Talking SIP™ Mobility

The Voice of the Next-Generation Network™

MVNA/MVNE/MVNO
Applications ▪ Media ▪ Billing
for Next-Generation Mobile Carriers

IVR Technologies

Key Features

Fully-Integrated.
Talking SIP Mobility converges mobile SIM-based voice, data and SMS services with a full suite of SIP-based next-generation VoIP enhanced services providing the ideal platform for building an MVNA, MVNE and/or MVNO service offering.

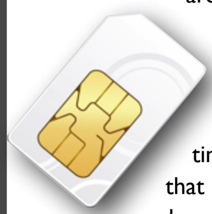
Cost-Effective.
Talking SIP Mobility is designed to run in the cloud, on virtual machines, or on-premise for unmatched deployment flexibility, and because it consolidates no less than 5 traditionally separate network elements it has one of the lowest CAPEX and OPEX overheads in the industry.

Future-Proof.
Talking SIP Mobility has a segregated architecture to allow new service applications to be readily developed and deployed on-the-fly, without subscriber interruption, to satisfy the ever-changing demands of the market.

Drives Revenue.
Through mature, widely deployed enhanced service applications, Talking SIP Mobility allows applications to be seamlessly bundled together to drive revenue to the network while reducing customer churn.

Scalable.
Talking SIP supports a wide range of session densities and scales in a just-in-time fashion. Quickly and easily grow your network to support any level of capacity with full redundancy in either a local or geographically distributed configuration.

Extensible.
Talking SIP Mobility utilizes an open Microsoft SQL Server database and a standards-based RESTful Web Services API to allow full access to subscriber and system information for reporting and analysis, as well as integration with internal and external business systems.



IVR Technologies is a leading software development company providing mobile and voice over internet protocol (VoIP) enhanced services and real-time billing solutions to mobile and next-generation carrier networks around the world. The Talking SIP™ Mobility solution is a field-hardened and globally deployed value-added services and real-time convergent billing solution that includes a rich suite of prepaid and postpaid voice, data, and SMS mobile SIM services. It is an agile, complete, end-to-end solution that can meet the ever-changing needs of the mobile provider through an architecture that segregates call processing from the application services to support rapidly changing and increasingly complex business models and cross-service bundles.

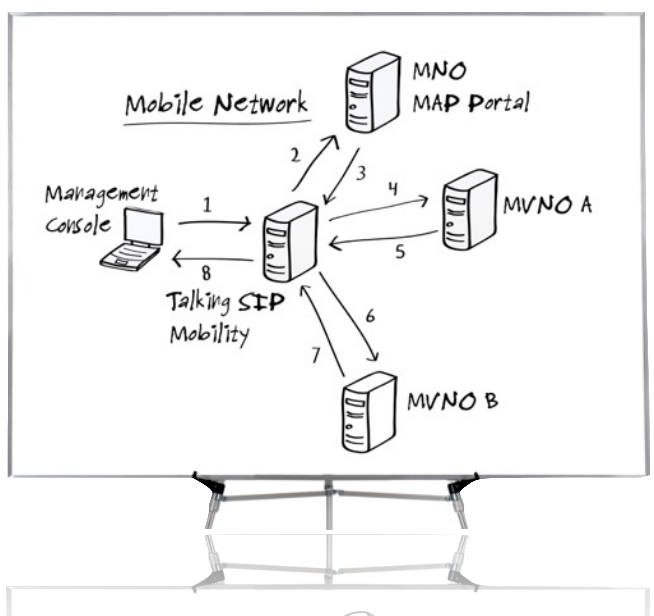
It is imperative in today's competitive mobile marketplace that providers can market and deliver in-demand and differentiated service offerings in order to gain market attention and traction. Talking SIP Mobility is designed and engineered to provide converged and differentiated service offerings with a time-to-market advantage to allow mobile carriers to quickly capitalize and execute on market opportunities.

The Talking SIP Mobility platform supports out-of-the-box e-commerce capabilities along with automated account charges, top-up and balance transfers in addition to shared family voice, data and SMS billing plans.

This flexible solution allows the rapid implementation of new service bundles and targeted promotions that can encourage and reward usage through a robust and mature prepaid and postpaid billing engine that supports bundled minute packages, multi-tiered rating, creative rules-based billing, and usage-based rating based on timing, wildcards, zones or E.164 destinations. Talking SIP Mobility provides a wide range of over-the-top mobile services, a full suite of revenue-generating VoIP applications, and a common subscriber web portal for account self-management across the full suite of enhanced services.

Designed to run in the cloud, on virtual machines, or on-premise, Talking SIP Mobility offers unmatched deployment flexibility to suit the network and technical expertise of the carrier. It is a turnkey solution that includes a converged session initiation protocol (SIP)/VoIP application suite with mobile network integration, SIM provisioning and account/system management.

As a full-service firm we provide 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 Mobility solution into your network.



Talking SIP™ Mobility Core Features and Overview

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Solution Architecture

Talking SIP Mobility 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 Mobility is designed to be readily deployed and easily managed through unparalleled support services and a rich management interface. Talking SIP Mobility 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 Mobility can meet the most demanding and challenging of integration requirements.

Talking SIP Mobility is engineered to meet the specific needs of today's mobile carriers with a platform that drives subscriber acquisitions, increases average revenue per user (ARPU), and greatly improves profitability and retention through converged services and subscriber empowerment.

As a fully converged mobile and VoIP services solution Talking SIP Mobility 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 Mobility is a competitively priced solution with unparalleled convergence, enhanced services, and real-time billing capabilities for the next-generation mobile carrier. Talking SIP Mobility 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 Mobility's just-in-time licensing model ensures that mobile carriers 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 Mobility provides an accelerated return on investment and allows scarce financial resources to be redirected towards marketing, subscriber acquisition and customer retention.

OPEX

Talking SIP Mobility automates common operational tasks and empowers subscribers to self-sufficiency through a full-service web portal. Talking SIP Mobility provides a very intuitive subscriber experience where they 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 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 Mobility are far more efficient than our competitors, which dramatically reduces the operational costs of your network.

Core Features

- Support for MVNA, MVNE, MVNO and M2M business models.
- Support for single unified account across multiple subscribers and/or mobile and voice services.
- Real-time authentication and throttling of data, voice and SMS consumption by device with capacity limits based on daily consumption, activation period, prepaid balance and/or credit limit.
- Supports CAMEL (call through and redirect) and USSD (callback) call flows for a seamless and intuitive subscriber experience.
- Turnkey solution for providing global roaming through multi-IMSI SIM prepaid/postpaid mobile voice and data services (fully branded travel SIM, data SIM, data roaming USB, world phone, smartphone, etc.).
- MVNA-MVNE architecture for the turnkey deployment of an unlimited number of MVNOs, each with its own branded web management and subscriber portal.
- Fully-integrated with class 5 enhanced services modules such as Voicemail and Find Me/Follow Me services for a complete converged offering.
- Integrated REST-based Web Services API facilitates customizations and the integration to third-party OSS/BSS systems and existing service provider web portals.
- Real-time convergent billing and class 5 value-added services for voice, data and SMS support across any GSM/SIM enabled device.
- Ability to readily create extremely flexible mobile plans that can drive market penetration and subscriber loyalty through custom tailored programs, free calling programs, and usage incentives.
- Customizable and fully branded web portals for MVNO mobile services management, subscriber self-management, e-commerce, account top-up, low account balance notifications and more.
- Bridges the GSM mobile network with SIP for on-net/off-net mobile 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

- XML 1.1: MAP (Mobile Application Part) XML messaging interface for mobile network signaling
- ETSI TS 123 078 CAMEL: For single stage call through call flows
- GSM 03.90 USSD: For two stage callback call flows
- 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
- Travelphones
- USB/SIM based data sticks
- Mobile softphones WIFI and 3G
- 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 voice, data and SMS consumption cut off
- Granular and flexible billing for voice, data and SMS services with the ability to set consumption limits by SIM 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
- Supports a wide range of disconnection fees, transit fees and mobile charges
- 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

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, 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 REST-based Web Services API provides for easy customization and integration with third party OSS/BSS systems and existing service provider web portals

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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