# Talking SIP™

The Voice of the Next-Generation Network™

## Key Features

**Market Ready.** The Voice Mail and PBX Module is ready for turnkey deployment. This enhanced service is a full-featured voice messaging, audiotext, PBX and automated call distributor system that delivers a market ready and highly competitive service offering for next-generation providers.

**Fully-Integrated.** The Voice Mail and PBX Module is seamlessly integrated into the other enhanced services of Talking SIP™ to help reduce customer churn while providing a consistent and comprehensive user experience.

**Drives Revenue.** In the highly competitive long distance market the Voice Mail and PBX Module offers a primarily inbound service that can be used to drive additional revenue to the network or it can act as a loss leader for other services.

**Scalable.** The Voice Mail and PBX Module supports both distributed and centralized deployment configurations. In a centralized deployment multiple Talking SIP™ nodes can be seamlessly deployed around a shared NAS/SAN device to allow message deposit and retrieval from local and remote locations.

## 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™. Talking SIP™ is a leading and fully-integrated application, media, billing, location and registration server for the next-generation network.

The Voice Mail and PBX Module offers full voice messaging, multi-tenant audiotext services, automated call distribution and PBX functionality. Features of this module 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; centralized or distributed message storage; and unlimited time of day/day of week greetings.

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 the Voice Mail and PBX 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 profitability to the next-generation network while helping to reduce customer churn.

The Voice Mail and PBX Module provides voice messaging, audiotext services, automated call distribution (ACD), as well as a rich set of PBX/MBX features to Talking SIP™ to help service providers create highly competitive and compelling service offerings that can be easily and efficiently scaled on-demand to satisfy future subscriber growth.

The following is a list of some of the features and functionality of the Voice Mail and PBX Module:

### Message Handling and Notification

- Message delivery to mailboxes, e-mail addresses and/or telephone numbers
- Support for distribution lists
- Auto forward and carbon copy feature
- Message waiting indication to an IP phone, e-mail address or telephone number
- Ability for multiple phones to monitor the same mailbox and to share the same message waiting indication
- Ability for a single shared mailbox to send different message waiting notifications to multiple telephones
- Silent record mode for call monitoring and a quick record mode for mobile note taking
- Ability to set an unlimited number of operator mailboxes
- Ability to automatically send a welcome message to new users
- Ability to customize welcome back message to users who have not logged into the system for an extended period of time
- Ability to assign a public password to a mailbox to restrict caller access
- Ability to rewind, pause and/or fast-forward during message playback
- Ability to support voice messages to e-mail via direct record, forwarding, distribution lists and/or speed dials
- Record and forward message functionality to allow a message to be recorded and forwarded to multiple destinations

### User Convenience

- New user setup wizard to guide users through setting up their account
- Toll saver mode for tariff savings during message retrieval that varies the number of rings based on whether or not a subscriber has messages waiting
- Option to return a call during message playback for messages left by external callers
- Numeric service to provide callers with the option to enter a callback number during greetings
- Ability to support automatic subscriber login based on incoming ANI with an option for a password for additional security
- Ability to have the system automatically call a subscriber when a new message is left to avoid long distance charges or mailbox polling
- Control of message length thresholds to help reduce incidences of dead air and hang-ups
- Ability to automatically play new messages upon login
- Ability to automatically save and advance messages during playback for hands-free mobile access
- Option to automatically play message information during message playback
- Extended absence mode with the ability to set a date and time range to allow the greeting to automatically expire
- Unlimited number of greetings by time of day, day of week and date range with the ability at the greeting level to turn off/on recording or require caller confirmation in order to record a message

### Integration

- Integration into any SMTP e-mail system for unified communication using the optional Intelligent E-mail Agent
- Full integration into Talking SIP™'s Debit, Find Me, Tandem, and Termination Modules
- Integration with the Debit Module for DISA-like functionality to provide users with the option to place outbound calls from the user menu

### Caller Control

- Ability to set call actions before a greeting, after a greeting or at the mailbox level
- Ability to assign one or more call actions to DTMF key presses of a mailbox

### Streamlined Management

- Automatic directory listing creation with the ability to dial by first name and/or last name
- Loop detection logic implemented to prevent endless-loop routing in the absence of user input
- Ability to automatically route incoming DNIS numbers to certain mailboxes
- Option for a DNIS prefix to allow endpoints to forward calls directly to voice mail to prevent re-presentment and to reduce the number of DNIS numbers required for voice mail routing
- Ability to partition the platform for multiple tenants
- Full call session logging of user account updates and activity to aid in customer service tracking and issue resolution
- Option to prevent messages from being left in mailboxes that have not yet been logged into
- Class of Services to centrally manage settings for groups of subscribers (e.g. maximum message length, maximum greeting length, message retention for new and saved messages, outbound origin, etc.)
- Ability to set an option to permit recording directly from an audiotext box for dynamic recording while navigating the audiotext tree

### Scalability and Extensibility

- Optional centralized NAS/SAN storage option to allow messages to be centrally stored and accessed by multiple Talking SIP™ nodes
- Ability to assign audiotext key presses to the Debit, Find Me, Tandem, and Termination Modules as well as directory access, subscriber login and prompt recording

## Opportunity

According to Research and Markets the Centrex/Hosted IP market is primed for substantial growth over the next four years. The demand in the SME market, the recognized sweet spot for these hosted services, has been accelerating for some time and will continue to perform at a year-over-year growth rate of 90% through 2009. The main drivers for the service include cost savings, an enhanced feature set including mobility and integrated messaging options, ease of use, greater control, convergence, and system access via a web portal.

Hosted IP services especially PBX applications hold significant growth potential and thus higher margin revenue opportunities for forward thinking service providers.

### 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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