP hint to.
+hint_application	Application to call when 'hint_exten' is dialed.
+endpoint/*	Variables to be passed directly to the endpoint.
+aor/*	Variables to be passed directly to the aor.
+inbound_auth/*	Variables to be passed directly to the inbound auth.
+outbound_auth/*	Variables to be passed directly to the outbound auth.
+identify/*	Variables to be passed directly to the identify.
+registration/*	Variables to be passed directly to the outbound registrations.
+phoneprov/*	Variables to be passed directly to the phoneprov object.