Talking SIP™

The Voice of the Next-Generation Network™

Key Features

Market Ready. The Conference Module is ready for turnkey deployment. This enhanced service is a full-featured reservation-based and reservation-less conference service that delivers a market ready and highly competitive service offering for next-generation providers.

Fully-Integrated. The Conference 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 Conference 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 Conference 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 for caller announcement from local and remote locations.

Conference Module

The Conference Module is an optional module that brings reservation-based and reservation-less conferencing to Talking SIP™. Talking SIP™ is a leading and fully-integrated application, media, billing, location and registration server for the next-generation network.

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.

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 Conference 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.

In today's global, outsourced economy with scarce resources and a greater awareness to one's carbon footprint, conferencing is an ideal way for people to meet and collaborate when face-to-face meetings are not practical, cost-effective or feasible.

The Conference Module provides high quality reservation-based and/or reservation-less audio conferencing with integrated billing to the Talking SIP platform. Conferencing allows your subscribers to join in a multi-party conference in an easy and convenient manner that helps to drive inbound revenue to your network.

The following is a list of some of the features and functionality of the Conference Module:

Conference Types

- Reservation-based conferences with or without a Moderator
- Reservation-based Listen Only conferences
- Reservation-less conferences with or without a Moderator
- Reservation-less Listen Only conferences

Features

- Supports ANI, DNIS, IP, remote-party-ID and P-asserted-identity authentication into a conference
- Supports silence, voice prompt, tone or a recorded name for participant announcement
- Supports silence or a tone for participant departure
- Supports several levels of custom branding
- Option to announce the number of participants in a conference
- Option to require the caller to confirm the conference access code prior to joining the conference
- Ability to block announce CLID/ANI from joining a conference
- User definable acceptable wait time for early arrival of participants
- Fall-through mode to check for a reservation-based conference and if one does not exist automatically create a reservation-less conference
- Support for multiple conference moderators
- Moderator mode to require a moderator's presence before a conference may begin and the ability to automatically end the conference when the last moderator leaves the conference

Reservation-Based Conferences

- Ability to set the maximum number of participants
- Ability to set a participant password for conference room access
- Ability to set a moderator password for moderator access to a conference room
- Ability to schedule conferences in the future with an automatic expiration
- Ability to set whether or not the conference is full-duplex or listen-only for participants
- Ability to set a custom hold-prompt at the individual conference level

Billing

- Real-time billing engine integration
- Ability to offer free conferencing or bill by the call and/or by the minute as well as vary the cost of the call over the call's duration
- Ability to vary the rate based on full or partial ANI/DNIS, access code, and/or based on the time of day, day of week or date range
- For prepaid accounts you may equalize the prepaid balance over the maximum number of participants for reservation-based conferences
- Ability to set up billing based on DNIS, device, caller or conference access code

Integration

- Full integration into Talking SIP™'s Callback Module to allow participants to be called and brought into the conference
- Full integration into Talking SIP™'s End User Web Interface to allow Conferences to be scheduled and managed
- Interface to allow third-party control (e.g. custom application or web service) to control a conference and/or its participants

Caller Control

- Ability for callers to mute and unmute themselves during a conference via DTMF key presses
- Ability for callers to obtain the current participant count during a conference via DTMF key press

Streamlined Management

- Supports the ability to extract the full or partial DNIS for conference room access
- Ability to automatic pre-pend a prefix to help streamline rating

Scalability and Extensibility

- Optional centralized NAS/SAN storage option to allow recorded names to be centrally stored and accessed by multiple Talking SIP™ nodes
- Supports centralized, decentralized and hybrid deployment models
- Centralized licensing to allow conference resources to be shared across the network
- Supports remote just-in-time license upgrades
- Support for automatic fail-over to another conference server in the event of failure or a lack of a timely response

Opportunity

According to Wainhouse Research audio conferencing is growing at a 28 percent compound annual growth rate (CAGR), with over 50 billion minutes of usage annually. Conferencing provides a natural complement to any voice service offering with the main drivers to the service being cost savings, convenience, ease of use, and control.

Contact us today to find out how to drive additional inbound revenue to your network and increase subscriber loyalty and retention through Talking SIP's Conference Module.

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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