



Intervoice announces application suite supporting SALT and VoiceXML

Ten applications and an alerting service

On June 15, **Intervoice, Inc.** (Nasdaq: INTV) announced the general availability of its Omvia Voice Express applications suite, which were previewed in part in January (TSN, February 2004, p. 1). Unlike the vertical segment approach taken by many packaged-speech-application vendors, the

Intervoice suite is organized by functionality, with four groups: (1) general information management, (2) customer acquisition, (3) billing, and (4) customer care. The General Information Module

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Nuance customers discuss deployments

Talks at V-World cite specific payoffs

At **Nuance's** V-World Conference in May, Chuck Berger, the company's president and CEO, in his opening keynote address cited data from **Apex Research**: "Previously, companies turned to speech to reduce costs and improve customer satisfaction. Now, companies are listing 'competitive advantage,'

'improved productivity,' and 'revenue generation' as additional drivers for deploying speech." The following is a sampling of applications and payoffs

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ScanSoft's new version of its DialogModules eases application development

Memory savings and improved performance on deployed systems

On June 8, **ScanSoft, Inc.** (Nasdaq: SSFT) announced the availability of OpenSpeech DialogModules 2.0 software, ScanSoft's application building blocks used to accelerate application development and deployment. Each DialogModule generates VoiceXML code that encapsulates field experience and best-practice voice user interface design for a specific dialog task that is part of an application. When the task is repeated in several parts of the application or in several applications—

for example, confirming a choice with a yes-no answer—the modules also assure that the user interface is consistent. Included among the OpenSpeech DialogModules are capabilities for collecting dates, time, and addresses; for example, rather than writing code to collect a correct spelling for a name, developers can use the Name DialogModule.

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ScanSoft offers two versions of auto-attendant solution

Standalone product and VoiceXML-based product for IVRs

The SpeechWorks Division of **ScanSoft** announced two versions of voice-activated auto-attendant (VAAA) solutions in June: (1) the latest release of SpeechAttendant 8.1, its turnkey auto-attendant solution (originally obtained in the acquisition of **Locus Dialog**); and (2) OpenSpeech Attendant, a VoiceXML-based packaged application for use on independent IVR platforms. Both solutions direct calls by spoken name or business

function, automating a text-based corporate directory for internal calls, external calls, or both. With both solutions, callers can use natural phrases such as, "May I speak with Martha Jones." ScanSoft calculates that SpeechAttendant customers typically realize a return on their investment in less than one year because of reduced need for operators to route calls and shorter calls.

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Editor's Notes

Speech output—a bottleneck in application management

William S. Meisel, Publisher & Editor

Developing telephone speech recognition applications has gotten easier because of better development tools and various levels of packaging. With that improvement, the generation of voice prompts—speech output—often becomes the bottleneck, particularly in updating an application with frequent changes. Developers have to schedule a professional recording session with the same person who recorded previous prompts and wait for the result to be delivered before the system is updated. The cycle for upgrading is typically a minimum of two weeks. The cost of recording prompts can also be an issue. As applications become more complex and embody branded “personas,” issues arise such as the continuing availability of the specific voice over years.

Contrast this process with the relative ease with which a Web site can be updated. The difficulty of updating a speech application encourages a view of automated telephone interactions with customers as limited and inflexible. Speech output issues are only part of the problem—testing a revised speech application is more difficult than testing a Web application. Nevertheless, generating prompts becomes an increasing bottleneck as speech development and testing tools improve.

Text-To-Speech synthesis (TTS) is a potential solution to this problem. In this month's VUI Visions column (p. 14), Roberta Ishihara, an independent consultant who manages **TMA Associates'** TTS resource site (www.TTS-Update.com), notes that the limitations of TTS are often overstated because the tuning tools available aren't used. On the other hand, Dr. Ishihara notes that tuning tools available for TTS are often not used in testing an application, leading to a mistaken impression that TTS does not generate sufficiently natural speech. Tuning based on testing is conventionally done for speech recognition, but not for TTS. Performing that tuning can minimize the difference between custom recorded prompts and TTS.

If TTS is already available for the parts of applications where recorded speech is not practical (for example, because of the variability of entries in a database), there is little additional cost in using that resource to generate prompts. In cases where TTS would otherwise not be used in the runtime system, it can be used to generate prompts offline, storing the generated speech as recorded audio. Used in this way, one TTS license can be used to generate audio files that are treated as if they are custom recordings. The prompts can be generated and tuned quickly by the developer, rather than requiring a professional studio and actor.

The boundary between TTS and recorded speech is blurring. For example, the **Microsoft** Voice Prompt Engine, part of the Microsoft Speech Server development tools, uses a database of recorded prompts created by **GM Voices**, a professional recording service. The Prompt Engine can concatenate speech segments in the database using technology similar to that used by TTS engines. The result is more natural than typical concatenated recordings. Over time, one can expect the size of the prompt database and its coverage to expand. (Custom recordings by the same actor are available from GM Voices to mix with the prompts generated by the database.)

Similarly, a number of TTS vendors offer the option of blending TTS and recorded speech in the same voice by using an actor to create the TTS that is available for custom recordings. They make available the voice talent and charge a fee for such custom recordings. This allows the option, for example, of a tailored introductory message, but generating most of the prompts using TTS.

Speech applications should someday be as easy to modify as Web applications. When that is the case, it will help companies consider call centers as opportunities to engage a customer rather than a cost of doing business. TTS and prompt databases are partial but necessary enablers of this transition.

New release from TuVox expands flexibility of its call center solutions

Sales “surge,” Roxio selects TuVox solution to get a competitive advantage

TuVox has in the past emphasized its ability to automate complex call-center applications with speech, such as answering frequently asked technical support questions. The company’s “Conversational Voice Response” system includes statistical analysis tools that can analyze text sources to automatically provide a start for such applications, supplemented by tuning tools that ease adjusting the application based on agent knowledge and field experience.

With a new release of its software, TuVox CVR 4.0, announced on June 28 and planned for availability later this year, the company has added more features to ease application deployment, including increased support for more conventional transactional applications. In addition, a new suite of “SmartGen” tools in TuVox CVR 4.0 makes it easier for IT organizations to create, maintain, and enhance their own speech applications without depending on outside professional services.

Larry Miller, TuVox president and CEO, said that the company has seen sales “surge,” with four consecutive quarters of sales growth and 270% growth in year-to-year revenue leading to record revenue in the latest quarter. He indicated there was continued momentum on the customer, partnership, and technology fronts. As an example, TuVox cited new customers such as **MCI** and **Roxio**, provider of digital media software and owner of the Napster music service. Customers previously named include **TiVo**, **Activision**, and **Definity Health**. The TiVo technical support application has been live for over two years and served over one million customers, Miller noted.

Miller indicated that the company’s marketing theme was “Speech within Reach.” Steve Pollock, TuVox executive vice president and co-founder, said that a key objective was to reduce the customer’s total cost of ownership (TCO) by improving the total cycle of speech application design, development, integration, testing, deployment, monitoring, tuning, extending the application, and repeating that cycle after the extension. “TuVox’s vision is to deliver ubiquitous voice self-service that is better than a live agent, such that companies and end-users will choose automation instead of pressing ‘0,’” Pollock said.

Pollock said that TuVox architecture focuses on the applications and tools, leaving optional the choice of “commoditized” infrastructure—

VoiceXML/SALT platform, speech engine, and telephony platform. The company has been selling with a number of platform vendors, Pollock indicated, including **Convergys**, **Verascape**, **Genesys**, and **Softel**.

New release

TuVox CVR 4.0 expands the types of calls that can be automated, allowing call centers to automate calls of any type or complexity, including knowledge calls, routing calls, and transactional calls. Additionally, TuVox CVR 4.0 enhances the built-in “Perfect Agent” features, which are intended to make the interaction with the customer more natural. These features include:

- *Personalization:* Perfect Agent can use a database to use the caller’s name, remember pertinent information from the last call, and similar features that personalize the call.
- *Natural Language Routing:* Callers can be routed with a broad prompt such as, “What are you calling about?”
- *Consistent caller control:* Perfect Agent builds consistent navigation controls into each application. The caller can use commands such as “wait,” “go back,” “start over,” “help,” and “repeat.”
- *Anticipatory messaging:* Choices offered to callers can depend on their last action, rather than a “blind” menu, as a built-in feature of Perfect Agent.
- *Flexible integration with a live agent:* Callers can ask for an agent or be automatically directed to one if they are having trouble. They can continue to use the automated system while they are waiting for an agent. Agents also receive any information the automated system has gathered.
- *Adaptation in the field:* The system monitors calls and stores information that allows easier tuning of the system after it is deployed, even on a daily basis.

As part of the new release, a real-time monitoring feature alerts call-center staff to problems and updates while the application is running.

TuVox’s SmartGen development tools are integrated with Perfect Agent to allow rapid changes and extensions to be made quickly. SmartGen was also enhanced as part of TuVox CVR 4.0:

- *Database integration:* A new “DataBrowser” feature expands call types that can be automated to include sophisticated transactional calls by providing callers with voice access to information in a variety of databases.
- *Avoiding recognition errors:* The new release includes a new Predictive Recognition feature that allows developers to test new grammars and words for confusability before deployment. The system works, Pollock indicated, by comparing phonetic spellings of competing alternatives to see which are most acoustically similar and warning the developer of possible recognition problems before they occur, as well as designing recovery techniques specific to anticipated problems. Pollock said that this feature can avoid exposing callers to problems that can be anticipated.
- *More dialog application components:* Modules have been added to support transactions such as order status, claim status, and shopping cart manager. In addition, a Product Selector module allows specifying a product by name, type, size, style, or SKU.
- *Automated dialog builders:* The prior release allowed building a dialog from knowledge-base articles. In the new release, dialogs can be generated from more sources: Visio diagrams, database tables, and Web services. Dialogs can also be created dynamically as the application is running, generated from databases.
- *Prompt Studio and library:* To address the issue of creating and managing prompts (see Editor’s Notes, p. 3), tools for managing prompts have been added including a Prompt Studio for prompt workflow management that keeps track of what prompts need to be generated, a prompt library that can store and access prompts that have been recorded, and a “concatenated prompt

manager” that can assemble a series of recorded items into one prompt.

- *Voice content library:* In addition to prompt management, the library maintains a collection of phrase grammars that can be re-used.
- *Test-script generator:* The tools can be used to automatically generate a script that can be used to test the system.

Miller added, “We’ve also expanded our application set to support SALT in addition to VoiceXML, increasing the non-proprietary options for call centers.”

Roxio

Roxio plans to automate up to 30% of its calls. Gary Schultz, director of international support, Roxio, said, “As our business grows, we want to continue to provide a high level of customer service while increasing productivity. Using TuVox’s conversational speech solution, we can deliver high-quality 24x7 service to our callers while continuing to offer a live agent option. In addition, using conversational speech applications to automate calls will benefit customers by reducing wait times during seasonal spikes in call volumes.”

Kim Weins, TuVox vice president of marketing, said that Roxio will first address customer service automation and then turn to technical support. In customer service, the first applications will be handling software registration, refunds, calls regarding downloads, and requests for CD Keys to activate software.

Roxio considers the call center automation a competitive advantage. It will be able to offer its customers a hassle-free call-in experience at no charge, while many of its competitors limit access to customer service conducted by phone.

Voxify automated agents help with hotel reservations for Pegasus

Initial application handles confirmation and cancellation calls

Pegasus Solutions, Inc. (Nasdaq: PEGS) provides hotel reservations-related services and technology. Pegasus’ customers include a majority of the world’s travel agencies and more than 50,000 hotel properties around the globe. The company’s representation services, including *Utell by Pegasus*, are used by more than 7,300 member hotels in 140 countries, making Pegasus the hotel industry’s largest third-party marketing and reservations provider.

On June 21, **Voxify** announced that Pegasus has successfully deployed Voxify Automated Agents to enhance the telephone services it provides to more than 4,200 *Utell by Pegasus* member hotels. Initially, the Voxify solution handles confirmation and cancellation calls and may be extended to handle end-to-end reservations.

Nasir (Kass) Kassum, Senior Vice President of Data and Voice Services at Pegasus Solutions, said,

“We set the bar high for ourselves and our vendors. Because of this, we initially allowed Voxify to answer only 50% of our confirmation and cancellation calls. But Voxify met the challenge, deploying within eight weeks and achieving a call completion rate of over 95%. With Voxify handling the majority of confirmation and cancellation calls, Utell member hotels are benefiting from enhanced automated services, while we have newfound resource flexibility to help drive better customer service.”

Voxify has other deployments in the travel industry, including solutions for **CanJet Airlines** and **Travelocity** division **World Choice Travel**. Adeeb Shanaa, Chief Executive Officer of Voxify, said, “Working with Pegasus furthers our ability to serve the travel and hospitality industry. Together we look forward to the opportunity of enabling

leading travel businesses to easily and quickly deploy speech-automated services, reducing costs and increasing customer satisfaction.”

Voxify’s “Conversation Engine” is at the core of its Automated Agents. Voxify builds its applications on a structure that models how agents converse with customers (TSN, May 2004, p. 1). The underlying engine models interactions that are general in the agent-customer context, modeling the goals of the interaction and behavior of callers rather than the specific phrases they use. The specifics of the particular application are inserted into this context, making the development and dialog management process more efficient. Voxify has announced partnerships with **BeVocal** (for hosting its solutions), **Vocomo** (an IVR system for deploying on premises), and **ScanSoft** (for core speech technology). (See TSN, June 2004, p. 8.)

SandCherry will offer LumenVox speech recognition with its platform

Integrated solution provides a VoiceXML development and deployment environment

On June 8, **SandCherry** and **LumenVox** announced a partnership to offer affordable speech and touch-tone solutions to service providers and enterprises. SandCherry will offer LumenVox’s Speech Recognition Engine (SRE) and LV Speech Tuner with SandCherry’s platform, in particular, its AppDev VoiceXML development toolkit, its AppPackage pre-integrated application solutions, and custom solutions using the SandCherry SoftServer speech services platform.

Edward Miller, president of LumenVox, noted that the joint solution can reduce the cost of customer and employee services or create new revenue-generating services. Charles Corfield, president and CEO of SandCherry, said, “LumenVox and SandCherry share

the vision that high-quality, speech-enabled solutions should be available to any size company. Both companies have focused on simplifying the development, integration, and deployment of speech services while maintaining affordability.”

LumenVox has developed its own speech recognition technology, with the view toward making the technology more affordable. The company’s suite of speech recognition software includes the SRE, LV Speech Tuner (that can be used to tune the recognition engine for specific applications), and the Speech Driven Information System (an application development and deployment environment).

Pronexus application development tool for SALT available

Offers starter kit with Microsoft Speech Server and Intel Dialogic boards

ON June 2, **Pronexus Inc.** announced the release of VBSALT, its rapid application development environment for the **Microsoft** Speech Server 2004, which uses the Speech Application Language Tags (SALT) proposed standard. First announced at the launch of Microsoft Speech Server in March 2004, VBSALT 1.0 allows IT Managers, software programmers and call center developers to rapidly create speech applications using VB.NET or other

languages supported by the Visual Studio.NET platform. Pronexus also announced in June the availability of ‘Get-Up to Speech’, a rapid application developer bundle for Microsoft Speech Server. The bundle is available for a limited time from Pronexus or its distribution partner **Cygcom** for \$1,695. VBSALT is also available from Cygcom as well as Pronexus.

VBSALT development environment

VBSALT provides developers with a tool for visual call flow creation in Visual Studio.NET. It is designed to maximize developer productivity and reduce the learning curve for developers new to speech. VBSALT lets programmers leverage existing development skills in the languages supported in Visual Studio.NET, and features a comprehensive range of telephony and speech building blocks, encapsulating best practices for voice user interface (VUI) design. The product dynamically generates the SALT and JScript code required to control Microsoft Speech Server. Since the low-level control of Microsoft Speech Server is hidden from the application programmer, developers can focus on the business logic and voice user interface aspect of their applications.

As an example of the use of VBSALT, **Vocantas Inc.** and **Globis Data Inc.** previously announced the signing of an exclusive partnering agreement that will allow real-time traffic information to be available to drivers via the telephone (TSN, May 2004, p. 6). The agreement is for the development of a voice information portal prototype that combines Globis Data's D.R.I.V.E.S.T real-time traffic data with Vocantas' Interactive Voice Response (IVR) capability to provide drivers with a voice-only, hands-free solution for live traffic information.

Starter Package

The 'Get-Up to Speech' bundle consists of the following components:

- Microsoft Speech Server 2004 Standard Edition, 180-day Evaluation Edition.
- Pronexus VBSALT 1.0.
- **Intel** Dialogic D/41JCT-LS Combined Media Board: This four-port, analog, converged communications board is fully compatible with Microsoft Speech Server and supports voice, fax, and software-based speech recognition processing in a single PCI slot.
- Intel NetMerge Call Manager: This telephony interface manager (TIM) is a software module that provides integration of the Microsoft Speech Platform with the D/41JCT-LS telephony board.
- Intel Dialogic System Release Software: This software package includes all necessary drivers for the telephony boards as well as a suite of installation, configuration, and diagnostic utilities that can further simplify telephony server development.

"The market is ready for speech; reducing costs and increasing customer satisfaction, employee productivity and revenue-generating opportunities are what businesses of all sizes are focusing on," stated Jeff Valliant, President & CEO of CYGCOM. "The Microsoft Speech Server is the platform that will allow all businesses—including small, medium and large enterprises—to efficiently develop and deploy the kind of speech solutions that were historically reserved for the few large corporations that could afford it, and this bundle is an excellent way to get started quickly."

Vocomo announces new version of its IVR platform

VoiceXML-based platform now supports more options

On June 8, **Vocomo Corporation** announced the immediate availability of VocomoVoice Response 2.0, VoiceXML-based IVR platform. Jim Ozimek, vice president of national sales at Vocomo, said that there have more than 100 deployments of version 1.0, which was released 14 months ago.

New and enhanced features introduced in the VocomoVoice Response version 2.0 include:

- *Operating system support:* The system is now fully supported on both Microsoft Windows and Linux.
- *Enhanced VoiceXML support:* The system is now fully compatible with either VoiceXML 2.0 or 1.0, including full support of supporting grammar formats SRGS 1.0 and ABNF, and SSML 1.0 (Speech Synthesis Markup Language).
- *Extended grammar support:* Formats JSGF, **IBM** BNF, and Nuance GSL are now supported; as well as ECMA Script Action Tags for JSGF, **ScanSoft's** SpeechWorks OSR Tag Format, and **Nuance** NLU.
- *Expanded support for speech technologies:* The release now supports the latest SpeechWorks products from ScanSoft, including OpenSpeech Recognizer 3.0 and Speechify 3.0; IBM speech

technologies; **Rhetorical** rVoice 4.1 TTS; and **LumenVox**'s speech recognition engine.

- *Operations and administration:* The release adds a Web-based administration console for system remote configuration and management; advanced call logging, statistics, and reporting; and an email alert mechanism.
- *Telephony connectivity:* The new release supports analog and digital telephony connectivity via **Brooktrout** TR1000 boards and **Intel** Dialogic JCT series boards; and Voice over IP using Dialogic's Host-based Media Processing software.

Vocomo's platform is available to integrators such as **Quality Call Solutions**, a provider of integrated

solutions for call centers. Johnny Lin, CEO of the company, said, "Vocomo's exceedingly capable turnkey IVR systems with their reliability and flexibility create business opportunities for us to develop VoiceXML applications and deploy them quickly on a consistent hardware/software platform."

Garret Blake, IT Manager for **Provident Funding**, a leading mortgage lender, emphasized the platform's compatibility with Provident's existing Web services. "Vocomo's sophisticated IVR systems easily tie the power of speech to the backend business database for voice driven data access that rivals the ease of use of online systems," he said.

Scotiabank tests Vocent speaker verification application

Multi-factor authentication a requirement for sufficient accuracy

Vocent first discussed its Vocent DecisionMaker speaker verification application in 2003 (SRU # 123, September 2003, p. 1), indicating that a key factor in successful deployments for call centers is employing multiple factors in the authentication process, rather than just depending on the biometric measure of the caller's speech characteristics. On June 7, the company provided further evidence of this assertion by announcing results of a long trial at **Scotiabank**, a \$200 billion Canada-based international financial services institution. As part of the announcement, the company discussed Vocent Confirmed Caller, a packaged application that incorporates Vocent DecisionMaker. The application uses either **Nuance** Verifier or **ScanSoft** SpeechSecure as underlying speaker-verification technology, as well as their speech recognition technology when required to gather supporting authentication information.

DecisionMaker integrates biometrics, data analytics, and content knowledge to make the best possible authentication decision. Scotiabank conducted a pilot project to evaluate a variety of technologies and methodologies for authenticating customers over the telephone.

The bank piloted Vocent's voice authentication solution as an alternative to content-based approaches, in which agents require customers to answer personal questions or recall personal identification numbers. The content-based approaches can be insecure and create customer discomfort by relying on personal information that

can often be easily discovered or memorized information that can be forgotten.

Vocent's multi-factor system integrates the ease of voice biometrics along with other factors, such as content knowledge and the caller's phone number, with the additional security layers of advanced logic, risk modeling, and dynamic dialog management. During the trial, Scotiabank successfully authenticated 95% of callers, while reducing false accept rates to below 1%. The Vocent system also helps to improve the customer experience by reducing the number of questions callers are asked during identity verification, asking questions only when the voice authentication or other information such as the phone number can't provide sufficient confidence in the identification. Scotiabank also evaluated authentication systems that use only voice biometrics, but—according to Vocent—discovered that such systems do not offer strong enough security or sufficient automation rates for large-scale consumer deployments.

"While voice authentication solutions offer very compelling benefits, key metrics for security and automation must be met," said Rick Davidson, Director, Electronic Banking, Channel R&D at Scotiabank. "Vocent's approach has proven far more robust than any of the other voice authentication systems we have tested, and is the only system we have tested that delivers the security and automation rates we require for consumer deployment."

Vocent DecisionMaker is now commercially available as part of Vocent Confirmed Caller and

Vocent Password Reset, which can be purchased directly from Vocent or through its expanding sales channel. Companies using Vocent solutions include

Marriott, US Bank, and Inovant (a Visa Solutions company).

US Cellular multimodal applications deliver Directory Assistance and ring tones

Phonetic Systems and V-Enable partner to deliver solution

Phonetic Systems and **V-Enable** previously announced a strategic alliance that provides for a global joint sales, marketing, and support relationship for multimodal solutions (SRU # 127, January 2004, p. 1). Phonetic Systems has its own speech technology embodied in its Voice Search Engine (VSE), which is optimized specifically for voice-searching very large directories or any size database with frequently changing information. V-Enable provides veANYWAY multimodal technology developed using X+V standards, delivered as carrier infrastructure software for mobile applications.

In June, the companies announced that **US Cellular** has launched two multimodal applications for their customers using its joint solution. SAY IT BIZFINDER and SAY IT TONEFINDER allow callers to use voice commands, instead of having to triple-tap buttons to enter text, in order to download information, directions, and ring tones. SAY IT. BIZFINDER and SAY IT. TONEFINDER are currently available on BREW-enabled phones.

SAY IT BIZFINDER is a multimodal directory assistance application that will allow callers to search, find, and locate business listings nationwide by simply speaking their input. The application can provide the conventional phone number, but it can do a lot more, like retrieve driving directions, maps, placing a call, and more. SAY IT TONEFINDER allows users to search, select, and download ring tones using their voice. For example, users can speak the name of an artist or song title and retrieve the ring tones that match their input, instead of having to type in the text on a small keypad. The combined applications received over twenty thousand hits in just the first week of availability.

Dipanshu Sharma, founder & CTO at V-Enable, said, "With more than 100 companies working on multimodality, V-Enable takes pride in bringing all of our vision of multimodality to market. In the initial weeks, we have learned that consumers are, by far, choosing multimodal interfaces to the traditional interfaces."

Kirusa multimodal platform to be used in Austrian research project

FTW to build several multimodal applications in demonstration project

Forschungszentrum Telekommunikation Wien (FTW, Telecommunications Research Center Vienna) is a joint research center for many telecommunications companies in Austria. It brings together departments from the **Vienna University of Technology**, several well-known telecommunications companies, some smaller enterprises, the **Austrian Research Center Seibersdorf**, and the **Association of the Electrotechnical and Electronics Industry**. In June, Kirusa announced that FTW will use Kirusa's Multimodal Solutions in its research program on mobile multimodal user interfaces. The research program, funded by leading carriers and information technology providers in Austria, involves building and showcasing several multimodal applications for delivery on a wide range of mobile devices on 2.5G and 3G wireless networks.

The use of multimodal applications in wireless networks has not grown as fast as earlier projections by some companies and analysts, in part because the wireless devices suitable for advanced multi-modal applications and the high-speed networks that propel the most data-hungry applications have not grown as fast as expected. Another hurdle is, however, determining which applications will motivate consumers. The research project is targeted at answering that question.

FTW's MONA project (Mobile multimodal Next generation Applications, mona.ftw.at) is funded by **Kapsch CarrierCom AG, Mobilkom Austria AG, and Siemens Österreich AG** together with the Austrian competence center program **Kplus**. The project is showcasing the value of multimodality in a messaging application, called MONA@Work, and in a multi user real-time game, MONA@Play. With the latter, consumers can entertain themselves on their

phones with a “Who Wants to be a Millionaire” style game.

Kirusa’s Circuit Voice multimodal platform, IP Voice Multimodal Platform, and IP Voice Multimodal Clients have been integrated into FTW’s MONA architecture. Kirusa’s multimodal solutions allow mobile users to simultaneously or alternatively use voice, text, graphics, keypad, and stylus to

interface with wireless services and applications. The solution integrates with industry standard technologies such as SMS, WAP, XHTML, VoiceXML, and HTTP, and supports all major mobile networks, including CDMA, CDMA 1x/3x, GSM, GPRS, UMTS, Flash OFDM, WiFi, and others.

Verizon Wireless and OnStar offer joint plan

Calling plan integrates handheld wireless phones with embedded in-car phones

OnStar provides wireless services integrated with an automobile, including voice-activated services that allow hands-free access to data and information. The service is available on more than 50 models from **General Motors** and select models from six other auto manufacturers. Customers who wanted to use the service were faced with having, in effect, one phone in the automobile and a separate wireless phone and service if they wanted portability.

On June 16, OnStar and Verizon Wireless, the nation’s leading wireless service provider, addressed this problem with the *America’s Choice with OnStar* service plan. With the new plan, customers can use their Verizon Wireless phones outside the vehicle and send calls from their handheld phones to OnStar-embedded in-vehicle phones. Customers can select from two options:

- Immediate Call Forwarding – Customers can forward all calls from their Verizon Wireless handheld phones directly to their in-vehicle OnStar phones.

- Conditional Call Forwarding – Customers can select to have the Verizon Wireless handheld phones ring four to five times before the call is forwarded to the in-vehicle OnStar phone. With this option, subscribers have the flexibility of answering either their handheld or in-vehicle phone.

Customers receive one bill. John Stratton, chief marketing officer for Verizon Wireless, summarized, “The America’s Choice with OnStar calling plan lets our customers use their Verizon Wireless minutes inside their OnStar-equipped vehicles, giving them access to our award-winning wireless service with the additional conveniences of hands-free voice-activated dialing and a single bill.”

OnStar’s improved digital hardware allows for more secure calling, improved speech recognition, and faster, continuous digit dialing. The embedded digital phone is compatible with CDMA wireless service.

VoiceXML Forum offers developer certification

Tests knowledge of VoiceXML and associated standards

When the **VoiceXML Forum** turned over the VoiceXML 1.0 specification to the **W3C Voice Browser working group** to manage the standard (resulting in the development of VoiceXML 2.0 by that group), they turned to other efforts to support VoiceXML. One was to create a Developer Certification program. On June 1, the Forum announced that testing for the certification was now available.

The VoiceXML Developer Certification Exam will be administered by Thomson Prometric (www.prometric.com), a worldwide provider of

comprehensive technology-based testing and assessment services. Testing is now available through the **Thomson Prometric** global network of testing centers. Developers can register for the exam at www.voicexml.org. The cost of the exam is \$150.00.

Jim Larson, manager of Advanced Human Input/Output, Network Building Block Division at **Intel Corporation** and co-chair of the **World Wide Web Consortium’s** (W3C’s) Voice Browser Working Group, noted, “The VoiceXML Developer Certification Exam will enable companies to audit

their development teams' familiarity with best practices and serve as an important credential for individuals seeking to join those teams."

The VoiceXML Developer Certification Exam measures a developer's knowledge of the W3C's Speech Interface Framework languages required to develop speech applications, including VoiceXML 2.0, Speech Synthesis Markup Language (SSML), Speech Recognition Grammar Specification (SRGS),

Semantic Interpretation Language, and the XML call control language (CCXML). It will also test how developers use these languages to create VoiceXML applications.

The VoiceXML Developer Certification distinction is awarded to anyone who passes the test. Certified developers may choose to be listed on the VoiceXML Forum website.

Interview with Roxann Swanson, Nortel Networks

Customers "demand" future speech applications be written in open standards

Roxann Swanson, Vice President and General Manager for Nortel Networks Enterprise Multimedia Applications, was interviewed by Bill Meisel in late June. Swanson is responsible for the technical development and product delivery of Nortel Networks Global Enterprise Applications, including the contact center, IVR, and messaging portfolios. Swanson has more than 25 years experience in the telecommunications industry. Prior to Nortel Networks, Swanson served as district sales manager for major markets at AT&T.

How do Nortel's products for the call center support speech technology?

Nortel Networks has taken a comprehensive approach to support speech in our customer contact portfolio. Our self-service platforms—the Media Processing Server 500 and 1000 (including our VPS/is solution which was a "Periphonics" product)—support speech applications and we have a dedicated speech server, Open Signaling Computing Analysis Resource (OSCAR), which is specifically architected for speech technology.

It is imperative to integrate speech seamlessly into the contact center, as Nortel Networks solutions do, to maximize the level of customer service and to assure that no information about the customer is lost. Nortel Networks call center management products, such as the Symposium Call Center Server, are tightly integrated with our speech enabled self-service solutions (a.k.a. IVR) which facilitates transfer of customer to an appropriately skilled agent and the delivery of customer information to the agent desktop.

How important is support of speech technology to customers?

All our customers are placing a great deal of value on speech. Speech enhances the two basic benefits of our self-service solutions; it improves customer service and reduces costs. Every survey I have seen reinforces the conclusion that a speech-enabled solution yields greater user acceptance. That means users prefer the speech-enabled solution. And the greater acceptance magnifies the financial benefits of off-loading calls from agents to the automated system.

Virtually every organization evaluating self-service solutions demands that the solution supports speech technology. About half of the ports we deployed the past couple of years have been speech-enabled and almost every RFP received in the past year has had speech as a requirement. Even those organizations planning to deploy a DTMF solution now demand that the underlying technology support advanced speech technologies.

How important are standards such as VoiceXML and SALT?

Standards are very important to our customers and to the market in general. Established standards are a sign of maturity, of general acceptance of this technology. Applications have greater portability, giving customers a greater choice of platforms, and standards create a greater community of development resources. Bottom line, the adoption of standards reduces the time and cost to deploy applications.

The VoiceXML standard in particular is important to customers since it allows them to leverage their investment in web self-service applications. Most of our customers tell us they want future applications written using standards such as VoiceXML. In fact, during a customer focus group earlier this year a number

of customers told us they demanded future applications be written in open standards even if it costs more to develop than using a proprietary GUI tool.

Nortel Networks is committed to using open standards—our products are compliant with the latest W3C standard for VoiceXML. In fact, every one of our application development personnel is either experienced in using VoiceXML or currently in training.

What do you feel distinguishes Nortel products from the competition?

Thousands of customers have selected Nortel Networks self-service solutions for a number of reasons. To paraphrase a common adage, the three biggest factors are experience, experience, and experience. Nortel Networks has unparalleled experience in designing and implementing both DTMF and speech-enabled self-service solutions which have a proven record of success in improving customer service and reducing costs. In most situations, we have experience deploying a solution identical or similar to what the customer needs and much of this experience is encapsulated in our portfolio of packaged application modules. That means we can deploy solutions quicker.

Other frequently cited factors are:

- *Breadth of portfolio:* a single source for all components of the solution, including the telecom system, the (Symposium) call center management system, the IVR system including the speech server, speech technology, solution design, and VUI design. We have strategic partnerships with industry leaders to assure best-of-breed solutions.
- *Technology evolution:* Nortel Networks has a proven record of evolving solutions with advancements in technology. For example, today most customers want their solutions developed using open-standard-based tools, such as VoiceXML for speech applications. Most suppliers have proprietary GUI tools. Nortel Networks customers can deploy new applications using VoiceXML—as part of our comprehensive Web-Centric Self-Service portfolio—and they can use these tools to interface with our GUI-based PeriProducer tools. They do not have to discard their investment in previous applications to satisfy current requirements. You will see that Session Initiation Protocol (SIP) will play an important role in future customer service solutions, and, Nortel Networks is the leader in SIP technology.
- *Ongoing support:* Nortel Networks provides full life-cycle support from presales solution definition to implementation to field support and ongoing application tuning. Nortel Networks will be around to support these applications for their entire life. We are a multibillion dollar company with over 115 years of stability.

Please provide examples of recent deployments and how they paid off for customers.

The most exciting deployment of the past twelve months was the comprehensive customer service application at one of the largest financial institutions in the world. It's our understanding that this was the largest single deployment in the history of speech enabled applications—essentially the size of 8-10 large, complex deployments. It's a unified interface to every one of the bank's customer service functions. The system is planned to handle over 600 million calls a year—4,000 toll free numbers are combined into one solution. Once a customer enters the system they can access any information relevant to their account(s) without having to re-enter, revalidate, and reorient themselves to different instructions.

This system is being systematically deployed throughout the bank's North American regions. Currently it is yielding over 70% self-service efficiency and every percent efficiency gain is worth millions of dollars in operating savings.

Any final comments?

Nortel Networks is committed to supplying world class self-service solutions, and that means we are committed to advanced speech technologies. All future developments, from Web-Centric Self-Service, to packaged applications to hosted applications will be speech-enabled.

Interview with Marty Parker and Tore Christensen of Avaya

Speech technology important in both call center and messaging solutions

Marty Parker, Director, Business Management, Messaging and Unified Communications, for the Enterprise Communications Group at Avaya, and Tore Christensen, a product manager for the self service portion of Avaya's Contact Center portfolio, were interviewed by Bill Meisel in late June, providing views of both the messaging and call center sides of Avaya's business. Parker was a General Manager and then VP Marketing for a leading West Coast Telecom Interconnect company, where his team brought the early voice mail systems to the marketplace. Marty was then CEO of both hardware-based and software-based voice-messaging startup companies. Marty received his BS from the Haas Business School of the University of California, Berkeley. Christensen has 24 years of experience in the telecommunications industry ranging from PBX software and hardware to call center and e-services applications. Tore has a BS in Electrical Engineering from the University of Minnesota, a MS in Electrical Engineering from Stanford University, and a Masters Certificate in Project Management from George Washington University.

Meisel: Avaya uses speech technology in both its internal communications applications and customer-facing call-center applications. Can you first summarize which internal products use speech technology?

Parker: The internal communications applications, which we put into the categories of messaging, unified communication, and self-service, all use speech technology (of course, beyond basic codec functionality):

- Avaya Messaging systems use text-to-speech (TTS) for text message playback, from both e-mail stores and Avaya Message Storage Servers, as well as for converting text names and message headers to speech formats.
- Avaya Unified Communication Center (UCC) uses speech recognition for a very powerful and flexible user speech command interface to all forms of business communications, including voice messages, text/e-mail messages, calendar appointments, tasks, directories, name dialing, conferences and more. In addition, UCC uses TTS for audible playback of all the information and service types listed here.

And in call center solutions?

Christensen: Avaya Interactive Response can be used for internal employee-facing self-service applications such as in Human Resources, but is more often used in customer-facing self-service applications. Avaya Interactive Response utilizes speech recognition, TTS, speaker verification, and dialog modules. It utilizes these technologies for a variety of self-service applications from stand-alone IVR applications to integrated call-center routing solutions.

Do these products use standards such as VoiceXML, and whose speech technologies do you support?

Parker: The use of VoiceXML varies by product type and applications. Our new Modular Messaging product uses VoiceXML for the Intuity AUDIX Touchtone Telephone User Interface that will be Generally Available in September 2004. This enabled a faster, more economic development cycle. Avaya Unified Communication Center uses VoiceXML for the Speech User Interface. This allowed speed and flexibility for development. In addition, some custom work has been done, leveraging the VoiceXML capabilities, for access to customer-specific data and services.

Christensen: Avaya Interactive Response supports a variety of application development and deployment models. It supports VoiceXML 2.0 applications development, both through standard VoiceXML, and also utilizing our Speech Applications Builder development environment. In addition Avaya IR supports our traditional scripting environment to allow legacy customers to migrate forward. Avaya resells and integrates with both **Nuance** and **Scansoft** speech products.

Can you give me some examples of customers using speech-enabled internal applications?

Parker: Avaya Unified Communication Center is being used by many types of companies for their mobile and field personnel, since it allows hands-free and eyes-free operation (critical in a vehicle). Examples include:

- An East Coast Consulting firm, **J. H. Cohn**, is billing upwards of an additional \$20,000 per year for their 300 consultants, producing a 6,000% ROI against the \$20/user/month operating cost.
- Sales forces in a large medical supply company, a large electrical products company, a large consumer electronics company, and a telecommunications services company are able to see two or more additional customers per week and shorten their sales cycles to produce revenue increases that return ROIs in the range of 4,000%.
- Field Customer Service Organizations, including many Insurance and Financial Services companies, are able to increase responsiveness as well as increase the number of customers visited per week. The hiring cost avoidance of the addition coverage is reported to generate up to 2,000% ROI.

Christensen: Avaya Interactive Response is used in conjunction with internal Contact Centers for speech-enabled applications. Avaya ATAC, utilizing a Natural Language Call Routing application, provides self- and assisted-service for pre-sales assistance.

Can you outline your call-center solutions that use speech technology?

Christensen: Avaya Interactive Response is an integral part of the Avaya Customer Interaction Suite. As such it can be used as a component in a larger contact center solution for Natural Language Call Routing, Self and Assisted Service, and Robotic Agents:

- *Natural Language Call Routing* is an application that is used in conjunction with contact centers to allow customers to interact with the system to direct their call to the correct agent using natural language instead of trees of menus. This also has a measuring system and a feedback loop to tune the application.
- *Self and Assisted Service* is an application that can provide self-service, but also can transfer the call and any partial results to an agent to assist the caller if desired.
- *Robotic Agents* is an application that integrates the IR with an outbound dialer to allow for automated alerts and proactive interactions.

Are the call center applications based on VoiceXML, and, again, which speech technology is used in these products?

Christensen: These call center applications are based on VoiceXML. The VoiceXML applications can also call existing legacy script applications that customers may have from their existing Conversant platforms.

Can you give examples of deployed call center applications?

Christensen: Avaya Interactive Response is used in conjunction with many large contact centers for speech-enabled applications:

- *HealthCare:* **Aetna** uses speech for benefits enrollment, change of address, card request, PCP referrals... ROI for the speech application was 4 months.
- *Financial Services:* **GE Consumer Finance** uses speech for credit card servicing, with 1000 port VoiceXML and speech application across four sites.

Any final comments?

Parker: The critical factor for speech applications is the business result, not the technology itself. While speech recognition demonstrations are impressive, it is the results produced by the complete solution—products, services, knowledge and methodology—that make the solutions compelling. Avaya has demonstrated market success in delivering and tracking these results, which is a driver of our market position and growth in this area.

VUI Visions

Does TTS = Truly Temporary Solution?

Roberta Ishihara, independent consultant

Each month, we ask an expert on an aspect of Voice User Interface design to address a specific issue in delivering a good experience to a caller. This month, Robbie Ishihara addresses the role of speech output, specifically text-to-speech synthesis, in VUI design choices. Dr. Ishihara has designed voice user interfaces for products on desktop, handheld, wearable, and telephony platforms in markets such as the financial, medical, messaging, and warehousing industries, and provides consulting services in these areas. She also manages TMA Associates' TTS Update web site (www.tts-update.com), a vendor-independent resource on TTS.

When you make a call to an automated service, whether it is to make an airline reservation, book a taxi, or obtain your checking account balance, what is it that you remember most about the conversation once you've hung up the phone? It's probably not the words and phrases that you spoke into the phone, but rather what you heard coming out of it. Although the capabilities of a speech recognizer are important, this observation underscores the value of an effective voice output strategy for a voice application.

Successful contact centers devote considerable resources to training, tracking, and retaining call center agents. Automating calls can help by freeing agents from routine requests so that they can concentrate on providing exceptional service to callers with special situations. In fact, asking call center agents to handle routine calls can be irritating, annoying, and morale-damaging for them. There is no question that using automation can be very valuable to a call center's operations. But it is essential that a company extend its call center's standards of efficiency and pleasantness through whatever automated interaction is put in place. Therefore, when incorporating speech technologies into automated solutions, it is often assumed that callers want and need to hear and interact with a human voice.

So in many cases where it is possible to anticipate all application prompts¹, companies decide to find, contract, and coach human voice talent (or contract companies that do so). If Text-To-Speech (TTS) synthesis is available to them, perhaps the companies will ask their developers to use TTS-rendered prompts as placeholders, until the human voice recordings are available. The development and maybe even demo phases of an application are run with TTS prompts, with a caveat that they are *only temporary* and that real, human voices will be substituted prior to production or deployment.

Unfortunately, since the TTS prompts are considered to be simply placeholders, no concerted effort is spent in trying to polish and tune them. As with speech recognition technology, out-of-the-box TTS is not the best that it can be, but is rather a starting point from which to improve. It is unlikely that any company would deploy a speech recognition engine without tuning. The same holds true of TTS engines. Nevertheless, I wonder how many companies have seriously investigated the use of TTS for their applications. Moreover, every time an in-progress version of an application is run with untuned, *only temporary* TTS prompts, the more the out-of-box TTS performance is heard, and the more it is judged unfairly to be less than acceptable for deployment. It is easy to hear errors and even slight aberrations from normal prosody (pitch, stress, intonation, cadence, etc.), all of which contribute to one of the perceived disadvantages of TTS—lack of naturalness.

These problems can often be corrected, but developers may be unaware that this option is available. There are some effective tools and methods for tuning, such as versatile developer's toolkits (by several providers), *sculpted prompts* (by **Rhetorical**), and domain-specific tags (by **ScanSoft**), to mention just a few. Instead, however, the decision is often made to use voice talent and recorded speech, despite its cost and logistical challenges.

But consider that almost all applications need to convey dynamic information during a dialogue. It might be an account number, a dollar amount, or a proper name. If a domain of information is completely variable, there is no choice other than to use TTS for it. In this case, issues arise as to how best to integrate TTS with

¹ There are, of course, applications where recording all instances of output isn't possible, e.g., some database entries or emails, and TTS is a requirement.

recorded speech. The most effective solution, though a costly one, is to use a single voice talent for both the TTS voice and the application's prompts.

Of course, if the dynamic information is finite (or composed of a finite number of parts), it can be recorded along with other prompts during recording sessions with the voice talent. Although the resulting interface might consist of human speech exclusively, it is not perfect. The challenge in this case becomes one of concatenating the segments of recorded speech in the most accurate, natural, and pleasing manner possible. Even if all the static and dynamic information can be anticipated and recorded in every context that will be required at runtime, the transitions between concatenated segments are usually still perceptible. Consider the typical example below, where the plus sign indicates a point of concatenation.

The account number + you said is + five + four + two + five. + Is that correct?

In fact, because such incongruence is unexpected by a caller who thinks he is listening to human speech, it emphasizes the artificial aspect of the interface. So it is not the case that simply using recorded human speech exclusively will result in a more natural, more accurate, or more pleasing interface than using TTS.

When a person calls in to a company, it is with a distinct purpose in mind. The goal is to obtain information or to complete a transaction. As long as the goal is accomplished, the caller will be satisfied. There always will be some people who resist talking to an automated interface, but they may not distinguish between TTS and concatenated recorded audio in this regard when they zero out to an operator. Given the high quality of TTS voices available today, callers who interact with TTS-rendered prompts can be happy callers if the dialogue is designed and executed properly.

In conclusion, when defining the output strategy for your application, consider carefully if, how, and when to use recorded audio and TTS. There are many advantages to using TTS over recorded speech, but one of the most compelling may be accuracy. This point is ironic because accuracy is often cited as a reason that TTS is avoided. But on a personal note, I have yet to hear a TTS rendering of my last name that is as unintelligible as what some humans have spoken to me. I would rather listen to a TTS engine pronounce my name than to a human call center agent mispronounce it, then ask me how to say it, and then apologize for getting it wrong in the first place. By using TTS for more than *only temporary* purposes, companies can benefit by enabling personalization and an unbounded domain of prompts, by eliminating jarring transitions with recorded speech, and by foregoing the use of voice talent services.

Intervoice (cont.)

Continued from page 1

allows customers to browse through products and services of interest while enabling them to generate information requests for future purchase considerations. The Customer Acquisition Module enables customers to order the product or service of their choice while also allowing their order to be fulfilled and confirmed. With the Billing Module, end users are provided with a method to pay for goods and services through the generation, collection, and confirmation of relevant billing information. Finally, in the event a customer needs service, the Customer Care Module provides service requests, service scheduling, service confirmation, returns/exchanges, and account information requests. Additional details on the module functionality are included in a table at the end of this article.

Intervoice also announced separately in June another part of the Omvia Voice Express Packaged Applications suite, the Omvia Notification service.

The notification service represents an alliance with **Emergency Email Network, Inc.** (EEN) to deliver a voice-enabled emergency event notification service for the government sector. Intervoice and **Apptera** have also previously announced that Intervoice will re-sell Apptera's suite of financial services applications under the company's Omvia Voice Express brand (TSN, June 2004, p. 9). Apptera applications include account information, funds transfer, name & address capture, ATM/branch locator, rate quote, and account payment.

The packaged applications support both Speech Application Language Tags (SALT) and Voice Extensible Mark-up Language (VoiceXML). Mike Segura, director of product marketing at Intervoice, said that current deployments of components of the suite that Intervoice can announce are all 24-port SALT versions. They are at **Mary Kay Cosmetics** (birthday congratulations notification to distributors), **Talbot's** (account information and gift cards), **Huntington Bank** (location services), and **NY Department of Education** (grades and other student account information).

Deployment options

The solutions reduce both cost and deployment time by including the application, documentation, on-line help, bug fixes, and upgrades in one package price. Segura said that modular construction allows adding and deleting features as needed. Omvia Voice Express applications can be in a number of ways—turn-key solutions, application templates, applications as a service (ASP), configurable applications, and developer applications (that a developer can use as a starting point).

Enterprises can configure an Omvia Voice Express application to their needs through tools that support both .NET and J2EE environments. They can choose speech technologies through the applications' support of the Omvia Voice Framework, which supports both .NET and J2EE environments, as well as speech technologies including **Microsoft** Speech Server 2004 [with either the Interoice Telephony Interface Manager (TIM) or **Intel** TIM], Interoice Media Server for VoiceXML, and Interoice Computer Telephony Integration (CTI) options for call center deployments.

Pricing

The packages are priced to be affordable for small-to-medium business, with some of the cost of customization removed. Pricing, as previously announced (TSN, April 2004, p. 11), is \$10,000-\$15,000 per application for four ports, \$20,000-\$59,000 for 16 analog or 24 digital ports, and "enterprise" (48 ports or more) pricing of \$30,000-\$40,000 per application (the last with an additional \$100-\$800 per port). As a service, the applications are available with usage pricing, starting at 8¢ per transaction plus 15¢ per minute (with a setup fee from \$10,000 and an annual database management fee starting at \$4,000). The notification (alert) pricing is \$10-\$40 per person per year. The Aptera applications range from \$43,200 to \$59,200 for up to 24 ports, and \$40,000 plus \$800 per port for 48 ports or more.

Notification service

The Omvia Notification service, provided in partnership with EEN, combines monitoring and detection with push technology to virtually any wireline, wireless, or Internet-enabled device, including voice telephones, fax, pager, email, PDA, SMS-text messages, or a combination of devices.

In August 2003, a major power outage struck simultaneously across dozens of cities in the eastern

United States and Canada – 60 million Americans were stranded without power - stopping trains, subways, elevators and the normal flow of traffic and life for several days. Lost revenues, lost income, lost tax revenues and overtime costs to emergency crews and other city workers cost the United States several billion dollars for this particular crisis. Also occurring last summer was Southern California's 2003 wildfire season in which fires struck 729,000-plus acres in five counties with estimated damages of over \$2 billion. In an effort to deliver a proactive alerting solution to respond to and better manage crises such as the August 2003 power outage in the Eastern US or the 2003 wildfires in Southern California, the Omvia Notification service provides voice-enabled alert notifications from local, regional, and national government sources to citizens. The voice-enabled service functions with state-of-the-art speech technology to process a request and then delivers the information verbally with a subsequent request for confirmation. The service can also send the request to the display on a hand held or Internet device. The system can be used for less-critical applications such as school closings and severe weather warnings to homeland security, natural disasters, and other emergency situations. When offered to public agencies or companies, this type of emergency notification would be provided on a subscription basis, with cell-phone style pricing for basic services and additional charges for use above a preset level. Businesses can also use emergency event notification during crises to support business continuity operations.

Interoice Voice Express Application Suite

General Information Management Module

- *Omvia Communicator*: Provides a speech-activated interface for name dialing and for corporate and shared address books, with standard interfaces to the telephone network.
- *Omvia Locator*: Provides speech-enabled location services for finding the closest retail outlet, kiosk, or other service by city, state, or zip code. Omvia Locator with Cross Street Finder adds the ability to find the closest location by cross street.
- *Omvia Rate Quote*: Gives quotes on mortgage interest rates. The application captures information such as property value, loan amount, property type, loan type, and term to provide custom quotes to callers and provides a library of mortgage questions that can be easily selected by the financial institution.
- *Omvia Name & Address Capture*: Captures first, middle and last names, zip code, city, state, street number, street name, and apartment number. Name & Address Capture shortens call durations using reverse phone

number lookup functionality based on a 10-digit phone number from Automatic Number Identification (ANI). It provides integrated third-party address verification and notifies callers of invalid primary and secondary address information.

- *Omvia Notification*: Available as a service, Omvia Notification provides voice-enabled emergency event notifications that combine monitoring and detection with push technology to virtually any wireline, wireless, or Internet-enabled communication device, including voice telephones, pagers, email, PDAs, SMS-text messages, or a combination of devices.

Customer Acquisition Module

- *Omvia Funds Transfer*: Advises callers about account eligibility, balance minimums, transaction limits, and insufficient funds. The application verifies transfer information before the transaction is completed, allows callers to change particular pieces of transfer information without restarting the transaction. The application supports transfers from external financial institutions. Funds Transfer also automatically provides account balances after transfer is completed.

Billing Module

- *Omvia Account Payment*: For credit card holders, callers can pay the current balance, last statement balance, minimum payment, or a custom amount. For mortgages and loans, callers can choose the monthly payment or a

greater custom account. The application supports mortgage and loan payoffs, payments from external financial institutions, and the setup of recurring automatic payments.

Customer Care Module

- *Omvia Survey*: Allows businesses to gauge customer satisfaction and conduct tele-voting initiatives through voice automation. Once a survey or tele-voting process is created, the application answers inbound calls and queries callers for responses. The user interface uses text-to-speech or pre-recorded prompts and records callers' verbal comments. Multiple surveys can run simultaneously and the caller will be directed to specific ones based on DNIS or by a Survey Identifier related to their specific caller account.
- *Omvia Identifier*: Supports automated changing of a password or Personal Identification Number (PIN). Identifier provides a comprehensive reset solution, including a consistent Voice User Interface (VUI), caller authentication, and back-end Reset functionalities that can be customized for use in a wide range of industries.
- *Omvia Account Information*: Allows callers to retrieve transaction history, balances, and status of a specific check while providing filters for reviewing transaction history based on date ranges and transaction types. The application also includes optional warnings for account balance minimums and activity thresholds.

Nuance (cont.)

Continued from page 1

in deploying speech applications, cited by Nuance customers in talks at the conference.

AAA Minnesota/Iowa

Joseph Alessi, vice president marketing and information technology, at **AAA Minnesota/Iowa**, an automobile club, said that the company implemented five applications: member maintenance, auto travel, number validation, call steering, and Estimated-Time-of-Arrival for service calls. The automation dropped the time spent on the call from an average of 3.5 minutes with an agent to 1.37 minutes, a savings of \$1.90 per call. Based on this success, the company plans to add other speech-enabled functions, including cross-selling and revenue generation, insurance policy updates (change of address, adding someone to policy, etc.), and possibly automating emergency road service calls.

Bell Canada

Belinda Bank, senior associate director, contact expertise centre, **Bell Canada**, discussed the company's 310-Bell call routing system. She indicated that there was a 79% reduction in IVR "executive complaints" in 2003 relative to the same period in 2002. Surveys reported a 5% improvement in overall ease of doing business with Bell Canada, a 14% improvement in getting to the right department promptly, and a 9% improvement in overall satisfaction with the call center. Call center employees expressed a noticeable improvement in the mood of callers when they reached an agent.

Citigroup UK

Chris Benton, vice president, **Citigroup-EMEA**, discussed Citigroup's deployment in the UK and Ireland. The company was outsourcing 20,000 calls per week and decided to bring the operation in-house

to support increased call volume, reduce costs, improve customer service, and provide 24/7 availability. They also wanted a system that could support business continuity with a redundant architecture.

The company instituted a “PeopleFind” application for 10,000 staff members, where employees can be contacted by spoken name using a **Lyrinx** solution. In January 2004, the system was handling 5,000 calls per day. In addition to the improved service for callers, the company gained a \$300,000 annual savings, for a 5-month ROI. The system handled 25% more calls with 80% less staff.

Lessons learned, Benton said, included:

- Source data can be an issue. Human resource files of employees can often have problems.
- Employees using hands-free “noise-cancellation” headsets to make calls sometimes had recognition problems.
- A good baseline metric for measuring improvement was “hang-ups.”

The company plans to add other speech-driven services. The business continuity features that allow contacting employees in the event of a power outage or other problem at one location will be enhanced. The existing solution will be expanded to add “find me” functions, a personal directory for out-dialing, additional sites in EMEA, conferencing, and possibly other unified communications applications.

Expedia

Sachin Jhunjhunwala, senior program manager, at **Expedia.com**, discussed applications that automated call steering based on itinerary number, automatic itinerary reconfirmation, and flight information. He said there was a 66% reduction in mis-routed calls. One-third of “comfort calls” (calls to confirm that reservations were still in place) were automated. The improvements generated a Return on Investment in two months, he indicated.

Suncorp-Metway

Andrew Mulvogue, General Manager, personal customer sales and service, **Suncorp-Metway** (Australia’s sixth-largest bank and second-largest insurer) said that the company was handling 4.5 million agent-based calls per year with 1,200 agents and 22 different agent skill sets. The company had a large volume of mis-directed calls and significant customer frustration with the touch-tone solution.

The natural-language call-routing system was integrated by **VeCommerce** and went live in December 2002. Customers respond to, “Please tell me which service I can help you with today.”

The system cost \$11.0 million, Mulvogue said, and is saving \$38 million per year. Call duration dropped from an average 62 seconds with touch-tone to 14 seconds with speech, saving over 2.5 million telephone minutes per year. Currently, 98.1% of transfers are successful, and mis-directed calls are as low as 2.5%.

TELUS

Bob Reczka, vice president of mass market operations at **TELUS**, the Canadian telephone service provider, discussed their use of Nuance Call Steering (which allows “natural-language” requests to be routed to the correct service or agent). The company decided, he said, to use a “non-branded,” neutral persona. The biggest challenge, he said, was the transition from touch-tone to speech—calls were initially split between the services. “It was like rewiring the house with the electricity on,” he said.

The deployment resulted in TELUS needing 1,250 fewer agents and a “dramatic” improvement in the fraction of calls answered within 20 seconds. Reczka reported that 81% of customers feel that their speech experience was the same or better than the previous system. More objectively, there was a 64% reduction in abandoned calls.

TELUS is also using speech automation for an internal application serving their 4,000 field technicians. Applications include ten applications including time reporting and attendance exceptions. The expected savings from this automation are about \$5 million per year.

T-Mobile

Gunny Markefka, in design management at **T-Mobile Deutschland**, the large wireless carrier in Germany, said the company employed 1,260 speech recognition ports from Nuance at “Tier 4,” the most advanced Nuance capability level, which supports natural language processing. The system was integrated by **Unisys**. The average cost per call with an operator for the automated applications was €4.62; with speech recognition, it dropped to €0.90. The system has been in operation since December 2001. Markefka said that their experience was that VoiceXML is not yet adequate for natural language applications.

United Utilities/Vertex

Rob Shorthouse, strategy and capability architect at **Vertex**, part of **United Utilities**, which delivers utility services on four continents (including water and wastewater infrastructure in the UK), said the company received 2.2 million billing inquiry calls per year and had limited success using touch-tone to automate those calls. The company used a menu-like structure with four options or “next” presented at a time. The result of this structured approach was that, while 5% of surveyed users said they preferred the speech recognition, 3% said they preferred touch-tone and 92% would prefer a customer representative. (Perhaps this suggests that being overly conservative with a speech dialog doesn’t convey the full advantages of speech.)

Vodafone Spain

J. Ignacio Casado, director of operational support in customer care, at **Vodafone Spain**, the wireless carrier, discussed an application for pre-paid customers, deployed as a hosted service by **Ydilo**. He said the number of agents serving this application was reduced by 50% while maintaining customer satisfaction. Two million calls were managed by IVR in Q1 2004. The system supports 480 simultaneous calls now, and is being expanded to 1,800. He said that only 6.4% of calls are diverted to agents because of recognition errors.

ScanSoft DialogModules (cont.)

Continued from page 1

The DialogModules are used in conjunction with ScanSoft’s SpeechWorks OpenSpeech Recognizer 3.0 on a wide selection of telephony platforms. “VoiceGenie is pleased to offer a flexible and high performance VoiceXML platform that is optimized for ScanSoft OpenSpeech DialogModules 2.0 execution,” said Rob Marchand, senior director of product management at **VoiceGenie Technologies**.

The latest release includes significant application development enhancements that result in lower infrastructure cost, major memory savings, and superior application performance. Enhancements in the latest release include:

- *Latest standards support:* The latest release supports multiple drafts of the VoiceXML 2.0 specification including the latest candidate recommendation from February 2003.
- *More efficient use of CPU resources and faster response times:* OpenSpeech DialogModules have improved caching performance (maintaining reusable results in memory) resulting in greater CPU savings and faster response time for callers. For example, OpenSpeech DialogModules 2.0 can now support more than 400 cacheable channels, a four-fold increase in performance relative to OpenSpeech DialogModules 1.2. As a result, speech solutions that employ OpenSpeech DialogModules 2.0 software can lower web application server infrastructure costs by 75%.

- *Improved application monitoring and analysis:* OpenSpeech DialogModules 2.0 improve application management and tuning by providing for server-side logging of information. Application developers can more easily access call logs.
 - *Dynamic change of prompts, grammars, and other properties:* Designers can dynamically change parallel grammars, properties, and behaviors. For example, designers can use text-to-speech software to spell back information using the Name and Address OpenSpeech DialogModules, incorporating the same voice used for the pre-recorded prompts.
 - *Wider Application Server support:* OpenSpeech DialogModules 2.0 support a wide range of J2EE application servers supporting the J2EE Servlet 2.3 and JSP 1.2 specification, independent of the operating system.
 - *OpenSpeech Xccelerator testing suite:* OpenSpeech DialogModules 2.0 includes a testing package for platform partners. OpenSpeech Xccelerator includes code, grammars, audio prompts, documentation, analysis tools, and metrics so that platform partners can quickly qualify and integrate OpenSpeech DialogModules on their platform.
- “Our professional services group relies on our patented OpenSpeech DialogModule technology for every customer engagement,” said Steve Chambers, president of the SpeechWorks division of ScanSoft. “OpenSpeech DialogModules

provide a consistent user interface design that has yielded superior application results in more than

750 speech deployments.”

ScanSoft auto-attendant (cont.)

Continued from page 1

SpeechAttendant is distributed throughout North America and Europe through channels such as **Bell Canada, British Telecom, NACR, NEC BNS, NextiraOne, SBC, and Telus**. Since SpeechWorks acquired Locus Dialog, the distribution has been broadened to include **Call Processing, eNabling Technologies, Servion, SSP Telecom, The Via Group, and Vitec**. The VoiceXML-based version will now allow standards-based platform providers to offer the solution.

SpeechAttendant 8.1

The turnkey solution, SpeechAttendant, has more than 1,000 installations worldwide and is handling over one-half billion calls annually, according to ScanSoft estimates. The latest release of the system, SpeechAttendant 8.1, is now available. LocusDialog originally used its own speech recognition, but the current release incorporates the performance and accuracy of SpeechWorks OpenSpeech Recognizer speech recognition and SpeechWorks Speechify text-to-speech software. The new release features the following:

- *Improved support for large directories:* SpeechAttendant 8.1 delivers enhanced performance for large directories, containing more than 100,000 names, with improved disambiguation techniques to offer a similar caller experience found with smaller directories containing fewer than 1,000 names.
- *Easier system administration:* SpeechAttendant automates some aspects of maintenance that have previously complicated system administration. The new release includes an extensive nicknames template that automatically associates common nicknames to proper names, such as adding “Bob,” “Bobby,” and “Rob” to “Robert” to reduce manual entries by the system administrator. The release also provides a large department-acronyms template that automatically imports department acronyms present in directory source files and converts them into formats that can be used by text-to-speech and speech recognition engines, such as converting “mktg” to “marketing” or “svcs” to

“services.” SpeechAttendant also now supports the HL7 protocol required by healthcare providers and hospitals to speech-enable access to patient rooms without relying on operators and touch-tone systems. In addition, SpeechAttendant 8.1 adds new system performance reports to the reports already available.

- *Improved out-of-the-box accuracy for wireless and noisy environments:* SpeechAttendant 8.1 incorporates technologies from OpenSpeech Recognizer software to discriminate between background noise and speech and dramatically increase accuracy rates for callers in wireless, hands-free, and noisy environments. It also features SpeechWorks’ LEARN capability that automatically adapts the system based on caller usage.
- *Improved fault-tolerance and support for complex multi-site deployments:* SpeechAttendant further enhances its redundancy and multi-site capabilities by allowing customers to accommodate real-time redundancy and load-balancing. In addition, SpeechAttendant 8.1 offers increased flexibility for deployments in complex, distributed architectures as speech-enabled auto-attendants become mission-critical systems.
- *Professional services support:* SpeechAttendant can be fully deployed in less than a week, the company indicates, depending on the size of the application. It includes prepackaged reporting and analysis tools so that system managers can easily monitor system performance and call traffic and further fine tune their application.

OpenSpeech Attendant

OpenSpeech Attendant 1.0 is based on the SpeechWorks Division of ScanSoft’s SpeechAttendant 8.1 architecture. OpenSpeech Attendant offers partners and developers a packaged, VoiceXML-based application. VoiceGenie is one partner that will offer its customers the application on its NeXusPoint VoiceXML-based framework. John Cameron, Executive Vice-President at VoiceGenie Technologies, said, “This combination provides additional value to customers that may want to extend their existing gateway investment with more complex speech services in the future.”

**Voice-activated auto-attendants:
Being a good solution to a problem isn't always enough**

If speech recognition were available when auto-attendants were developed, this would be the natural way to direct calls. For historical reasons, we accept having to know extensions or individual numbers for each person at a company, or to accept spelling a name on a keypad and then wading through the alternatives suggested. If usability were the only criteria, every business phone would be answered by a Voice-Activated Auto-Attendant (VAAA). This has encouraged a number of companies to offer the solution, but it has not proved to be as attractive to buyers as one might expect if usability were the main criteria. In practice, there is little to motivate changing something that people aren't complaining about. The cost savings (fewer receptionists, less printing of directories, etc.) are minimal relative to other costs in most organizations (the call center being a prime example).

In addition, the classical sales channel for auto-attendants also make it hard to sell. Most have been sold as an incremental feature for a PBX system. The distributors selling such systems are not always well-trained in explaining why a buyer should consider something different than they are accustomed to. Salespeople often just give up on the evangelical sale required, and focus on selling their primary product.

All of this could change when enough phones are answered by a VAAA. People would get used to the idea, just as they did with voice mail systems, and come to expect such systems. This will happen when there is enough penetration of the market, but getting to that point is difficult if the solution is hard to sell.

The solution to this problem is probably to add features to a VAAA that make it even more of a solution for internal applications. For example, a VAAA can include "find-me" or "follow-me" features that let an employee be reached on a specified schedule at several phone numbers, including a wireless or home phone, without divulging those numbers to others.

Some companies are using such systems as part of their business continuity planning. By mirroring the system at more than one location, it is easy to re-establish employees when one location can't be used and simply have the employee call in and change the number associated with their name.

The solution should be accepted for its core utility. It may take adding other business solutions to the core name-dialing functionality, however, to create wide adoption.

- Bill Meisel, Editor

News Briefs

Nortel survey shows momentum toward converged networks

On June 1, **Nortel Networks** [NYSE/TSX: NT] announced survey results that show businesses that are deploying or planning to deploy converged networks are at unprecedented levels. The comprehensive survey, conducted by **Mindwave Research** this spring, featured 430 members of the International Nortel Networks Users Association (INNUA). More than 27% of respondents have already deployed a converged networking infrastructure, and a full 79% of businesses have converged or plan to converge the disparate elements of their communications networks within five years.

Malcolm Collins, president, Enterprise Networks, Nortel Networks, commented, "Converged networks are being implemented at an aggressive pace for businesses of all sizes in all economic sectors. Put simply, companies recognize the emerging applications, cost savings, and productivity enhancements enabled by a single networking infrastructure and are poised to realize these benefits."

The top three cited benefits are the ability to deploy integrated applications with greater ease (42%); minimizing costs and complexity associated with moves, adds, and changes (35%); and the ability to deploy enhanced voice functions (34%). The most pressing challenges are that the cost benefits are not perceived as being compelling enough at this time (48%), lack of budget (47%), and the requirement of systems to manage and troubleshoot IP voice quality (42%).

For Wireless Local Area Networks (WLANs), only ten% of respondents have rolled out a WLAN infrastructure enterprise-wide, but a substantial 55% have deployed a limited WLAN or plan to within the

year. Increasing employee productivity through mobility was the top reason cited for deploying WLANs and the biggest challenge was weakened security. From a security perspective, network and data security was the most significant challenge for 20% of respondents.

Nortel addresses this trend with its new Architecture for the Converged Enterprise (ACE), announced in June. ACE will provide customers with the building blocks to implement a converged infrastructure that Nortel indicates is easy to manage, secure, and efficient.

Telus deploys IP Unity for its IP-One messaging services, will add speech recognition and TTS

In an example of the move toward IP networks for telecommunications, **TELUS**, the largest telecommunications carrier in western Canada, and **IP Unity**, which provides IP service solutions for blended networks, announced that TELUS has deployed IP Unity's Harmony6000 platform to serve as the voicemail and unified messaging engine for its IP-One Telephony Services. In mid-2003, TELUS became the first Canadian telecommunications provider to launch carrier-grade hosted and managed Internet Protocol (IP) Telephony Services. Since then, the company has deployed IP services to business subscribers in the nation's seven most populous provinces. The carrier has selected and deployed IP Unity's Harmony6000 Media Server to upgrade its first generation voice messaging platform to the latest functional requirements and cost performance targets, and to rapidly migrate its voicemail subscriber base to all-IP based messaging by the end of 2004. "TELUS has achieved a watershed accomplishment by being the first North American ILEC to deploy a suite of IP telephony services in its core network," emphasized IP Unity CEO Arun Sobti.

Intel releases upgrade for its Host Media Processing Software

Intel's Host Media Processing is a technology used to perform media processing tasks on general-purpose standard high volume (SHV) servers with Intel Architecture processors but without specialized digital signal processing (DSP) hardware. As speech is transmitted in IP format, telephone interface cards will be less cost-effective. The HMP software enables customers to build full-featured, scalable, and cost-effective software-only IP media servers for interactive voice response (IVR), voice mail, unified messaging, and conferencing. The software includes speech integration using Intel's continuous speech processing APIs.

The company has released a new version, NetStructure Host Media Processing Software Release 1.1 for Windows. The new release adds support for Windows Server 2003, "semi-automatic" startup of system service, changes to improve quality of service (minimized latency and high voice quality), program control of the volume of RTP (Real-Time Transport Protocol) sessions, and some bug fixes.

Release 1.1 of HMP is suitable for IP environments only, and can be used to create components for converged IP voice networks and the modular network. HMP can also be enabled in a time division multiplex (TDM) environment, but a telephony interface card that supports HMP is required. Public Switched Telephone Network (PSTN) connectivity for HMP is available through media gateways such as the Intel NetStructure PBX-IP Media Gateway. According to Intel, PSTN connectivity cards designed for HMP are in development and planned to be available in 2005.

Montreal public transportation agency to use Nuance TTS to speak route and reservation data

On June 15, **Nuance** announced that **La Societe de Transport de Montreal (STM)**, Montreal's public transportation organization, has deployed Nuance Vocalizer 3.0 text-to-speech (TTS) software in its bilingual door-to-door transportation information line. This application enables disabled passengers to call into STM's automated traveler information line and hear needed reservation and route data spoken back to them by Nuance Vocalizer TTS in Canadian French or English. STM has also deployed two features that use Nuance Vocalizer: AUTOBUS, which allows users to obtain the next three arrival times at any given bus stop, and STM.INFO, which provides a variety of information to customers.

Serge Belanger, information systems manager for STM, said, "Since our new transportation information line became available last fall, many travelers have begun to rely on the service for fast, automated information. We attribute part of our success with the new system to the accuracy and reliability of Nuance Vocalizer text-to-speech software. Riders love how they can quickly access route or drop-off location

information easily from any phone... We're now planning to offer riders more voice-automated services, including detailed itineraries and route change information."

Nuance Vocalizer is deployed at STM with the **Cisco IP Interactive Voice Response (IP IVR)** product, a component of Cisco IP Communications. Cisco IP Communications includes IP telephony, unified communications, IP video and audio conferencing, and contact center applications—enabled by Cisco AVVID (Architecture for Voice, Video and Integrated Data).

Voice.Trust to use Vocalocity Voice Gateway in turnkey authentication solution

Voice.Trust, a European provider of speaker authentication solutions, and **Vocalocity**, a provider of VoiceXML and SALT platform software designed exclusively for OEM and VAR partners, announced a strategic alliance. Voice.Trust will embed Vocalocity's Voice Gateway in the stand-alone version of Voice.Trust Server 5.

The Voice.Trust Server provides an authentication step to secure access to resources, sensitive data, and secure transactions. Vocalocity Voice Gateway is the embedded communications software platform for the application. The integration of both of these products supports applications such as Password Reset, Two-Factor Authentication, Remote Access, Single Sign-On, PKI-Management, Secure FileSafe, and Caller Authentication.

Gearworks' mobile workforce management software, with VoiceXML, used by Bradco Supply

Gearworks, a provider of mobile workforce management software, announced that **Bradco Supply**, one of the nation's largest roofing materials manufacturers and distributors, has selected and fully deployed Gearworks' flagship product, etrace, running on **Nextel Communications'** national packet data network, to improve the productivity and service levels of its national materials delivery fleet. Bradco is using etrace to collect and report the location, time and speed of individual delivery vehicles, ensuring the right product arrives on time, every time. etrace is also enabling Bradco to improve productivity by helping drivers make an additional stop each day.

Etrace is a wireless and Web-based mobile workforce management solution that runs on GPS and Java-enabled phones, such as the Nextel i58sr by **Motorola**. Etrace incorporates functionality such as configurable SmartTalk voice menus, based on VoiceXML technology, and WorkZone location-aware reporting. SmartTalk enables workers to gather and process data to close out jobs in the field, create invoices, print receipts, and take payment.

Vocera signs VAR agreement with SAIC for voice-activated communication system

On June 3, **Vocera Communications** announced a strategic reseller agreement with **Science Applications International Corporation (SAIC)**, a diversified high-technology research and engineering company. The agreement with SAIC will give Vocera additional opportunities in the government and military sectors.

The Vocera Communications System consists of Vocera Server Software, residing on a customer server and using **Nuance** server-based speech recognition, with Vocera Communications Badges, which are small devices worn by users which operate over a wireless LAN (802.11b). With voice commands, Vocera instantly connects people to other people or groups. Targeting mobile personnel in hospitals, retail operations, government facilities, and other in-building environments, The Vocera Communications System provides hands-free, voice-activated communications throughout any 802.11b-networked building or campus.

In November 2003, the U.S.S. Coronado (AGF 11), the U.S. Navy's sea-based battle lab, installed the Vocera Communications System to facilitate instant communication among select members. This was the first military installation for the company.

Apptera launches partner program

On June 7, **Apptera**, which offers packaged speech applications, announced the Apptera Partner Program (APP), which will initially include a group of more than 15 platform providers, service bureaus, voice

application hosting companies, and service companies. Apptera and its partners will cooperate to integrate, sell, and market complete packaged voice solutions for enterprises of all sizes.

Members of *APP for Resellers and Service Bureaus* receive software, full technical training, demonstration capabilities, sales and marketing support, and support services at discounted prices. The charter members include **Intervoice**, **Information Technologies Australia (iTa)**, **NEC**, **Premier Technologies**, and **Tele-Direct Call Centers**. In May, Apptera named Intervoice as its first strategic partner to resell the company's suite of financial services packaged applications under the Omvia Voice Express brand.

iTa was the first Australian speech applications developer to demonstrate Apptera's flagship product, Financial Services Suite, using Australian English speech engines. The suite includes speech-enabled Account Information, Funds Transfer, Name & Address Capture, Account Payment, Mortgage Quote, and ATM/Branch Locator functionality.

Members of *APP for Technology and Services Companies* receive Apptera software (where necessary), full technical support, and training. Among the charter members are **Genesys Telecommunications Laboratories, Inc.**, **Intel Corporation**, **Nuance**, **SandCherry**, **ScanSoft**, **Visa International**, and **Walsh Media**.

DTSC to be Persay VAR in Taiwan for speaker verification in call centers

On June 10, **Persay Ltd.**, which provides speaker verification systems for call centers and remote services, and **DTSC (Delhum Technology & Service Corporation)**, a leading provider of call center solutions in Taiwan, announced a partnership agreement for providing biometric speaker verification solutions for Taiwan's call center market. Persay's speaker verification products VoacIPassword and FreeSpeech use the customer's voice and speech characteristics for authentication. Persay's products enable secure and convenient customer authentication within the IVR session or in the background of a conversation with the call center agent.

DTSC provides integrated voice/media recording and quality/security monitoring solutions for call centers, financial institutions, telecom operators, government agencies, military, and large corporations. DTSC's customer base includes most of Taiwan's banks and telecom operators.

Emerging Technologies & PACC Electronic Engineering announce the formation of Speakatel for the delivery of Arabic and multilingual speech recognition solutions

PACC Electronic Engineering and **Emerging Technologies** announced today the signing of a joint venture agreement whereby a new company will be formed and named **Speakatel**. The new company will focus on the delivery of Arabic and multilingual speech recognition solutions and services to the local and international markets with initial launch from offices in Dubai, UAE; Cairo, Egypt; and Kuwait city, Kuwait.

Crosstalk

Telematics commentary

In this month's Commentary in *Speech Recognition Update*, Bill Meisel looks at Telematics, the key role of embedded speech, and what motivates a large company like **IBM** to pursue this market aggressively.

Sprint uses operator-aided speech recognition for "captioned" telephone service for hearing disabled

Sprint (NYSE: FON) provides telecommunications relay services (TRS) for the deaf and hard of hearing and has launched CapTel Relay Service (Captioned Telephone Voice Carry Over) in Texas. An operator listens to the call, but rather than type what the caller is saying, the operator repeats what is said into a

speaker-dependent, large-vocabulary speech recognition system on a PC, which transcribes the speech to text. For more on this, see this month's issue of *Speech Recognition Update*.

Financial Notes

Intervoice first-quarter revenues rise 9.1% over Q1 2003

On June 22, **Intervoice** (Nasdaq: INTV) reported earnings for its fiscal first quarter ended May 31 of \$0.08 per share, in line with analysts' estimates, and a large increase relative to \$0.03 earned in Q1 2003. Revenues rose 9.1% year-over-year to \$41.9 million, near analysts' estimates. The company also reaffirmed its Q2 revenue outlook of \$41 million to \$47 million. Cash balances increased by \$4.6 million to \$45.5 million during the quarter, and the company reduced its debt by \$0.5 million to \$12.6 million.

Company executives were upbeat in a telephone earnings call. Craig Holmes, executive vice president and CFO, cited "large solutions orders" and "the significant increase in solutions backlog." The solutions backlog was \$43.2 million on May 31, 2004, up 36% from February 29, 2004. David Brandenburg, CEO and Chairman of the Board, cited speech technology as one of the significant trends that will cause the business to expand, noting that only about 15% of Intervoice's installed touch-tone IVR base had upgraded to speech technology so far, and that essentially all of the new orders included a speech application or the requirement that the platform support adding speech. Bob Ritchey, Intervoice president, noted that the company had won a number of customers from competitors, in part based on the company's strong offerings in speech.

Unveil Technologies raises \$6.5 million in venture financing

On June 15, **Unveil Technologies**, a provider of conversational voice self service applications for contact centers, announced that it has raised \$6.5 million from investors **Sevin Rosen Funds**, **TD Capital Ventures**, and **Solstice Capital**. This latest investment will enable the Company to expand staffing in all key functional areas, commercialize new technological developments, and extend operations into selected international markets. As part of this effort, the Company will be adding to its world class team of speech and language scientists and developers. Unveil's speech solution, the Unveil Conversation Manager, blends technology and agents in a collaborative relationship aimed at reducing costs while improving customer satisfaction.

Fred Bamber, General Partner of Solstice Capital, said "Unveil's patented Adaptive Learning technology and method of mixing speech with live agents resonates with organizations that are wary of deploying traditional voice systems. This combination of technology and architecture eliminates the legacy conflict between automation and customer satisfaction. We expect Unveil to transform the call center landscape of agents, computers, and customers."

Nortel Networks provides update on status of restatements and related matters

Nortel Networks [NYSE/TSX: NT] has indicated the necessity of restating some past earnings downward, and an announcement on June 2 suggests more restatements may be necessary. Nortel said it continues the work to restate the financial results reported in each of its quarters of 2003 and for earlier periods including 2002 and 2001 as announced previously. The detailed review is now underway with respect to the third and fourth quarters of 2003.

Bill Owens, president and chief executive officer, Nortel Networks, said, "We are making progress toward completing the restatement of our financials, but this is a complex process and there is significant work yet to be done. This has been a very challenging time for the Company, our investors, customers and employees. We will get through it in good shape. At the same time, we are encouraged to be seeing good business momentum and support from our customers with growing demand for our next generation networking solutions."

Highlights of some recent company announcements include:

- Continued success in the cable multiple system operator (MSO) market segments with Voice Over IP (VoIP) wins announced with Charter Communications (USA), Telenet (Belgium) and VTR (Chile);

- Advances in 3G wireless solutions—selected as national 3G provider to Pelephone in Israel for CDMA 2000 1X EV-DO equipment and implemented UMTS-based 3G services with Orange for the Cannes International Film Festival;
 - Gains with service provider voice over packet solutions with a contract win with Bell South;
 - Helping enterprises leverage next generation networking solutions to transform their business including working with the FedEx Institute of Technology to support the State of Tennessee’s “Workforce of the Future” program and upgrading the China State Tax System in Beijing; and
 - Launching the Multiservice Provider Edge, a new IP edge networking solution that features IP networking reliability for service provider networks.
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People

Wes Hayden Appointed President and CEO of Genesys

On June 4, **Genesys Telecommunications Laboratories, Inc.**, a subsidiary of **Alcatel** (NYSE: ALA, Paris: CGEP.PA), announced that Wes Hayden has been appointed president and CEO, effective immediately. Mr. Hayden has more than five years’ tenure at Genesys as senior vice president, Americas, where he led a profitable and growing sales organization, as well as previous sales and management experience at Informix, Sun Microsystems, Digital Equipment Corporation, and Applied Data Research. Mr. Hayden replaces Laurent Philonenko, who has left the company. Mr. Hayden holds a B.S. degree from the University of Illinois, Champaign-Urbana and an MBA from the Kellogg Graduate School of Management at Northwestern University.

Preferred Voice Appoints Carlos D. Blanco as Vice President of Products

On June 22, **Preferred Voice, Inc.** (OTC Bulletin Board: PFVI), a provider of telephone speech recognition solutions, announced that it has appointed Carlos D. Blanco as vice president of products. Mr. Blanco’s initial focus will be the web enablement and user experience of Preferred Voice’s existing products. Mr. Blanco began his carrier at Microsoft Corporation from 1991 until 1998 where he designed and implemented various pieces of Microsoft’s Office and Access products. From 1998 until 2001 he was a founder and Chief Technology Officer of Ememories.com, a web enabled consumer product Company that allowed users to create photo memory books on line that could be stored and shared with other on line users. From 2002 until the present he has been a Microsoft .NET consultant designing and developing web e-commerce applications for direct to consumer products.

Tuvox names Terry Shough senior vice president of worldwide sales and Rick Davison to head western sales

On June 11, **TuVox** (p. 3) announced the appointment of two executives to lead corporate sales. Terry Shough joins TuVox as senior vice president of worldwide sales and Rick Davison as vice president of western region sales. Industry veteran Shough brings over two decades of sales management experience and success at call center automation companies that include Edify, Intervoice, and Nortel-Peripherals. Davison brings 18 years industry experience at customer self-service technology companies, most recently at Edify Corporation.

For Further Information on Products Mentioned in this Issue

Company	Location	Product Mentioned	Telephone
Aculab plc	Milton Keynes, UK	Telephony boards with speech recognition	+44 (0)1908 273 800; www.aculab.com
Apptera	San Francisco, CA	Telephone speech applications	(650)635-0600; www.apptera.com
Artisoft, Inc.	Cambridge, MA	Computer telephony solutions	(617)354-0600; www.artisoft.com
Avaya Communications	New Jersey	Enterprise telephony solutions	(908)953-6000; www.avaya.com
Brooktrout Technology	Needham, MA	Telephony hardware and software	(781)449-4100; www.brooktroutinc.com
Cisco Systems, Inc.	San Jose, CA	Internet infrastructure and IP telephony	(800) 553-6387; www.cisco.com
Convergys Corporation	Cincinnati, OH	Customer care and employee-benefit solutions	(513)723-7153; www.convergys.com
CYCOM Integrated Technologies	Markham, ON, Canada	Telecom system integrator	(905)946-0677; www.cygcom.com
Delhum Technology & Service Corporation (DTSC)	Taiwan, R.O.C.	Voice recording and security equipment	+886-2-2712-4567; www.dtsc.com.tw
Edify Corporation (S1 subsidiary)	Santa Clara, CA, and London	Enterprise self-service solutions	(408)982-2000; +44-181-263-2710 (UK); www.edify.com
Emerging Technologies	Abu Dhabi, United Arab Emirates	Systems integrator	+971-2-6817666 ; www.em-t.com
Forschungszentrum Telekommunikation Wien (FTW)	Vienna, Austria	Telecom research center	www.ftw.at.
Gearworks	St. Paul, MN	Mobile workforce management software	(651)209-0350; www.gearworks.com
Genesys Telecommunications Laboratories, Inc.	Daly City, CA	Call routing and contact center solutions	(888)GENESYS; www.genesyslab.com
Globis Data Inc.	Kanata, ON, Canada	Traffic information	(613)271-1657; www.globus.ca
IBM Pervasive Computing	Somers, NY	Speech recognition and multi-modal technology and platforms	(561)862-2267; www.ibm.com/software/voice
Information Technologies Australia	Brisbane, Australia	Telephone customer service	+61 7 3350 0700; www.itaus.com.au
Infotalk Corporation Ltd.	Hong Kong, China	Multi-lingual telephone embedded speech recognition	+852 2190 9600; www.infotalkcorp.com
Intel Communications Group	Parsippany, NJ	Telephony products and software	(973)993-3000; www.intel.com/design/network
Intervoice, Inc.	Dallas, TX	IVR and telecommunications solutions	(972)454-8712; www.intervoice.com
IP Unity	Milpitas, CA	IP telephony platform and apps	(408)582-1100; www.ip-unity.com
Kirusa, Inc.	Berkeley Heights, NJ	Multi-modal platform	(908)464-4467; www.kirusa.com
Lumenex	Washington, DC area	IVR software developer	(410)561-7388; www.lumenex.com
LumenVox LLC	San Diego, CA	Speech recognition engine	(858)707-0707; www.lumenvox.com

Companies mentioned in this issue

Company	Location	Product Mentioned	Telephone
Lyrix, Inc.	Tewksbury, MA	Unified communications software and services	(978) 851-5300; www.lyrix.com
Microsoft Corporation	Redmond, WA	Microsoft Speech Server	(206)454-2030; www.microsoft.com/speech
NEC Business Solutions	Australia	Telephone solutions	+613 9262 1111; www.nec.com.au
Nortel Networks	Bohemia, NY	Telephony and networking systems for service providers and enterprises	1-800-4NORTEL; www.nortelnetworks.com/solutions
Nuance Communications	Menlo Park, CA	Telephone speech recognition software	(650)847-0000; www.nuance.com
Onstar	Troy, MI	Automobile service with speech recognition	www.onstar.com
PACC Electronic Engineering	Egypt	Telecom solutions	+20-10-11 16 423
Pegasus Solutions	Dallas, TX	Hotel reservations-related services and technology	(888)431-0700; www.pegs.com
Persay	Woodbury, NY, and Tel Aviv, Israel	Speaker authentication technology	1(516)677-7291; +972-3-7678666; www.persay.com
Phonetic Systems Ltd.	Billerica, MA	Speech-enabled directory and database search	(978)439-3600; www.PhoneticSystems.com
Preferred Voice	Dallas, TX	Voice-enabled telephone service	(214)265-9580; www.preferredvoice.com
Premier Technologies	Melbourne, Australia	Customer interaction solutions	+61 3 9200 7777; www.premier.com.au
Pronexus	Ottawa, Ontario, Canada	Speech recognition computer telephony	(613)271-8989; www.pronexus.com
Quality Call Solutions	Santa Clara, CA	Customer service center solutions	(408)367-8804; www.quality.com
Rhetorical Systems	Edinburgh, UK (US: Framingham, MA)	TTS	+44 131 525 6800 (UK); 1(508)820-0700(US); www.rhetoricalsystems.com
SandCherry, Inc.	Boulder, CO	Communications software	(720)562-4500; www.sandcherry.com
ScanSoft, Inc.	Peabody, MA	Speech technology and OCR solutions	(978)977-2000; www.scansoft.com
Science Applications International Corporation (SAIC)	San Diego, CA	Navigation service using distributed speech recognition	1-800-430-7629; www.saic.com
Softel Communications	Norcross, GA	Telephony integrator	(770)613-5313; www.softel.com/ or www.softeleurope.com
Sprint Government Systems Division	Herndon, VA	Phone call captioning service	www.sprint.com/gsd; www.sprintrelay.com
Tele-Direct Call Centers	Sacramento, CA	Call center services	(800)342-4815; www.tele-direct.com
TELUS Corporation	Vancouver, BC, Canada	Telecom products and services	(604) 482-7969; www.telus.com
The Emergency Email Network, Inc.	Jacksonville, FL	Emergency alerts	(904)371-3217; www.emergencyemailnetwork.com
Tuvox Incorporated	Los Altos, CA	Call center solutions	(650) 623-0210; www.tuvox.com
Unisys Corporation	Paoli, PA	Telephone solutions and voice application development	(215)986-4321; www.unisys.com
Unveil Technologies	Waltham, MA	Natural language processing	(781)890-7333; www.unveil.com
VeCommerce	Sydney, Australia	Voice-activated telephone applications	+61 2 9428 9951; www.vecommerce.com.au
V-Enable	San Diego, CA	Wireless software infrastructure company	(858)824-1888; www.v-enable.com
Verascape	Downers Grove, IL	Turnkey voice-enabled systems	(847)919-0873; www.verascape.com

Companies mentioned in this issue

Company	Location	Product Mentioned	Telephone
Verizon Wireless	Bedminster, NJ	Wireless telephone services	www.verizonwireless.com
Visa International	Foster City, CA	Telephone speech recognition services	(650)432-5769; www.corporate.visa.com
Vocalocity	Atlanta, GA	VoiceXML and SALT platforms	(404)487-1200; www.vocalocity.net
Vocantas	Ottawa, ON, Canada	Voice-enabled telephone self-service	(613)271-8853; www.vocantas.com
Vocent	Mountain View, CA	Voice authentication solutions	(650)938-3663; www.vocent.com
Vocera Communications, Inc.	Cupertino, CA	Wearable wireless communicator	(408)790-4100 ; www.vocera.com
Vocomo Software	Cupertino, CA	Voice telephony applications	(408)253-8626; www.vocomosoft.com
Voice.Trust	Munich, Germany	Authentication applications	+49 89 127 16 0; www.voicetrust.de
VoiceGenie Technologies	Toronto, Ontario, Canada	VoiceXML gateways	(416)736-0905; www.voicegenie.com
VoiceXML Forum	New York, NY	Voice eXtensible Markup Language	(732)465-6486; info@voicexml.org; www.voicexml.org
Voxify, Inc.	Alameda, CA	Automated agents for call centers	(510)545-5000; www.voxify.com
W3C Voice Browser Working Group (World Wide Web Consortium)	—	Standards effort including VoiceXML	dsr@w3.org; www.w3.org/voice
Walsh Media	Oak Brook, IL	Voice recording and voice interface service	(800)359-6158; www.walshmedia.com
Ydilo	Madrid, Spain	Application development and hosting	+34 91 252 8400; www.ydilo.com

TMA Speech Index™

The TMA Speech Index (TSI), initiated by TMA Associates in the March 2003 issue of *Speech Recognition Update* (published at the end of February reporting on the mid-February value of the index), tracks the weighted stock prices of a representative number of companies whose business may be affected by speech technology. The index was initiated with a value of 100 as of February 19, 2003.

Company	Stock symbol	Weight	5/19/04	6/16/04	
			Price	Price	change
Alcatel	ALA	1	13.91	13.84	-0.5%
AT&T	T	1	16.96	16.24	-4.2%
AT&T Wireless	AWE	1	13.97	14.34	2.6%
Avaya	AV	2	15.09	15.56	3.1%
BellSouth	BLS	1	25.12	25.70	2.3%
Cisco	CSCO	1	21.36	23.88	11.8%
Convergys	CVG	3	14.49	15.02	3.7%
IBM	IBM	1	87.05	90.38	3.8%
Intel	INTC	1	27.11	28.12	3.7%
Intervoice	INTV	4	12.62	13.54	7.3%
Microsoft	MSFT	1	25.62	27.32	6.6%
Nortel	NT	1	3.47	4.09	17.9%
Nuance	NUAN	5	4.21	4.87	15.7%
ScanSoft	SSFT	4	4.86	4.97	2.3%
Sprint	PCS	1	17.75	17.42	-1.9%
Verizon	VZ	1	36.09	35.74	-1.0%
West Corp.	WSTC	2	24.69	26.44	7.1%
Z-Tel	ZTEL	3	1.41	1.37	-2.8%

TSI		241.40	5.3%
S&P500		1133.60	4.1%
NASDAQ		1998.23	5.3%

Change since initial value of index (2/19/2003)

TSI	100	241.40	141.4%
S&P500	845.13	1133.60	34.1%
NASDAQ	1334.32	1998.23	38.9%

About the TSI: Monthly values are calculated based on stock prices on the third Wednesday of every month. The index is calculated by taking a weighted average of the % changes in each stock, and applying that % change to the previous value of the index. (In other words, the index changes each month by the weighted average change of the individual stocks.) The weights are set between 1-5 based on the subjective judgment of Bill Meisel, the editor of this newsletter to reflect the degree to which the stocks behavior is related to speech technology products or services. (This is not a linear weight; the smallest weight is one-fifth the largest weight, and does not reflect the % of revenues affected by speech technology for a large company.) The TSI can be viewed as the behavior of a stock portfolio where the investment in each stock is weighted relatively by the weights shown; e.g., if the sum of weights of all stocks listed is 36, a stock weighted 1 would be 1/36 of the value of the portfolio. Stocks may be added or deleted from the index from time to time; after a change, the average weighted % change is computed and applied to the previous value of the index, preserving comparability. *The TSI is not intended to be a recommendation of the stocks included in the index, but is intended only to track their behavior.*

Calendar

Benchmarking for Web Sites & Voice Response, Las Vegas, Nevada

September 9 – 10, 2004

Sterling Audits, 877-504-5145,

www.sterlingaudits.com/workshops.html

SpeechTEK 2004, New York Marriott Marquis

September 13-16, 2004

(877)993-9767; www.speechtek.com/conference

INTERSPEECH 2004- ICSLP, Jeju Island, Korea

October 4-8, 2004

icslp@icslp2004.org, www.icslp2004.org

Bill Meisel's **Telephony Voice User Interface Conference**, San Antonio, Texas

February 9-11, 2005

TMA Associates, www.tmaa.com/conference, (818)708-0962

TMA Associates' Telephony Voice User Interface Conference
 San Antonio, Texas, Feb. 9-11, 2005
 www.tmaa.com/conference

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