IN THIS ISSUE

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Talking SIP 3.3 Delivers Find Me (One Number Locator/Simultaneous Ring) Services, Universal Call Transfer along with Additional Class 5 Features

IVR Technologies is pleased to announce the release of Talking SIP Version 3.3.

With a widely deployed customer base and the phenomenal growth of the 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 in reducing their management overhead and improving business efficiency. One of the ways we have achieved this objective in Talking SIP 3.3 is by doubling the maximum session capacity per server/blade from 240 callers to 480 callers when running on the latest processor technology, greatly reducing the network footprint, hardware cost and administrative overhead of the solution.

Talking SIP version 3.3 further enhances our Class 5 Feature set by adding many popular call control options that put your subscribers in control of their unique call handling requirements with centralized configuration control through easy web access. In the traditional PSTN networks call control was limited to hard to memorize DTMF key strokes. With the next-generation VoIP network these features are supported via the traditional DTMF input method, rich voice guided prompts, as well as via the ubiquitous web which all help to improve subscriber retention and usage. Having the ability to set Caller ID and Caller ID blocking parameters, selective call forwarding (by date range, time of day or day(s) of the week) and selective call blocking (silent disconnect, busy, fast busy, do not disturb, out of service) via the web greatly enhances your subscribers' control and empowerment.

Available as an option is our Find Me module that provides one number locator services that can sequentially or simultaneously ring multiple phones. The Find Me module, when coupled with our End-user Web interface provides subscriber control over which phones to ring, when, how and in what order. It includes options for music on hold, call screening, and confirmed answer. The Find Me module further expands Talking SIP's value proposition as well as its ability to help service providers grow their subscriber base through innovative and high-margin services.

We are extremely excited with the release of Talking SIP 3.3 as it is the culmination of a tremendous amount of time and effort from our team and we look forward to our customers deploying it and being able to realize its full benefit and potential.

Talking SIP 3.3's New Features

A Brief Overview of the New Features and Functionality

Talking SIP 3.3 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** – New calling features include station-to-station dialing, off-premise stations, call forward (all calls, busy, do not disturb, no answer and out-of-service) and anonymous call blocking/rejection.
- **Universal Call Transfer** – An innovative feature that allows the subscriber to easily transfer a call from one device (e.g. a landline) to another (e.g. a mobile) with preset DTMF keystrokes that can be set at the system or user level.
- **Flexible Session License Allocation** – This new white list functionality allows the service provider to restrict certain Talking SIP nodes to a subset of available session licenses. In addition, this functionality allows a more secure framework by requiring all nodes to be explicitly provisioned.
- **Reduced Administrative Overhead** – Quickly turn up new Talking SIP nodes with the copy node wizard that allows for the configuration settings of one Talking SIP node to be copied to another with just the click of the mouse.
- **Optional Find Me Services** – The Find Me module provides subscribers with the ability to have their calls hunt them from one device to another based on the user defined criteria (e.g. date range, time of day, day of week) or simultaneously with the ability to ring an unlimited number of user devices all at once.
- **Web Enabled Subscriber Call Control and Administration** – Subscribers can now manage from a single web login who can and cannot call a user on the system as well as what destinations a user can and cannot dial; they can even specify how inbound callers are blocked (e.g. silent disconnect, busy, fast busy, do not disturb, out of service and more). When coupled with our optional Intelligent Email Agent the system can even send an email to the subscriber informing them whenever a blocked caller tries to dial their number.

CUSTOMER SPOTLIGHT

Vx Telecom is one of the UK's fastest-growing telecoms and IT-enabled services companies, delivering great value voice, mobile and internet solutions to both business and residential sectors. The company currently operates in Australia, Canada, New Zealand, Sri Lanka, UK and USA and is growing at a rate of 30% a month. Customers appreciate the exceptional call quality and competitively-priced packages we offer, the ease with which the technology can be installed and used, and the excellent follow-up and loyalty services.

"Talking SIP has provided an extremely versatile, value-for-money solution for our company, allowing us to operate in six countries worldwide from our hub in the United Kingdom," said Raj Jeyaraj, Managing Director of Vx Telecom.

"Using a single provider has meant we can take huge leaps forward in all areas of our business, from call processing to billing, and this has enabled us to offer a very comprehensive service. The customer support team is friendly and helpful, and always quick to respond to any queries. We've been very impressed by the whole package."

Implementing Class 5 Features That Promote Independent Subscriber Call Control, Drive New Sources of High Margin Revenue and Improve Subscriber Retention

Today's service provider is pressured by declining per minute rates and declining profit per call not to mention increased competition. So how do you best compete? Do you base your value proposition on price? Do you base it on a features matrix? Or, do you base it on providing your subscribers with turnkey telecom solutions that promote user defined call control and value added services that attract and retains users?

We have seen the price per call fall in favor of bundled packages of minutes in US/Canada and we now see this trend evolving in Europe. So how do you re-build the value of a call and make your profits soar like in the 90s? It is our belief that to achieve revenue, profit margin and subscriber growth you must give your customers a reason other than price to subscribe to your services.

Offering attractive packages of destinations and competitive rates is important, but also tying them into pre-packaged solution sets that target the needs and wants of subscribers, from the basic home user to the power business user, is also necessary for any service provider offering retail services.

Custom tailoring services and features to your clientele that are easy-to-understand and are just as easy to remember and utilize will help to attract and retain customers. Your customer will in turn gladly promote your services within their circle of influence.

Providing these customers with web access for the self management of their services will empower them and provide them with a level of control and flexibility they have never experienced before.

Whether you are targeting the residential market, the business market, or both, Talking SIP provides you with features you need to grow your business.

Customizing calling plans has never been easier with the added ability of universal call transfer between any devices, at any time, coupled with the Find Me services and user defined access lists. With these new features your customers now have the ability to manage how they communicate, with whom they communicate and when they communicate.

MARKET RESEARCH

"The number of worldwide residential and SOHO VoIP subscribers will grow to about 172 million in 2010"
Infonetics

"Wholesale VoIP origination and termination revenues will experience a 30% compound annual growth rate (CAGR) through 2010. Local wholesale VoIP is the fastest growing segment and will account for 27% of wholesale revenues by 2010, up from 7% in 2006."
In-Stat Research

"Global telecom services revenue reached US $1.4 trillion in 2005, representing an 8.2% increase over 2004. Up to 2010, growth is expected to experience a five-year compound annual growth rate (CAGR) of 3.6%. Revenues are expected to increase by US $275 billion to reach almost US $1.7 trillion."
MarketResearch.com

CUSTOMER TESTIMONIAL

"IVR Technologies' Talking SIP server has been a phenomenal billing platform for Globecomm Network Service Corp. (GNSC). Our primary business is voice wholesale to Middle East countries. Originally we were looking at Talking SIP for a prepaid calling card service. GNSC had been using a billing platform that was antiquated for its wholesale service and was planning to continue to use the platform; however maintenance of the old billing platform came to a head and therefore GNSC fully migrated over to the Talking SIP system.

The Talking SIP system is currently handling 3,000,000 minutes per month with more room to grow. It has helped GNSC consolidate its voice billing system into a single platform for both prepaid and wholesale service. Talking SIP is remarkably scalable and redundant. For a system of its price, the Talking SIP system has remarkable features - features that other billing systems will charge as options.

The most important aspect of IVR Technologies is its customer support. They are both very responsive and very quick to help. The level of after sales service and the capabilities of Talking SIP is enabling GNSC to look for larger markets knowing that the back-end system is ready to cope with the increase work load."

Gary Lim, Globecomm Network Service Corp. (GNSC)

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.

Change the Amount Being Billed in the Middle of a Call

In today's competitive market landscape, oftentimes it is not sufficient to charge your customers based on a simple rating structure consisting of just a per call charge and per minute rate. Based on your cost structure, traffic patterns, and/or the terms of the program that you are offering you may want to raise or lower the cost of the call after a certain amount of time is consumed to encourage customers to talk for either a longer or shorter amount of time.

Talking SIP provides the ability for tiered rating. You can set up a rate plan where the rating parameters for the call change as the caller continues to talk. Each rate tier allows you to specify the duration of the tier, a per tier charge, and the per minute rate for that tier. The per minute rate allows you to adjust the rate up or down, and the per tier charge allows you to apply a charge to the account as the caller crosses a certain threshold.

For example, say you want to charge the customer $0.25 per minute for the first three minutes. Once the customer talks for three minutes, you'd like to charge an additional $0.10 and change the per minute rate to $0.30 per minute thereafter. This can easily be accomplished using rate tiers thereby allowing you to increase your profit the longer the customer talks. Each rate tier also allows you to specify a new set of billing increments and minimum seconds billed. This means that while you may have been billing in 60/6 billing increments when the call began, you can change the parameters to bill for an additional 120/60 after a certain amount of talk time. This further allows you to control the profit of the call whether it is for a short or long duration.

With tiered rating you also have the flexibility to add a disconnect charge in each tier that can be configured as either a flat charge or as a percentage of the amount billed for that tier. You can also modify the definition of a minute for just that tier allowing you to define a minute as something other than 60 seconds (54 seconds for example). These parameters can be configured to be included or omitted when the minute balance is spoken to the caller.

Talking SIP's ability to provide tiered rating allows you the ability to change the amount billed in the middle of the call. This gives you extreme flexibility to implement creative rating plans that help to drive revenue through your system and increase profitability.

Gain Greater Control Over Voice Quality with Codec Classes

Talking SIP now provides control over the voice quality of a call by allowing you to specify which codec will be selected and used when answering or placing calls.

There are many situations where being able to control the codec used for a call is important. For example, when a user first calls into the system, you may want to ensure that the prompts are played at the highest possible quality. Or, there may be a situation where you want to force a specific codec to be used for technical reasons. For example, when placing a callback call, Talking SIP uses 3PCC (Third Party Call Control) to send an INVITE to a device without specifying a codec, as it expects the answering device to respond with the codec it wants to use. However, some SIP devices may not yet support this feature, so it is necessary to specify a codec when sending an INVITE request under these circumstances.

Specifying a codec is accomplished in Talking SIP through the use of Codec Classes. Codec Classes are collections of codecs that specify the preference of codecs that will be used to answer an inbound call or will be offered during an outbound call. By creating different Codec Classes with different listed codecs, you can provide callers with different levels of voice quality. Users calling over high bandwidth networks can utilize higher bit-rate codecs and other users calling from international destinations over dial-up or low bandwidth networks can utilize more efficient lower bit-rate codecs.

As always, Talking SIP provides ultimate flexibility in where Codec Classes may be used. They can be assigned at the Device Module Map, DNIS Module Map and/or Outbound Route level so the codecs that are used can be matched with either the device or the application servicing the call.

With Codec Classes, Talking SIP provides an easy way to control which codecs will be used when answering or placing calls. This gives you control over the voice quality of a call by allowing you to adjust the codec for bandwidth limitations, as well as being able to overcome any technical issues that may be related to codec selection.

Utilizing Rate Intervals to Realize Additional Revenue or Attract Additional Customers

The telecommunications market is as competitive as ever which makes it challenging to attract and retain customers. With rates as low as ½ a cent per minute and in some cases flat rate, margins are getting squeezed from all sides.

How can the next-generation carrier stay competitive while remaining profitable? While Talking SIP offers many innovative ways to attract and retain customers through innovative and in-demand enhanced services many carriers' main line of business is traditional calling card where consumers are only interested in placing cost-effective long distance calls. How does one increase margins in these instances when competing with rates that are quickly approaching zero?

Fortunately, Talking SIP allows tremendous flexibility in being able to recoup additional revenues by charging daily, weekly or monthly administration/maintenance fees that can be based on usage. In addition, Talking SIP allows flat rate or percentage-based disconnect fees to be charged to callers with the option of being able to set whether or not these fees are disclosed to the caller.

Talking SIP also provides a nifty feature that allows the dimension of time over which callers are charged per minute rates to be altered, using a setting called the Rate Interval, which can be set at the Rate Schedule or Rate Plan level.

Using Rate Intervals, providers now have the ability to specify that a caller minute is 50 seconds rather than 60 seconds thus charging customers a 20% premium over posted rates. Alternatively, a provider can set a caller minute at 70 seconds thus providing a 15% discount to rates for callers. Providers can set whether or not changes to the Rate Interval (default 60) are disclosed to the caller and these changes can vary based on destination, time of day, day of week, date range and/or customer.

With flexible options like administration fees, disconnect charges and Rate Intervals, providers now have the ability to market competitive rates while being able to recoup additional hidden or disclosed fees to increase margins.

Note: Please check with local regulatory bodies about the ability and/or legality of varying the dimension of a minute before adjusting these settings.

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