Talking SIP™

The Voice of the Next-Generation Network™

Key Features

Fully-Integrated. Talking SIP™ consolidates an application, media, billing, registration and location server into a single, fully-integrated solution.

Cost-Effective. Talking SIP™ utilizes standard off-the-shelf Intel-based hardware resulting in a cost-effective solution devoid of any per-server licensing cost.

Future-Proof. Talking SIP™ has a segregated architecture to allow new and customized enhanced service applications to be readily developed and deployed on-the-fly without caller interruption.

Drives Revenue. Through mature, widely deployed enhanced service applications, Talking SIP™ allows providers to develop customer tailored applications that are seamlessly linked together to drive revenue to the network while reducing customer churn.

Scalable. Talking SIP™ supports a wide range of session densities, scaling from a single T1/E1 to thousands of sessions anchored by a centralized database.

Extensible. Talking SIP™ utilizes an open Microsoft SQL Server database to allow full access to subscriber and system information for reporting and third-party integration.

Our flagship product, Talking SIP™, provides next-generation networks with innovative and in-demand voice guided enhanced services, interactive voice response, and media streaming along with real-time prepaid and postpaid billing. Talking SIP™ is a turnkey, software-based and fully-integrated application, media, and real-time billing solution for SIP-based service providers and carriers.

Using open standards and off-the-shelf Intel-based hardware, a centralized Microsoft SQL Server Database, and a tiered secure management console, Talking SIP™ supports a wide range of network densities in both centralized and decentralized configurations without per node licensing costs.

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.

Available enhanced service applications and options for Talking SIP™ include:

- Prepaid/Postpaid Calling Card
- Voice over Broadband (VoBB) with Class 5 Features
- Callback and Click to Call
- Reminder/Wake-Up Services
- Voucher Recharge
- Credit Card Subscription and Recharge
- Toll-Free/Toll/DID/Local Access Termination Services
- Registration and Location Services
- Find Me/Follow Me/Simultaneous Ring Services
- End User Web Interface

Contact us today to learn more about Talking SIP™ and why it is the voice of the next-generation network.

IVR Technologies

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Performance and Scalability

- Supports over 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 server or dedicate servers 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
- Supports the leading voice codecs (all compression and decompression are performed in the edge device (e.g. IAD, IP Phone, Gateway, etc.)
- 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 2976 The SIP INFO Method
- RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- 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
- RFC 3725 Best Practices for Third Party Call Control (3PCC)

DTMF Support

- DTMF via RFC 2833
- DTMF via SIP INFO Method (Cisco Implementation)
- DTMF via SIP NOTIFY Method (Cisco Implementation)

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.)
- Multiple low water mark warnings
- 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 with or without caller disclosure
- 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 127 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

- 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
- Customizable call flows, branding and user menus on a group or individual basis
- Customizable language selection menus

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

High Availability

- Uses state-of-the-art redundancy and load balancing licensing technology
- Supports database mirroring and fail-over solutions for data redundancy in mission critical environments
- Centralized management over one of more Communication servers

Open Architecture

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

Extensibility

- Separate application and call processing engines to allow applications to be readily created or modified in response to market demand and then remotely deployed to a live server without any caller interruption

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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