Talking SIP™

The Voice of the Next-Generation Network™

Key Features

Market Ready. The enhanced services provided by Talking SIP™ are mature, field-proven, and in-demand to reduce time to market while driving revenue to the network.

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

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 help reduce 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.

Enhanced Services

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.

With its advanced product architecture Talking SIP™ satisfies multiple network requirements in a single, integrated solution. Networks in the past required a separate application server, media server, registration/location server and billing server but today it can all be achieved by Talking SIP™.

With leading, highly flexible and extensible enhanced services that are mature and field proven Talking SIP™ reduces time to market while providing in-demand services that drive revenue to the network while reducing customer churn. Existing enhanced services can be readily modified and wholly new ones created to maintain competitive advantage while being able to offer tailored and differentiated services to the market.

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 Talking SIP™ and why it is the voice of the next-generation network.

IVR Technologies

800/900 Termination Module
The 800/888/877/900 Termination Module allows the termination of toll or toll-free calls to a local phone number or SIP device. This module allows prompt and efficient redirection of toll-free numbers to various termination numbers without the traditional delay and overhead. The 800 Termination module also provides full billing and accounting of all traffic that passes through each toll-free number.

Debit Card Module
The Debit Card Module provides calling card functionality to Talking SIP™. Prepaid and postpaid accounts are authenticated and authorized manually by DTMF key presses or automatically based on the caller's ANI, IP address, Remote-Party-ID or SIP credentials and then prompted for the desired destination. The caller's destination is rated and routed, where Talking SIP™ then connects the caller through to the desired destination. The call is tracked to detect when the caller wishes to make another call without having to be re-authenticated and if the account balance is depleted during the call it will be automatically disconnected. Once the call is disconnected the account balance is updated, a call detail record written and a billing entry created.

Service Charge Module
The Service Charge Module is used to process Billing Packages (Billing Packages are used to automatically perform certain billing or replenishment operations on accounts) to allow certain charges (e.g. first use charges, immediate charges and reoccurring monthly charges) to be applied to accounts. This module also supports the optional Credit Card Verification Server module, to allow charges to be automatically billed to account holders' credit cards.

Tandem Module
Tandem Switching allows Talking SIP™ to provide class 4 tandem switching where the platform answers calls and authenticates, rates, routes and then rapidly terminates the call through the platform. The Tandem module utilizes Talking SIP™'s billing engine to maintain billing and call detail records for each call on a prepaid or postpaid basis. Account authorization can be performed based on a specified account, DNIS, ANI, Remote-Party-ID, SIP credentials or IP address. This module also supports Class 5 features like Busy Call Return, Call Blocking, Last Caller and Last Called Party.

Voucher Recharge Module
The Voucher Module allows service providers and telecom operators the option of providing their customers with the flexibility to transfer balances from one account/voucher to another account. This module can be configured as a stand-alone application or integrated with the Debit Card Module whereby the caller is provided with the option of transferring a balance whenever he/she logs into the system. The Voucher Recharge module helps to increase customer retention by providing customers with the option of being able to transfer balances to a familiar account number as opposed to having to memorize a new account number and/or reprogram telephone speed dials whenever they wish to extend their credit.

Optional Modules

Advertising Module
The Advertising Module is used in conjunction with the Debit Module to allow advertising/content messages to be streamed to callers, using a third party Advertising (or In-Call Media) Server like Voodoovox (www.voodoovox.com), as an additional source of revenue and/or to subsidize calling for end subscribers to encourage usage and/or to help provider's remain competitive.

Callback Module
The Callback Module provides callback services to Talking SIP™. A caller dials a shared or dedicated access number and then hangs up after a certain number of rings before incurring any toll charges. The system authenticates the caller based on the incoming ANI or the dialed number and then calls the caller back at a preset telephone number or from where the initiating call was placed. Callback requests may be configured by a multitude of ways such as an inbound call, a web page, an e-mail message, an SMS message, a click to call link, as well as by a simple but secure API.

Conference Module
The Conference Module brings reservation-based and reservation-less conferencing to Talking SIP™. Conferencing helps to bring friends, family, colleagues, partners, suppliers and customers together by bridging time zones and geographic boundaries in a timely, natural and convenient manner. When face-to-face collaboration and communication is just not feasible, Conferencing provides a great alternative that drives inbound revenue and increases profit margins while cultivating customer loyalty and increasing subscriber retention.

Credit Card Recharge Module
The Credit Card Recharge Module allows callers the flexibility and convenience of being able to recharge their prepaid (debit) accounts using a credit card and a touch tone telephone. This self service recharge mechanism helps to increase customer retention by allowing customers the convenience of being able to recharge their existing account while also being able to control when their account is recharged and by how much.

End User Web Interface
The End User Web Interface provides HTML-based account management where empowerment is the key. By empowering end-users with the flexibility to view account balances, check messages, review call histories and recharge their accounts, there are fewer resources needed to service the customer. Customer satisfaction is also improved as end-users appreciate the convenience of being able to review their accounts from any web browser, on any platform, 24 hours a day, 7 days a week.

Find Me Module
The Find Me Module, also referred to as One Number Locator, Follow-Me or Simultaneous Ring, is an optional module that provides the convenience of a single telephone number for callers to use to reach subscribers where Talking SIP will automatically call multiple phones simultaneously (e.g. to simulate a PBX hunt group) or sequentially (e.g. office, then mobile, and then voice mail) to connect them. Custom branding, access and answer passwords, custom ring tones, call screening, on-hold music and destination specific timeouts are just some of the amazing features of this module.

Intelligent E-mail Agent
The Intelligent E-mail Agent allows system reports such as traffic analysis, call summaries or billing records to be extracted from Talking SIP™ and automatically e-mailed at predefined intervals. System administrators can also use this agent to execute stored procedures or custom SQL statements to assist in the maintenance of the network and/or database.

Registration and Location Services
This option allows Talking SIP™ to act as a Registrar Server and a Location Server. With this option SIP endpoints are able to register with Talking SIP™ as well as be challenged when utilizing services resulting in the most secure authentication method. In addition, nomadic SIP endpoints as well as SIP endpoints residing behind dynamic IP addresses may be located automatically in order to facilitate PSTN to IP and IP to IP calling.

Reminder Module
The Reminder Module provides appointment, reminder and wake-up services to Talking SIP™. Once authenticated the caller is prompted for the type of reminder they would like to create or the system can be configured to automatically select a particular reminder type (e.g. wake-up). The caller is prompted for the desired reminder time, which is confirmed and then the call is disconnected. When the reminder's date and time arrives Talking SIP™ places an outbound call to deliver the message -- if the message cannot be delivered the system will automatically re-queue it for later delivery.

Voice Mail and PBX Module
The Voice Mail and PBX Module is an optional module that brings voice mail, PBX/MBX, auto-attendant (ACD) and audio-text services to Talking SIP. Seamlessly integrated into the other enhanced services, this module's features include distribution lists, message delivery to mailboxes, e-mail addresses and telephone numbers, support for message waiting indication and stutter dial-tone, directory dialing by first name and/or last name, auto-forwarding, auto-carbon copying, partitioning, toll saver mode, hands-free message retrieval, message callback, classes of service, optional unified messaging, distributed or centralized message storage, and unlimited time of day/day of week greetings to help service providers drive additional revenue to their network while reducing customer churn.

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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