Talking SIP™

The Voice of the Next-Generation Network™

Applications ▪ Media ▪ Billing

Voice over Satellite Services for Next-Generation Satellite Operators

IVR Technologies

Key Features

Market Ready.
IVR Technologies' Talking SIP solution delivers voice services designed for satellite operators targeting maritime, aviation, defense, oil, gas and rail in both the enterprise and government segments. With a robust and turnkey suite of applications that support direct calling and callback services, Talking SIP is market ready, robust, highly scalable and provides deployment flexibility by being able to be run on premise or in the cloud on dedicated or virtual hardware.

Reduces OPEX.
Talking SIP reduces OPEX costs associated with operating a satellite service by adding revenue generating services that can build new profit centers or can help offset satellite operational expenses through the direct sale of services to employees and crew members.

Drives Revenue.
Through mature and widely deployed enhanced voice services, Talking SIP enables satellite operators to quickly launch targeted and branded services that are custom tailored to their clientele and that are seamlessly linked together to drive new revenue sources over the existing satellite network.

Builds Morale and Productivity.
Whether at sea, in the air or working at a remote site, crew members want to stay connected with family and friends as well as be able to conduct personal business while out of traditional coverage. Voice communications help keep morale and productivity high by providing a deeper and more meaningful medium of human contact that simply cannot be achieved through messaging or email.

IVR Technologies is a leading software development company providing voice over internet protocol (VoIP) enhanced services and real-time billing solutions to satellite operators around the world. The Talking SIP™ solution is a field-hardened and globally deployed value-added services and real-time billing solution that includes a rich suite of prepaid and postpaid voice service applications. These services are designed with features that help keep remote workers connected to family and friends while helping companies maintain productivity by keeping morale high.

It is imperative in today's competitive marketplace that satellite operators market and deliver in-demand and differentiated service offerings in order to gain market traction and competitive differentiation. Talking SIP is designed and engineered to provide differentiated service offerings with a time-to-market advantage that enables satellite operators to quickly capitalize, execute and win business globally. Talking SIP can be used to facilitate, encourage and/or govern usage with controls in place to ensure accountability and limit fraud as well as abuse.

Designed to run in the cloud, on virtual machines, or on premise Talking SIP offers unmatched deployment flexibility to suit the network and technical requirements of the satellite operator.

As a full-service firm, IVR Technologies provides pre-sales engineering support and professional on-site services for software installation, system configuration, training, customizations and network integration to ensure the streamlined and rapid deployment of the Talking SIP solution in your network.

The following are some of the in-demand and turnkey Talking SIP applications designed with satellite operators in mind:

Calling Card

The Calling Card application provides secure prepaid calling capabilities where the subscriber is authenticated into the system and then prompted for the destination number they wish to reach. Voice prompting can be in any supported language helping to personalize the service to the subscriber helping to make them feel more at home even though they are far away. When a balance drops below a certain threshold the subscriber is notified via a whisper prompt, and includes options for automated recharge.

Prepaid calling card is an important service for satellite operators to offer across their customer base as it helps drive new revenue and improves the morale and welfare of their employees and crew.

Auto-Attendant, PBX & Voice Messaging

Provide your enterprise customers with a custom branded auto-attendant with PBX features and robust voice messaging. Setup extension dialing with delivery of calls to remote sites either directly or via the find me/follow me features in order to track down users and deliver their calls anywhere they are. Ideal for reaching crew on airplanes, ships and any other remote site where satellite connectivity exists. With the auto-attendant and PBX module the company's PBX can follow and adapt to the business' growing and dynamic presence that may not be fixed to any particular physical site.

Conferencing

Talking SIP's conference services are market-ready and packed full of advanced conferencing features and third-party control. The conferencing service supports reservation and reservation-less based conferences with partitioned billing, moderator or peer modes, participant limits, custom branding, custom on-hold music and several announcement and departure notifications. Integrated into the Callback features of Talking SIP and with powerful database and web integrations it supports the creation of web based conferencing portals and conference management to allow attendees to join a conference automatically or by calling a shared or dedicated access number into the conference.

Talking SIP's real-time billing engine is tightly integrated into conferencing to provide the ability to set up free conferences or bill for them based on a flat rate, a per minute rate or a varying rate over the length of the conference.

Solution Architecture

Talking SIP delivers an advanced solution architecture that utilizes a common operating system and database with full redundancy across every component. Unlike our competitors who rely on a variety of operating systems and databases for their solution, as well as big iron and less than intuitive management and subscriber facing interfaces, Talking SIP is designed to be readily deployed and easily managed through unparalleled support services and a rich management interface. Talking SIP facilitates multiple levels of third-party access by customers and partners which fosters extensibility through a standards-based, intuitive and secure interface, whether that be through direct access using the open Microsoft SQL database or via the standards-based RESTful Web Services API, Talking SIP can meet the most demanding and challenging of integration requirements.

Talking SIP is engineered to meet the specific needs of today's satellite operators with a platform that drives revenue and greatly improves profitability and retention through converged services that empower employees/crew members and boosts morale and improves mental wellness by facilitating meaningful connection to family and friends.

As a fully converged mobile and VoIP services solution Talking SIP reduces the number of network vendors and technical touch points for your business while achieving greater business efficiencies, subscriber value and network profitability through its consolidated and converged network architecture.

CAPEX and ROI

Talking SIP is a competitively priced solution with unparalleled convergence, enhanced services, and real-time billing capabilities for the next-generation satellite operator. Talking SIP provides the features and functionality of no less than five traditionally separate network elements, which greatly reduces the capital requirements, the network footprint and the system complexity of the solution.

Talking SIP's just-in-time licensing model ensures that satellite operators only need to invest in infrastructure as it is needed, and its shared/pooled licenses allow new markets to be tested and cultivated gradually without huge capital outlay requirements. With licensing costs which are often a third of comparable solutions, Talking SIP provides an accelerated return on investment and allows scarce financial resources to be redirected towards marketing, subscriber acquisition and customer retention.

OPEX

Talking SIP automates common operational tasks and empowers subscribers to self-sufficiency through a full-service web portal. Talking SIP provides a very intuitive subscriber experience where employees and crew members can sign up for services online via credit card and fully manage their accounts and recharges with zero burden on the human resources of the satellite operator — this end user empowerment reduces the operational costs of the network and provides a better user experience through anytime access and self-care.

As a Microsoft Windows Server and SQL Server-based solution there is a vast global network of Microsoft engineers who are eager and ready to help you manage the solution and your network. Microsoft solutions reduce OPEX in many operational areas of the network including power consumption, rackspace requirements, technical resources, and customer service burden. Whether it be through virtualization, cloud-based hosting, or dedicated servers the computing resources required for Talking SIP are far more efficient than our competitors, which dramatically reduces the operational costs of your network.

Core Features

- Offers network consolidation by providing functions traditionally requiring up to five separate network elements in a single unified product for reduced complexity and service/support touch points
- Supports direct, indirect, inbound (via local, toll-free or toll (e.g. 900) access number) and callback to support triggering an inbound call in instances where there is a cost differential advantage with inbound vs. outbound calls (e.g. VSAT)
- Fully-integrated with leading class 5 enhanced services modules such as Voicemail and Find Me/Follow Me services for a complete converged offering
- Support for multiple business models (Group, Corporate and MLM) and parent/child hierarchical level control
- Support for a single unified account across multiple users, divisions and/or voice services
- Real-time authentication and throttling of consumption by account/device with capacity limits based on daily consumption, activation period, prepaid balance and/or credit limit
- Agile Web Portal for account, rate and cost management via a web browser, mobile phone or tablet providing secure and partitioned access to Talking SIP
- Automated or manual caller authentication as well as the option of being able to authenticate and authorize service by credit card
- Extensible and brand-able End User Web Interface providing end user empowerment for sign-up and self service account management, e-commerce, account top-up, low account balance notifications and more
- Real-time convergent billing and class 5 value-added services across any SIP enabled device
- Ability to readily create extremely flexible rate plans that can drive market penetration and boost employee and crew morale through custom tailored programs, free calling programs and usage incentives
- Bridges the satellite network with SIP for on-net/off-net network origination/termination via VoIP for reduced origination/termination cost, higher redundancy, and improved profitability

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Performance and Scalability

- Supports over 2,000 sessions (4,000 call legs) in a single physical server, virtual 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

- Smartphones
- 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 real-time consumption cut off
- Granular and flexible billing for voice services with the ability to set consumption limits on a daily basis, prepaid balance or credit limit.
- On-net to on-net, on-net to off-net, off-net to on-net billing support so you can charge differently based on how the call is routed.
- 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
- 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

- Capable of routing calls to multiple outbound carriers and/or SIP Trunking providers with automatic route advance
- Support for third-party least cost and/or service based routing (LCR) redirect servers
- 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
- Session Border Controller (SBC) compatible for SIP softphone services in order to provide far-end NAT traversal services to the network and for security, redundancy and scalability

Extensibility

- Separate application and call processing to allow remote service deployment without caller interruption

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
- Basic-like telephony service creation language to extend existing services or create wholly new ones

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, RemoteParty-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 or more Communication servers

Open Architecture

- Microsoft SQL database for open and complete access for analysis and reporting purposes
- Integrated optional REST-based Web Services API provides for easy customization and integration with third party OSS/BSS systems and existing service provider web portals

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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