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## Talking SIP 3.1 Debuts - Representing the Third Generation of SIP Based IVR and Real-time Billing Solutions

IVR Technologies is pleased to announce the release of Talking SIP Version 3.1.

Talking SIP 3.1 represents the culmination of a tremendous investment by IVR Technologies in the development of our third generation architecture to further entrench IVR Technologies' leadership position of providing industry leading and revenue-generating applications with the utmost in real-time billing control and flexibility, without sacrificing ease-of-use and low administrative overhead.

Talking SIP's signaling, media processing and application scripting components have been completely redesigned to reach new milestones in scalability, performance and control for the next-generation network. We are extremely excited about this release as it provides an amazing foundation upon which to build voice 2.0 applications that highlight the empowerment, flexibility and control that broadband access and voice over IP have to offer.

Success will not come from PSTN replacement applications that most CLECs and service providers offer today, but rather it will be assured by being able to truly capitalize on broadband connectivity and IP based communication with a service offering, level of control and flexibility that are simply impossible with PSTN-centric solutions.

Telecommunications is changing forever and it is our voice you are hearing.



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## Talking SIP Version 3.1 Now Available

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Talking SIP 3.1 is built around a completely redesigned application, media and signaling architecture to serve as the foundation for rich, in-demand applications that capitalize on the ubiquity, access and richness of broadband connectivity.

The following is a brief overview of the features and functionality embodied within this release:

- **International and Web Callback** - This optional application allows providers to drive revenue by allowing their customers the ability to trigger callback calls to get a remote dial-tone via SMS, email, telephone, web page or integrated into third party applications via our API to keep customer traffic within your network even when they are traveling out of city, state/province or country. With many creative billing options, this application can even navigate through PBXs and ACDs for use from hotels, motels and corporate offices.
- **Reminder/Wake-up Service** - This optional application allows providers to offer wake up and reminder services to their customers for the small office/home office as well as to the hotel, motel and timeshare markets.
- **Improved Performance** - Talking SIP 3.1 offers improved performance, reduced CPU load and a reduced memory footprint to allow the application to make better use of the host system's resources and to allow greater levels of responsiveness and scalability. In addition, Talking SIP provides greater access to signaling and media to allow greater control and performance tuning for different hardware systems in addition to voice caching to help optimize text to speech playback.
- **Improved Tones** - Talking SIP 3.1 adds the ability to play customized ring tones in two-stage calling applications like the Debit Module. This new feature allows better internationalization by allowing providers to configure country specific tones like ringing, busy and fast busy to provide their customers with additional levels of comfort when using the system. These tones can be customized to allow opportunity for branding as well as personalization.
- **Asynchronous Voice Prompts** - Talking SIP 3.1 now supports asynchronous voice prompts to help streamline call flow and reduce hang times. Now you can be playing voice prompts to the caller while background operations like a credit card or voucher recharge is taking place.
- **Multi-leg Billing for Callback** - Talking SIP 3.1 offers the industry's most accurate multi-leg billing engine with dynamic balance updates and on the fly maximum call durations to ensure that callers receive the maximum call time while enforcing all of the credit limits of their accounts.
- **Supports any Codec** - Talking SIP 3.1 supports any voice or video codec between calling and called parties to allow proprietary and/or bandwidth appropriate voice and video codecs to be used as well as dynamically changed during a call.
- **Open XML interface Added for Credit Card Processing** - Our optional Credit Card Verification Server module now supports Verifone's PC Charge Software, LinkPoint as well as an XML interface to allow third party integration to any banking or financial institution. This module allows for automated web based subscription, as well as web, automated and telephone based recharge.
- **Performance optimization** - Completely new SIP stack and communication agent that optimizes the media and call signaling capabilities for advanced applications and new media type support e.g. video and fax

Growing your business, improving the subscribers' experience and lowering maintenance and administrative overheads are all benefits of this new release. By deploying the new callback module service providers can now attract the mobile and web user, two previously untapped subscriber markets. Talking SIP 3.1 also provides for more granular and flexible IVR prompting for ring back tones/messages, asynchronous prompt playback during non-blocking operations and improved balance handling for better

voice quality and improved internationalization. This release also builds upon the dynamic and highly flexible real-time billing engine to provide unmatched support for multi-leg billing and disconnect charges so that creative billing programs and multiple service branding can be achieved for customer tailored marketing programs.

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 continual quest to provide the leading enhanced services and real-time billing platform in 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.



## Differentiate Yourself from the Competition with Innovative and In-demand Applications

With the release of Talking SIP 3.1 come two new optional applications to provide advanced callback and notification services.

Service providers looking to reach new markets to help grow their business' revenue and profitability will benefit greatly from the added functionality that our Callback module provides. In addition to supporting standard telephone-based callback triggers the Callback module supports advanced call triggers which can be placed from a service provider's web site, by a click to call link or by embedding our Callback API into a third party application. This service can simplify the calling process for your on-the-go and mobile subscribers by providing email and/or SMS triggered callback features where the user simply sends an email or text message with the callback number and destination and Talking SIP handles the rest with prompt callback and connection to their desired party.

The opportunities are endless for discovering new revenue streams such as providing private click to call services for personal ad web sites, links for Craigslist advertisements or Ebay auctions or any web page, search engine results or e-mail signature where you want to provide cost-effective user initiated calling that turns browsing into personal or business opportunities.

Today, most service providers offer telecom services to wired or wireless subscribers -- why not open your network and services to web based calling and the global market? This web initiated callback services allows you to reach new markets with very little upfront cost and almost immediate returns.

The Callback Module supports tremendous billing flexibility to allow providers to offer extremely creative billing packages such as time of day discounts, cost reductions for longer or shorter duration calls, bundled minutes as well as rating zones based on where the call originated and where the caller is dialing.

With this level of billing control and configurability designing service plans that take into consideration termination to typically free on-net destinations (e.g. softphones or IP phones) and those off-net destinations where a per minute fee is required (e.g. PSTN or mobile phones) can easily be created. This process can be further automated by adding our optional Credit Card Verification module that can automatically process credit card transactions such as monthly service fee billing and settlement of metered charges.

Building new revenue streams through new service offerings delivered by Talking SIP, is further achieved by the release of our Reminder/Wake UP/Appointment notification module.

With the Reminder module the service provider can provide its subscribers the ability to create and deliver various notification types such as reminder messages, wake up calls and appointment notifications. Attract business clients like hotels, motels and timeshares where they can now offer their guests expected wake-up services but now on a hosted basis. Corporate customers now have the ability to notify their staff/departments of upcoming events and corporate wide announcements. Political parties, charities and sporting teams often require outbound notification services which you can provide using this new module to build your business through multiple consumer and business/enterprise channels. With web based access and an automatic scheduler, the customer is empowered to create and manage their reminders without the need for customer service assistance. Many different notification delivery methods are possible such as to a softphone, IP phone, PSTN connected phone or mobile phone- the recipient also receives the option of snoozing the notification for a predefined period of time (e.g. 5 minutes).

### **Conclusion**

It is with new and innovative applications that today's next-generation VoIP service providers will grow subscribers and revenues, profitability and a more efficient business operation. The Callback and Reminder applications represent the first of many new innovative and caller empowering applications that we are developing upon our new Talking SIP 3.1 architecture.

IVR Technologies continues to be your resource for media rich applications coupled with unmatched real-time billing control integrated into the most turnkey, affordable and performance based solution in the industry.



## Talking SIP 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 the product.



### Support Multiple Dialing Prefixes for International Customers

Talking SIP can be deployed in a centralized or decentralized fashion as well as a hybrid model depending on the network topology and the comparable costs associated with providing rich IVR services at the edge or at the core. In addition the licensing model of Talking SIP allows customers the flexibility to add additional nodes without incurring additional licensing cost. As providers increase their network's footprint by targeting international customers, they will undoubtedly encounter the need to support multiple dialing prefixes/patterns within the same node, for example to allow 7 digit dialing to different cities where the 10 digit translation will be determined based on where the caller is calling from. Another common need is to be able to simultaneously support multiple international dialing prefixes where some customers are used to dialing 011 for international calls while others are used to dialing 00.

Talking SIP by default supports a domestic dialing prefix (1) and an international dialing prefix (011) but by setting `SYS_PREFIX_MAP` to 'Yes' you can utilize the `PREFIX_MAP` table to create flexible dialing patterns that allow the caller to dial in the manner in which they are accustomed, while being able to perform dynamic number translations before and after rating. This allows a common rate plan to be used that is fed from different source numbers in an effort to streamline rate management by avoiding the need to create different rate plans to common destinations that only vary by the dialing prefix. The `PREFIX_MAP` allows dial patterns to

be set based on the node, originating device, routing code, originating area code, ANI, partial ANI, DNIS and/or partial DNIS to provide great control and flexibility as to when they are invoked.

The `PREFIX_MAP` table can also be used to remap telephone numbers, for example long distance access numbers to toll-free access numbers, provide global speed dials as well as implement globally blocked destinations across multiple rate plans.

The `PREFIX_MAP` table supports single character wildcards (?), multiple character wildcards (\*), inclusion ranges and sets (e.g. [1-3] will match 1, 2 or 3) as well as exclusion ranges and sets (e.g., [^345] will exclude entries with a 3, 4, or 5 within a specified position) for ultimate control and flexibility. You can also specify the number of leading digits to be trimmed from the incoming number as well as the desired digits to add before termination, with the ability to perform even more digit manipulations at the Outbound Route level.

Talking SIP is configured for streamlined initial deployment with the ability to enable lower level control as needed that ensures that the next-generation provider can operate their network in the most efficient manner possible while being able to offer customer-centric services that improve customer satisfaction and retention.



### Offer Customers Bundled Minutes for a Set Monthly Charge

Talking SIP offers a variety of flexible ways to bill your customers and drive revenue through the system. One of these methods is to provide a set number of minutes for a monthly charge and then per minute billing once the caller has exhausted those packaged minutes. This billing model is similar to that employed by many wireless carriers.

To automate the billing process on a monthly basis for a plan like this you must run our Service Charge Module which is included with Talking SIP. The service charge module is a daemon that runs in the system tray and can be configured to run at a time when your network is experiencing its lowest call volume. The service charge module is the application that will actually

In Talking SIP the “bucket” of minutes assigned to the caller are referred to as “Packaged Minutes.” At the account level, the caller not only has a balance which keeps track of the monetary balance of the account, it also has a packaged balance which keeps track of the balance of minutes that the caller has remaining. The packaged balance can be assigned either at the time the account is generated or at a later date.

Each destination in the caller’s rate plan has the option of whether it will be billed from the caller’s packaged minutes or whether metered billing should occur based on the caller’s account balance. This provides for the ability to restrict certain destinations from the calling plan. For example you may want to allow your customers to call any domestic number with their packaged minutes but bill all international calls at different per minute rates.

charge the customer for these packaged minutes and reset the customer’s packaged balance. The packaged minutes can be configured to reset to an initial value, or you can allow them to “roll over” into the next month. It is also possible to configure the service charge module to work in conjunction with our optional Credit Card Verification (CCV) Module to charge the customer’s credit card and update the packaged balance only if the credit card purchase was authorized.

The configuration of these calling plans occurs in a “Billing Package” which is tied to the account’s rate schedule. The billing package is where you specify when and how much your customer is billed and in what fashion you want the packaged minutes reset each month.



## Take Advantage of Flexible Rules for Mapping Calls to Service Applications

To increase revenue and expand their service regions, providers frequently make arrangements to terminate traffic from new sources, whether from a new carrier or a recently added device, such as a gateway. Since each new source of traffic has unique requirements, it will need to be mapped to a particular application module to service the incoming calls.

Determining which application to use is accomplished by either directly assigning an application to a specific device (via the Device Module Map table) or by mapping DNIS numbers to a particular module (via the DNIS Module Map table).

While previous versions of Talking SIP allowed multiple DNIS numbers to be mapped to the same module, each unique number required an explicit entry in the table. In addition, other mapping scenarios were more difficult to implement, such as splitting the traffic from a single device into two groups and mapping each group to a different module (e.g. when terminating class 4 Tandem calls, where the DNIS is always unique).

Talking SIP 3.1 provides some new, more flexible mapping rules, making application selection easier and more flexible than ever.

First, both the Device Module Map and DNIS Module Map tables have been enhanced to allow an Origin to be used as selection criteria. An Origin is simply a name that is assigned to an incoming call to identify from 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, or by using various combinations of these methods.

Second, the DNIS Module Map table has been enhanced to allow wildcards and ranges to be entered as DNIS numbers, such as:

- An explicit DNIS entry (e.g. 5551000).
- An inclusive range of valid DNIS numbers (e.g. 5551000 - 5552000)
- A fixed DNIS entry with wildcard values (e.g. 555??6)
- A variable length DNIS entry that optionally begins with a certain series of leading digits (e.g. 555\*)
- A variable length DNIS entry that optionally ends with a certain series of digits (e.g. \*555)
- A default entry that is used as a catch-all whenever a matching DNIS cannot be found

By providing several new ways of mapping incoming calls to various applications, Talking SIP simplifies the job of adding new sources of revenue generating traffic to the system, making maintenance and configuration easier and giving you more control over device partitioning for various services.

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