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Industry Recognition



IVR Technologies has been named to the 2004 "pulver 100" - which is the VoIP industry's premiere listing of privately held growth companies that represent the future of the communications ecosystem. IVR Technologies has been named to the list due to its market experience, outstanding customer service and innovative product offerings.

The pulver 100 list, first introduced in September 2002, includes private companies in the communications sector that have substantial real-world deployments and enjoy significant growth rates. The value chain characterized by listed companies differs substantially from the vertically integrated telecom model of the last century. The companies prospering in the new environment have followed the computer and networking industry model with open interfaces, connectivity decoupled from services, and software decoupled from hardware. They have little or no dependence on the traditional telephone network, equipment, or service providers.



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.

Technical Tip 1



Supporting Multiple ANIs for Authentication in the Debit Module

To streamline callers authenticating into Talking SIP's Debit Calling Card module, a module level option called `DEB_ACCOUNT_ANILOOKUP` can be used. When this setting is set to 'Yes' it instructs Talking SIP to attempt to match the caller's incoming ANI with the `ALIAS` field(s) of an account in the system. If a match is found, the associated account number is used to automatically authenticate the caller. This functionality helps to increase profitability by reducing the amount of non-revenue time ("hang time") that a caller spends logging into the system prior to making their revenue generating outbound call.

If you wish to allow an account holder to use multiple ANIs to automatically authenticate into the system (e.g. a home number, an office number and a cellular number), and with `DEB_ACCOUNT_ANILOOKUP` set to 'Yes', you can perform the following:

1. When editing or creating a new account, select the 'Additional' tab, where you will find a new section called 'Account Aliases'.
2. For each ANI that you wish to allow the caller to use to automatically authenticate into the system, create a

new entry and set the 'Alias' field to the ANI that you wish to use for automatic authentication. The 'DNIS' field will typically be left at its default value - an asterisk (*), which allows the Alias value to be used for authentication regardless of the number used to access the platform. If you wish to allow the same ANI to be used to authenticate two different accounts, one or both of the accounts must have a specific DNIS access number specified. Note: The ANI specified in the `ALIAS` field must match exactly to the ANI being received by Talking SIP.

With the aforementioned configuration, callers can now have the freedom and flexibility to access your platform from multiple locations while being able to be automatically authenticated into the system so that they can quickly and readily reach their destinations. If a matching ANI/Alias cannot be found in the system the caller will be prompted to manually enter their account number via DTMF key press, unless `DEB_ANI_AUTHENTICATION_ONLY` is set to 'Yes', in which case the caller will be disconnected.

Technical Tip 2



Take Advantage of Flexible Multi-language Support

It is common for service providers to want to provide the ability for their customers to hear prompts in their native language. Talking SIP not only allows for this to occur, but provides a myriad of ways to accomplish this. Here are some of the most common

ways that Talking SIP supports multiple languages on the same platform.

employed, Talking SIP has already authenticated the caller by the time the first prompt is played and knows what that account's language setting is. If ANI authentication is not used, then the caller will hear the first prompt in the default system language and all subsequent prompts in the account's preferred language.

In many cases, however, the caller is using a prepaid card, and the service provider does not have any

First, the language can be specified by DNIS. If the service provider is targeting a certain ethnic market, then those accounts can be directed to call a specific access number that has been configured to answer with a specified language. In this case, the caller will hear all prompts, including the initial greeting, in the specified language.

But what if the service provider wishes to provide this functionality by account instead of by DNIS? No problem! Talking SIP also allows for each account, be it prepaid or postpaid, to be configured for a specific language. This option is particularly helpful when used in conjunction with ANI authentication. The reason for this is that Talking SIP doesn't know which account is calling in when it prompts the caller to enter his/her card number. Therefore the initial prompt for the account number needs to be played in the default system language. However when ANI authentication is

knowledge of who the caller is or what his/her preferred language may be. In these cases, Talking SIP allows for each DNIS to be provisioned with a language group. Each access number can be configured to play a language choice prompt allowing the caller to select the preferred language via DTMF. Each language group can be configured separately to play a different prompt while mapping different key presses to different languages. This provides the caller with the ultimate choice, even when nothing is known about the caller at the time of answering the call.

By using the methods outlined above, Talking SIP provides extremely flexible support for multiple languages on the same platform. The result is that the caller enjoys a uniquely rich call experience in their language of choice, while the service provider maintains an easy to manage and flexible system to meet the exacting needs of their customers.

Technical Tip 3



Digit Manipulation For Outbound Calls

In many circumstances, service providers need explicit control over how an outbound call will be placed to a terminating device, whether it is for the purpose of authentication or the application of specific routing logic. Talking SIP provides the ability for service providers to control how outbound calls are routed out of the system to different destinations. Each destination is encapsulated in an Outbound Route, which specifies the device used to terminate the call when it is placed. Outbound Routes can be grouped together into a Route Group, which is used by the system when an attempt is made to place an outbound call after the destination number has been successfully rated. Each route will be tried in order from first to last until an available route is found. Once found, the formatting rules of the Outbound Route can be used to identify the call to the terminating device and/or modify the destination number before the call is placed.

When identifying the call to the terminating device, we may wish to substitute specific values for the caller's identity (by 'identity', we essentially mean the ANI from the terminating device's perspective). Using some powerful formatting capabilities through the use of special keywords (such as "ANI" or "InboundIP"), Talking SIP allows the identity of the caller to be specified as:

- The ANI from the inbound device (after it has potentially been formatted as described in the previous article entitled 'Customize the Formatting of Incoming ANI!')
- The DNIS as it was presented by the inbound device.

- The IP address of the inbound device.
- The account number of the card holder.
- An explicit, user defined value or literal.
- The Origin code of the call, which is defined by the inbound device's configuration settings to identify the location or 'origin' of the call before it is rated.
- A combination of any of the above values.

These same keyword formatting capabilities (and a few others) can be used to manipulate the destination number in various ways such as:

- Specifying a specific digit or a series of digits that are to be stripped off of the front of the destination number before it is dialed by the outbound device. This is useful, for example, when an area code (NPA) needs to be stripped from the destination number before it is dialed because it is local to the outbound device.
- Specifying a digit or a series of digits that are added in front of the destination number before it is dialed by the outbound device. This is useful when an access code, such as "9," needs to be dialed before the destination number for all outbound calls on a given device.
- Specifying additional digits that are to be added to the end of the destination number before it is dialed by the outbound device.

By providing some extremely flexible and powerful formatting capabilities, Talking SIP allows you to have total control over how calls are routed out of the system as well as how they are presented to other SIP devices for termination.



Talking SIP Version 2.1.2 Now Available

We are pleased to announce the release of Talking SIP Version 2.1.2, to all Talking SIP licensees who are within a valid support agreement.

The following is just a brief list of some of the newly added features and enhancements that are contained within this version:

- Over 13 new, mostly graphical reports for traffic and licensing analysis.
- Comprehensive change to the underlying database architecture and structure for greater performance and scalability.
- The ability to authenticate based on the Origin/carrier code with the Tandem module for secure networks.
- Improved handling of info-digits and added the ability to block calls by info-digits (e.g. block all pay phone originated calls).
- Improved debug logging without requiring an application restart even for low level logging.
- Improved internationalization for date formats with call records and financial transactions now logged in universal time and displayed in local time for globally deployed networks.
- True multiple ANI authentications by allowing multiple aliases to be assigned to a single account without having to use Parent/Child account structures, which were previously required to implement this functionality.
- Added HTML based e-mails for clearer more professional presentation in the Intelligent E-mail Agent.
- Greater control over how data is exported from Talking SIP as well as the ability to readily import and export rate plans in .CSV format, when using Microsoft® Excel to manage rates.
- Added the ability to automatically import all 70+ reports in a single operation across multiple categories on **new databases**.
- Added the ability to assign security rights to provide/limit access to certain Sales Groups.
- Improved usability and clarity when Editing accounts.
- System tray balloon tips added for greater readability and notification of critical system errors and events.
- Origin added to the System Monitor to allow calls to be identified on how they were originated into the platform.
- Improved performance when having Talking SIP accessing system/module settings.

It is through valuable and greatly appreciated customer feedback and industry research that new features are added to Talking SIP to enhance the solution. If there is functionality or enhancements you would like to see in the product, please feel free to e-mail feedback@ivr.com -- your comments and suggestions are always welcome.



IVR Technologies' Customer Satisfaction Survey Results

It is our continued focus to provide our customers with the highest level of positive experience with both our product and pre/post sales support. In order to independently measure our customers feedback and as a benefit of IVR Technologies being a Certified Microsoft Development Partner, we were provided by Microsoft through the third party services of TNS, to conduct a Customer Satisfaction Survey. This survey was conducted in the month of October and limited to only our deployed customer base.

We appreciate the time and effort of those of you who participated. The survey provided us with valuable feedback on how we have performed to date in the areas of pre/post sales support, ease of doing business, product quality, strengths and improvements and customer loyalty. Once all of the results were tallied an overall satisfaction number is calculated and referred to as the NSAT score. $NSAT = \text{very satisfied \%} - (\text{somewhat dissatisfied} + \text{very dissatisfied \%}) + 100$. An NSAT score of 120+ is considered good, 81-119 is average. We are pleased to announce that IVR Technologies scored a 175 ranking it at the upper tier of net customer satisfaction!

We look forward to conducting additional customer satisfaction surveys in the future and have already learned and taken action based on this initial feedback that is evidenced in our latest Talking SIP version. Thank you again for your continued support of our company's product, services and people.

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