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## Talking SIP 3.2 Released

*Talking SIP 3.2 Offers Registration and Location Services, Class 5 Features, Additional Granular Billing Options and Recording Capabilities*

IVR Technologies is pleased to announce the release of Talking SIP Version 3.2.

With a widely deployed customer base and a quickly growing VoIP market, IVR Technologies has been quick to identify features and functionality that are important to our customers not only in building their brand and staying competitive, but also to reduce their management overhead while improving business efficiency.

In version 3.2 customers can now apply daily, weekly, monthly, bi-monthly and yearly account service charges, recharges and other fees on an account-by-account basis. Class 5 Features like call blocking, call masking and call return, to name but a few, have also been integrated into version 3.2. With the user independent prompt recording module, custom branded voice prompts and a full range of system prompts can now be recorded in any supported codec on-the-fly.

Available as an option is our Registration and Location Services that make it possible to register your SIP devices (IADs, IP Phones, Softphones, etc.) directly with Talking SIP for authentication and on-net/off-net call routing, provisioning and automated customer signup. The Registration and Location Services Module, when coupled with our End-user Web interface and e-commerce options, truly makes Talking SIP an unbeatable solution for deploying next-generation services in any SIP-enabled network.



## Talking SIP 3.2's New Features

### A Brief Overview of the New Features and Functionality

Talking SIP 3.2 brings highly sought after features and functionality to the market in a tightly integrated and turnkey configuration.

The following is a brief overview of some of these exciting new features and functions:

- ◎ **Class 5 Features** – Calling features such as last number redial, last called party, last calling party, call blocking, call return, caller ID masking and automatic busy number redial have been added.
- ◎ **Integrated Prompt Recorder** – A new standard module of Talking SIP, that allows for application, system and custom branding prompts to be recorded on-the-fly in any supported codec and by any authorized user.
- ◎ **User Defined Billing Fee and Surcharge Intervals** – Easily configure the Service Charge module to automatically apply daily, weekly, bi-weekly, monthly, bi-monthly, and yearly account fees and charges.
- ◎ **Registration and Location Services Option** – This optional module allows Talking SIP to act as a Registrar and Location Server for any SIP endpoint.

- ◎ **Rating Based on Calls or Minutes to Date** – Set which Rate Plan is selected based on the number of calls or minutes an account has made to date.
- ◎ **AutoDialer Feature** – This feature allows a DNIS number to be shared across multiple accounts to automatically dial different user specified destinations whenever the DNIS is dialed – think of this feature as DNIS-based speed dials.

As we continue to advance Talking SIP, it is taking on a greater and vital role in the network as it provides more core functionality that allows other equipment to be redeployed or decommissioned or avoids additional equipment purchases altogether.

It is through valuable and appreciated customer feedback, as well as industry research and vision, that new features are added to Talking SIP in our continued quest to provide the leading enhanced services and real-time billing platform to the industry. If there is any functionality or enhancements that you would like to see in the product, please feel free to e-mail [feedback@ivr.com](mailto:feedback@ivr.com) -- your comments and suggestions are always welcome.

## CUSTOMER SPOTLIGHT



Infocom Technologies Incorporated, a pioneer in the Philippine Internet Service Provider (ISP) industry is charting new markets and technologies where no ISP has gone before. The company is focused on ensuring that its clients experience Excellent Services and Total Customer Satisfaction through its Internet, Customer Service Outsourcing and IP Exchange Businesses.

"Talking SIP has allowed Infocom to provide different prepaid and postpaid products to our local wholesale customers. Since our service offering is IP-based, we are able to extend our clients' reach beyond geographical borders. It also fits in well with Infocom's strategy of being a technology enabler and solutions provider."

Reynaldo Sulit,  
IPX Business Manager



## Streamline Network Management, Reduce CAPEX and Drive New Sources of Revenue

As the value proposition of Talking SIP continues to build, we are driven by one of our main goals which is to provide a solution that integrates more and more core functionality related to IVR, applications, media and real-time billing into the solution.

With our advanced product architecture and continued application enhancements and creation, Talking SIP is a product that satisfies multiple network requirements in a single, integrated solution.

Networks in the past required a separate application server, media servers, registration/location server and billing server but today it can all be achieved by Talking SIP. Service providers can now work with a single, integrated solution, which consolidates several network elements and greatly reduces administrative overhead by allowing you to communicate with a single vendor instead of multiple vendors resulting in reduced time to market and the time required to resolve interoperability issues.

Cost Reduction and accelerated return

on investment are at the forefront of every service provider's decision making process when building and/or enhancing their next-generation network. Talking SIP is unmatched in its ability to deliver on these important requirements. As a software only solution, Talking SIP delivers its services to any point of presence from either a centralized or de-centralized configuration using standard off-the-shelf server hardware, making it one of the most affordable solutions in the market. With integrated solutions for today's most popular and profitable services such as Voice Over Broadband, click to call, calling cards and tandem switching, multiple markets can be targeted using a single solution helping to quickly grow your subscriber base.

With the release of Talking SIP 3.2 we have been able to further enhance the product's value from a feature, functionality and management perspective – with Talking SIP 3.2 targeting new sources of revenue has never been easier.

## MARKET RESEARCH

The VoIP applications market will show sustained growth, passing through \$500 million by 2010.

**Juniper Research**

VoIP is gaining traction because it has the potential to lower telecommunications costs across the food chain (from the service provider down to the end user at both home and work), and support applications such as unified communications that improve employee productivity. [www.newtelephony.com](http://www.newtelephony.com)

The 8th annual VoIP industry report by iLocus further reveals that that there were estimated 37.5 million VoBB subscribers worldwide as of end 2006.

**iLocus**

An estimated 1,079 billion minutes of VoIP traffic was carried by service providers worldwide in 2006.

**iLocus**

### CUSTOMER TESTIMONIAL

"KBT Systems is a Canadian wholesale provider of VOIP infrastructure. KBT services large tier I telecommunication companies throughout the world and has experienced tremendous growth over the past 10 years. Our ability to source and utilize best of breed vendors has been instrumental in our growth. KBT considers IVR Technologies to be such a best of breed vendor by their products and through their dedication to outstanding service and support, which has contributed greatly to KBT's ongoing success.

"We consider IVR Technologies to be an industry leader in the VoIP application, media and billing space, and we look forward to continuing our great working relationship. We would recommend IVR Technologies to any next generation provider, large or small, without hesitation or reservation."

**Stephen Mundy,**  
President, KBT Systems Corp.



### Technical Tips



As a recurring section of our newsletter, our goal is to provide you with informative tips that help you to discover and utilize very powerful features of our product. We are confident that these tips will surely add value to your business and help you to become more efficient in the use and management of Talking SIP.

### Registration and Location Services

With the release of Talking SIP 3.2, registration and location services are now supported as an additional software option; however the configuration of these services has been seamlessly integrated into the TMC to make it as easy as possible to manage.

The first decision you need to make is how each device will handle an incoming SIP REGISTER message. By default, all of your current devices will have their registration type set to "Deny," meaning that REGISTER messages coming from that device will be rejected with a '403

Forbidden' error message. In order to accept incoming registration messages, there are two other options, "Allow" and "Challenge," which you can use.

The "Allow" option tells the device to accept incoming REGISTER messages and respond with a 200 OK, although the registration username and password will not be authenticated. This model is useful for situations where you have several IADs, softphones, or SIP phones that require registration in order to place calls, but you do not want the burden of provisioning registration credentials for all of these devices.

By setting the device's registration type to "Challenge," Talking SIP is configured to challenge the registration request. This is accomplished by sending a '407 Proxy Authentication Required' response message causing the registering device to send a new SIP REGISTER message with registration credentials provided therein. These credentials must already have been provisioned in the TMC by adding a registration account alias for a given account in the system. This alias allows

the registration username and password to be specified, and those credentials must match with those provided in the SIP REGISTER message in order for Talking SIP accept the REGISTER message and respond with a 200 OK. If the credentials do not match, Talking SIP will reject the REGISTER message with a '403 Forbidden' response.

Once a caller is properly registered using this method, the system can be configured to use the caller's registration credentials to automatically authenticate the caller, negating the need to enter an account number via DTMF while offering the most secure authentication method. In addition, calls can now be routed to this registered endpoint no matter where the physical device resides, thus providing location functionality to the system.

## The Power and Flexibility of Origins

When an inbound call is first accepted by Talking SIP, there are various stages that it goes through before finally being routed out of the system to another destination.

As the call proceeds through these different stages, it is often necessary to identify and keep track of where it originated in order to make decisions that will affect the call flow, such as which service application will be used to process the call (e.g. Debit, Tandem, Callback, etc.), which Rate Plan will be used to determine the call charges, and which device will be used to route the call to its ultimate destination.

Typically, the full or partial ANI and/or DNIS will be used to identify where the call originated. In many circumstances the ANI or DNIS will contain a digit prefix (called a Routing Code) that will have been added by the inbound carrier or device to identify it. Instead of examining the ANI or DNIS at every point in the call flow where decisions need to be made, Talking SIP introduces the concept of an Origin.

An Origin is simply a name that is assigned to an incoming call based on various criteria to identify where it originated. A variety of methods can be used to define an Origin, such as by specifically assigning an Origin to a device, by mapping special ANI prefixes to different Origins, by associating an Origin with a unique or partial ANI or DNIS number as mentioned above, or by using various combinations of these methods.

Once an Origin has been defined, it can be used throughout the lifetime of the call in several places, such as:

- ⊙ At the Device and Module Map level, when determining which service

application (e.g. Debit, Tandem, Callback, etc.) will be used to process the call

- ⊙ At the Rate Schedule and Rate Plan level when determining which Rate Plan or Rate Plan entry will be used to rate the destination number
- ⊙ At the Route Group level, when determining which Outbound Route will be used to place a call to a terminating device

By allowing an Origin to be assigned to an inbound call, Talking SIP provides a powerful and flexible, yet simple way to identify a call's point of origin. This can streamline the call flow and ease administration and maintenance, because while the definition of an Origin can easily be set in one place (e.g. via new or modified association with ANI or DNIS digits), the name of the Origin can remain the same everywhere else it is referenced in the system.

## Take Advantage of Flexible Rules for Rate Plan Selection

When rating destination numbers, Talking SIP provides a very flexible model for determining rate selection. The fundamental building block of rating in Talking SIP is the Rate Schedule.

A Rate Schedule can be assigned directly to a single account or to a Class of Service referenced by one or more accounts and determine how calls will be rated. It can contain one or more Rate Plans (sometimes referred to as 'rate decks' or 'rate tables'), which can be dynamically selected at run time based upon a variety of selection criteria, such as:

- ⊙ Priority, which can be set to override the default order in which the rate selection criteria are evaluated to force a specific Rate Plan to be selected over another

- ⊙ Day of week (including date ranges)
- ⊙ Time of day (including time ranges)
- ⊙ Origin
- ⊙ A partial or explicit DNIS entry, which can be used to set the Origin for rate selection via the DNIS\_FORMAT table
- ⊙ A partial or explicit ANI entry, which can be used to set the Origin for rate selection via the ANI\_FORMAT table

In addition to the above criteria, Talking SIP 3.2 adds two new features, which enhance the rate selection process even further:

- ⊙ A call threshold, which allows different Rate Plans to be selected based on the number of calls an account has made to date. For example, the first call made by an account could use Rate Plan 'A', the second could use Rate Plan 'B' and the third could use Rate Plan 'C'.
- ⊙ A minute threshold, which allows different Rate Plans to be selected based on the number of minutes an account has been billed to date. For example, when the account is first used, it could use Rate Plan 'A' and after it has been billed 10 minutes, it could use Rate Plan 'B'.

By utilizing and combining these selection criteria, Talking SIP provides an almost unlimited number of ways to determine how rate selection will be performed.

Service providers now have the flexibility to partition rates based on a variety of conditions specific to their business needs, so they can pursue new and creative ways of generating revenue by organizing rates targeted to their diverse customer base and their varied service offerings.



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