Talking SIP™

The Voice of the Next-Generation Network™

Key Features

Market Ready. The Trunking Module is ready for turnkey deployment. This enhanced service offers a full-featured and scalable solution that delivers a market ready and highly competitive SIP Trunking offering for next-generation service providers.

Fully-Integrated. The Trunking Module is seamlessly integrated into the other enhanced services of Talkingg SIP™ to help reduce customer churn by providing a consistent and comprehensive user experience.

Drives Revenue. In the highly competitive long distance and interconnection market the Trunking Module offers a tremendous opportunity to extend the network's footprint to drive additional revenue.

Scalable. The Trunking Module is an extremely efficient and scalable Class 4 SIP Trunking switching system that allows calls to be aggregated to a centralized or distributed Talking SIP™ network while capitalizing on a common back end network for termination that ensures all network elements are fully and effectively utilized.

Trunking Module

The Trunking Module is an optional module that brings SIP Trunking capabilities to Talking SIP allowing next-generation Internet Telephony Service Providers (ITSPs) the opportunity to provide Class 4 switching and termination services to the growing enterprise market. Enterprise customers are now realizing the phenomenal cost-savings, flexibility, efficiency, and reliability that SIP Trunking offers their businesses and Talking SIP™'s Trunking Module lets next-generation service providers capitalize on this market opportunity.

The Trunking Module offers real-time authentication, authorization, routing and billing to allow providers the flexibility to service a broad range of customers with the piece of mind that real-time safeguards are in place to combat fraud, eliminate receivables and minimize exposure with a solution that is field-proven, reliable, and scalable. With creative rules-based billing and support for third-party least cost routing (LCR) services the Trunking Module allows providers to create highly competitive plans that attract customers while being able to reduce costs and increase profit margins. The Trunking Module also supports a release link trunking option to utilize Talking SIP™'s real-time authentication to validate the call and then transfer to the call to another endpoint for actual termination.

The Trunking Module can be seamlessly integrated into Talking SIP™'s Class 5 services like conferencing, call queueing, IVR, automated call distribution (ACD), international callback, prepaid calling card and one-number locator services to drive additional sources of revenue to network as well as increase customer loyalty.

Powering leading providers' networks around the world, Talking SIP™ is a robust, scalable, market-proven solution, and it is one of the easiest to install, turn up, and manage. Contact us today to learn more about the Trunking Module and why Talking SIP™ is the voice of the next-generation network.

IVR Technologies

Talking SIP™'s innovative applications and integrated real-time billing engine provide unmatched flexibility for designing high margin integrated services that drive revenue to the next-generation network while increasing profitability and customer loyalty.

Talking SIP™'s Trunking Module allows next-generation Internet Telephony Service Providers (ITSPs) to capitalize on the growing SIP Trunking market and provide cost-effective, high-quality and reliable termination services to enterprise customers and their PBXs. By using virtual connections rather than dedicated circuits like T1s and ISDN-PRIs enterprises can realize significant savings and efficiencies on their local and long distance calling while ITSPs can extend their network's footprint to drive revenue, increase profit margins and reduce customer churn.

The following is a list of some of the features and functionality of the Talking SIP™'s Trunking Module:

Performance and Scalability

- Supports up to 2,000 sessions (4,000 call legs) in a single server or blade (approx. 24,000,000 minutes per month based on a 5 minute average call duration)
- Supports 10,000+ sessions by easily and seamlessly integrating multiple server chassis/blades in a unified network
- Ability to mix multiple services on a single instance or dedicate instances to specific services based on network/business requirements
- Provides simultaneous access to services for subscribers located anywhere in the network
- Fully integrated application, billing, registration, location and media server for streamlined deployment and management
- Completely software based without any costly DSP resources or third-party hardware required
- No additional licensing cost to deploy additional servers in the network for centralized, decentralized or hybrid networks
- Centralized license pool to ensure the most efficient allocation of globally deployed network resources

Protocol Support

- RFC 3261 SIP: Session Initiation Protocol
- RFC 2327 SDP: Session Description Protocol
- RFC 3264 An Offer/Answer Model with the Session Description Protocol
- RFC 3550 RTP: A Transport Protocol for Real-Time Applications

Open Architecture

- Microsoft SQL database for open and complete access for analysis and reporting purposes

Multi-Vendor Device Support

- Gateways
- Softswitches
- Session Boarder Controllers
- Proxy Servers
- Softphones, SIP phones, and IADs

Billing and Costing

- Integrated pre/postpaid real-time billing and costing engine with call cut off
- Three (3) additional billing models and credit caps for corporations, groups, and multi-level marketing (end user, reseller, wholesaler, etc.)
- Blocked, flat rate, single rate and multi-tier rating (vary rate based on call duration)
- Rating by ANI, NPA, NPA+NXX, DNIS, Account, Account usage to date, Device, time of day, day of week, date/time range and/or destination
- Costing by ANI, NPA, NPA+NXX, DNIS, time of day, day of week, date/time range and/or destination
- Surcharging by DNIS, ANI, Info Digits, Account, Account usage to date, Device, time of day, day of week, date/time range and/or destination
- Supports a wide range of disconnection fees
- Ability to vary the scale of a minute at different intervals of a call.

Reporting

- Centralized Reporting Engine to allow reports to be managed from within the database so all additions, updates and deletions are immediately reflected on the agents' screen
- Includes over 136 pre-configured, customizable reports
- Optional fully integrated Report Designer

Customer Service Management

- Multi-level password protected access
- Audit logging of customer account transactions
- Account usage credit/debit with audit trail
- Billing and Invoice statement generation

Routing

- Support for third-party least cost routing (LCR)
- Real-time call processing, call cut off and routing
- Full digit manipulation of the inbound ANI and origin/carrier codes
- Full digit manipulation of the inbound DNIS
- Full digit manipulation to the terminating device with CLI/ANI manipulation
- Routing and rating by time of day/holidays/ CLI/number dialed

Custom Call Scripting

- Ability to deploy multiple instances of the same enhanced services module
- Ability to customize call flow

High Availability

- Centralized management over load balanced and/or redundant communication servers
- Redundant License Manager architecture and support for database mirroring

System Security

- Up to 40 digit account numbers generated in fully random or most efficient mode
- Additional account reference methods such as an alias, a sequence or a reference number to reduce the need for account number disclosure
- Ability to prevent or define simultaneous account use
- CLI logging of invalid authentication attempts for risk management
- Authentication via variable length account, account+pin, SIP Credentials, Remote-Party-ID, ANI, DNIS, Carrier Code or IP address

Opportunity

SIP trunking for toll bypass delivers great value, which has customers latching onto it like fly paper. Today 13% of businesses use SIP for 100% of their WAN traffic, and 42% of businesses will send all their toll traffic on SIP trunks by 2018, according to a new SIP study by The Eastern Management Group.Provider VoIP and IMS, Infonetics Research.

Contact us today to find out how to drive additional revenue to your network and increase subscriber loyalty and retention through Talking SIP™'s Trunking Module Module.

For More Information

Please contact us to find out more information about our products, receive a quotation or locate a value added reseller in your region:

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