

Overview of Session Management for VoIP Services

When the SAE activates a service session, it authorizes the session with authorization plug-ins; it may use the admission control plug-in (ACP) to perform call admission control and allocate bandwidth; and it installs the policy required for the service on a JUNOS interface.

VoIP and multimedia service sessions are typically established in multiple phases that require changes to installed policies and authorized bandwidth while the service session remains active. To support VoIP sessions, the SAE allows changes to active service sessions. These changes include:

- **Controlled bandwidth.** If bandwidth demand increases, the authorization plug-in must authorize the change.
- **Policy parameters.** Only parameter substitution values can be changed. Policy parameters can include classifiers, such as destination address and port, and actions, such as rate-limit profiles.
- **Session and idle timeouts.** All attributes that can be set for initial service activation can be set for service session modifications.

Accounting and Tracking

Accounting information is preserved across service session changes. Accounting information for a complete service session includes the sum of counters for all service session segments.

When the ACP receives an interim update request, it compares the upstream and downstream bandwidth in the request with the current values. If the bandwidth has changed, ACP modifies its counters based on the difference between the current and new values.

Tracking plug-ins are informed of service session changes through an interim update message. The interim update is sent even if regular interim updates are disabled. If the controlled bandwidth changes, the interim update message contains the new bandwidth settings.

VoIP Call Setup

Initial setup of a VoIP call requires changes to bandwidth and to the endpoint address during call setup. The setup sequence for a VoIP call can follow this pattern:

1. The subscriber attempts to establish a call.
2. The gatekeeper (or Session Initiation Protocol [SIP] proxy) performs local admission control.
3. The gatekeeper allocates a Codec for the call; for example, 64 kbps.
4. The gatekeeper activates the VoIP service on the SAE with 64 kbps bandwidth and a destination address of unknown.
5. The SAE performs admission control, activates a service session, and installs policies on the router.

6. The gatekeeper negotiates call parameters with the remote endpoint.
7. The gatekeeper modifies the VoIP service with negotiated parameters; for example, 32 kbps, destination address 10.10.3.4, and UDP port 5678.
8. The SAE creates new policies that reflect changes to the traffic classifier and rate-limit profile, and then removes the existing policies from the router and installs the new policies.
9. The SAE sends interim updates to the ACP and tracking plug-ins.

- Related Topics**
- Overview of Global and Local Parameters
 - For information about configuring and managing policies, see the *SRC-PE Services and Policies Guide*
 - Configuring Policies and Services for VoIP
 - Activating VoIP Services for Assigned IP Subscribers