CONNECT. CONTROL. SEE CLEARLY.

TeleFinity SBC

SIP session border controller

Connect configured SIP peers. Control calls and media.
Manage operations from one web portal.

A CONTROLLED CONNECTION BETWEEN YOUR SIP ENVIRONMENTS



Two-way SIP signaling. Media can be direct or optionally relayed through TeleFinity SBC.

Built for controlled SIP interconnection.

Guided SIP connectivity

Create paired carrier and application routes with endpoint validation, shared-source DID checks and review before saving.

Flexible call routing

Use exact or prefix DID routes, primary or round-robin targets, transport selection and optional outgoing SIP domains.

Admission and destination control

Apply source rules, digest authentication, caller checks, concurrency caps, CPS limits and international-call permissions.

Controlled media paths

Anchor RTP when needed, advertise network-specific SDP addresses and bridge the supported RTP/SDES-SRTP profiles.

Live operational changes

Apply supported settings to new calls while existing calls retain their configuration. Trunk revocation ends its calls.

Integrated call diagnostics

Inspect active calls, per-leg RTP totals, retained SIP messages and completed call records through the authenticated portal.

Guided administration | **Defined calling policies** | **Correlated call diagnostics**

CAPABILITIES AT A GLANCE

Specifications & deployment

Plan the connection, calling policies and operational requirements together.

CAPABILITY	SUPPORTED FUNCTIONALITY
SIP signaling	UDP, TCP and TLS; separate inbound and outbound legs; call setup, answer, cancellation and release.
Routing	IPv4 source IP/CIDR; optional port/transport; shared-source DID checks; exact/prefix rules; primary and round-robin targets.
Capacity controls	Trunk, shared-connection and target concurrency; CPS and per-IP rates; global overload admission. Ringing calls count.
Call features	Supported in-dialog signaling, hold/resume, optional session timers and duration limits. Qualify transfer and peer compatibility.
Media paths	Optional RTP relay; symmetric RTP learning; network-specific IPv4 SDP addresses; separate RTCP or negotiated multiplexing.
Media protection	SDES-SRTP AES_CM_128_HMAC_SHA1_80; distinct leg keys; configured RTP/SRTP conversion. Requires TLS, relay and valid media addresses.
Codec handling	Audio/video codec policies; relay without live transcoding. Both peers need a permitted common codec.
Announcements	PCM WAV playback to both legs: negotiated PCMA, PCMU, G.729, Opus, AMR-NB or AMR-WB. Confirm codec availability.
Access & security	Source allow/deny lists; allowed methods; digest replay and abuse controls; caller/destination rules; certificate validation and refresh.
Administration	Guided setup and web forms; password plus email-code sign-in. All portal accounts are administrators. Working SMTP is required.
Diagnostics & records	Active-call snapshots; per-leg RTP RX/TX totals; searchable SIP history; completed CDRs and CSV records.
Deployment	Windows or Linux hosting; protected remote portal access. Confirm connectivity, storage, media ports, certificates and email delivery.

DEPLOY WITH A CLEAR SCOPE

Qualify carrier interoperability and sustained capacity on the target deployment. Protect SIP history, call records and backups as sensitive operational data.

Scope. IPv4 SDP media. WebRTC/ICE/DTLS-SRTP, T.38 relay, general registration, live transcoding and replicated call failover are outside this scope. Live edits apply to new calls; trunk revocation, source-policy changes, maintenance and restarts can affect active calls.

Plan your SIP interconnection.

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Feature availability and interoperability depend on configuration and deployment qualification. No measured capacity, availability or certification is implied.