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INTERNET TELEPHONY®
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Volume 8/Number 2 February 2005

Service Provider Survival Guide Debuts! (page 48)

The VoIP Authority Since 1998™

10 Things Service Providers Must Know

Also In This Issue:

- VoIPeering: New Monthly Column!
- Pannaway's Mike Skubisz On The Future of VoIP
- IP PBX ROI

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Triple Play Fever Catch It!

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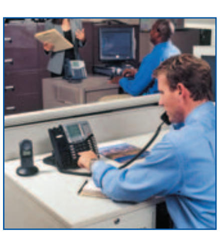

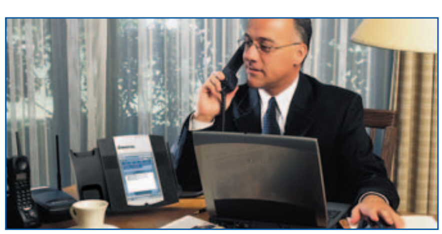

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The VoIP Authority

By Greg Galitzine

February, Short But Sweet In Miami

February is by far my favorite month of the year. My mom gave birth to her youngest son (yours truly) in February. My beautiful wife gave birth to our amazing twins in February. Internet Telephony magazine was born in February. And my favorite trade show of the year takes place in February. (Hint: the show's initials are Internet Telephony Conference & EXPO!)

This year's event in Miami (February 22–25) is shaping up to be a fantastic occasion. As always we're expecting a packed Exhibit Hall, and we are confident that we have put together a conference program that is unrivaled in this industry. Internet Telephony Conference & EXPO is where people come to learn all about purchasing and deploying VoIP and IP Telephony, and the conference tracks at the Miami event feature a wealth of new content, much of it geared towards the service provider market, with topics such as VoIP Peering, UNE-P to VoIP migration strategies, Triple Play sessions, and more. The February issue, which you hold in your hands, also features several new additions to the Internet Telephony universe.

This month, Hunter Newby, chief strategy officer at telx — an interconnect which maintains facilities in New York City and Atlanta — debuts his new monthly column entitled VoIPeering, which will cover the ins and outs of peering and the important role peering plays in the VoIP market. Peering is the concept of interconnecting networks allowing IP (and thus VoIP) traffic to be carried between service providers and companies without the need to engage a middleman, or in this case, an additional service provider.

Also debuting this month is the *Service Provider's Survival Guide to a Successful VoIP Migration*. This five-part series will offer ongoing commentary from Volo Communications' CEO Shawn Lewis on a number of topics relating to the need for CLECs to migrate to VoIP in order to stay alive. This series is certainly something CLECs will want to pay close attention to. The ground rules regarding UNE-P are changing and in many cases VoIP might be the only road to survival. Compelling stuff.

And of course, if you hadn't noticed by looking at this month's cover, our main feature for February is Triple Play. This issue is chock full of information relating to the technology that many analysts forecast will be the hottest thing to hit the VoIP landscape for 2005. We feature a full-on TMC Labs review of Pannaway's triple play service as well as a conversation with Pannaway's CTO Michael Skubisz and his views on the future if VoIP and Triple Play.

So I hope to see you all in Miami, where I look forward to hearing about all the interesting and profitable business ventures and relationships that will be "born" during the event. There's simply no better place to be. Miami. In February.

-Greg Galitzine, ggalitzine@tmcnet.com

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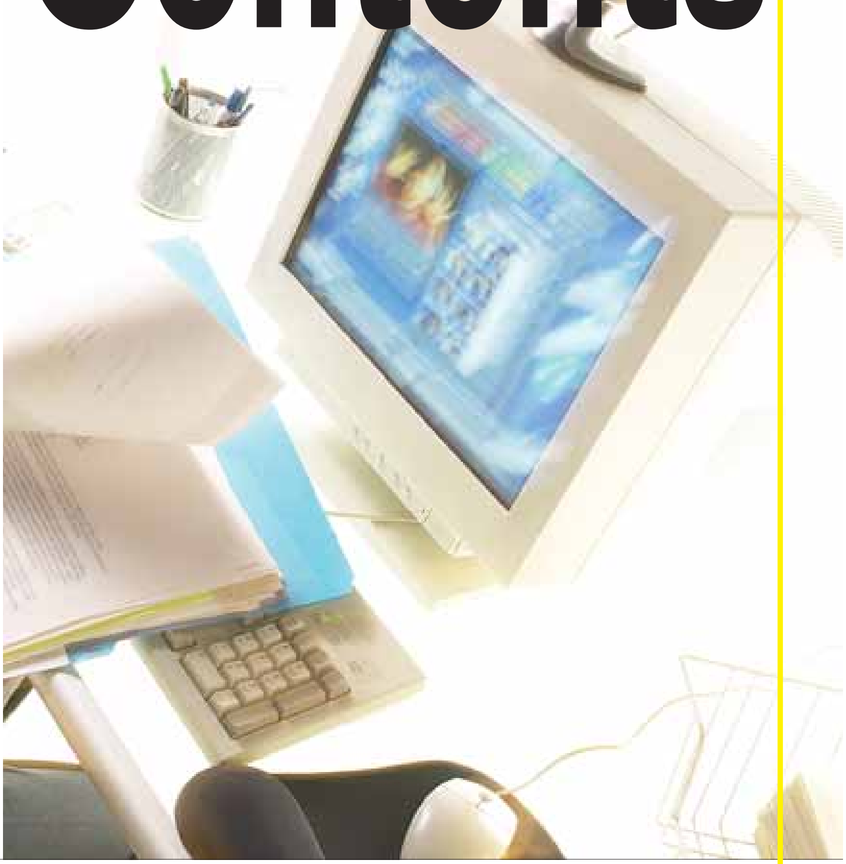

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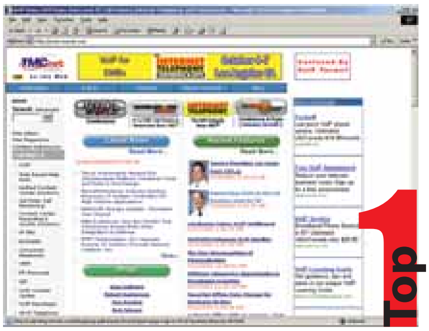

Top 10

Top 10 Visitors to TMCnet.com (by U.S. State)

1. Virginia	6. Texas
2. California	7. Washington
3. New York	8. Illinois
4. New Jersey	9. Florida
5. Massachusetts	10. Colorado

QUOTE OF THE MONTH:

"There's a good reason for State and Federal taxing bodies to be concerned about a loss of revenues with the lack of VoIP taxation. After all, telecommunications is already one of the most comprehensively taxed services in the United States. In fact, aside from "sin" taxes (on products like tobacco and alcohol) and fuel taxes, it is difficult to find an industry levy with a higher burden or more complicated tax regime than the taxes, fees and surcharges on telecommunications and related services."

– James E. Nason, Partner, Deloitte Tax LLP

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With over 3.6 million page views generated by 350,000 unique visitors per month, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

Here's a list of several articles currently on our site.

Dial Thru Expands VoIP Services in Iraq

Dial Thru International, a VoIP service and Internet telephony products provider, announced that it now has six fully operational Points of Presence (POPS) in Iraq
http://tmcnet.com/82.1

Deloitte Identifies Top Trends in Telecom For 2005

Deloitte's Technology, Media and Telecommunications (TMT) industry group announced its predictions for the global telecommunications industry in 2005, forecasting a year of significant milestones, as well as difficult questions
http://tmcnet.com/83.1

Alias Show Stars Packet8 VoIP Phones (Oh, And Jennifer Garner)

In addition to Alias, the Packet8 Videophone has also been incorporated in episodes of 24 — The Series, King of Queens, Law & Order SVU, Will & Grace, Stargate Atlantis, Medical Investigation, Eyes, Entourage and Navy NCIS.
http://tmcnet.com/84.1

WiMAX: All The Way To China

The growth of electronics equipment production in China has been widely described as the most fundamental shift in the world electronics industry.
http://tmcnet.com/85.1

Getting A Network Ready For VoIP

As a writer and editor covering the VoIP space, Al Bredenberg gets a large number of interesting communications from media relations people and various experts in companies that work in this market space.
http://tmcnet.com/86.1

TMC's IP PBX Channel

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By Rich Tehrani

The Best VoIP Investments Of 2005

Often, when I get asked the same question over and over, it makes sense to share the answer in a forum such as this column, thus letting as many people as possible in on my response. The particular question that I will address here has to do with investment, and identifying the opportunities that our industry has to offer. Indeed, what are the areas that are ripe for investment? How would you position yourself to profit from the hyper-growth of the VoIP market?

I purposely stayed away from obvious companies such as Net2Phone (news - quote - alert), VocalTec (news - quote - alert), Avaya (news - quote - alert), Cisco (news - quote - alert), Nortel (news - quote - alert), etc. I stayed away because there is already tremendous analyst coverage or the company does much more than just VoIP (define - news - alert). I am really focusing more on areas of VoIP that I believe will do better than average.

Each of the following ideas will be augmented with supporting reasons as well as potential pitfalls. It also should be noted that I have some sort of business relationship with most of the companies mentioned as they are all potential/actual customers, etc. Some of these companies are private and some public and some private ones mentioned below aren't looking for capital.

Furthermore, we need to realize that communications costs are plummeting and products such as Skype are getting millions of people accustomed to "free" telephony. In such an environment, it is obvious that the price of VoIP service will continue to fall as more people get used to downloading software from the Internet, buying a headset or any number of other VoIP capable products and then making calls.

The price of VoIP service will continue to fall as more people get used to downloading software from the Internet.

Peer To Peer

In such an environment, service providers will have to differentiate themselves with sticky features such as distinctive ring tone, second and third lines, virtual numbers, conference calling and more. A war of differentiation and services will have to occur because if we continue to have a price war until no one pays for telephony, no one will be left.

Peer to peer products from companies such as Skype (news - alert), Nimcat Networks (news - alert), and Popular Telephony (news - alert) are in a great position to benefit from the interest being paid to this space. The latter two companies focus on the enterprise and work with phone vendors to embed their technology into phones that eliminate the need for a central PBX (define - news - alert). Both the consumer and enterprise model make great sense.

The downside to this technology is the potential for ILEC and Cable companies to intentionally disable the quality of VoIP calls. This is a very real threat to our industry and the FCC needs to do their part to ensure an adequately competitive landscape so the VoIP market can flourish.

VoIP Peering

Many people get peering mixed up with peer-to-peer. They are not the same or even similar but can work together. Peer-to-peer products communicate with one another without the need for a central server while peering is the interconnection of networks. Companies that are strong in VoIP peering are Stealth Communications, telx, Terremark, and Infiniroute Networks.

Open Source Telephony

Two of the more visible companies in this space are Digium (news - alert), the maker of Asterisk open PBX, and PingTel (news - alert). Digium was first to go open source and Pingtel has been around longer but went from focusing on making IP phones to IP PBXs to now open-source software. The business models for these companies is similar to Red Hat in that they give you a free product and you choose to pay for support, consulting, some ancillary equipment (if needed), etc.

The downside here is if there is a move towards more hacking attacks on Linux servers. These products would be at risk and as we all know it is much worse to have your phone system go down than your e-mail system. This last statement is not true at TMC anymore but for many other companies phone systems continue to be more important than e-mail servers.

Government Suppliers

The government is spending on VoIP like never before. Each year, more conferees sign up to our Internet Telephony Conference & EXPO. When you talk with these conferees they tell you that VoIP is becoming more and more critical in

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the government and military sectors. A number of companies sell to this market including Net.com and Telecommunications Systems.

Triple-Play

The companies in this space are doing a great job coming out with products that are compelling for service providers to install and sell services with. Surveys tell us consumers want a single bill. Bundling is a great way to reduce churn. All service providers need to consider the triple-play opportunity and see if they can offer it in the future or be part of a broader offering.

Cable companies have the easiest time with this as they have already mastered TV and broadband, and VoIP is an easy addition. ILECs are going to have a tough time gaining share in the TV market in my opinion. It will take a while for them to get their systems up and running so it is premature to rule them out. There is the possibility that WiMAX (define - news - alert) providers will appear as well offering triple-play service. WiMAX is potentially the cheapest way to provide triple-play services.

The question is, will we even need triple-play providers in the future? A concern is that sooner or later TV will be streamed to users over broadband connections that can be viewed on any TV. In other words, TV may not be TV in the future… Or at least not the way we look at it today. You may just subscribe to Yahoo! TV and stream your channels. This would make the triple-play a double play as you don't need a TV provider if you have Yahoo!

The incumbent providers have pretty much blocked access to the fiber they are laying down and they may block access or reduce the quality of streaming video as well. We will have to wait on this one.

This is where the tight relationship between SBC (news - quote - alert) and Yahoo! (news - quote - alert) comes in. Imagine if Yahoo! were to work a deal with SBC so you need SBC service to access Yahoo! TV. Watch this area closely. By the way both www.yahootv.com and www.yahoo.tv are already taken by Yahoo!

One final note: it is possible that the company with arguably the best wireless network, Verizon, will just clean house by providing a superior quadruple-play offering and continuing to market their strength in the wireless market.

Consumer Electronics

You may not need to read this column to know that this market is hot but what you may have missed is how every product is becoming VoIP enabled. All telephony devices will have to support VoIP whether they are wired or not. I predict MP3 players, mobile video games, and 10 other categories of products I can't even imagine will somehow integrate and leverage VoIP.

There will be a continual worldwide upgrade of all wireless phones whether they be cellular or the 2.4 GHZ cordless kind found in our homes. They will all need to support WiFi (define - news - alert), Bluetooth and WiMAX and in that order. The cordless phone in your home will be thrown away as often as your cell phone. This of course assumes you don't just have a single device.

There will also be a tremendous need for ATA devices that convert analog to VoIP. As the VoIP market grows, these

On Record Pace!

On January 14, 2005, we crossed an important milestone in our year on year growth. As of that date, we were officially 100% higher in pre-registered attendance for Internet Telephony Conference & EXPO on a year-over-year basis. To put that in perspective, this is what was said in the VoIP Industry about the last show we held in Miami in February 2004:

"This place is a mad house!"
–Debora Glennon, Nortel Networks

"I haven't seen a show this busy in three years."
–Mark Straton, Senior VP, Siemens

"...it was the most active conference that I have been at in nearly three years…"
–Howard Kendall, Intertex

The total attendance at last year's event was 3,629. If current trends continue, we will have over 7,000 attendees. No VoIP show comes anywhere near us in terms of attendance. In the TMC tradition of under-promising and over-delivering we would rather be conservative and predict we will have 4,500 attendees at this show.

More testimonials can be found online here:
http://tmcnet.com/80.1

Many people ask me how the VoIP market will progress in 2005 compared to 2004. If our pre-registration numbers are any indication and the trend of growth continues, the industry could be up 100 percent year-over-year! The best part of all of this is we are seeing a surge in service provider attendance and thousands of enterprise and government attendees as well. There is also a dramatic uptick in the reseller attendees and we expect our conference track for resellers to be standing room only!

Internet Telephony Conference & EXPO is still the only VoIP industry marketplace, where buyers and sellers come together. We are 100 percent dedicated to generating buyers. We are not an industry insider/regulatory event by nature. People come to this show because this is where the sellers are and many of our exhibitors will not be found at any other event. You will kick yourself if you aren't there to witness this for yourself!

See you in Miami: February 22–25, 2005. IT


You just bought your entire company
Siemens optiPoint IP phones.
Good call.
That's because its high-fidelity voice quality is not only better than competitors, it's better than traditional phones. In fact, independent tests by CT Labs and the Tolly Group give the phones top scores for “excellent speech quality” and optimal flexibility with “greater management functionality than rival products tested.”
The optiPoint family of IP phones also offers easy-to-use features that fit into a LAN environment just like a standard data device. Like a side-car module, support for the widest range of voice compression and crucial security enhancements. And with optiPoint IP phones, you could even see your infrastructure costs and connection charges go down. So not only will you be making a sound investment, you'll be protecting it. Because you'll always be able to stay up to date and compatible with other SIP platforms, simply by adding the newest features with software downloads.
For more details, talk to your Siemens representative, visit hellodirect.com, or see Siemens Online at http://enterprise.usa.siemens.com.
To download a free copy of the CT Labs and Tolly Group Test Reports, visit http://enterprise.usa.siemens.com/go/opti
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devices are necessary. Eventually the phone itself will deal with the conversion.

HMP or host media processing will eventually be standard in all equipment, meaning specialized DSPs won't be needed. Phones and cameras will continue to merge and we won't need tapes, disks, etc. Everything is going to plain old flash memory.

VoIP Chips

These chips are finding their way into everything from routers to hubs to cell phones to PDAs. You name it, there is a reason to put a VoIP chip in it. Please note HMP comments from above.

Hosted Communications

This market has finally arrived. The worst is behind us and there was an extensive weeding out process that left the strongest providers. The value of hosted providers will only increase. I don't see a downside here unless major players such as Salesforce.com have a disastrous financial problem.

SIP

SIP (define - news - alert) is good. It is becoming great as more companies and service providers support this open standard. Anything SIP has potential to do very well. The market for SIP products is still young and there is ample opportunity for many players.

Downside to this one is a new standard or protocol eclipsing SIP. It happens all the time. It happened to H.323. So far this is the standard. Let's see what happens next. Also Skype could become the defacto standard for VoIP in the future. It is possible and at the pace of downloads they are receiving so far (over 50 million so far) they may just be on everyone's desktop and make SIP irrelevant. Niklas Zennstrom basically mentioned this in his keynote at a recent Internet Telephony Conference and EXPO.

The VoIP space is ripe for investment. I have identified several technology areas that should serve as a starting point for those looking for the next big thing in VoIP. But keep your eyes and ears open, the market landscape is constantly changing, and the next great opportunity in Internet telephony might be waiting just around the corner. Good luck! IT

VoIP, Broadband, And Regulation

The entrenched telecom players are ramming legislation down our throats on the federal and state level. They are eliminating competition at all costs and in many cases doing so with government support. Where is the huge CLEC market that was supposed to give us all this competition? Instead of a multitude of CLEC success stories, we see and hear negative news.

While the FCC is also a champion of VoIP, these are conflicting interests and empowering ILECs to control transmission facilities and reduce the competition in the UNE-P market is bad for all of us. These are the exact reasons I am looking forward to the UNE-P (define - news - alert) to VoIP Summit at Internet Telephony Conference & EXPO this month. CLECs are scrambling to stay in business in this changing regulatory environment we all live in. The question is: what happens next? If we don't ensure fair treatment for all VoIP service providers, we can see the VoIP service market fade away as quickly as the CLECs.

If you are in the IT or telecom space you know how far behind the U.S. is falling behind in broadband. You can't be a global power in anything without first-rate broadband infrastructure. We need choices, fair competition, and a fair and evenhanded regulatory environment. I urge you to read the following blog I posted (tmcnet.com/81.1), which contains links to many pertinent industry organizations and opens the door to a wealth of information. IT

EXPO Coverage

Read about these Internet Telephony Conference & EXPO exhibitors. Editorial coverage of these exhibitors appears in this issue on the page noted.

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Letters

Mistaken Peeve

It was with great interest that I read the December 2004 column by Marc Robins entitled "Pet Peeves." His first peeve of focus in particular, "#1: Waiting for the Next Killer App," is an appealing subtitle of interest, after all. Unfortunately, the details of this peeve were so very wrong and misleading they motivated me to urgently write this response as soon as I was finished reading it.

Marc begins by bemoaning admittedly tired references to some future killer app for IP Telephony, an interesting premise which caught my attention. His conclusion is essentially that the killer app exists already buried within the "wow" factor of VoIP. Primarily, he lists second tier features like multiple phone numbers as well as physical and logical phone portability as supporting evidence. This conclusion is amazing and obviously wrong. A killer app is a compelling market mover. It must be a utility that is so compelling to a large cross section of users that it develops its own market momentum — typically in excess of expectations. While there's no doubt that those buried in the technology appreciate the unique aspects of the VoIP feature set, the examples Marc gives do not come anywhere close to the level of a killer app

Don't get me wrong. I am a major proponent of VoIP — both currently and especially in the future. But the idea that VoIP's obscure features (to the lay person) represent a killer app is just absurd. There is a steady growth to Internet Telephony, and we can expect it to continue. But don't base your business strategy on the assumption that VoIP technology currently has an embedded stickiness that is so compelling as to rise to the level of killer app. It does not (currently). It is obvious and clear that the overwhelming primary reason most users move to Internet Telephony is for cost effectiveness of voice services — especially with respect to long distance cost savings. I'm sorry that is not sexy or representative of the long awaited killer app, but it happens to be both true and evident. You would be hard pressed to find a market study from anyone which reasonably shows that most potential mass-market target customers of VoIP are even aware of Marc's listed benefits, let alone motivated to switch by them. In fact, I suspect the listed features are not even that sticky. That is, if you offered all Internet Telephony users a circuit switched (POTS) service with unlimited long distance for $10 per month, most would quickly drop their VoIP service (with minor exception).

The right answer for today is that VoIP is cost driven. Thinking too much of your product's bells and whistles is a mistake.

Does that mean I believe there will never be a killer app? No. In fact, I began reading the article in the hopes Marc was going to offer enlightened prognostication. Although he didn't, I might make this suggestion. The future killer app may be in the extended cost savings of true Fixed Mobile Convergence (FMC). That is, soon, local phone companies may be offering a VoIP-based home (and later business) service that is supported over a Dual Mode Mobile phone (WiFi-GSM). Motorola offers an enterprise-only solution that does this today, although scaling that to carrier grade turns out to require a very different network solution. Once this occurs, users will have a combined mobile and landline/DSL-based service for roughly the price of just the mobile service (perhaps a bit more). Again — it will be cost savings driven. But the feature benefits will also be more profound. To implement such a capability correctly, the carrier really needs to utilize an IMS-based solution (versus UMA-hybrid or UMA-only). This means all the cool multimedia services that are emerging on the mobile networks will become directly available and compatible with the home-users phone and DSL services. That would be an interesting subject to discuss and may in fact be a compelling area for a future killer app.

– Peter Naleszkiewicz
Siemens – Mobile Networks Division
Atlanta, Georgia

The views expressed above are the personal views of Peter Naleszkiewicz. Peter is a senior technical program manager for Siemens — Mobile Networks division. Siemens does offer a leading carrier-grade IMS solution on which future FMC-VoIP solutions are being based.

Marc Robins responds:

I'm happy to see my column has produced such a strong reaction, albeit an extremely misguided and misinformed one. You seem to be, in fact, the very person I describe whose nose is stuck to a tree and thus blind to the forest. (And I've been called wrong before, like when top executives from AT&T (quote - news - alert) mocked me back in 1998 for publishing a magazine — this very magazine — devoted to a dog and pony show.) I didn't write or imply "the killer app exists already buried within the "wow" factor of VoIP." Rather, I suggested that in order to recognize the existent sticky apps, one must rediscover the sense of wonder — the "Wow" if you will — that attracted us to the technology in the first place.

Where you, Peter, only see "obscure," "second-tier" features, I see the makings of "compelling market movers." In fact, I'm in pretty good company in this regard. Most if not all of the foremost independent industry thought leaders, including publishers, editors, analysts, market researchers, and conference producers, feel strongly that the true appeal and value of VoIP is not in its low cost but in the really cool things you can do with it. This sentiment is echoed by the hundreds of lay users I hear from regularly, and I'm happy to say is being recognized by some leading industry players: Cablevision (quote - news - alert), for example, which markets it's Optimum Voice cable telephony service in the metro New York area where I live, just started airing commercials designed to educate viewers about such features as Web-based feature control and access to voice messages. And these ads contain not one mention of price.

No one, including yours truly, disputes that the discounted

pricing for VoIP (define - news - alert) is a major attraction for many consumers. It's a moot point. Many low-income and price conscious consumers — the same demographic that shops in Wal-Mart (quote - news - alert) for its everyday low prices — will undoubtedly be attracted first by low price, second by feature set. But your assertion that low price constitutes a killer app is just ridiculous (just as you posit that VoIP users would switch to unlimited POTS service for $10/month: Hello…? Remember AT&T's withdrawal from the consumer circuit-switched long distance market? The reality, Peter, is that no carrier could ever offer such a price point and hope to survive.)

In fact, rather than creating a "market momentum in excess of expectations," the current discount pricing campaigns are showing signs of running out of steam: rumors are percolating that customer acquisition rates are slowing dramatically. There are also indications that VoIP providers are starting to back away from the price-war front — introductory rates for many services are coming to an end, and sign-up and termination fees have been adopted by some in order to bolster short-term revenues. This isn't surprising and simply reflects business realities: While low-cost offerings have helped to raise mass-market awareness and create powerful incentives to sign up, they have also created dangerously low — or non-existent — profit margins for providers. A dirty little secret in the VoIP industry is the current high cost of customer support relative to traditional POTS service, as some implementations are still buggy and many new subscribers are technically inept and thus require costly handholding.

Regarding your mention of FMC, I wholeheartedly agree that this is an area ripe for exciting new capabilities, if not new killer apps. In fact, I've written about this several times in the past, in work I've done for BusinessWeek and for this magazine — most recently in my July 2004 Mind Share 2.0 column *Bridging the Wireline/Wireless Divide*. You're apparently a new Internet Telephony magazine reader and clearly a newbie to my columns, so you're forgiven for not being aware of this. However, the fact that you singled out FMC as the ONE technology area you feel is killer app laden gave me pause (and a few chuckles) due to the fact that you're not exactly an independent, dispassionate person in this regard. After all, you do work for the mobile network division of Siemens, which offers products for FMC solutions.

– Marc Robins
President, Robins Consulting Group

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Industry
NEWS

VTech Delivers Fully Integrated VoIP Phone

VTech Communications, Inc., a wholly-owned subsidiary of VTech Holdings Ltd., has announced a fully integrated VoIP broadband cordless 5.8GHz telephone, expandable up to four handsets, configured with Vonage's leading broadband service.

"VTech has introduced revolutionary products for more than 25 years, so we are excited to partner with them to drive the much anticipated era of broadband telephony," said Jeffrey A. Citron, chairman and CEO of Vonage Holdings. "Together, we're making it more convenient and economical to connect with people all over the world."

Market research from IDC and TIA forecasts nearly half of PC households will adopt broadband by 2007, with 17M consumers using VoIP services in 2008.

"Consumer interest in VoIP is increasing because it's significantly less expensive than regular phone service, offers equal or better clarity and is simple to set up, as long as you have the right equipment," said Donna Silbert, vice president of product management at VTech. "With VTech's first integrated VoIP telephone, we selected Vonage as the service provider because they have the experience, expertise and track record of offering premium service that is often better quality than the traditional landline. Vonage shares a similar vision with VTech, which is to migrate consumers seamlessly to VoIP services, delivering the tools they need that are easy to use."

VTech ip8100 Cordless Broadband Phone

VTech's ip8100 cordless broadband phone is a fully integrated VoIP telephone, containing a terminal adapter built into the 5.8GHz model. It includes two handsets and one base station, and is expandable up to four handsets, which allows users to distribute the service through the home without additional wiring. The VTech ip8100 plug and play solution does not require a separate router and it works with either a broadband cable or DSL modem. The phone also features a compact vertical design, fitting on crowded desktops where space is at a premium. VTech's ip8100 boasts 5.8 DSS technology, offering enhanced clarity and security. Other features include hands-free handset speakerphone, 50 name/number phonebook and 10 ringer melodies.

VTech 4106 VoIP Terminal Adapter

The VTech VoIP Terminal Adapter enables users to convert their current corded or cordless telephones into a VoIP telephone system so they can take advantage of the new calling features and low rates that VoIP service providers offer. The VTech 4106 connects to a cable or DSL modem, as well as a corded telephone handset or a cordless telephone base station. Unlike many other TA's in the market, the 4106 has both LAN and WAN ports which allow users to share their Internet connection with a PC without the need to install a router. The 4106 also has a small form factor, enabling traveling executives and other consumers to take it with them on the road and save money on all calls. Other features include parental controls and firewall features. Like the ip8100, the VTech 4106 requires subscription to Vonage services.

http://www.vtechphones.com

http://www.vonage.com

ClearOne Intros MAXAttach Conferencing System

ClearOne Communications Inc., the developer of the MAX Wireless conferencing phone, recently introduced its new MAXAttach wired conferencing phone system.

The MAXAttach system is a conferencing solution comprised of two complete phone units that link together. This provides even sound distribution in medium to small sized rooms by doubling the number of speakers and microphones available in single-unit conferencing phones — without doubling the cost.

In larger rooms, one or two additional MAXAttach phones can be added to deliver unprecedented sound quality compared to competing products, with up to three additional loudspeakers and four additional microphones for better audio coverage. The result is clear, intelligible audio for natural and effortless conversations in virtually any size conference room.

All connected MAXAttach phones have their own controls so that participants, regardless of where they are seated in the room, can conveniently dial a call, mute microphones or adjust volume settings without having to stand up and reach across the conference table.

MAXAttach uses audio processing technologies from ClearOne's XAP line of professional conferencing products to deliver clear, natural sound for interactive, productive group communication. Key technologies include:

- Gentner Distributed Echo Cancellation — improves audio quality by eliminating echo.
- Noise Cancellation — removes background noises, such as those from projectors, laptop computers and HVAC systems, making speech audio easier to understand.
- Full-duplex sound — allows participants on both ends of the call to engage in highly interactive, productive conversations.
- First-microphone priority — eliminates hollow "tunnel" sound by activating only the microphone closest to the person speaking.
- Three microphones per phone unit — provide 360-degree audio pickup to ensure participants are heard clearly at the far end of the call.

MAXAttach is currently available for shipping.

http://www.clearone.com

Inter-Tel Releases Axxess Software Version 9.0

Inter-Tel Incorporated has announced the release of Version 9.0 software for the Inter-Tel Axxess Converged Communications Platform. The converged platform is designed to enable customers to deploy Internet Protocol (IP), wireless, digital, or analog communications depending on their dynamic business requirements.

Axxess Version 9.0 software enhancements include the ability to offer Private Networking and IP endpoints on the same Inter-Tel Internet Protocol Resource Card (IPRC). Axxess v9.0 also offers Virtual Local Area Network (VLAN) Tagging support for Inter-Tel Model 8600, Model 8622, Model 8662 and Model 8690 IP endpoints. VLAN Tagging is designed to enable administrators to segment a single physical network into multiple virtual networks for simpler management, increased performance, improved voice quality, and enhanced network security. Additionally, this release includes support for Phantom Stations, which enables traveling associates to create virtual stations that do not require a physical endpoint, since these users are typically not in the office.

Axxess Version 9.0 software also adds barge-in capabilities and enhances the Silent Monitor feature to allow our customers to monitor and participate in agent calls for quality assurance and training efforts. Numerous diagnostic enhancements and new administration and development tools are also included in this release, designed to streamline system management and increase the productivity of network administrators.

The new features and functionality of Axxess v9.0 software are designed to enable a business to run its system more cost-effectively, ease the management of administration, and improve quality assurance and training initiatives.

http://www.inter-tel.com

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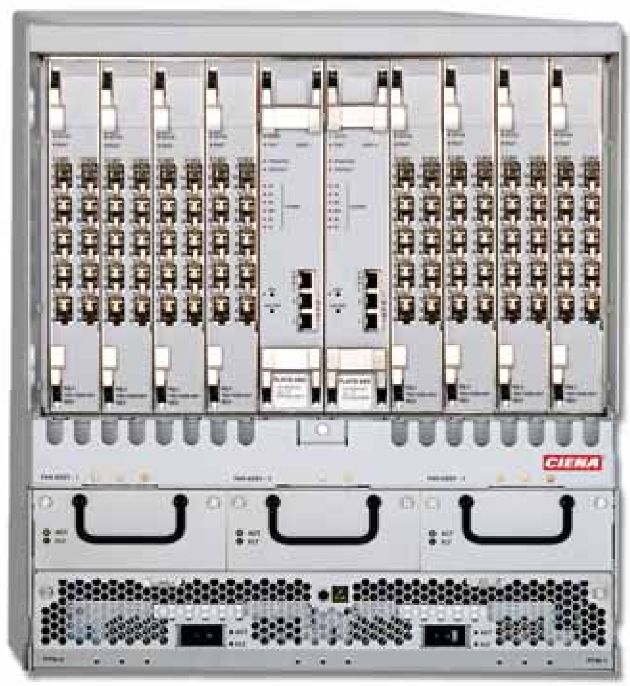
Ciena Unveils Carrier Ethernet Switch For Triple Play

Ciena Corporation recently announced the CN 4350 Ethernet Services Provisioning Switch. The CN 4350 is a carrier-grade Ethernet platform specifically designed for telco and cable service providers to deliver triple play services to residential and commercial customers over a converged packet network, including high-value Ethernet private line and local-area network (LAN) services. According to Ciena, the CN 4350 is designed to lower combined capital and operational costs by up to 70 percent as compared to certain competitive offerings.

Ciena's CN 4350 is already delivering a variety of video, voice and data services in multiple customer networks, including a deployment with ARMSTRONG Cable, a U.S. cable operator with subscribers across five states. The platform is generally available and additional deployments and trials are underway with other telco and cable service providers, including Adelphia Communications Corporation.



"We've been tracking the gradual addition of Ethernet capabilities into optical equipment, so the extent of Ethernet functions in a WDM chassis in Ciena's CN 4350 Ethernet Services Provisioning Switch is a pleasant surprise for us," said Michael Howard, principal analyst at Infonetics Research. "Ciena's CN 4350 gives telco and cable service providers a platform specifically designed to support Ethernet and triple play applications, which can be used as an operationally simpler alternative to deploying MPLS all the way to the customer."

Designed to deliver voice, video and data over a converged packet switched network, the CN 4350 enables Ciena's customers to broaden their services portfolio without increasing operating expenses and to improve services margins without complicating the network. Ciena's CN 4350 combines full mesh 160 gigabits per second fabric, VirtualWire VLAN switching, GbE and WDM transport interfaces, and carrier-grade reliability to provide the same service performance and operational simplicity of time-division multiplexed and gigabit Ethernet multiplexing, but with the cost-effectiveness and bandwidth efficiency of Ethernet switching. In contrast to general purpose data-switching architectures, the CN 4350 delivers absolute QoS and assigns each service its own "virtual wire" with guaranteed end-to-end bandwidth and performance, which is critical for concurrently switching and transporting best effort and QoS-sensitive services at 100 percent utilization to ensure optimal performance and service level requirements.

Currently deployed in service provider central offices and cable headends, Ciena's CN 4350 includes a number of features that make it a true carrier-grade Ethernet switch with no single point of failure for non-stop triple play services. The platform approaches 99.9999 percent availability by supporting non-service affecting software updates, a SONET-style embedded signaling channel for performance monitoring, integrated optical protection switching, and hardware-accelerated Layer 2 restoration.

The CN 4350 supports fully redundant power, fans, switch fabric and controller modules, has separate data and control planes, and is NEBS Level 3 certified. Ciena's fault-tolerant switching and control complex, which provides persistent forwarding and maintenance of the spanning tree topology during switchover, provides highly reliable converged services without the cost and administrative complexity of dual routers at every location running virtual router redundancy protocol (VRRP).

http://www.ciena.com

ARRIS Announces Industry Milestone With Comcast

ARRIS recently announced the successful completion of multiple Cable Modem Termination System (CMTS) hitless software upgrades with Comcast. Accomplishing a "hitless upgrade" has been an important goal within the industry. This capability enables operators to more seamlessly perform the software upgrades needed to launch new high-speed and voice services. Results can include: increased network reliability, enhanced customer experience, and reduced operator costs.

Comcast and ARRIS leveraged the ARRIS Cadant C4 CMTS hardware redundancy features and automated software upgrade process in their hitless upgrades to the Cadant C4 CMTS Release 4.0 software.

"At Comcast, we are focused on continuously advancing our network, rolling out new built-for-broadband services, and ultimately creating the best broadband experience for our customers," said Dave Fellows, Comcast chief technology officer. "ARRIS' industry milestone fully supports this vision, enabling Comcast to support all of our new service launches with regular software upgrades that we expect to be transparent to the customer. We value ARRIS' understanding of the important need to balance technology advancement with the customer experience, and we are pleased to have worked with them on achieving this industry milestone," continued Fellows.

http://www.arrisi.com

Microsoft Intros Solution For Hosted Messaging And Collaboration

Microsoft Corp. has announced Microsoft Solution for Hosted Messaging and Collaboration 3.0, a next-generation hosted service designed to addresses the requirements of small and medium-sized businesses (SMBs). The new offering, which is made available through Microsoft's service provider and hosting partners, builds on the strengths of Microsoft Solution for Hosted Exchange 2003 by adding support for hosted versions of Microsoft Office Live Communications Server 2005 and Windows SharePoint Services. With this solution, SMBs can enjoy the robust, enterprise-class e-mail, shared calendaring, contacts, document collaboration and instant messaging capabilities typically used by larger companies.

Microsoft Solution for Hosted Messaging and Collaboration is a hosted version of Microsoft's family of productivity server products: Microsoft Exchange 2003, Live Communications Server 2005 and Windows SharePoint Services. For service providers, the solution offers a scalable, multiservice platform through which they can deliver a rich set of productivity services to small and medium-sized businesses, as well as enterprises and consumers.

"Service providers recognize the huge untapped opportunity of offering 'software as a service' to small and medium-sized businesses," said Pascal Martin, general manager for Hosting Solutions at Microsoft. "With Microsoft Solution for Hosted Messaging and Collaboration, we are enabling our service providers to run a profitable business and address their SMB customers' needs."

The new solution offers the following key benefits:

• ***Faster time to market for hosters of all sizes***. The new solution includes an upgrade to Microsoft Provisioning System based on Windows Server 2003 and a new deployment tool that automates 85 percent of the installation process.

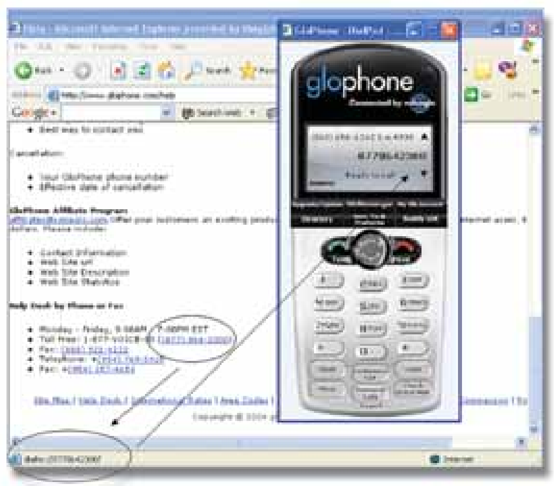
• ***Maximized operational efficiency and cost-cutting***. Using key products such as Microsoft SQL Server 2000 and Microsoft Operations Manager 2005, service providers are able to control their capital expenditures and reduce the cost of operation and centralized management through new reporting, filtering and security services.

• ***Increased revenue from launch of services***. The new solution integrates two enterprise products — Windows SharePoint Services Hosting and the new Live Communications Server 2005 Hosting — making it easier for service providers to differentiate themselves and add new revenue opportunities.

www.microsoft.com/serviceproviders/hosters

Voiceglo Debuts Click-To-Call Feature

Voiceglo announced that it has enhanced its Web- and PC-based applications, GloPhone and GloConnect, with a new feature entitled Click-To-Call. Click-To-Call is designed to allow users to dial any number they see on a Web site in one easy step using Voiceglo's browser-based telephone service. This convenient feature turns any phone number listed on a Web site into an active link, allowing GloPhone and GloConnect users to simply place their cursor over the desired phone number, click their mouse and make a phone call to the listed party. The Click-to-Call application dials the number within seconds, and users can participate in a phone conversation directly from their computer.



With GloPhone and GloConnect, the new Click-to-Call option works with any published U.S. or Canadian telephone number on the Internet. VoiceGlo plans to add international destinations shortly.
http://www.glophone.com

InfiniRoute Launches VoIP Peering Service

InfiniRoute Networks, Inc., has announced the availability of its Voice over IP (VoIP) peering service for Wireless Carriers. InfiniRoute's Managed VoIP Peering (MVP) service, launched in 2004, is a carrier- neutral VoIP Peering service designed to integrate and manage voice and IP routing for wireline, wireless, and emerging carriers.

InfiniRoute's MVP service now provides mobile carriers with the ability to terminate and receive international traffic using cost-effective, high quality VoIP technology. Carrier benefits include:

• Greater access to global markets. Mobile Network Operators without extensive TDM networks can now economically reach international termination partners without significant capital expenditures.

• Lower cost of operations. Mobile networks can leverage VoIP to deliver international traffic at lower costs over a scalable and inexpensive packet-based infrastructure.

• Reduced technology risk. InfiniRoute's proven solution lowers capital expenditures associated with emerging technologies.

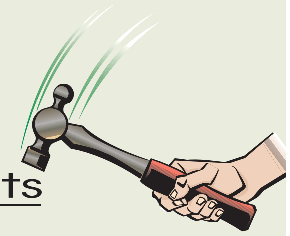
• Faster time to market. The fully managed service, combined with InfiniRoute's VoIP expertise, enables mobile network operators to enter the global VoIP telecommunications market.

http://www.infiniroute.com

NGT Announces 6DegreesIP Enhancements

New Global Telecom (NGT) has announced that the company has completed rolling out product release 3.2 of its 6DegreesIP managed wholesale VoIP product suite. Release 3.2 introduces new product components as well as enhancements to the product suite.

6DegreesIP release 3.2 includes the introduction of a Trunk Replacement product offering. This solution provides CLASS 5 features for the PBX trunk component of a converged voice and Internet service. Connectivity to the end user PBX is provided through standard integrated access devices (IADs) over dedicated T1 links. This Trunk Replacement product significantly enlarges the addressable enterprise market for NGT's service provider customers.
http://www.ngt.com



Quick Hits

Carrier Equipment Study Released

Mercator Capital recently completed a research study on recent trends in the VoIP carrier equipment market. The combined revenue generated by Class 5 lines, Class 4 lines and Media Gateways in Q3 2004 was $169 million. According to the report, in Q3 2004, Carrier IP Centrex equipment generated $7.7 million in revenues. The Session Border Controller segment generated $15.5 million, while the Media Server segment generated $12.5 million.
http://www.mercatorcapital.com

Vonage Crosses 400,000 Lines

Vonage, the leading broadband telephony provider, today announced it has exceeded 400,000 total lines on its network, doubling its subscriber base in less than six months since reaching the 200,000 line mark. The company ended 2004 with more than 390,000 lines in service having added 115,000 lines in Q4 2004 alone.
http://www.vonage.com

Operators Deploy Tekelec Gateways For 3G

Tekelec announced tier-one mobile operators in North America and Asia are deploying the Tekelec 8000 Wireless Multimedia Gateway (MG) with Universal Mobile Telecommunications Service (UMTS) capabilities. These customers will leverage Tekelec's product architecture and flexibility to increase network capacity while seamlessly migrating to third-generation (3G) networks.
http://www.tekelec.com

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TriLink Gateway Streamlines Installation, Improves User Experience

Westell Technologies, Inc., a provider of broadband/DSL access products, gateways, and conferencing services, announced a new broadband access device — TriLink Gateway — designed to combine multiple voice and data technologies into one unit including VoIP, ADSL Internet access, wireless networking, and alternate access such as Fiber to the Premises (FTTP).

TriLink Gateway integrates multiple broadband technologies into a single device that enables self-installable wireless/wired networking for home or small businesses without the need to purchase multiple devices. As a result, the company outlines several product benefits for customers:

- Easy to use and install: No need to purchase and connect individual conflicting devices to access multiple applications.
- Built-in ADSL with integrated modem and router makes turn up simple and reliable.
- Dedicated WAN/LAN Ethernet port supporting access to FTTP applications provides a path to the future.
- Two integrated VoIP access ports with POTS backup eliminate hardware clutter and complexity.
- 54 Mbps WiFi wireless networking in addition to four-port wired networking offers flexibility and local connectivity options.
- Remote device management via the DSL Forum's TR-069 protocol reduces operating costs and improves users' experience.

Product features for the new TriLink Gateway include: four Ethernet ports with managed switch, one dedicated WAN/LAN Ethernet port for WAN access and/or fiber applications, 54 Mbps WiFi 802.11g wireless service, built-in QoS for managing multiple, simultaneous applications, optional USB port, and two RJ11 jacks for use with conventional telephones to access VoIP. The product also includes a TR-069 client to provide manageable upgrades from a remote management system.
http://www.westell.com

Wireless VoIP Solution Reduces Costs, Enables Deployment

Quintum Technologies and Wi-LAN, Inc., have entered into an agreement to offer a secure wireless high-speed communications solution to businesses, telecom service providers, and government enterprises.

The solution is designed to enable service providers and WISPs to penetrate new markets by allowing them to bypass the incumbent telco to provide value added services such as voice and high-bandwidth data services to customers who need secure and robust networking. Wi-LAN's wireless network products along with Quintum's patented MultiPath Architecture provide the ability to simply integrate into existing voice and data networks while providing failover and redundancy that cannot be achieved utilizing traditional VoIP gateways.

Enterprise customers can securely connect multiple locations with both data and VoIP Services. Quintum's ability to integrate with existing PBX and IP networks, coupled with Wi-LAN's simple to deploy products allow for a highly available and low impact deployment.

"The Quintum Tenor products provide Wi-LAN with an easy, cost effective solution for our customers to deploy wireless Voice-over-IP networks," said John Seliga, VP of marketing of Wi-LAN. "This unique solution allows us to deploy Tenors on both the customer premises and service provider access points, minimizing the disruption to the existing network infrastructure that is common with traditional VoIP gateways."

"Wi-LAN's VoIP solutions offer a much needed alternative to traditional land line PSTN networks," said Charles Rutledge, VP of marketing for Quintum Technologies. "Tenors work seamlessly over both types of networks and offer flexibility to service providers that no other VoIP equipment can."
http://www.quintum.com
http://www.wi-lan.com

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The Ultimate Performance Machine for IP-Based Service Delivery

Pannaway Technologies' intelligent Personal Branch Gateway (PBG™) has captured the imagination and streamlined voice, video and data service delivery for telcos worldwide.

Clearly, the PBG's extreme styling, ADSL2+ horsepower, and "fiber to the curb" access speeds are for those who appreciate the thrill of unleashed data speeds up to 100Mbs, along with advanced voice options like distinctive ring and lifeline support.

More than a modem, the PBG is technologically engineered to capture the very essence of IP-based services – without the joy ride of having to modify or replace an existing network infrastructure.

Go ahead: Drive IP-based service transport in your marketplace today.

SCN™
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The Pannaway Service Convergence Network (SCN™) is the industry's first managed end-to-end IP solution for secure, converged broadband transport services.

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Legerity Introduces Integrated Voice Solutions For VoIP

Legerity, Inc., a provider of analog/mixed-signal integrated circuits (ICs) for voice and data networks, has announced availability of the newest members of its VE880 VoicePort Series, the Le88111, Le88116, Le88131 and Le88136, used for providing VoIP connectivity to analog phones. These VoicePort products combine SLIC and codec functionality into a single device. The new single channel devices are designed to increase design flexibility, improve system performance, and reduce system BOM cost.

The new VoicePort devices, with pin-compatible options for 100V or 150V, as well as narrow band or wide band, provide designers the flexibility to develop one application with population options for any market worldwide. In addition to design flexibility, VoicePort devices are highly programmable, with an eye towards simplifying software development, minimizing development time, and speeding time-to-market.

"Legerity has developed a truly unique, highly integrated voice solution for the VoIP market. Designers can now implement the most cost effective single channel voice solution in a matter of days, instead of months or years," said Mike Stibilia, vice president of marketing and applications for Legerity. "Designers benefit from the easy to use features of VoicePort without sacrificing performance."

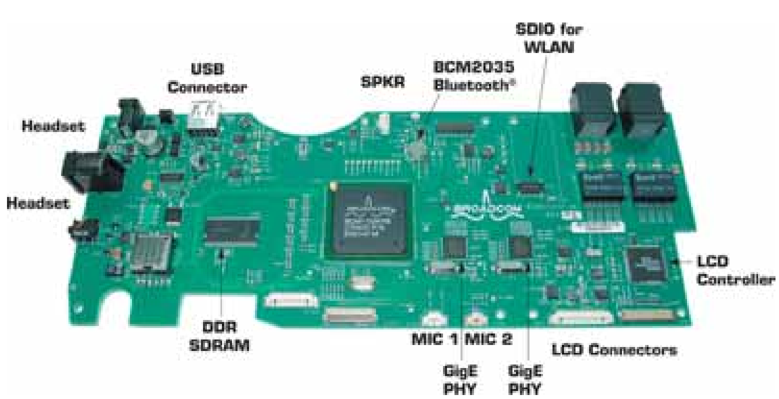
The Le88111, Le88116, Le88131, Le88136 are available for production orders today. These VoicePort products are targeted toward consumer premise equipment (CPE) such as voice enabled cable modems, DSL modems, gateways, and media terminal adapters (MTA). In addition, these devices are ideal for FTTx, WLL, PBX, and other SME applications. These FXS devices have a complementary FXO device, Le88010, which is also available. The Le88111 is targeted at $2.50 per device in high volume.

http://www.legerity.com

Broadcom Unveils Gigabit Ethernet IP Phone Chip

Broadcom Corporation recently announced an Internet Protocol (IP) phone chip that incorporates Gigabit Ethernet (GbE) switching, end-to-end security, advanced quality of service (QoS) techniques and increased processing performance in a single-chip design. This highly integrated IP phone chip features new Ethernet switching capabilities that enable deployed IP phones to achieve maximum throughput over Gigabit Ethernet networks. GbE technology is now becoming widely deployed in enterprise networks to support the increased bandwidth requirements of converged voice, video and data communications. Broadcom's new Gigabit Ethernet IP phone chip is the successor to the company's 10/100 Mbps (Fast Ethernet) IP phone chip, which has been widely deployed in enterprise IP-PBX systems from leading equipment vendors worldwide.

"The deployment of Gigabit Ethernet in enterprise networks is rapidly taking over existing Fast Ethernet installations due to its ability to provide more bandwidth in support of voice, video and data services to the desktop, at compelling price points. IP telephony in the enterprise will grow at a CAGR of 43 percent over the next five years," said Sean Lavey of IDC. "The Gigabit Ethernet IP phone chip from Broadcom is an ideal solution for next-generation IP phones as the enterprise network evolves to Gigabit Ethernet."



The Broadcom BCM1103 IP phone chip is designed to enable manufacturers to build IP phones with integrated Gigabit Ethernet switching and other essential features to support: improved voice quality; guaranteed voice security; and graphics-driven applications that demand more processing power.

The chip integrates a 10/100/1000 Mbps Ethernet switch and two 10/100 Mbps Ethernet transceivers, allowing for the development of traditional 10/100 Ethernet IP phone designs without the additional cost of adding external transceivers. With the addition of external Gigabit Ethernet transceivers, manufacturers can easily upgrade their designs to create Gigabit Ethernet IP phone models.

http://www.broadcom.com

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Ixia Announces Packet Over Cable Testing Solution

Ixia has announced the latest release of IxVoice, its comprehensive analysis solution for converged telephony testing. This release of IxVoice introduces a number of enhancements designed to improve and simplify the verification of packet data over cable solutions, including support for traditional PSTN and VoIP technologies and related protocols, such as MGCP, COPS, and SS7.

Ixia's IxVoice incorporates a Packet over Cable test suite that makes it possible for equipment manufacturers and cable operators to quickly verify conformance, performance, and interoperability of their solutions. The Packet over Cable test suite can emulate analog phones as well as the traffic from the CMTS devices, eliminating the costs associated with acquiring and managing a large test bed that traditionally would include thousands of cable modems and MTA's (Media Terminal Adapters).
http://www.ixiacom.com

TI, AT&T In VoIP Pact

Texas Instruments and AT&T have announced that they are working together to provide VoIP equipment designers and manufacturers new platforms for the development of VoIP products compatible with AT&T's global IP network. The combination of TI's VoIP software and silicon technology and AT&T's broadband telephony solutions is designed to quickly enable communications equipment manufacturers to develop new equipment, such as terminal adapters and VoIP gateways, for use in managing voice services through broadband connections.

As part of the agreement, TI-based reference designs for voice gateways will be compatible with AT&T's CallVantage Service, via AT&T's certification lab testing. One of the first TI-based VoIP products that will be available through AT&T's CallVantage Service is VTech Communications' VoIP6322 Dual Line Corded/Cordless Broadband Telephone System. The 2.4GHz DSS VoIP6322 can be used with both traditional and broadband phone services and includes a corded base station and one cordless handset.
http://www.ti.com/broadband
http://www.att.com

TRENTON Expands CompactPCI Product Line

TRENTON Technology Inc. announced its CP10 single-board computer with the Intel Pentium processor and the Intel E7501 chipset. The CP10 has many key features, including a front-access PMC slot that supports PCI or PCI-X cards, local storage and IPMI options, dual 10/100/1000Base-T Ethernet ports and provisions for up to a 2GB plug-in DDR200/266 memory DIMM. The optional Rear Transition Module, RTM20, provides rear access to the SBC's I/O and status LEDs.

The CP10 is currently available at processor speeds of 1.8GHz and 1.6GHz and supports processors with a 400MHz system bus. Future processor versions, with faster processing speeds and a 533MHz system bus, will also be supported on the CP10. The Intel E7501 chipset supports Intel's Hub Link interface technology for optimum data exchanges between the processor, system memory, I/O controller and the Ethernet networks.

http://www.trentontechnology.com

Ingate Unveils High-Capacity SIP Products

Ingate Systems has recently taken the wraps off of two new products: the Ingate Firewall 1600 and Ingate SIParator 60.

Both are designed for organizations with high demands for capacity, throughput, reliability, and VoIP. Both the Firewall 1600 and SIParator 60 are only 1U high, fitting into an ever-more-crowded network operations center. Both feature six interfaces, two of which can be used at Gigabit speed; and each supports up to 360 simultaneous RTP sessions (e.g., VoIP calls).

The Firewall 1600 and SIParator 60 can be upgraded with a crypto accelerator, which makes the VPN even faster and relieves some pressure on the processor. The products are managed by the network security administrator via a graphical html-based user interface. Ingate Firewalls and SIParators include an encrypted Virtual Private Network (VPN) termination module. In addition, both products have a new, easy-to-read display that communicates the status of the unit.

"The Firewall 1600 and SIParator 60 offer customers unique solutions for bringing SIP-based VoIP, IM, presence, and a host of real-time person-to-person IP communications to large-capacity enterprises," said Olle Westerberg, Chief Executive Officer, Ingate Systems. "These products deliver IP PBX functionality, support for Microsoft Office LCS 2005 and SIP-enabling technology — combined with the very best in network security."

Ingate Firewall 1600

Included with the 1600 are 10 SIP user licenses that can be used for the registration of SIP user agents, such as phones and soft clients, on the SIP registrar. Five SIP traversal licenses also come standard, allowing up to five calls to traverse the firewall at the same time. Additional SIP user licenses and SIP traversal licenses can be purchased at any time.

Like all Ingate firewalls, the 1600 boasts the company's security technology, offering IP PBXs a secure solution for VoIP, not only within the company, but also globally. The full SIP proxy inspects the SIP packets before sending them on. TLS encryption ensures privacy when communicating, making eavesdropping, call hijacking and call spoofing harder to do. All Ingate Firewalls support stateful inspection and packet filtering. In addition to full support of SIP, Ingate Firewalls have a proxy for all standard protocols, including TCP, UDP, FTP, and DHCP.

Ingate SIParator 60

Included with the SIParator 60 are 50 SIP user licenses that can be used for the registration of SIP user agents, such as soft phones and soft clients, on the SIP registrar. Twenty-five SIP traversal licenses also come standard, allowing up to 25 calls to traverse the firewall attached to the SIParator at the same time. Additional SIP user licenses and SIP traversal licenses can be purchased at any time.

The SIParator 60 can be configured as a part of the DMZ or in a standalone mode. In both cases, the benefits of SIP-based communications can be added to the network simply and easily.

Both new products also boast full support and compatibility for Microsoft Office Live Communications Server (LCS) 2005, allowing voice, video, IM and presence applications to work outside the LAN.

http://www.ingate.com

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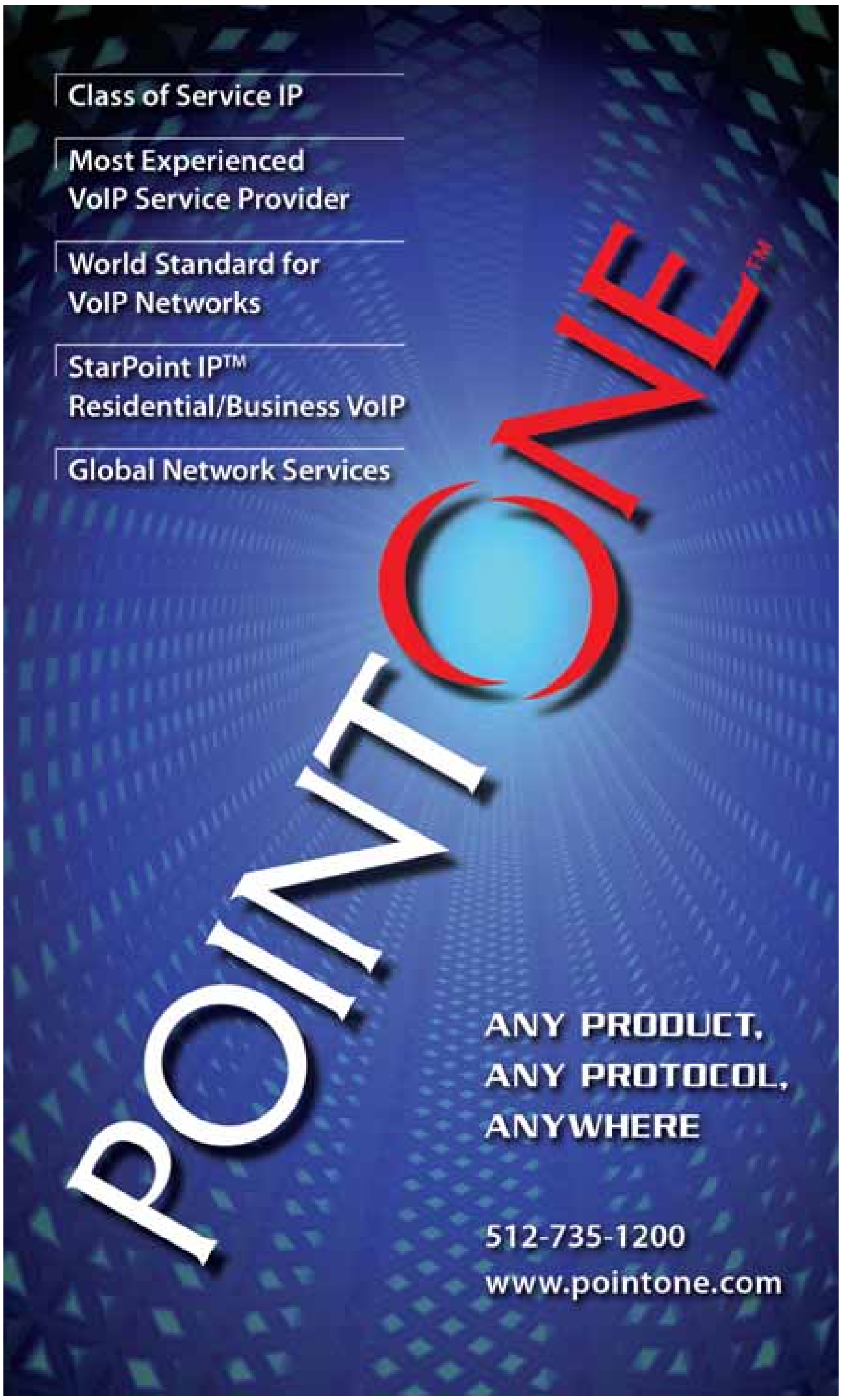
ASC Provides Free VoIP Recording Solution

ASC has announced that a trial version of its new VoIP recording solution, EVOip, will be available for free to any company that requests it. The full working solution, offered for use through December 2005 with no stipulations, allows companies to determine its value as a business tool. Completely software-based with no proprietary hardware required, EVOip may be installed on a Windows 2000 / XP PC to record, store, search and replay VoIP call data.

Günther Müller, Chairman and CEO of ASC, said, “ASC stands firmly behind its products and is pleased to offer our IP recording solution for free and independent evaluation. We re-invest about 15 percent of our revenues into research and development, one reason why we’re always a step ahead of the competition. EVOip fills a vital need for companies to ensure protection from liability and full legal compliance.”

Calls are categorized by selected parameters such as date, start/end time, call duration, channel or IP address; and then stored on the company’s server. The solution works with leading VoIP solutions, supports enterprise-wide applications and is currently tested and approved for Alcatel, Avaya, Cisco, DeTeWe, innovaphone, Mitel, NEC, Nortel, Siemens and Tenovis Networks.

http://www.asctelecom.com



Telrex Announces Release Of CallRex Version 3.0

Telrex has announced the release of CallRex Version 3.0. CallRex Version 3.0 includes many new and unique features including; look-back call recording, support for Citrix terminal services, the ability to export multiple calls, improved trigger filtering, automatic deletion of recorded calls according to pre-defined criteria, remote polling of recorded phone calls, improved recording quality, re-start capability for remote data collectors, improved call compression, improved record-on-demand capabilities, the ability to automatically receive software updates online and many other additional innovative features.

CallRex works with numerous IP PBXs, and currently supports products from 3Com, Mitel, Avaya, Cisco, ShoreTel, Nortel, Siemens, NEC, Zultys, Artisoft, and many other softswitches and gateways. CallRex is sold through a network of resellers.

http://www.telrex.com/callrex.htm

IBM Selects EADS' Centergy For Sprint Project

EADS recently announced a new project working with IBM Global Services to unify all 40 Sprint PCS contact centers worldwide under a single platform. According to Jay Lassman, Research Director for Gartner, Inc., "This approach is consistent with Gartner studies that project cost savings for large call centers by consolidating networks and centralizing applications, while at the same time, integrating virtual call center sites where appropriate."

The first phase, which will be fully integrated this year, will initially include 12,000 agents in seven countries, including approximately 6,500 agents working from remote locations.

To power the project, IBM has purchased EADS' Centergy Contact Center solution which includes Centergy Remote and Centergy Reporting applications. Centergy can handle more than 1.5 million calls per hour in peak periods. With this capacity, IBM can meet Sprint PCS' requirements, including the need to handle 250 calls per second.

http://www.eadstelecom-na.com
http://www.ibm.com/services

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CrystalVoice Meets 3Com Testing Requirements

3Com Corporation and CrystalVoice Communications, Inc., recently announced that CrystalVoice has met the interoperability testing requirements of the 3Com Voice Solution Providers Program (VSPP). CrystalVoice Remote Extension is the first product certified for both the 3Com NBX and 3Com VCX 5.0 IP Telephony modules through the VSPP.

The new certification confirms that CrystalVoice's Voice over Internet (VoI) software solutions are fully compatible with the 3Com NBX and VCX IP Telephony modules, providing enterprise customers with an advanced portfolio of real-time telephony, presence, messaging, and conferencing solutions. CrystalVoice Remote Extension delivers an important component to a distributed or mobile work environment through combining CrystalVoice's advanced Acoustic QoS technology and SIP support, which provides high-quality voice communications over the Internet, with its secure, scalable, high-availability architecture. CrystalVoice's solutions are designed to deliver productivity enhancing, cost-effective solutions for remote and mobile employees, interoffice communications, contact centers, collaboration, conferencing and interactive on-line learning.

"The recent lab testing expands the relationship between 3Com and CrystalVoice to build upon the success of the CrystalVoice solution for the 3Com NBX IP telephony platform," said Pat Rudolph, 3Com vice president of solutions architecture. "This certification confirms the CrystalVoice Remote Extension software solution as interoperable with both the 3Com NBX IP telephony system for small business and medium enterprises and the VCX IP telephony system for large enterprises."

"CrystalVoice is proud to complete the rigorous testing process and receive the 3Com VCX system certification," said Steve Zola, CrystalVoice president and CEO. "This certification is a testament to CrystalVoice's partnership with 3Com, and the company's commitment to providing industry leading, enterprise-class IP Telephony solutions to the enterprise. The company intends to offer its customers and partners the most advanced solutions available to meet their voice communications needs."

http://www.CrystalVoice.com
http://www.3com.com/voip

New Features For deltathree's Global Reseller Program

deltathree, Inc., announced the launch of new advanced features and services for its Reseller Program. deltathree, which offers resellers around the world the tools necessary to deliver customizable, comprehensive VoIP calling services, is enhancing its offerings with a range of new features, including inbound calling, call forwarding, voicemail, Web call and Web and SMS callback features. In addition, deltathree has expanded the range of devices that it has certified are interoperable with its services, and recently launched new pricing plans and an enhanced look and feel for its VoIP services.

Among the new features being introduced is an expanded choice of inbound local phone numbers. The service provides customers with a choice of receiving calls via these phone numbers from a variety of countries, including the United States, Austria, Belgium, France, Germany, Ireland, Israel, and the United Kingdom. These numbers are available to end users without regard to where they are physically located around the world.

deltathree's new customizable Web-based and SMS-based callback features offer two ways for end users to make VoIP calls from anywhere to anyplace in the world. These callback features allow the user to decide if the call should occur immediately or at a later time.

deltathree's VoIP programs offer its reseller partners all of the tools they need to deliver comprehensive VoIP calling services and build revenue streams. Through the Reseller Program, resellers can participate in the VoIP revolution that is transforming the telecommunications industry. The Reseller Program is ideal for a broad range of potential deltathree customers, including: calling centers/shops, Internet Cafes, local hardware and software distributors, local ISPs, calling card and callback companies, telecom consultants, Web portal and entrepreneurs.

http://www.deltathree.com

By Marc Robins

Testing Is Fundamental

No doubt about it, the enterprise market for IP telephony gear is on a tear. According to market watchers such as Synergy Research, IP PBX (define - news - alert) equipment revenues are expected to approach the $10 billion mark this year. And 2005 is also expected to herald a major milestone for this equipment sector, when revenues should surpass those of traditional TDM-based PBXs.

As the spread of IP telephony and converged networks into the enterprise is opening the door wide to an array of new enhanced applications and services, as well as dramatic cost savings and productivity increases, the growth of converged networks is also bringing with it a unique set of challenges.

New converged networks will need to serve "double duty" when they begin to support real-time voice (and possibly "triple duty" with new video applications), whether an enterprise is converting to a pure IP solution or making due with legacy equipment and a gateway. Such new demands can quickly overwhelm these networks if proper planning and testing isn't conducted prior to deployment.

If personnel tasked with network management fail to ensure that the network and equipment are up to the new tasks at hand, more and more companies will start running the risk of experiencing an array of quality of service (QoS) issues. These QoS issues can result in an unacceptable degradation of voice and video traffic quality, and include packet loss, excessive latency levels, jitter, clipping, and excessive levels of echo. Indeed, an untested IP telephony deployment is simply asking for trouble, as these QoS issues can seriously impact the quality of corporate communications at minimum, and lead to complete network failure in the worst-case scenario.

Essential Predeployment Considerations

In order to get the most out the new IP telephony gear on the market, and to avoid the risk of communications disruption and network failure, there are a number of essential considerations one must make before choosing and deploying a new enterprise IP telephony solution.

1. Get a Handle on Network Performance. What is the network currently capable of? What loads and traffic types can be placed on it before performance is impacted?

2. Understand the Level of Interoperability. Are data and voice equipment interoperable with each other, as well as any legacy equipment kept in the mix?

3. Determine the Impact of Security Equipment. Firewalls and other network security equipment can seriously affect the quality of voice and video traffic by introducing additional jitter and latency, and in some cases make effective communications impossible.

4. Determine the Impact of Software Upgrades and New Hardware. When making changes within the network, what is the impact these changes will have on network performance?

5. Assess Real-World Performance Versus Vendor Representations. Many vendors make claims such as "guaranteed interoperability" with Product Type A or guarantees regarding load capacity or scalability. But what happens when products are deployed in the real world?

6. Right Size the Equipment Investment. What is the amount of equipment and the scalability you will require? Having a certain amount of foresight is a requirement as you begin the migration.

Testing Is Fundamental

Comprehensive lab testing, or at the very least pre-deployment testing, is increasingly viewed as the most reliable way to ensure a converged network will operate successfully in its new configuration. A number of companies, including Spirent Communications (www.spirentcom.com), offer innovative testing solutions to address this increasingly important market requirement.

Spirent's recently introduced Abacus 5000 IP Telephony Migration Test System combines IP telephony and PSTN testing in a single platform. This solution offers real-time call statistics and protocol analyzers for identifying all types of QoS issues. The system works by generating real voice streams and simulating enterprise traffic loads for an accurate analysis of voice quality impact. It also offers the ability to test interoperability of a number of devices and switching schemes, including analog, TDM and VoIP (define - news - alert) traffic. The call generation feature supports a number of popular IP telephony protocols, including SIP, H.323, Megaco/H.248, MGCP, and RTP.

On the service front, Spirent has started to offer a new integrated IP Telephony Assessment Service for companies that recognize the need to perform network testing, but don't want to incur the additional cost of purchasing a testing system. The Spirent IP Telephony Assessment Service is designed to provide analysis of both IP and PSTN networks in separate and converged configurations. First, a complete assessment is conducted of a company's requirements in terms of desired applications and scalability, as well as the type of network and services being implemented. Then a customized network analysis follows, which includes voice quality assessment and connectivity analysis based on location. A summary report is then provided that features detailed call quality statistics and charts, as well as problem identification for additional testing. Follow-up testing can address specific QoS problems like latency and jitter. IT

Marc Robins is Chief Evangelism Officer of Robins Consulting Group, which offers an array of services to the IP telephony industry. He has been involved in the telecommunications industry as a reporter and analyst, trade show producer and publisher, and marketing executive and consultant for more than 24 years. For more information, call RCG at 718-548-7245 or e-mail robinsconsult@optonline.net.

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By Tony Rybczynski

Be Your Own Browser

PC-based browsing was the key enabler behind the Internet boom. But, how often would you like to get Web-based information or even undertake Web-based transactions, but don't have access to a PC or PDA (define - news - alert) or are outside the WiFi (define - news - alert) hotspot coverage area? Wouldn't it be nice to give customers broader access wherever they are and whenever they want? Now you can with Voice XML (define - news - alert) and advanced speech technologies! Phones are available just about everywhere in the world, are always on (and cell phones have longer battery life than WiFi devices), and don't have to be booted up.

What Is Voice XML?

XML (eXensible Markup Language) — a standard of the World Wide Web Consortium (W3C) — is the most widely accepted, platform-independent standard for building structured documents for Web applications. Voice XML is a dialect of XML developed to write voice dialogs for self-service solutions, and is also being steered by the W3C.

In the Web world, Voice XML is to voice interfaces what HTML is to visual displays. HTML applications are accessed via a graphical Web browser with display, keyboard, and mouse. In contrast, Voice XML applications are accessed via a voice-capable device that accepts audio and touchtone keypad input and delivers audio output... such as a telephone. In short, Voice XML empowers users to interact with the Web through any wired or wireless phone — making Internet content available to anyone with a phone. For users, Voice XML enables them to interact with the application in the most natural way, namely by speaking and listening. Voice XML scripts use a combination of speech and touch-tone commands to exchange data between people and machines, independent of the vendor's hardware. Application developers can create audio dialogs that use speech and touch-tones as input, and deliver synthesized speech or digitized, pre-recorded audio as outputs.

Natural language advanced speech applications actually can make it easier to get information and complete a transaction.

Voice XML documents can perform a variety of functions, such as user prompting, natural language speech recognition enabling the user to provide multiple pieces of information in a single utterance, event-handling features (e.g., timeout or unrecognizable input), branching ("if-then-else") and Boolean ("and, or, not") logic, and progressive prompting to better handle invalid responses to a prompt.

It's The Application, Stupid!

Voice XML is well-suited for applications that require relatively little input from the user and deliver highly targeted output that generally is available from an HTML Web interface. In many ways, natural language advanced speech applications based on Voice XML actually can make it easier to get information and complete a transaction. Because it's interruptible, the user can just say what he or she wants minimizing the time spent to complete the task at hand.

The simplest Voice XML application is for information retrieval, such as account balances, airline flight information, or weather from a Web site. Voice input can often handle large vocabularies much easier than touch-tone. Free-form street addresses for a city, or stock quotes for a specified company and period are good examples. Speech is also well-suited for single utterance input of multiple related items. For example, the user could say, "What is the interest rate for a 20-year fixed-rate jumbo mortgage?" Voice XML is also naturally suited for customer service applications, such as parcel shipment tracking and online banking, and for accelerating call center services. In addition, personal name dialing, one-number "follow-me" services, teleconferencing set-up, and other telephony features can be voice-enabled through Voice XML. Because security features that apply to the Web, such as firewalls and encryption, can be applied to voice applications as well, Voice XML can be used to create secure intranet applications that voice-enable internal processes, such as supply ordering, HR self-service, and corporate news. This underscores an important benefit of Voice XML: it enables customers to share their Web-based infrastructure and resources with voice processing applications. Finally, Voice XML can unify voice and electronic channels, for example, by allowing users to read their voice mails, dictate their e-mails, and originating and terminating pager messages on the phone.

Summary

Voice XML is a powerful yet simple language for building dialogs that blend the voice world with the Web to enable innovative new self-service applications. For IT, voice applications can now be constructed with widely available Web application development tools. For the user, just say what you want! IT

Tony Rybczynski is director of Strategic Enterprise Technologies at Nortel. He has over 30 years experience in the application of packet network technology. For more information, please visit http://www.nortel.com.

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By William B. Wilhelm, Jr., Esq.

Federal Preemption: What Now?

On November 12, 2004, the Federal Communications Commission (FCC) issued a historic decision and preempted the Minnesota Public Utilities Commission (MPUC) from applying its traditional "telephone company" regulations to Vonage's (news - alert) Voice over Internet Protocol ("VoIP") (define - news - alert) service. Soon afterwards, a Federal Court of Appeals for the Eighth Circuit upheld a permanent injunction prohibiting Minnesota from regulating Vonage. Despite these two favorable rulings, the California Public Utilities Commission nevertheless chose to file an appeal of the unanimous FCC order. What now? This article explores the basis for the FCC decision and considers its impact on state and federal regulation of IP enabled voice and video services.

The FCC decision arose from a 2003 MPUC finding that Vonage was subject to state utility regulation. As a practical matter this meant that Vonage would have to comply with the same tariffing, certification, and 911 rules that apply to traditional carriers. As a broadband VoIP provider, Vonage can never know which of its customers are actually using the service in Minnesota, nor can it know whether they are making intrastate or interstate VoIP calls. Furthermore, it is impossible for VoIP companies to comply with state utility rules because they can not technically offer their services in a manner similar to switched wireline telecommunications carriers. As a result, Vonage appealed the MPUC order both at the FCC and in federal court. Although a federal district court soon issued a permanent injunction — a ruling that was recently upheld by the Eighth Circuit — the FCC continued to evaluate the merits of the petition filed with the agency.

In a 5–0 decision, the FCC acted to preempt the MPUC from regulating VoIP providers such as Vonage. In issuing its preemption order — the FCC found that while Vonage's VoIP service resembled traditional telephone service provided over the PSTN, there are fundamental differences between the two types of services.

The FCC found that while Vonage's VoIP service resembled traditional telephone service provided over the PSTN, there are fundamental differences between the two types of services.

- Vonage does not know where its users are located because service can be accessed over any broadband Internet connection and users can access the service from any broadband connection anywhere in the world;
- The service requires the use of specialized software or customer premises equipment;
- The service combines voice with a suite of integrated features and capabilities; and
- While the Company uses telephone numbers, the telephone number is not necessarily tied to the user's physical location, rather, the number correlates to the digital signal processor of the Vonage customer's equipment so as to allow for the routing of the call over the Internet.

In reaching its ruling, the FCC applied Commerce Clause jurisprudence. Under this reasoning, a state regulation violates the Commerce Clause if:

1. It has the practical effect of regulating commerce outside of its borders; or
2. If it is excessive when compared to the local benefits of the regulation; or
3. In certain instances, commerce, by its unique nature, demands cohesive national treatment.

The FCC determined that the MPUC order runs afoul of the Commerce Clause under any of the three tests. The MPUC has the practical effect of regulating interstate commerce due to the inability to segregate the intrastate and interstate components of Vonage's service. Even if it were possible to segregate out an intrastate component of the Vonage service, the efforts would be costly and impractical with no clear benefits achieved though economic regulation of the service. Finally, due to the unique nature of the Vonage service that cannot be constrained by geographic boundaries, inconsistent state economic regulation would cripple the service.

In a similar vein, the FCC determined that Vonage offers a "jurisdictionally mixed" service and as a result it can not be subject to state regulation. When separating a service into interstate and intrastate communications is impossible or impractical, the FCC has the authority to preempt state regulation that would thwart or impede the lawful exercise of federal authority over the interstate component of the communication. The FCC concluded that it was impossible to either directly or indirectly identify the intrastate portion of a Vonage communication. Using telephone numbers as a proxy to determine whether a call was local or interstate fails because telephone numbers associated

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with rate centers in Minnesota can be used by customers located outside of the state. Similarly, using billing address or address of residence as a proxy for determining jurisdiction fails because whenever Vonage's service is used from a location outside of the state, the interstate communication would wrongly be deemed as intrastate.

While the pending litigation in the Ninth Circuit creates some additional uncertainty, VoIP providers should be heartened by both the Vonage ruling and the Eighth Circuit holding. In the meanwhile, companies should keep their eye on the Ninth Circuit case and continue to look to the FCC for further clarification concerning what social obligations VoIP providers will be required to comply with. IT

The FCC determined that Vonage offers a "jurisdictionally mixed" service and as a result it can not be subject to state regulation.

William B. Wilhelm is a partner in the firm of Swidler Berlin Shereff Friedman, LLP. For more information, please visit http://www.swidlaw.com. The preceding represents the views of the author only and does not necessarily represent the views of Swidler Berlin Shereff Friedman, LLP or its clients.

By Hunter Newby

The ENUM Reality

There are many different implementations, but one thing is the same: ENUM (define - news - alert) is changing the way voice communications is managed, and in the process, is undoing the traditional telephone company business model. For those who are unaware, there are several commercial ENUM registries in the world, all currently being utilized to varying degrees. Economics and adoption rate will determine which registries succeed and which fail.

So, what is ENUM? The acronym stands for Electronic NUmber Mapping and refers to a system of mapping public telephone numbers to Internet URI's (Uniform Resource Identifiers), basically linking a number to an address. This system works well as part of a database that can store private numbers (such as those of an enterprise) and/or public numbers (such as those of a CLEC or MSO.) The purpose of the database, or registry, is to act as a central point for the call to query. The query will return a "yes" or "no," essentially to confirm if the number being called is in the database. If it is, the call is connected or peered via either the public Internet or a private IP network. The wonderful benefit of ENUM is that it can make these identifications and subsequent connections directly between the parties and their respective networks without the need for additional networks or routing. Thus, it is very efficient and it eliminates the costs associated with legacy, outdated systems and business models. Since this technology exists and can be implemented by almost anyone with a need, most if not all of the ENUM calling today is settlement-charge free.

So, who is using ENUM today? There have been discussions, but no conclusions as of yet, regarding two ENUM consumer classifications: carrier and user. Traditional carriers are commonly known as companies that are in the business of selling telecommunications services. They include RBOCs, I/CLECs and IXCs. Users are typically consumers of telecommunications services and fall in to another two groups: enterprises and end users, or individuals. Enterprises consist of businesses, educational and research institutions, and government entities. There are examples across the board of each of the above using ENUM whether knowingly or not, but those who have actually implemented ENUM have done so out of pure economic gain.

> **It would make sense for the traditional local carriers to implement ENUM from a network efficiency standpoint.**

ENUM Economics

How does ENUM impact telecom economics? There are three basic ways to save money with ENUM:

1. Carriers reduce inter-carrier settlement fees.
2. Enterprises eliminate per-minute metered phone call fees intra and inter-company by building their own VoIP (define - news - alert) networks, or by buying a flat rate service.
3. End users move from a per-minute metered service to a flat rate.

The enterprise user category is at a tremendous advantage because they have no business model to charge someone for calling them, so they have nothing to lose and a lot to gain by implementing ENUM.

It would make sense for the traditional local carriers to implement ENUM from a network efficiency standpoint, but from a revenue standpoint, many still have their reciprocal compensation or inter-carrier settlement fees to protect. Since ENUM is basically a reciprocal traffic model enabler, this does not work so well for them at the moment. Increasingly, cable MSOs are getting into the voice business. This is due in large part to their broadband access to the home, but also due to the economics of VoIP. Since they have no legacy revenue models for per-minute metered voice, they offer their services for a flat rate to the end user. In order to raise the probability of profitability, the MSOs need to minimize their cost to terminate the calls their customers generate. ENUM plays right in to this. The MSOs don't care about reciprocal compensation because they don't have much of it, if any. Therefore, they are looking to establish these types of peering relationships. Their approach to buying is also changing. When negotiating with an IXC for domestic call termination, if they are presented with a rate per-minute, they counter with, "Oh, no, we don't pay per minute. We're going to send you X number of megabits of voice. What's our rate per meg?"

IXCs are a bit of a different breed. Reciprocal traffic can occur between those that own or control the network access to the voice device (out-dated business models aside). IXCs are traditionally the middle piece and don't own the access on either side. Since the cost to terminate a domestic U.S. call is so low now — well under a penny per minute — it can be considered "on-net" and no longer need be defined as long-distance. That doesn't help the pure domestic U.S. IXC revenue model. What is still considered long-distance, or "off-net," are international calls. VoIP has been used in the international minutes business for almost 10 years now and it originally acted as a cloaking device to bypass international settlement agreements between carriers (what was originally known as a leaky PBX (define - news - alert), and is now known as a Grey Route) to create an arbitrage and lower costs. It was the difference between a circuit switched

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minute and a data packet. One was regulated, or metered per minute, the other was not. The same thing holds true today domestically, but the rate per minute to most international destinations is still too high to play the flat rate game to the end user. Those places are still "off-net" and billed per minute although many at the lower VoIP wholesale rate.

What is well worth mentioning though is that the more developed the country, the more telecom infrastructure they have. Where there is oversupply, there are low costs to terminate. In those countries, such as the UK, France, Germany, Canada, and parts of Asia, the terminating rate is so low that some voice service providers have begun offering flat rate plans that includes those destinations. If you're an IXC on a traditional revenue model, that's spooky. It's like a big game of RISK; battles are fought and countries become "on-net." An end user flat rate model works when the cost to terminate a traditional phone call being billed per minute to the provider offering the service, i.e., an MSO, or VoBB, is so low that even if the end user makes 120 hours of calls in a month, at a rate of $0.005 per minute to terminate to the provider and a service plan rate of $35 monthly, the carrier breaks even. Do you talk on your home phone more than 120 hours per month? Probably not. It's probably closer to a few hours a week. Starting from that point, the provider of the voice service then looks to further reduce costs by finding other networks it can directly connect and pass calls to, avoiding even having to pay the $0.005. As end users make the logical decision to move from their traditional local and long-distance providers to these new providers, keep in mind that it is the economics and not necessarily the technology that is making it happen.

Conclusion

As it is with anything that is new and being widely adopted simultaneously and globally, there are a few kinks to be worked out. There are some interoperability issues that need to be addressed between network operators' signaling, but this new system makes so much sense that these challenges are being met and overcome. There are even new business models coming to life around ENUM that solely deal with resolving these issues. They are successful because there is demand and the economics make sense. When was the last time that happened in telecom?

In the next issue there will be an analysis of the currently available ENUM registry services operated by e164.org, VeriSign, and The Voice Peering Fabric (http://thevpf.com). We will identify each registry's user groups and benefits. If you own, operate, or are aware of a commercially available ENUM registry and would like it to be reviewed, please e-mail hnewby@telx.com. IT

Hunter Newby is chief strategy officer at telx. For more information, please visit http://www.telx.com.

By Ben Guderian

Is WiFi Telephony All Grown Up?

The WiFi telephony (define - news - alert) market continues to gain momentum with announcements coming almost daily about some new product or player looking to capitalize on the convergence of wireless and VoIP (define - news - alert). Does this mean WiFi telephony is all grown up?

The enterprise market is welcoming the technology. WiFi telephony (also referred to as voice over WiFi) is garnering more attention from the consumer market, which represents a much larger opportunity but not without its own challenges. Requirements differ significantly between residential and enterprise markets. In the home, low cost, ease of use, and aesthetics are key. In the enterprise, PBX integration, reliability, and total cost of ownership are much more important. WiFi telephony is still in its early adoption stage, but enormous opportunities exist in both the home and workplace markets as the technology and standards mature.

Bringing WiFi Telephony Home

In the residential market, the ubiquitous mobile telephony solution is the good old cordless phone. It's cheap, literally a snap to install, and comes in a plethora of colors, form factors, and features. We've seen the home cordless market tout the virtues of 900 MHz, then 2.4 GHz, and now 5 GHz radio technology. Interference — from consumer wireless devices such as baby monitors, home WiFi networks, wireless speakers, your neighbor's cordless phone — has been the primary challenge for the cordless phone market.

Using a WiFi telephone at home seems to make sense. You can leverage a single WiFi access point for both your wireless data devices and your cordless phone. All you need is a way to get your telephone calls onto your WiFi network, and now with consumer VoIP services such as Vonage (news - alert) and ATT CallVantage (quote - news - alert), the pieces are falling into place for residential WiFi telephony. In fact, Net2Phone has already announced the availability of a WiFi telephone compatible with their VoIP service.

Most people that are willing — and able — to make WiFi telephony work at home today can probably be categorized as early-adopters. Combining the two nascent technologies of VoIP and WiFi is still too complex today for widespread adoption, and WiFi handsets still cost several times the cost of a typical cordless phone. But it won't be too long before integrated, plug-and-play residential WiFi telephony solutions hit home with enhanced features and capabilities to make them competitive with the common cordless phone.

WiFi telephony is still in its early adoption stage, but enormous opportunities exist in both the home and workplace markets.

Before There Was WiFi

The enterprise market is where WiFi telephony continues to proliferate, where improvements in employee productivity and responsiveness drive the ROI and skilled IT staff can deal with implementation issues. But just as cordless phones preceded WiFi telephony in homes, other wireless voice technologies have been used in enterprise applications with various degrees of success.

Enterprise telephone systems providers have had limited success selling wireless telephony solutions to their mainstream customers due to the high cost of the available solutions. In North America, adoption of enterprise wireless telephony has been limited to markets with a strong need for mobile communication in the workplace, predominantly in healthcare, retail, and industrial applications. In contrast, the adoption rate of enterprise wireless telephony in Europe and some parts of Asia has been much higher thanks to having a standards-based wireless voice technology. The Digital Enhanced Cordless Telephone (DECT) standard is used for both business and residential applications, allowing broad industry support and lower equipment costs. About 2.5 million DECT handsets were sold into enterprise applications in 2003, with more than 25 million sold as residential cordless phones. But DECT has not been available in North America due to different radio spectrum allocations, although some DECT-based products have been modified to use the unlicensed 2.4 GHz band used by 802.11b/g devices. More DECT-based products are expected to hit the U.S. shores next year as additional unlicensed spectrum becomes available.

The WiFi Versus DECT Debate

So is DECT a viable alternative to WiFi for enterprise telephony? It depends on a few factors such as cost, network integration, and long-term convergence objectives. The lower cost of DECT solutions can be attributed to DECT's tech-

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nology and market maturity. DECT has been around for more than a decade with standards that have driven down the cost of DECT components. In most cases today's WiFi solutions are more expensive due to the rapid pace of innovation and continuing enhancements to WiFi standards. But WiFi component and equipment costs have already seen drastic reductions due to widespread adoption in both enterprise and residential markets. WiFi technology also has the advantage of supporting both voice and data applications, whereas DECT is not an IP-based technology and therefore has very limited use as a wireless data network. Enterprises that deploy WiFi networks to support data applications can leverage that investment to support WiFi telephony as well with little or no additional infrastructure cost.

Just as consumer VoIP services are a primary driver for WiFi telephony in the home, enterprise VoIP adoption is also a driver for WiFi adoption in the office. VoIP and WiFi are a perfect match, both sharing IP-based protocols. DECT is a time division multiplex (TDM) technology with fixed, dedicated channels for each wireless phone call. While TDM technology is very efficient for voice-only applications, it doesn't work well for packet-based data transport. On the other hand, IP network technology is not the most efficient transport for fixed bandwidth, real-time applications like voice, but the amount of bandwidth available and the benefits of a single, converged network more than makes up for the inefficiency.

It is clear that VoIP technology will replace traditional circuit-switched telephony in the enterprise. It is no longer a question of if, but when enterprises will migrate to IP-based telephony solutions. This is where WiFi holds a clear advantage over DECT, or any other proprietary TDM-based wireless solution. WiFi allows for end-to-end IP telephony making it the most elegant — and cost-effective — technology for enterprise wireless in the long run.

We'll probably see both traditional and WiFi-based wireless solutions deployed in the home and office for the next few years. Traditional wireless technologies like home cordless and DECT will continue to be attractive in markets where VoIP adoption will be slower, such as SOHO and smaller enterprises. WiFi telephony market growth will continue to build on the success in vertical markets, but will really start to take off along with general enterprise WiFi adoption. IT

Ben Guderian is director of marketing at SpectraLink Corp. For more information, please visit the company online at http://www.spectralink.com.

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VoIP is More Than An Opportunity for Service Providers - It's a Lifeboat!

Clearly, VoIP is the future of telecommunications. If you are a service provider not currently offering VoIP service, you better be planning to offer it — or you should be planning a new career.

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Triple-Play Becomes Quadruple-Play, Quintuple-Play, etc.

To succeed, service providers will have to provide voice, video and data to their customers, and perhaps even mobility services and data services like anti-virus. Like any great opportunity, you can't effectively deliver these products to your customers, however, until you've tackled a number of issues, including your underlying service delivery platform, access technologies, bandwidth issues, implementation challenges, management and billing and finally, user interfaces and CPE.

VoIP E-911

The positive press friendly to VoIP that we witnessed for the past year will vanish the moment someone is injured or worse because there is a problem with VoIP and e911 connectivity. I consider this a stumbling block that needs addressing on our way to achieving VoIP 2.0.

VoIP Security

There can be no greater prize for a hacker than to be able to crack your network and listen in on your customer's conversations. Furthermore, all the same issues any data network faces such as hacking, spoofing, viruses and spam are threats to stable and consistent VoIP service delivery. Deploying VoIP is one thing; another is effectively securing it in such a way that it is at least as reliable as the PSTN. In many cases, adequate security in the network is the deciding factor for an organization considering VoIP deployment. You need to know what types of attacks VoIP networks are most susceptible to and what points in your network are most vulnerable.

VoIP Peering

I am convinced that in a few years, virtually all VoIP service providers will peer with each other in order to save money. This largely misunderstood shift in telecom has potential to change the way VoIP works. Peering is the concept of interconnecting networks allowing IP and subsequently, VoIP traffic to be carried between service providers and companies without the need to pay a middle-man, or in this case, an additional 'long-distance' service provider. By using session border controllers placed neatly between the service providers, you can provide a 'translation service' between your caller and the recipient who uses a different provider.

UNE-P To VoIP

Yesterday's CLECs exist because of a FCC ruling called unbundled network access-platform, or UNE-P for short. This rule specifies the rates that incumbent carriers can charge CLECs to lease their lines. Recently, the FCC has rethought this concept allowing ILECs to raise these rates and subsequently reduce competition severely. UNE-P's demise is VoIP's gain, but the shift is not without it's challenges for CLECs. The biggest question for CLECs now is whether you should seek to build your own VoIP infrastructure or purchase/lease service from a new breed of wholesale carriers.

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In addition to the conference program, the Exhibit Hall at INTERNET TELEPHONY® Conference & EXPO showcases the newest switches, routers, servers and other equipment you need from the industry's leading manufacturers -- come find the solutions you need to profitably deliver VoIP services. Over 100 exhibitors will have displays for you to visit and demo products.

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We're so confident that this is the best VoIP education you'll find, we actually guarantee it. If you do not feel the sessions you attend made you better prepared to tackle your VoIP project than you were when you arrived, simply stop by the conference registration counter at the show and we'll issue you a free pass for any future INTERNET TELEPHONY conference. (No requests will be honored after the conference ends.)

I look forward to greeting you at in Miami.

Rich Tehrani

Rich Tehrani

TMC President and Conference Chairman

A Service Provider's Survival Guide To A Successful VoIP Migration

This month, the editors of Internet Telephony Magazine are proud to introduce a five-part editorial series entitled, "A Service Provider's Survival Guide to a Successful VoIP Migration." Technical, market insight, and editorial commentary will be provided by Volo Communications' CEO Shawn Lewis, along with several other leading voices in the VoIP industry. This issue's feature is entitled "Is There Life After UNE-P (define - news - alert)?" the first installment of the Survival Guide series. Topics that will be addressed in subsequent issues include:

March: What To Look For In A Wholesale VoIP Partner

Industry analysts believe the wholesale market opportunity creates a large business opportunity for new, next-generation wholesale carriers. The market for wholesale broadband voice services in the U.S. is characterized principally by quality, price competition, and least-cost advantage.

But is that enough? How do you know which wholesale partner is the right wholesale partner for you? We will address what you should look for in a wholesale partner, including technology (customizable applications, service creation environment, scalability); automated provisioning (service provider interface, end user controls, real-time CDR); and fulfillment (rapid deployment, end point equipment, and customer support) — all critical to your successful VoIP business.

April: Next Generation Networks: Should You Build Or Buy?

As the wireless industry has shown, disruptive new technology with better product and service features has the effect of luring customers to regularly change carriers. To minimize this risk of churn, carriers must continually expand their service offering in order to retain their existing customers. With the proven acceptance of packet telephony, the incumbent carriers are again faced with a disruptive technology that has a lower cost of service, and heavy CAPEX requirements.

Legacy incumbent service providers have huge capital investments in equipment geared toward transporting traditional circuit switched voice calls. They are quickly finding their technology is outdated. The technology they invested in does not address the convergence of voice and data technologies without significant CAPEX expansion. The market; however, is demanding that they offer new IP-based services, forcing them to invest in and build entirely new networks or to purchase the services wholesale. By purchasing services wholesale, the capital investment is eliminated, and the service provider can spend its time, energy and financial resources on acquiring customers and adding profitable revenue, rather than building its own next-generation network.

The article will explore the many benefits of buying network services from wholesale providers versus building your own network and address the following:

- CAPEX considerations.
- Avoid investing in rapidly outdated technology.
- Invest in marketing to acquire customers.
- Building a revenue base before building a network.

May: VoIP Applications: To Host Or Not To Host? — That Is The Question

The overall market for VoIP services is rapidly increasing. VoIP has been described as the next "killer application" for telecommunications services. Consumers are demanding more services, more features, and more choices, at a lower cost. The true killer application is not VoIP, but how service providers will bundle and host these advanced services. IP Centrex features are in high demand by end users, and service providers have to either host the services at their facilities or at the customer's premises.

Hosted IP Centrex services enable service providers to immediately provide PBX functionality for phone service in SOHO and small/medium enterprise business environments. Phone service is provided over IP from the customer to the Internet, a provider's private IP network, or VPN. All standard features of PBX functionality, as well as more advanced calling features, are provided to the customer.

Hosted IP Centrex provides an immediate cost savings to both the provider and the customer, reducing CAPEX costs, sustained OPEX costs and additional telecommunications and other service charges immediately. In many cases, Hosted IP Centrex also adds flexibility and feature sets not found in many smaller Key or PBX systems. This column will address:

- OPEX considerations.
- Speed to market.
- High-demand features.
- Rapidly refresh product offering.

June: VoIP Product Packaging — It's Not Just About Price

Industry analysts project that VoIP traffic will grow to over 1 trillion minutes in 2005. There are several reasons behind the dramatic growth of VoIP, including the proliferation of broadband Internet connections in the residential and small business markets, and the promise of rich features at a lower cost.

However, existing VoIP customers are already discovering that price is not the only factor to consider when selecting a VoIP carrier. The type of advanced services available and the quality of telephone service are quickly coming into play.

Service Providers who offer the richest features with some form of guaranteed Quality of Service are gaining attention. For instance, estimates show the U.S. hosted VoIP marketplace will grow to $4.5 Billion in 2006. In June, we will address:

- Expanding product capabilities by bypassing central office limitations.
- Avoiding pricing wars through high-demand features.
- Creative new bundles for small business with PBX and UM features.
- Residential bundles; Competing with the incumbents.
- Retention from win-backs. IT

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Is There Life After UNE-P?

By Shawn M. Lewis

When the FCC mandated the Unbundled Network Elements Platform (UNE-P), it was envisioned that competitive local exchange carriers (CLECs) would drive revenue in their key markets prior to investing in expensive switching infrastructure. It was viewed as a departure from the "build it and they will come" syndrome associated with the Telecommunications Act of 1996.

While the UNE-P model has been a successful one for many CLECs, the fact that CLECs are still dependant on local access facilities from the incumbent local exchange carriers (ILECs) continues to hinder their long-term viability. In addition, last year's court rulings drastically limited the future of the pure UNE-P CLEC. There is a lot of talk about migration to a UNE-L strategy that requires a significant capital investment and requires a geographically targeted (and more costly) marketing effort. And what about the new-age high-demand enhanced features?

So, we ask ourselves… Is there life after UNE-P?

One answer is a low-cost, high-quality VoIP solution that speeds time to market, assures carrier-grade quality, and enables early market entry without the CAPEX requirements associated with building a new network.

VoIP addresses cost of goods pressures.

CLECs have to face facts. UNE-P wholesale pricing is going up, putting increased pressure on cost of goods sold (COGS). The FCC has asked that CLECs negotiate with the ILECs directly "in good faith." In reality, only a few of the larger UNE-P CLECs, such as ZTEL and Sage Telecom, have negotiated UNE-P contract extensions with certain ILECs while most have not. Regardless of the extensions, the costs are going up while retail pricing is going down. This trend is a major threat to the survival of the UNE-P CLEC.

VoIP is an attractive option for numerous reasons. By migrating to a VoIP strategy, a CLEC can realize an immediate reduction in their COGS and in their customer provisioning and moves, adds, and changes costs. Another great benefit that VoIP brings to the CLEC is that they can offer a "bring your own access" product to their customer. This means that the customer may have a broadband access connection from a different provider and still have VoIP service from the CLEC. This completely eliminates all of the local access facility constraints as well as the costs. The ensuing result is that CLECs can reduce retail pricing while increasing their maintained profit margin.

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It's not what you charge the customer that really counts. It's what it costs the customer that matters.

A UNE-L strategy requires high CO density that most CLECs just don't have.

There is talk that several of the largest UNE-P providers are planning their transition by geographically concentrating their marketing activities around central office (CO) concentration, and then installing softswitching and broadband access gear in central offices to provide next-gen services. These plans still require that the local loop be provided via UNE-L, ROI is risky, and broadband VoIP is an equal or greater threat to UNE-L as the courts are to UNE-P. It is virtually impossible to achieve a return on investment for any collocation facility with less than 800 to 1,000 lines in service. VoIP technology is not CO specific. CLECs have numerous options to connect to their customer via IP. By eliminating the CAPEX in CO equipment, CLECs can price their products attractively to rapidly gain market share.

Retail pricing pressure: It's about the "net" cost of service to the customer.

There is a misconception about retail pricing for local voice service. It's not what you charge the customer that really counts. It's what it costs the customer that matters. Consumers of both wireline and wireless service have long complained about taxes and surcharges on their bill. A very common customer service call goes something like this: "I was told that my unlimited plan was only $49.95 per month but my bill is $65.00!" Although this is likely to change over time, VoIP services are not subject to the taxes and regulatory fees that local and long distance service is. That said, if the provider's COGS is reduced 25 percent and there is another 25 percent in fees that are eliminated, the provider could reduce the retail price and the cost threshold to the customer by up to 50 percent. This makes for a sticky customer that is not likely to be "won back" by the ILEC.

VoIP services offer high-demand features, eliminating "feature limitations" of traditional voice technology.

For years, ILECs offered Centrex services to their small to medium business customers as add-on enhanced features that they could only get from large-scale, expensive on-premises Primary Branch Exchange (PBX) systems. As the PBX costs came down, customers scrambled to have their own system. In today's world of hosted IP-based applications, customers can abandon those legacy systems for more robust features that increase their productivity and promote a more mobile, fluid working environment. Multi-branch offices, remote SOHO workers, and telecommuters are now enjoying the benefits of having their PBX functions as an integrated part of their VoIP service. CLECs that don't offer these rich features will get caught flat-footed and inevitably find their customer base churn away. **IT**

Shawn Lewis is the CEO of Volo Communications, a wholesale provider of advanced voice and data services and applications including broadband VoIP service. Mr. Lewis also wrote the first two patents for softswitch and media gateway technologies. For more information, please visit http://www.volocommunications.com.

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Maine Energy Company Selects Networked Telecom Solution

Downeast Energy is a family-owned company, delivering energy products and building materials to households in Maine and New Hampshire since 1908. Headquartered in Brunswick, Maine, Downeast Energy supports 12 locations with 400 employees and 60,000 customers.

"In a region like Maine where winter home heating can be a life or death situation, our telephones literally can be the lifeline for our customers," said Chris Turner, director of IT for Downeast Energy. A Toshiba customer since 1983, Downeast Energy turned to NorthStar Communications of Cumberland, Maine, with whom it has a 20-year relationship, when it was time to upgrade their system and network its 12 locations throughout Maine. With up to 1,600 incoming calls per day in the Brunswick office alone, the stakes were high.

"Our challenge was to design an integrated solution that permitted each user and each location to utilize the technology that best served their particular customer service needs," said Bill Fogel, president of NorthStar (news - alert). "Some locations would benefit from IP applications, while traditional PBX was a better fit for others. Everyone wanted centralized voicemail and instant internal connectivity."

To network its communications between its 12 sites, Downeast Energy chose four Toshiba Strata CTX670 business communications systems (one at its headquarters), five Strata CTX100 systems, and Toshiba (quote - news - alert) IP phones at three smaller locations. Toshiba's Stratagy iES32 Voice Processing system delivers centralized voice mail for all 12 sites.

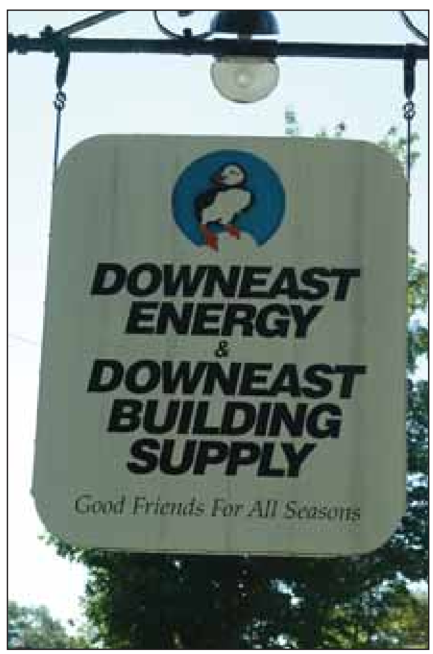
Improved Communication Inside And Out

Twenty-year company veteran Patti Weaver relies on Toshiba's PC Attendant Console to answer and route all of the incoming calls to the company's headquarters office in Brunswick. Turner said, "We don't want to use an automated attendant, as we believe in personalized customer service. Having the Toshiba PC Attendant makes it possible for Patti to answer the hundreds of calls we receive, making sure every customer feels important."

Weaver was a key decision-maker for the telecommunications system. Turner added, "Patti had the final vote, because we respect her experience in knowing what would work to maintain customer satisfaction over the telephones. She's been very pleased."

Downeast Energy also benefits from Direct Inward Dialing (DiD) and four-digit extension dialing between its 500 employees regardless of location. Turner said, "Extension dialing improves internal communication while reducing overall costs since we rely on our own WAN (define - news - alert) for the calls, rather than analog tie lines or public lines."

Using Toshiba's Stratagy iES32 Voice Processing system, NorthStar Communications installed a centralized voice mail system that supports all loca-



"We've greatly improved our employees' ability to communicate with each other and our customers."

- Chris Turner
Downeast Energy

tions. Turner said, "By centralizing voice mail, we can communicate with groups of people with broadcast messages. We've greatly improved our employees' ability to access their voice messages and communicate with each other and our customers."

Remote Administration With Toshiba WinAdmin

There's a saying in Maine, "You can't get there from here…," which can be absolutely true in the wintertime. Turner said, "Having the ability to remotely program changes, such as moves, adds, and changes, to our Toshiba system is a major benefit when temperatures dip and roads are closed. We can still make changes for our offices hundreds of miles away, without leaving our desks, using Toshiba's WinAdmin program."

In addition, Turner has his voice mail programmed to page him when calls come in after hours, allowing him to handle emergencies quickly, a functionality that is new with the Strata CTX.

He also appreciates the system's reporting capabilities and the ability to track call volumes and determine usage needs and requirements.

Seamless Migration Path

Downeast Energy purchased its first Toshiba Perception telecommunications system in 1983, migrating to Toshiba's Strata DK systems as the company grew. It migrated its Strata DK systems to the Strata CTX. "Toshiba's migration to the Strata CTX systems allowed us to retain much of the investment we had put into our earlier equipment, through seamless re-use of many telephones and CO and station cards at a savings of $20,000, while still allowing us to take advantage of the latest technologies and capabilities," Turner said. "There was a great comfort in knowing that NorthStar Communications would be handling our migration. Their team knows our system inside out. Despite all the technology, this is still a people business."

Not having to retrain its employees to learn a new telephone system was a huge time and cost savings, according to Turner. "Our employees were pleased with the smooth migration. Folks were just as proficient on the cut-over day as they were the day before."

Figure 1. Downeast Energy's Patti Weaver, with NorthStar's Jason White, relies on the Toshiba PC Attendant Console to handle up to 1,600 incoming calls a day.

Cost Savings Abound

Today, by avoiding the public lines and relying on its own WAN for extension calling, Downeast Energy eliminated long-distance charges between employees, a savings of more than 20 percent. Using their ATM backbone and VoIP cards in every PBX, the WAN also provides a cost savings of about 50 percent over using leased lines, along with delivering VoIP dial tone to the smaller sites via Toshiba IP telephones. By using VoIP instead of installing three complete telecommunications systems, Downeast Energy's overall costs are reduced significantly, yet the smaller offices get the benefits of the communications system.

The Bottom Line

"NorthStar has given us unparalleled support, and Toshiba's products are in a class of their own delivering quality and reliability," Turner said. "Both companies delivered superior service in helping us migrate and be able to use our system at its ultimate performance. The combination of NorthStar Communications and Toshiba is unbeatable."

Turner summed it up, saying, "We wanted a business communications system that works like the utility companies — you turn on the switch, and it works — and that's what we've got." IT

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A Triple Play Top 10

Service Provider Considerations For Platform Selection

By Ken Meaghe

The "Triple Play" (define - news - alert) has been a major topic of discussion and hype for a number of years now while all the real action has been around delivering best-effort high-speed Internet. Times have changed. The combination of regulatory changes and trends, along with the competitive landscape, has progressed to a point where all service providers are making firm plans on how to best move forward with the triple play. The decisions they make will impact their ability to compete over the next decade and as such, are fundamental to their future business. They need to move forward with plans on how to deliver multiple services over a single infrastructure. A multiservice network isn't built using a best effort network model so a significant shift is required. This is a complex optimization equation balancing investment with revenue potential and there is no "one-size-fits-all" solution.

Here are the Top 10 things a service provider needs to consider when selecting a platform for delivering triple play services.

1. It all starts and stops with the "user experience."

The starting assumption has to be that the service offering must be as good as — or better than — what a customer is able to get through traditional service providers for video, voice, and Internet service today. Customers are not going to tolerate anything less than parity with existing benchmarks. The channel surfer is not going to accept long channel change times or issues with picture quality. The voice caller is not going to put up with long delays in conversation.

There will be a willingness on the part of the consumer to move to service providers that offer new features that come from the integration of voice, video and Internet into a single bundle that interwork and enhance the overall experience. This is where the key to new service revenue and subscriber growth lies.

2. Building a differentiation capability.

Service providers cannot focus too narrowly on the service offerings and bundles, as they exist today. The objective needs to be extended to include building a capability to deliver new and differentiated services. The current service offerings have been commoditized, and so will a "bundle" of these services quickly become a commodity. Convergence will enable a service provider to roll out new services that provide differentiation. Differentiation is the answer to new sources of revenue and customer retention. This, however is much more than a platform decision, it has to be built into the organizational model as well if the service provider is looking to build a sustainable competitive advantage.

A service provider that appreciates this will build an infrastructure and organizational capability to deliver new services, harvest and renew them over and over again.

3. Standards-based interfaces and protocols will reduce risks and costs.

It is essential to build a network solution consisting of a set of components that have standards- based interfaces and protocols. There are a large number of moving parts in a network that will deliver the triple play and to reduce the technology and vendor risk, a standards-based approach must be taken. The additional benefit is that costs are easier to control if there are multiple system choices available.

Even with a standards-based approach to building a service delivery platform for the triple play, interoperability between the various components will still be an issue for service providers and vendors. Standards compliance does not necessarily equal interoperability. It takes time to work out the subtleties of the interfaces and protocols. Service providers and vendors will have to work

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together in this area to achieve full interoperability.

4. End-to-end quality of service (QoS)

QoS is probably the most talked about piece of the triple play equation and perhaps also the most misunderstood. In simple terms, it represents the ability to ensure that a subscriber is able to get what they have paid for.

The network must be able to reserve the required end-to-end network resources necessary to deliver the service requested by a subscriber at the time the service is requested. This is very different from just being able to prioritize traffic flows in a best effort network and becomes increasingly important with interactive services where delay, jitter, and packet loss may render a service inoperable.

For Internet telephony, people are not going to tolerate long delays in their voice conversations such that it interferes with the rhythm of conversation. Nor will they put up with too much echo. For video delivery, the use of higher compression techniques will cause video to suffer significantly from packet loss. The delivery platform must have deterministic characteristics for end-to-end delay, and must not drop high-priority packets under congestion conditions.

The service provider needs to think about the trends in services offerings and what that means with regards to network performance parameters such as delay and packet loss.

5. Subscriber/Policy Management

Subscriber management or policy management is also going to change significantly in this next-generation network. In the traditional "best effort" Internet model, subscriber management represents the classic PPP termination point where the "AAA" was performed and subscriber traffic was passed into the Internet. In the multi-service network, subscriber management must be extended to include policy management. There is great debate within the industry as to whether the "BRAS" function should be centralized or distributed.

For the service provider, what is important is getting revenue for the resources used to deliver the subscribed services and gaining additional revenue by offering new services that subscribers can select even temporarily when network resources are not fully utilized.

The subscribers want easy access to the services that they have subscribed to and to have the ability to select enhanced services if the network resources are available. Since they will select different service bundles and performance attributes, the network must have the ability to shape/police traffic based on a subscriber's service agreement and have the ability to change those policies on the fly in order to maximize the revenue potential of the network.

6. Flexibility for deployment and evolution

Having the capability to work into legacy networks and a variety of softswitches gives the service provider an important degree of flexibility in terms of how to migrate and evolve the network. Working into the existing voice infrastructure is especially important to the incumbent providers today while service providers without existing infrastructure are likely to deploy softswitching technologies. In either case, a platform that offers flexibility in this regard will mitigate the risk of selecting technologies that may become obsolete quickly as technologies and the services that they are based on continue to evolve.

Triple Play Makes Itself At Home

By Greg Galitzine

Back in 2004, I had the chance to speak with the good folks at iPlay3, a consortium made up of NetCentrex, Envivio, and Highdeal. Together, these vendors offer an integrated triple play solution featuring voice and video over IP, and a pricing/rating solution that leverages any type of broadband IP infrastructure including Fiber to the Home (FTTH), DSL, Cable, and Wireless. The iPlay3 solution is designed to enable voice, video, and OSS services from the service provider's headend or central office. Utilizing the service provider-owned IP network, the services are delivered to a set-top box in the subscriber home.

The VoIP portion is delivered using NetCentrex' MyCall Residential solution, a carrier-class VoIP platform leveraging all types of broadband networks. MyCall Residential delivers Class 5 telephony features, regulatory capabilities, and interactive services.

Well, NetCentrex (news - alert) recently announced a triple-play-focused partnership through which Ephrata, WA-based HomeNet Communications will deliver VoIP services enabled by MyCall. HomeNet will use the MyCall voice solution to deliver VoIP solutions to deployment customers and will also market HomeNet video and data services to round out a fully integrated triple-play service offering. HomeNet utilizes an ASP-based model, which allows smaller deployments a scalable way to redistribute services utilizing optical fiber interconnections.

"HomeNet Communications and NetCentrex are logical partners with tremendous synergies in our integrated triple-play deployment, vision and strategy," said Kelly Ryan, chief executive officer of HomeNet Communications. "This partnership will provide HomeNet Communications with an avenue to increase its customer base significantly as we work with the largest VoIP provider in the world."

"What IPlay3 and HomeNet offer, and what today's triple play providers need are integrated service platforms, not network complexity. Smooth customer interfaces and proven backoffice interoperability result in high service take rates and improved ARPU," states Joe Savage, President of market analysis firm Telecom Think Tank. IT

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At the physical layer, the service provider is faced with numerous physical plant and deployment scenarios. A platform that can easily address a variety of deployment needs will reduce the number of variables required to deliver the total solution. On the access side this may mean supporting a variety access technologies, xDSL and Fiber (PON and FTTC), as well as a number of network facing technologies such as SONET and Optical Ethernet. A single platform or product family that can address a variety of deployment needs will be easier from a deployment, maintenance, and interoperability perspective.

7. The Profitability Equation — Revenue Versus Costs

In November 2004 The Wall Street Journal published an article entitled "*Meet the New TV Guy*" in which Ed Whitacre discussed SBC's evolution to being a provider of TV services. In this article, he describes a $100 service bundle that includes TV, Internet, and voice and wireless services. In order to operate at a price point of $100 it will be crucial that service providers consider the Total Cost of Ownership and move beyond the CAPEX piece of the equation. While everyone may not agree with a $100 price point for the bundle, it is certain that with increased competition, the lowest possible operational cost model will be critical. Consumers have a certain amount of disposable income to spend on these services. This discretionary spending will go to the service provider best able to balance the value/price equation. Given the competition for this disposable income, a low operational expense model will be increasingly important. Service providers should take an "Activity-Based-Costing" approach to refining operational expenses.

8. Scalability — more than just size that matters

Every vendor claims to offer products that scale easily to support multiple users and services. Service providers need to spend time understanding how the network should scale to support thousands of subscribers using multiple services and how performance and operational costs may be impacted by subscriber growth and load levels. Vendors make a lot of claims about system performance and scale and service providers need to be careful about what is being quoted and under what conditions. They need to determine how the system will be used, what the performance needs to be under various conditions to ensure that the selected solution is capable of delivering a positive user experience. Scalability claims may be impressive but the service provider needs to get behind the numbers to ensure they closely match their needs. The next step is testing the solution and stressing it to reflect the real world application.

9. Distributed intelligence

In order to deliver multiple services in this network, more intelligent endpoints are going to emerge. The networks of the past had central control elements that were very large, costly, and usually single purpose. They also suffered from being expensive and difficult to upgrade to provide new features.

Video is the significant driver for change in the broadband network today and the most significant impact is on network bandwidth. In order to economize on network resources, distributing a multicast capability into network elements makes sense. By having the capability to join and leave multicast groups made by elements within the network, the service provider is able to minimize network bandwidth and channel selection times.

In order to support the variety of multimedia applications and their evolution, it is likely that the media processing and control will be distributed on a service basis. There are multiple control planes that evolve here. The first is the network control plane that manages and reserves network resources required to support a specific service and the second would be the service control plane (IGMP for Video, SIP (define - news - alert) for communications). They will have to work together to deliver the service but they are likely to be distributed into various network elements.

10. Demarcation point between service provider and the customer becomes blurred

As the service infrastructure becomes IP-based and subscribers receive multiple service offerings over their home networks, the demarcation point between the service provider and the customer is going to blur. A set-top box will be used for video delivery and its network interface will be an Ethernet port that is connected to a hub or router that is part of the customer's home network. Other home networking gear including wireless local-area networks (LANs) are already an important element of the home network and if not engineered properly, may impair the service offering.

Like it or not, service providers, are going to have to deal with customer network issues. Service providers need to consider this carefully as they move to a single IP network to deliver multiple services especially as home networks become increasingly complex and varied. Types of networking equipment and the quality of it will all factor into delivering a quality experience.

Conclusion

This article has tried to highlight the top ten issues facing a service provider deploying a network for the triple play. There are others, but one thing is clear, the triple play changes the game significantly from the best-effort Internet model that has been deployed over the last several years. The technology exists to build a true multi-service network than can deliver a quality end user experience. Service providers need to focus attention on building a capability to deliver new and differentiated services over a common infrastructure. Only by building such a capability will they create a sustainable competitive advantage for the future. **IT**

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INTERNET TELEPHONY® EDITORS' CHOICE AWARD 2005

Pannaway Service Convergence Network (SCN)

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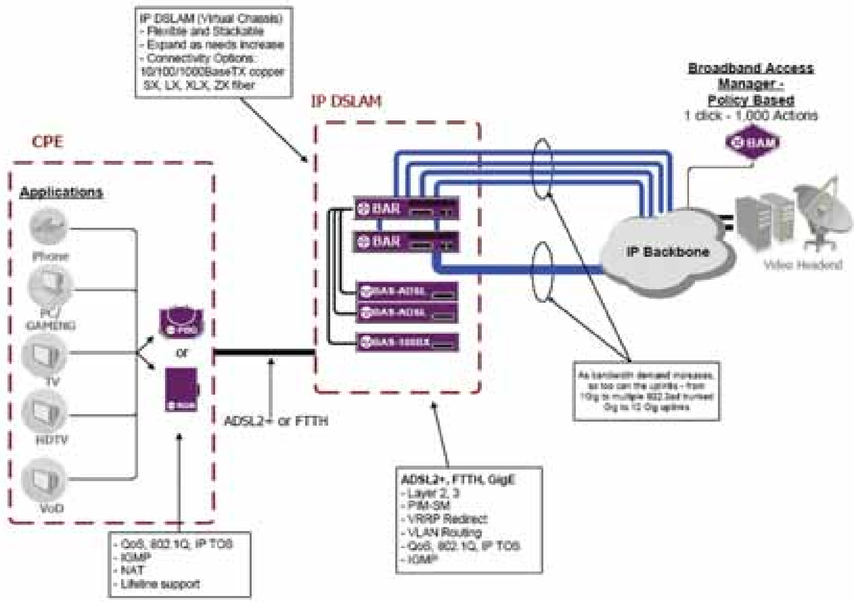
Web: http://www.pannaway.com

Price: ~$800/subscriber



Triple Play Is Hard

A "triple play" in baseball is one of the most challenging feats to accomplish. Similarly, the "Triple Play" of voice/video/data has been more hype than actual deployments, making it seem as though Triple Play is facing technical challenges or other hurdles. In my Top 10 VoIP Predictions for 2005 (tmcnet.com/69.1), I mention that Triple Play is going to take off, so what's the hold up? Why haven't I seen actual deployments and why am I not being offered a trial account for a product review? I WANT MY TRIPLE PLAY!

Well, New Hampshire-based Pannaway contacted me recently and offered me an industry "scoop" — the first Triple Play product review ever. I was very intrigued when I talked with Pannaway about their feature-set. Pannaway goes beyond the Triple Play hype by offering a Triple Play solution that works today and better still they have actual deployments not trials or "promises" of Triple Play. Just as an example, Cross Telephone, a carrier based out of Oklahoma is one such Pannaway customer and they currently have 9,000 subscribers with voice, data, HDTV, and Video On Demand (VOD) — all over IP. When they evaluated various Triple Play services they determined it would cost approximately $3,000/subscriber to overhaul their existing ATM infrastructure. Cross discovered Pannaway's solution which leverages Cross's fiber network allowing them to augment existing DLCs and initiate a controlled roll-out of an IP-based solution — rather than an ATM solution — for approximately $800/subscriber including premise equipment.

Pannaway's Triple Play Architecture

Pannaway's architecture consists of the Broadband Access Manager (BAM), Broadband Aggregation Router, Broadband Access Switches (BAS-ADSL2+ & BAS-100BX), Call Control Manager (CCM), and Network Media Exchanges (NMX-SS7 & NMX-PRI), and the CPE devices, Personal Branch Gateway (PBG) & Residential Gateway NID (RGN). Pannaway essentially is an IP-based solution that rides on top of high-speed IP technology, such as fiber (Fiber-to-the-Home) or the latest DSL technology, namely the ADSL2+ standard. ADSL2+ not only has more bandwidth than other DSL flavors, ADSL2+ lines can reach much further from the Central Office (CO), delivering 20–24 Mbps in many real world deployments, thereby increasing the radius that carriers and service providers can offer Triple Play services using standard copper wiring loops already deployed to the majority of homes. In fact, according to some ADSL2+ reach versus rate charts I have, Pannaway was able to deliver five video streams over an ADSL2+ connection @~5,000 feet, four video streams @~7,500 feet, three

RATINGS (0–5)
Installation: 5
Documentation: 5
Features: 4.95
GUI: 5
Overall: A+

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Figure 1: Streaming TV video feed with CallerID popup.

video streams @~8,000 feet, two video streams @~12,500 feet and one video stream @~14,000 feet. In the Cross Telephone deployment, Pannaway claims they were able to deliver high-quality Triple Play digital services at distances exceeding 8,000ft — very impressive.

The Pannaway Service Convergence Network (SCN) architecture is engineered to economically replace the fundamental features that would be served by a DSLAM, DLC, and IP/Gigabit Ethernet gateway. In order to serve these primary functions, the SCN architecture was built around a suite of standards-based interfaces designed to provide high performance, redundancy, and scalability in the IP video world.

Data and video services require large amounts of bandwidth and high security, both of which Ethernet supplies in a highly scalable, cost-effective manner. Ethernet frames are the best solution for secure IP transport and have proven to be more suitable for Quality of Service (QoS) management than packet-over-SONET structures and more efficient than chopping IP packets into ATM cells. The Pannaway architecture relies upon IP packet level techniques to provide consistent QoS priority enforcement. Pannaway's QoS approach incorporates policy-based management with standard Layer 2/3 QoS mechanisms including 802.1Q, IP Type of Service (TOS), and IP Differentiated Services (DiffServ) to deliver a framework, which allows various IP-based services (i.e., VoIP, VOD, IPTV) to be quickly and easily provisioned and deployed.

I should point out that traditional DSLAMs supporting IP Ethernet interfaces are built on ATM VCC and cell switching-based architectures. The inherent flaw of this commonly used solution is that it requires multiple PVCs per subscriber for the delivery and provisioning of services. ATM-based IP delivery architectures have also proven to be cost prohibitive and unsuitable for the delivery of bandwidth intensive multicast applications.

Pannaway's end-to-end IP-based architecture makes switching decisions based on packet level information and is capable of performing packet and port-level inspection while port-to-port switching is routinely handled without the need for a Broadband Remote Access Server (BRAS). Pannaway does not require multiple PVCs per subscriber port and provides QoS delivery based on packet queues. According to Pannaway, "Pannaway's highly scalable packet-based architecture reduces service provisioning and configuration expenses through its ability to eliminate per subscriber/per service/per ISP virtual circuit provisioning. Pannaway's IP design requires only one-time specification of service policies eliminating most of the service provisioning and configuration efforts inherent to ATM (i.e., ATM VC per subscriber, per hop)."

Triple Play Is Simple…

If you think about it, simply running voice and video over IP is not very complex to do, right? Just take Vendor A's voice solution, Vendor B's TV/video over IP solution, slap them together over a broadband pipe and bam you've got a Triple Play solution, right? Wrong! There is much more to it than that.

Triple Play The Right Way With Pannaway…

For one, it's better to have an integrated solution that offers some nifty features. For instance, Pannaway's solution can sense an incoming call at the CO, "digitally splice in" the CallerID information into the video stream and then the subscriber can see a CallerID popup on their television — very cool stuff (Figure 1). Pannaway has built voice and SIP into

PROS	CONS
Fully integrated solution. Future-proof design. Unique E911 capability.	Lack of integration partnerships. Need to embrace wireless solutions in-home.

both flavors of its premise devices (RGN and PBG) and into the Remote Terminal (RT) BAS thereby making integrated features such as TV call popups a reality.

The PBG and RGN come with three orderable uplink interfaces, including ADSL2+, Ethernet 10/100TX, Ethernet 100 BX, with a flexible design for future interfaces so you don't become outdated. They include a stateful inspection firewall, NAT, and VPN support. The PBG which is ADSL2+-based supports two Ethernet interfaces and the RGN, which is their fiber CPE device supports six. Both support DHCP and DNS server and client, RIP v1 and v2, IGMP. Most importantly they support QoS and VLAN support with 802.1p and 802.1q with up to eight priority queues. Prioritizing traffic means that if someone in the house starts downloading a massive file or uses a bandwidth-hungry application such as eMule, you won't cause the streaming IP video to break-up or cause voice breaks in any current VoIP call. This is critical for any Triple Play solution.

One other critical feature when evaluating a Triple Play solution is whether or not it supports E911 or what happens in the event of a power or Internet outage. Most VoIP solutions today do not support E911 and even the ones that do lose phone service if the electricity or Internet goes offline or if the broadband pipe fails. Many carriers refuse to choose any solution that does not guarantee phone service in the event of power failure. Well, Pannaway handles "Lifeline POTS" in a very interesting and unique way that separates it from many of its competitors. If the power goes off, Pannaway's premise devices will automatically switch over to analog/POTS (since it's DSL and not cable, DSL already has analog/POTS built-in). But here's the real kicker — when it switches over to POTS/analog, your phone number stays the same! Unlike competing solutions, this is no kludge whereby you have one primary phone number using VoIP and a secondary backup phone number using POTS in the event something bad happens. It's all seamless to the end-user — one phone number both for inbound and outbound calls even when the power fails — and all of the inherent features realized by SIP-based VoIP including distinctive ring tones and call forwarding stay intact.

From a technical standpoint, anytime electrical power is lost, a relay trips at the customer premise device (CPE), automatically connecting the phone line directly to Pannaway's BAS in an RT or CO. The BAS then provides phone service to the subscriber. Figure 2 illustrates the scenario of a 911 call for help after a hurricane, earthquake, or some other disaster has cut all the power lines.

Pannaway came to TMC Labs and set up all their equipment. Within an hour that had all their gear and features set up (premise device, RT, CO, flat-screen LCD TVs, call features, on-screen call pop-ups, distinctive ring, Lifeline POTS, VoIP, etc.) Basically I had a full Triple Play solution on a single lab workbench. It was quite a sight to behold and to think that the equipment I was looking at could power thousands of subscribers with high-speed data, TV, VOD, CallerID popups on TV, and more.

Show Me The Money…

First, they demonstrated two video streams going to two separate LCD TV displays. Since we weren't actually connected to a live video feed with multiple channels they used a few simulated live TV feeds via MPEG2 encod-

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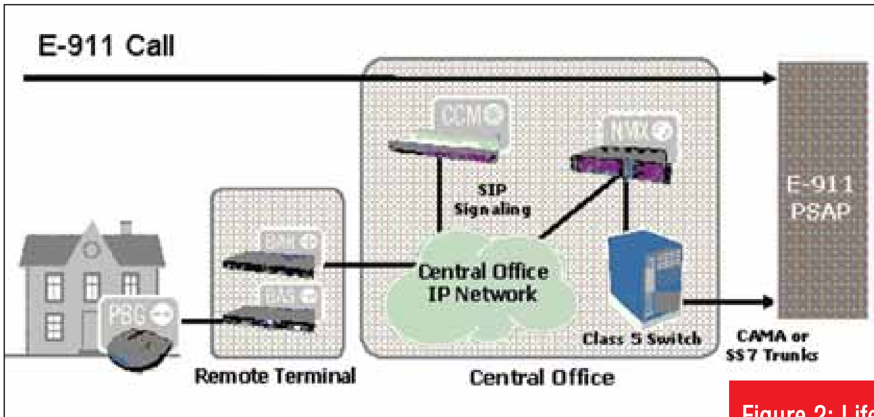

Figure 2: Lifeline POTS support architecture.

ed files. Pannaway also supports MPEG4.

Next, they demonstrated a VoIP call and the quality was superb utilizing the G.711 codec. I asked about other low-bandwidth codecs and they said most carriers want G.711 since it offers "carrier-class sound quality." Next, Pannaway demonstrated their TV screen-pop capability. We initiated another call and then saw CallerID information pop onto the bottom of the TV screen while we were still watching the simulated live TV video feed. Pannaway told me that with their architecture they can also pop up new e-mail notifications or the actual e-mail itself.

We also examined Pannaway's highly intuitive Web-based administration tool CCM, which both the service provider and the subscriber utilize. The service provider obviously has access to more administrative interfaces whereas the subscriber has the ability to access call logs, calling features, voicemail, set call forwarding schedules, or perform self-provisioning of new services — all over the Web. The subscriber's Web interface can be seen in Figure 3.

All of the tests and demos went very well. After demonstrating that their Triple Play solution wasn't "smoke and mirrors" we discussed some of Pannaway's other features, which are listed below.

Features:

- Every port is voice, data, and video ready.

• Web-based subscriber access to call features, voicemail, phonebook, etc., as well as self-provision new services.
• Scales from 16 to 252 ADSL2+ ports per cabinet.
• Zero-touch provisioning.
• Powerful service and management control for activating/de-activating voice, video, data on a per subscriber basis.
• Detailed device viewing and inventory management.
• Fault management with alarms (SNMP support).

VoIP Benefits:
• Line Blocking, Class 5 offload, Advanced Class-like features.
• Selective Call Blocking based upon calling number, time, or day.

IP DSLAM Features:
• All forwarding decisions are based on packet information at the OSI networking model layer 2 or higher.
• Every port is capable of deep packet inspection and queuing.
• Layer 3 multicast routing supports packet replication at wire rate.
• Layer 3 multicast routing tables support 1,024 groups.
• Stateful IGMP Snooping:
  ◗ Supports multiple multicast/IGMP clients at the premise regardless of the CPE.
  ◗ Multiple IGMP clients at a given customer premise can 'channel surf' without ill effect (IGMP Quick Leaves guarantee sub-second channel change times).
• Port to port switching is performed without the need for an external BRAS or router.
• Highly scaleable design supports multiple Gigabit Ethernet and 10 Gigabit Ethernet interfaces at wire speeds.
• Distributed internal packet switching fabric delivers non-blocking performance in excess of 20 Gbps.

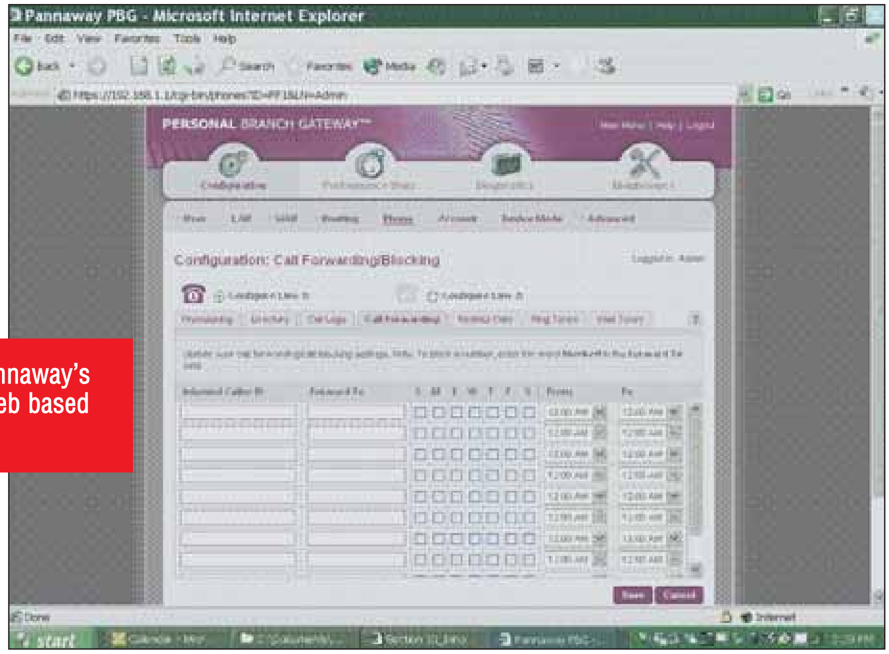

Figure 3: Pannaway's advanced Web based call features.

• Highly resilient architecture supports Layer 2/3 redundancy including Virtual Routing Redundancy Protocol (VRRP).
• Support for multiple access technologies including ADSL2+, Active FTTx (100Mbps), Gigabit Ethernet, and 10 Gigabit Ethernet.

SIP-based call features (even in Lifeline mode)
• Caller ID
• Caller ID Blocking
• Call Waiting
• Caller ID on Call Waiting
• Call Forwarding
• Three-Way Calling
• Customer Originated Trace
• Toll/900/976 Blocking
• Distinctive Ringing
• Distinctive Call Waiting
• Selective Call Rejection
• Selective Call Forwarding
• Do Not Disturb
• Automatic Recall
• Speed Calling

Room For Improvement

One area where Pannaway can improve industry acceptance for its SCN is to develop integration partnerships with leading-edge vendors to solve specific problems faced by telcos. One example that is industry-wide is the in-house cable problem. The majority of the homes in rural America have coax cable, not CAT 5 which is a necessity for the delivery of Triple Play services. It also requires a tech to come and wire the home with CAT5 cabling for HDTV delivery to rooms with televisions. Although, there are new wireless technologies that claim wireless HDTV capabilities in the home. For instance, FOCUS Enhancements' announced its UWB chip-set goals are to transmit wireless video ranging from 880 Megabits per second (Mbps) at 8 meters to 37 Mbps at around 40-plus meters. These capabilities would enable multiple HDTV transmissions, seamless interoperability with the MBOA UWB Standard, and home or small office personal-area networks larger than ten feet to wirelessly connect personal/digital video recorders, TVs, set-top boxes, DVD players, printers, computers, etc.

Conclusion

The IP protocol has already established itself in the data world, followed soon after by the voice (VoIP) world, now IP is poised to take over the video world. Access networks that are built on IP will allow for seamless interoperability with the IP-based equipment that will make up the service creation platforms (i.e., digital video headends and

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softswitch servers) and dominate traffic flow in the core (i.e., routers and switches).

Pannaway claims that their Broadband Access Switch (BAS) is the industry's only SIP-based DSLAM/DLC replacement that couples ATM to IP interworking with support for E-911 Lifeline POTS and advanced calling features. One key advantage of the Pannaway 100 percent IP-based solution is the reduced operational and equipment costs of a truly converged IP platform that deploys the triple play of voice, video, and data from the same platform. The Pannaway BAS is a stackable IP DSLAM with a low per-port "Triple Play" delivery cost as compared to competing solutions. For scalability multiple BAS's can be interconnected via Gigabit Ethernet interfaces to support thousands of subscribers. Delivering digitized voice, high-quality video and high-speed data services simultaneously over a single broadband connection, the BAS enables telcos to quickly offer new revenue-generating services including parental control features, HDTV, and VOD, in combination with call features that include on-screen call pop-ups and distinctive ring.

Unfortunately as with all good things, my experience with the Pannaway product came to an end. It was with a heavy heart that I learned I could only test it for one day. Normally, we keep products in the labs for 30 days but alas, this equipment was their customer/trade show demo, so we were only able to keep it for a day. It was very difficult to say goodbye to the Pannaway equipment. Pannaway's Triple Play demo certainly left me desiring a Triple Play solution for my home in Connecticut (where SBC is the ILEC). All I have to say is "Hey SBC, remember me? I've been a Vonage customer for 2+ years, but before that I was your SBC customer for 10 years, but you overcharged me leading me to switch to Vonage. Here's your chance to earn my business again. Go deploy Triple Play. Then come and get me! I'll be waiting. Call me." Seriously though, Pannaway's Triple Play solution, which is based on all industry standards (SIP, Diffserve…) is a superb choice for service provider and carriers looking to deploy Triple Play and I would not hesitate to recommend it. IT

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From Voice Processing To Data Routing And Beyond

The Evolution Of Residential VoIP At The Processor Level

2004 was a turning point for the residential VoIP (define - news - alert) market. In North America, we witnessed independent service providers like Vonage (news - alert) lead the charge, while other suppliers, such as AT&T (quote - news - alert) and Primus (news - alert), began rolling out new VoIP services as well. Traditional telecom service providers and cable multi-service operators (MSOs) also launched testing and early deployment phases. As the focus now shifts from the feature-rich and lucrative enterprise market into the consumer space, a new series of challenges faces VoIP providers. Retail channels such as Best Buy and Staples are offering VoIP products from manufacturers like Linksys (quote - news - alert) and Motorola (quote - news - alert), resulting in commoditization and pricing pressures that will challenge providers, OEMs, and solution providers across the board.

As the industry takes steps to move VoIP deployments across a broader range of the market, a variety of end product configurations are emerging, including solutions that not only integrate the voice functionality but the combination of the voice gateway and home router as well. The early adoption of terminal adapter products, providing only basic analog to IP conversion for VoIP service, is being replaced by a much more feature rich voice gateway solution that encompasses both voice and data router functionality in a single solution. At the end of the day, service providers will live or die by features and capabilities — and many of the processors that are out there today cannot handle this. Price wars and entry into uneducated retail channels are setting the performance bar too low.

Residential Processor Performance Requirements

Voice/Telephony Processing

While some residential voice gateways are single POTS or FXS (RJ11) channels, the most common configuration in the residential VoIP space today is two full-featured voice ports. These ports must support voice and fax relay (the ability to transmit fax information reliably). Fax is most commonly characterized by the T.38 protocol. Voice encoding requirements always include the G.711 PCM vocoder but most service providers also require low bit rate vocoders (G.729ab, G.723.1) for low-bandwidth broadband connections, such as DSL lite. For full voice quality, each channel should have the complete, robust voice processing system that includes excellent echo cancellation, voice activity detection, adaptive jitter buffer/voice playout system, tone detection and generation, RFC 2833 for DTMF relay, a variety of caller ID variations and support for supplementary services such as call forwarding and call transfer.

Data Routing

From an architectural perspective, a residential voice gateway sits behind a broadband modem or fiber/ethernet device. While it provides interfacing for two full voice capable channels, a large number of other functional requirements exist. Among the most significant is WAN to LAN routing. At a minimum, the speed requirements must be the maximum rate capable by the modem. Key routing related features

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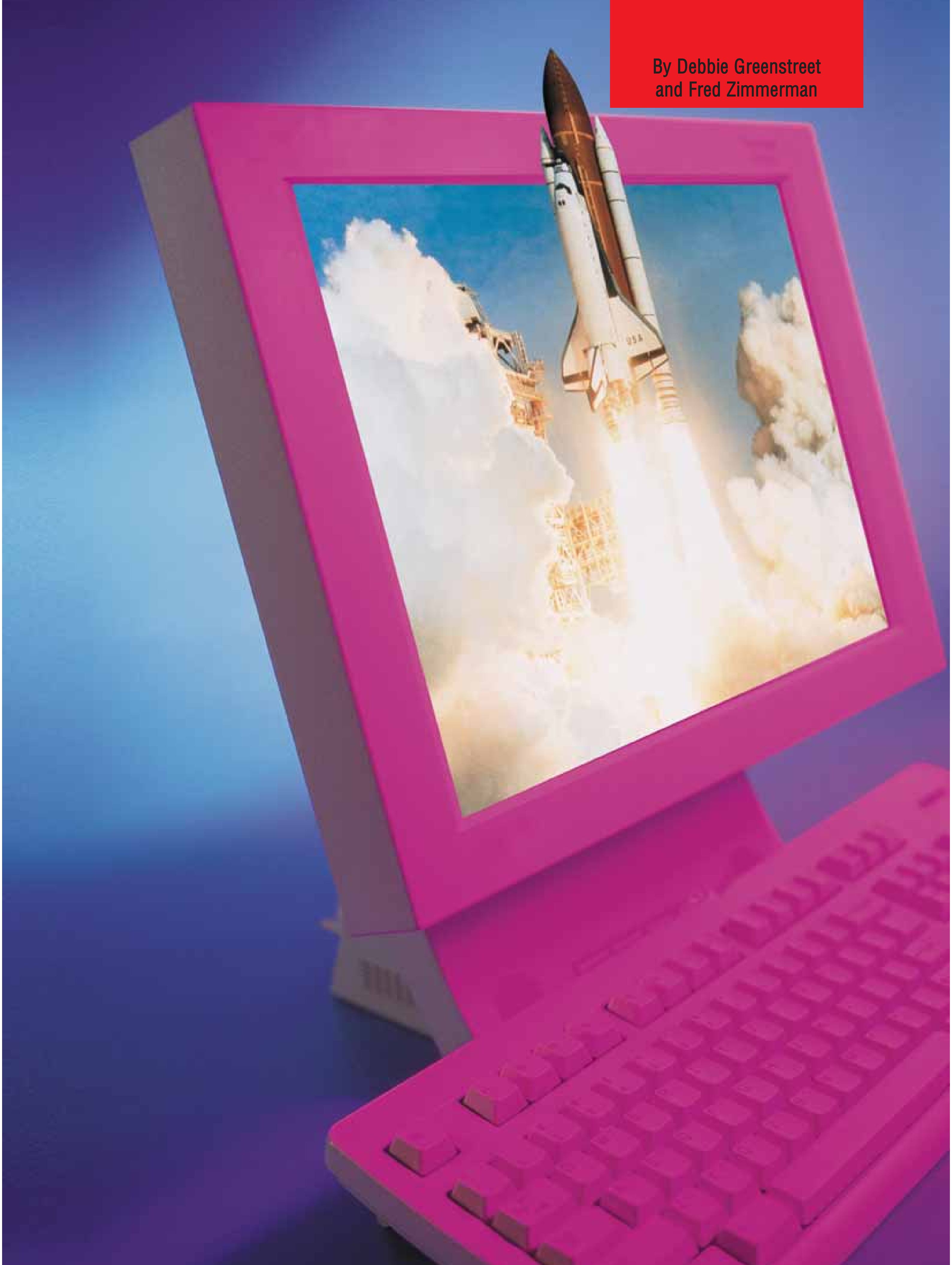

By Debbie Greenstreet
and Fred Zimmerman

such as Network Address Translation (NAT), device firewall, Quality of Service (QoS) mechanisms that support transmit prioritization of real time related packets (such as voice), and authentication and voice security features must be incorporated in the voice gateway design. The inability to handle these requirements will result in poor quality voice.

Data routing directs data from the external WAN network to a properly addressed computer IP address on the internal LAN. For a residential router, another function included is a firewall to protect the internal LAN from corruption or sabotage through the WAN.

Network Address Translation (NAT) (define - news - alert) is an Internet standard that enables IP addresses on private LANs to be separate or hidden from corresponding public IP addresses. The NAT function provides the necessary address translations so that data can pass back and forth from the LAN to the WAN and vice versa, while acting as a protection mechanism by shielding the internal IP addresses from public view. IPv4 is the most common type of Internet protocol in use today. There are a limited number of IP addresses available in IPv4 as the allocation of these address are mostly reserved, leaving little for private consumption. NAT has been the enabler, allowing multiple PCs or devices on a home network to appear as a single IP address to the public network, therefore consuming only a single address. This provides a layer of security by isolating the internal addresses from public access and also allows for internal addressing schemes and management without conflict to the public IP addressing model.

A firewall in the residential voice device is an element that serves as an enforcer of security between two networks. It determines which traffic to block and which to allow access to in an internal or private network. Firewalls are configured to protect against unauthenticated logins from the "outside" world and keep internal network segments secure.

Some of the features supported by a firewall include:

- Protection against remote login without approval
- SMTP session hijacking
- Operating system bugs allowing remote access
- Denial of service attacks
- E-mail bombs
- Hacker designed macros
- Viruses
- Spam
- Dubious source routing

Depending on how vulnerable a network is, designers may want to enable protection against all external attacks. However, this maximum protection will take up extra CPU cycles and therefore, reduce performance of the data router.

QoS implementations can vary, although they all typically include some type of service tagging or queuing to allow the prioritization of packets in high traffic conditions. In the case of voice gateway routers, the primary purpose is to ensure that voice packets have priority over data packets since voice packets are delay sensitive. If they arrive too late, or with significant delay in transmission, voice quality suffers.

Security, with respect to device authentication and service provider provisioning, is a key requirement in residential VoIP applications as well. It is critical that only qualified (paying) customers are allowed access to a service provider's network and services. In addition, providers are now starting to request security features for the voice payload and the voice call setup as well. These new features obviously require additional processing horsepower to execute.

Wireless LAN access point router functionality is becoming an increasingly important requirement in residential voice gateway devices. This level of integration allows customers to essentially have multiple "boxes" in one device. WLAN requirements usually include 802.11b/g functionality, as well as the access point software to properly route data packets between wireless home computers and the broadband modem. Of course, the performance of the access point solution plays a key role in the voice processor performance as well.

Processor Architecture

The processor selected for residential VoIP applications must handle a large number of simultaneous operations and functional real-time requirements, as detailed previously. The majority of processors in use for VoIP today are System-on-a-Chip (SOC) processors that have integrated additional circuitry such as ethernet, TDM, and memory, with direct relevance to the application.

The processor architecture must be able to handle four to five simultaneous data streams, including:

- The wide-area network, which is typically the broadband interface;
- The local-area network interface, which can be a single PC connection or a three- to five-port Ethernet connection;
- Two channels of telephony (voice); and
- [Often], a WLAN interface.

At the same time, the processor must also handle routing and application level functionality. Ideally, the processor should only inspect and tag for action these data streams, otherwise, it risks becoming loaded with data movement and therefore, is unable to perform data manipulation and processing.

The majority of the VoIP processors utilize a dual processor approach with both RISC and DSP integrated to share the processing load. These devices typically include the DSP function for voice/telephony processing and the RISC processor for network and telephony protocol processing (along with general device management), in a single, integrated solution. The RISC processor will also be used to perform routing functionality. These integrated devices are available in a variety of speeds and performance levels; some designed for basic voice gateway functionality, while others have the capacity to perform routing as well. This architecture is opti-

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mized for VoIP applications and often has the DSP and the RISC processor running at different frequencies, offering peak performance for robust voice and data routing.

Some SOCs utilize a single high-speed RISC processor that executes not only the telephony, network protocol, and router functions, but also the voice processing operations as well (which typically run on a DSP). This requires higher speed, and hence, more expensive and higher heat dissipating processors. To run the voice function on the RISC, it usually takes three extra instructions in RISC versus DSP for voice processing. Conservatively, it typically takes at least twice the amount of RISC MHz than it does DSP MHz. Significant care must be taken in sizing the overall solution functionality. If there are not enough MIPs at any given time, voice quality will be affected.

In analyzing the overall processor capability, the speed of the processor clock is only one key criterion. The other critical piece resides in the internal architecture configurations relating to data flow. It is important to minimize choke points in the architecture by relieving the RISC or DSP from the unnecessary task of data movement. Therefore, a distributed DMA architecture retrieves and delivers data to the processor without intervention. When the processing is complete, it moves the data to memory for queue to other peripherals. High-speed switching architectures, bus widths, data bursting capabilities, and intelligent peripherals that can directly move and process data without constant CPU monitoring, play a critical role in overall system performance. This function allows the processor to perform the processing, command and control functions that it is designed for.

Some of the architectural features supporting optimized data flow include:

External Memory Interfacing — External memory interface can affect performance and amount required if it is not designed well. Memory bus width of 16 or 32 bit, clock speed, and bank interleaving capability — features offered on all PC chipsets — can improve performance by up to 40 percent if combined with optimized software.

Processor Cache Sizing — Cache running at processor speed reduces slower external memory fetches and increases performance. Presence and sizing for instruction and data cache can be optimized for the typical processing and traffic requirements.

Internal Bus Switching — Switched central resource architectures with crossbar functionality can remove blocking effects that allow multiple data streams to flow simultaneously, as well as allow concurrent control register accesses. This enables data to move between any two peripherals that are not involved in data transfer.

Peripheral Configurations — Distributed DMA control and data bursting capabilities can maximize throughput by autonomously moving data with CPU intervention. Dedicated peripheral interfaces can be configured to require no CPU support after initialization, and maximize efficiency by efficiently performing specific tasks without CPU intervention.

Implementation of the capabilities mentioned above can significantly improve processing speed and reduce CPU loading. Optimization of the voice and data processing flows will lead to lower delay and jitter effects, therefore resulting in optimized voice quality.

Future Requirements

Looking at the near-term horizon, voice related security requirements are one of the most critical aspects of widespread residential deployment. Security mechanisms for the voice payload, which typically include encryption, authentication, and key exchange, are particularly important. A leading candidate for this functionality is the Secure RTP (SRTP) standard. In addition, the signaling portion of the voice functionality requires secure mechanisms as well. Since most residential voice gateways are SIP-based, SIP TLS is the most popular solution for voice call setup security. Both of these functions require additional processing power to execute.

Looking at the near-term horizon, voice related security requirements are one of the most critical aspects of widespread residential deployment.

As wireless carriers begin their rollout of VoIP with connectivity to existing wireless networks, the likely trend will be to include support for wireless-based vocoders, such as GSM-FR and EFR, and EVRC, in addition to traditional VoIP vocoders, such as G.729ab and G.723. Since wireless vocoders require more processing horsepower to execute than the traditional VoIP vocoders, care must be taken to include processor overhead for future growth.

Conclusion

As consumer mindshare and interest in VoIP continues to grow, new equipment configurations and applications will continue to emerge. OEMs will require optimized processor configurations that can be utilized across an assortment of product configurations. The R&D investment in platform and application software, as well as investments in field hardening and interoperability, must be leveraged across future product lines. Selecting optimized VoIP-enabled SOCs, which are designed for the requirements of residential VoIP gateways, will lead to reduction of overall BOM costs and provide the necessary flexibility for new and emerging VoIP applications. IT

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Software Media Servers Step Up

Software-based media servers are the next iteration in the SIP (define - news - alert) community's drive to move to a pure-IP services infrastructure that will allow service providers to increase revenues and lower infrastructure costs. Not only will software-based media servers increase profitability, but they will help to make technology choices future-proof.

Media servers play a crucial role in the delivery of advanced IP network services. An industry-wide movement to deploy IP-based, converged network infrastructure has opened up new markets and revenue opportunities for traditional service providers with their existing legacy networks, and for emerging carriers with IP-based, next-generation networks.

Media servers have, until recently, been almost exclusively hardware-based. Hard-coded DSP logic was the only solution for providing the ability to handling the media processing workload in phone networks. But software-based media servers can be deployed as a stand-alone IP-based media server solution to support SIP-based entities throughout a SIP-enabled network or in combination with an application server running SIP-based services. While used in its purest application to enable call signaling for call control, the SIP protocol by design was built to be extensible; many new innovative services will emerge, with SIP evolving from call control to more generic event control where events could be triggered by instant messages, voice messages, presence state changes, or even video sessions.

While hardware-based software media servers will continue to thrive in large-scale telephone networks, software-based media servers can supplant DSP-based hardware servers for "commodity IVR" functionality. "Commodity IVR" is the small set of features required for prepaid and most other IVR (define - news - alert) applications, including:

- Streaming pre-recorded prompts.
- Detecting DTMF entries.
- Standard IVR controls (e.g., barge-in, digit timers).
- Recording and playback.

Unshackling Service Providers

These software-only media servers will unshackle service providers from reliance on proprietary (read: expensive) hardware for base functionality. Utilizing off-the-shelf Linux-based systems, software-based solutions can support a variety of media processing functions, including announcement generation, DTMF detection and generation, message play and record, conference recording, audio bridging for small n-way conferences, and other advanced capabilities. Currently, software-only media servers are deployed operationally with prepaid IVR systems supporting more than 300 million minutes a month of calls. A single Intel processor today can support 400 full duplex IVR sessions with DTMF detection.

Traditional carriers are using PSTN/IP media gateways to perform Internet offload functions and in so doing defer expensive upgrades on their TDM switches; emerging carriers along with some traditional carriers (both wireline and wireless) are using media gateways to convert PSTN voice calls to "packetized" voice and route these calls to lower cost IP network infrastructure (IP trunking). Carriers are utilizing these media gateways to deploy proven revenue-generating services such as calling card services, conference calling, broadband telephony, and voice mail services. These solutions are incorporating an IP-based services architecture that includes IP media servers and SIP-based application servers connected to media gateways.

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By Ken Osowski

Service Logic Runs on Application Server

The IP-based media server takes DSP logic that ran deep inside traditional service nodes and makes these functions available to the service logic execution environment running on an application server. These functions include interactive voice response capabilities such as playing prompts and collecting DTMF digits, automatic speech recognition (ASR), bridging calls together for conferencing applications, as well as recording greetings, announcements, and messages to a server.

As call flow logic executes, the application server requests these functions when needed from the media server using either SIP or MGCP protocols. This decomposed service node model speeds up application development and allows these components to scale independently so that new services and capacity can be easily added.

As noted earlier, most implementations today rely on hardware-based media servers. But the capability of software-based media servers is growing by leaps and bounds. Because software runs on industry-standard hardware running Linux, it offers a lower cost-per-port than DSP-based IP media servers. As faster hardware becomes available, carriers can scale capacity without expanding their hardware footprint and without disrupting services. Common hardware components also means hardware spares can be reduced, eliminating the need to inventory dedicated, DSP-based media server hardware platforms. Carriers have the added flexibility of choosing a variety of cost-effective Linux platforms.

Software-based media servers run media handling logic to process media streams — usually written in C++ — directly on general purpose processors running off-the-shelf operating systems such as Linux. Hardware-based media servers run this same logic in firmware on DSPs that are designed to perform analog signal processing functions; they are optimized to perform media pro-

IMS: The Catalyst For Service Convergence Across Wireless & Wireline Networks

By Grant Henderson

VoIP (define - news - alert) has quickly become a mainstream technology that is changing the face of modern telecommunications. VoIP is now the foundation of every leading wireless, wireline, and cable/MSO operator's next generation network architecture, as well as a critical part of these operators' corporate strategy for cost reduction, differentiation, and increased competitiveness. The IP Multimedia Subsystem (IMS), as defined by the wireless industry's 3G Partnership Project (3GPP) is the most recent refinement of the enhanced services architecture — so much so that even wireline carriers are looking towards the IMS as the basis for achieving the nirvana of carrier operations: a single enhanced services architecture for delivering any service, using any media, to reach any customer, regardless of how they connect to the network.

Mobile Network Architecture & Standards

Historically, enhanced services have been deployed as a collection of point solutions and were delivered using a vertically integrated approach: an announcement server for network announcements, an IVR server for touch-tone services, a conference server, a prepaid calling card server, and a voice-mail server.

In 1998, a group known as the Third Generation Partnership (3GPP) was founded by a global consortium of standards bodies to define a next-generation mobile architecture that built on the success of GSM. 3GPP architects recognized that the traditional approach of deploying vertically integrated enhanced services was inefficient and set out to develop a horizontally layered architecture for enhanced service delivery. This new architecture was designed to minimize costs for the service provider, while supporting the universal delivery of enhanced services, regardless of the access technology used by subscribers to reach those services. The 3GPP subsystem responsible for enhanced services is known as IMS.

The key components of the IMS are shown in Figure 1. The Serving Call State Control Function (S-CSCF) provides the centralized "brains" of the IMS through interactions with the various service platforms. The S-CSCF maintains session state, while interacting with the gateways, service platform elements, and charging functions for the overall orchestration of service delivery. The Home Subscriber Server (HSS) continues its role as the main data storage for all subscriber and service-related data. Interworking with the PSTN and Internet is provided by various gateway elements defined by the 3GPP — namely the Media Gateway (MGW) for the PSTN or earlier-generation wireless networks, and the Gateway GPRS Support Node (GGSN) for connectivity with the Internet.

Enhanced Service Delivery Within The IMS

The key elements of the IMS architecture as it pertains to enhanced service delivery are the Application Server (AS), the Multimedia Resource Function Controller (MRFC), and the Multimedia Resource Function Processor

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(MRFP). Figure 2 shows their relationship and their respective functionality.

The Application Server (AS) is responsible for the execution of service-specific logic, for example call flows, database dips, and user interface interactions with subscribers. The AS delivers value-added services to the IMS, such as push-to-talk, ringback tones, prepaid calling card, multimedia conferencing and multimedia messaging service logic. The Multimedia Resource Function Processor (MRFP) — more commonly known as the IP Media Server — provides adjunct media processing for the application layer such as audio mixing, DTMF digit collection, content recording and playback, and codec transcoding. Unlike traditional service delivery approaches however, the MRFP is not dedicated to a single application but provides media processing as a shared resource to a multitude of applications. The Media Resource Function Controller (MRFC) provides a media resource broker function between the AS and MRFP resources in the IMS, and can be implemented as part of an application server or as a separate network element.

In an IMS-compliant service delivery platform, the enhanced service components communicate via industry-standard and open interfaces such as SIP. This allows service providers to mix and match best-in-class Application Servers with best-in-class MRFP/Media Servers, and eliminates some of the traditional cost and feature inertia associated with legacy service node approaches to enhanced services.

Figure 1: IP Multimedia Subsystem (IMS).

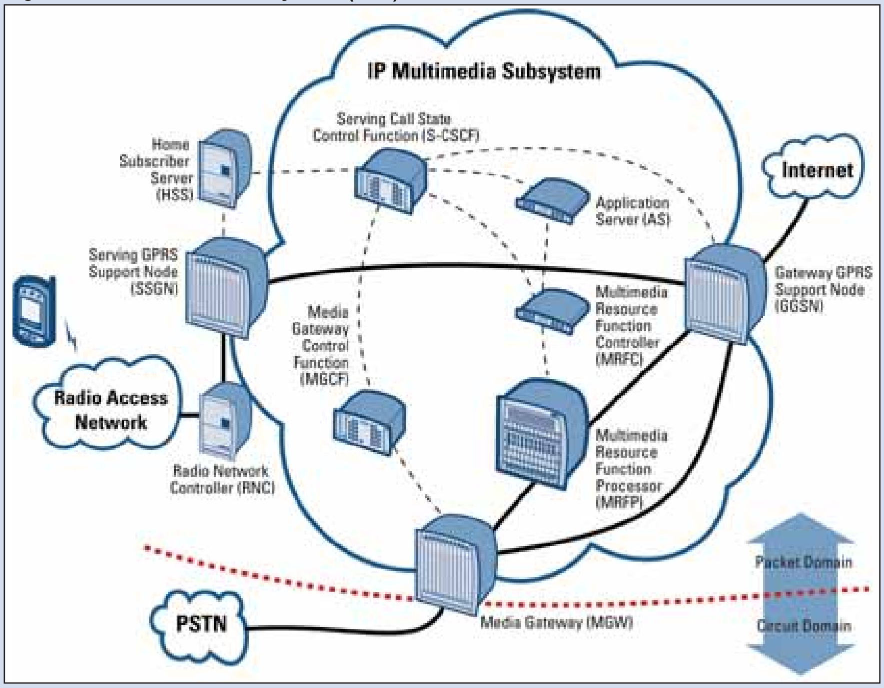
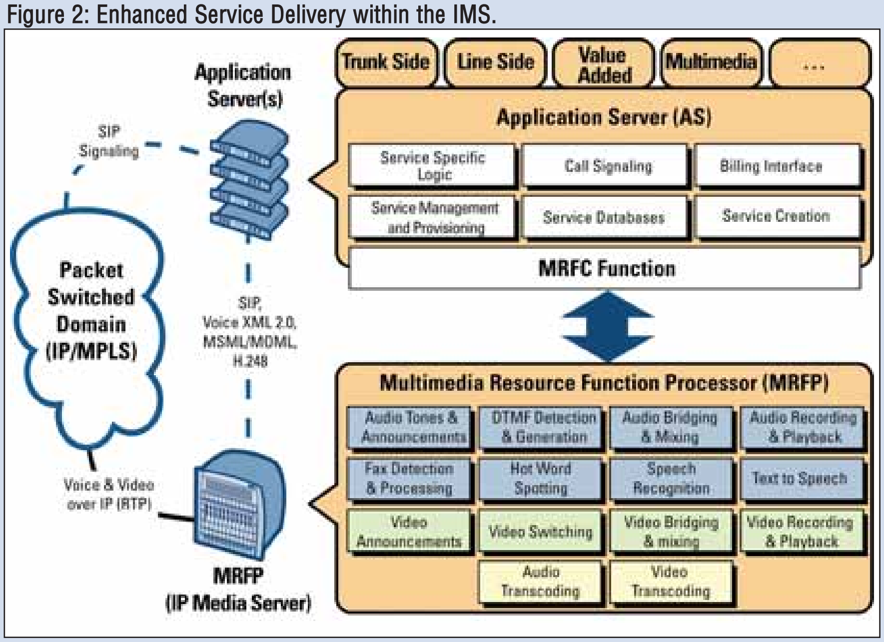


Figure 2: Enhanced Service Delivery within the IMS.



Applying 3GPP IMS To Fixed Line Networks

Historically there has been very little opportunity to share technologies across fixed and mobile networks, especially at the enhanced services layer. However today, the architectures defined by the wireless operators (i.e., 3GPP IMS), wireline operators (i.e., Multiservice Switching or Softswitch), and cable network operations (i.e., PacketCable Multimedia) share a great number of common goals, principles, and standards including the following:

- SIP is commonly specified for the end-to-end signaling.
- Application and call signaling logic is separate from the media processing functionality.
- Media servers are intended to be multi-service, multi-protocol, sharable resources for any voice or video service.
- SIP, along with XML-based scripting languages, are gaining momentum as the preferred control signaling between the Application Servers and MRFPs/Media Servers.

This level of common objectives and standards has resulted in a relatively clean mapping between the various functional elements across the standards. Although the nomenclatures differ, the functional intent and scope of various elements in the 3GPP IMS, Multiservice Switching, and PacketCable cessing functions and can scale to thousands of "ports" in a relatively small footprint. But software media servers can deliver up to 800 ports of IVR in a 1RU form factor, and given that this processing logic is written to run under Linux, it can be more readily changed to add new feature sets without requiring that DSPs are replaced to introduce new features.

In the case of a prepaid calling card application, a subscriber would typically call an 800 number provisioned on a media gateway to get access to the service. Based upon the number called, the media gateway "forwards" the call to the application server by issuing a SIP invite request. The application server then starts prepaid service logic that plays an introductory service greeting and prompts the subscriber to enter their PIN code. During this part of the call flow, the application server asks the media server (usually under MGCP control) to play the prompts ("Please enter your PIN code") and collects the subscriber's PIN code that was entered using DTMF input on the subscriber's phone. The "bearer channel" or voice path that enables the subscriber to hear the voice prompts is established between the media gateway and media server as RTP streams. In this type of application, the software-based media server can be used to play the welcome prompt, collect PIN information about the subscriber, or play "whisper prompts" to the subscriber telling them how many minutes they have left on their card.

In the case of residential broadband telephony with integrated voice mail, the software media server is used to perform IVR functions in conjunction with an application server. When a caller reaches a subscriber's voice mailbox or the subscriber accesses their messages, IVR menus are presented to the caller by application server service logic requesting the media server to play a prompt, to collect DTMF to navigate messages or collect password information. The media server also records

Figure 3: The table shows how functional intent and scope of various elements in the various architectures can be identified.

Function	Wireless 3GPP Term	Wireline Softswitch Term	Cable/MSOs PacketCable Term
Call Agent/ Switching Logic	S-CSCF	Softswitch	CMTS
Bearer Channel Interworking Unit	IMS-MGW	Media Gateway	Media Gateway
Enhanced Services Logic	AS / MRFC	Application Server	Application Manager
Media Processor	MRFP	Media Server	Audio/Media Server

Multmedia architectures can be mapped as shown in Figure 3.

The commonality of these architectures offers service providers deploying VoIP a relatively easy migration path for adding mobility to their plans. In addition, since enhanced services developed to the IMS architecture can easily span fixed and mobile networks, equipment and software vendors gain a larger addressable market with a single common product development effort.

Industry Movement Towards Wireless Standards

A number of VoIP industry leaders, including British Telecom and Sprint, are ensuring that support for next-generation wireless services is a key consideration in their next-generation enhanced service architectures.

In the case of service providers like Sprint who operate both fixed and mobile networks, adopting a wireless-centric architecture is natural given their desire to create a common network infrastructure for both businesses, and the desire to deliver a common set of value-added services to customers regardless of whether they use a fixed or mobile handset.

In the case of service providers like British Telecom, who currently offer only fixed-line services, adopting a wireless-centric service architecture helps future-proof their network and enables them to enter the mobile market as new VoIP-friendly access technologies such as WiMax emerge. In fact, British Telecom is taking a leadership position in helping drive the industry towards this outcome through its involvement in the Fixed Mobile Convergence Alliance (FMCA) and its highly publicized BluePhone initiative.

Conclusions

The 3GPP initiatives around the IP Multimedia Subsystem (IMS) architecture have received a great level of interest amongst telecommunication carriers — even amongst the traditionally distinct wireline and cable network segments of the carrier market. The common principles and similarities shared between the various next-generation service architectures offer an unprecedented opportunity to achieve true convergence across services, media, and access technologies. Industry-wide adoption of unifying standards such as IMS will allow service providers to increase the applicability of their service offerings to broader market segments and achieve scale economies in the delivery of those services, while allowing subscribers to experience a level of unification that was simply impossible in earlier-generation networking technologies. IT

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voice messages to the message store or plays them back using RTP media streams. The audio streams are mixed by the media server to create a conference call or possibly bridge in an operator to assist the subscriber.

Scaling Gracefully As Needed

One commercially successful prepaid calling card service provider is using these technologies to scale to over 300 million minutes of monthly long distance voice traffic. This deployment takes software-based media processing functions and combines it with application server logic that runs the prepaid calling card application. This allows the service provider to scale very gracefully by adding one server at a time to increase its overall capacity. This eliminated the need to manage a hardware-based media server resource as yet another network element that needs to be maintained and capacity managed separately.

Software-based media servers are very real and readily support a breadth of applications today. Will hardware media servers ever get replaced by software media servers? For commodity IVR and voice messaging functions the answer is yes. Because it is possible to run these software-based media servers on industry-standard hardware running Linux, service providers are able to benefit from a low cost-per-port.

For proven, carrier-deployed, revenue generating services such as calling card, voice messaging, and broadband telephony, software media servers will emerge as the most economic and flexible alternative to DSP-based media servers. Intel processors running Linux will continue to deliver more raw MIPS for media processing tasks and provide to carriers an all-software alternative to deploying next generation communication services. IT

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Media Gateways… Oh How Far They've Come!

By Thomas Howe

What came first, the chicken or the egg? The answer in VoIP is clear: it was the gateway. In 1996, at the beginning of the VoIP revolution, VoIP gateways were the first available commercial product. The business model was very simple and lucrative. Two gateways were attached to traditional phone lines, and then they were connected in a point-to-point network. At first, a single clear channel T1 line was used to connect the boxes. Using the compression from the gateways, service providers could fit six times the phone calls into the bandwidth required by one, resulting in a large cost savings. At that time, gateways were pretty stupid devices, but that was fine since they were connected in static routes. No intelligence required.

About four years later, sometime around 2000, it became clear that adding intelligence in the middle would be a good idea. The intelligent thing in the middle was called a softswitch, and provided large amounts of intelligence to the network. You could intelligently route calls between the gateways, choosing destinations based on cost, quality, or anything else you can imagine. The gateways were easier to configure now, too — you didn't have to set up a dozen static routes. All you had to do was to point each gateway towards the softswitch. Billing was better as well. Billing could be done at one place, in the center of the network, easing deployments. This model also worked well for adding applications, as the softswitch would direct calls to some central device that would provide the service.

Today, as we start the first real deployments of large VoIP networks, a new complication is appearing. Service globalization demands network infrastructures that optimize not only network performance, but politics, organizational structure, and legal considerations. Instead of a one size fits all solution, service providers need flexible solutions that fit their unique needs. Standard solutions don't always work from a business or technical angle. Far from making networks simpler, VoIP has added yet another set of protocols, making the alphabet soup of telecom a little thicker. Flexibility is required to optimize today's networks, and handling intelligence at any point in the network is a key ingredient. As an example, a simple modification in tariffs may change termination and origination strategies overnight. International long-distance providers may see tremendous incentive to replace TDM connections with IP. A flexible solution that makes VoIP and TDM terminations transparent could mean the difference between a decent profit and a hefty loss.

Intelligent media gateways are the natural evolution of first-generation VoIP gateways, except that they are more flexible, they handle media processing functions and can host small VoIP applications. With them, the network planner has a powerful tool to handle dynamic and complex networks.

So what makes an intelligent media gateway intelligent?

To understand intelligent media gateways, a quick discussion of first-generation media gateways is in order. A media gateway is used to connect older, PSTN networks and equipment to VoIP networks. The role of the first-generation gateway is simply one of conversion. Calls from a T1 line are converted into VoIP calls such as SIP or H.323 (define - news - alert). Inbound VoIP calls are converted into traditional T1 calls. There are two distinct parts of any phone call: the signaling and the media. Gateways are responsible for converting both sides of the calls. First-generation gateways typically work with application servers or switches. In addition, media gateways tend to be "closed" devices. That is, their architecture doesn't lend itself to simple upgrades, upwards or downwards scalability or integra-

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tion of third-party technology. In fact, you had to choose your gateway very carefully since you may not be able to upgrade them easily, if at all.

Intelligent media gateways bring us into the second generation of gateway, and solve some real world operational issues that could simply not be solved with first-generation gateways. They can be used any place a traditional media gateway would, and will provide the same functions. But, unlike first-generation gateways, they have features that give them a much greater degree of flexibility. The most important feature of intelligent media gateways is that they depend on open standards as much as possible. This not only includes call control and management protocols, but also extends to hardware chassis and backplanes. By having an entirely open gateway platform, service providers can easily choose third party technologies to include in the gateway. In the future, you may even be able to mix and match line cards from two intelligent media gateway vendors. This openness naturally solves another pressing problem of gateways: scalability. Scalability is a critical feature because it reduces business risk — service providers no longer need to guess about network growth patterns, and overspend on initial purchases.

Another important function is the ability to handle any combination of protocol conversion. First-generation gateways are fairly limited in what they can do: convert TDM to IP, and IP back to TDM. They cannot handle IP to IP or TDM to TDM conversions. When added to a gateway, the ability to convert any protocol lets the network administrator keep his internal network clean and consistent while accepting anyone else's VoIP traffic, without regard to formats. This is critical for larger service providers, who want to have clean G.711 and SIP-based internal networks, but must accept H.323 or G.729 traffic from smaller providers.

Intelligence is not something you would expect to see in a media gateway. However, when it is added, you add a whole new world of flexibility. Instead of depending on a single vendor, physical location, or source for application intelligence, you can easily mix and match. Here's an example: imagine that your long-distance traffic from Mexico will be heavily tarriffed if you don't terminate the call in the country. The rest of your operation is located in New York. If you use an intelligent media gateway in Mexico to route the local calls, and terminate there as well, you can optimize your network expenditures without reproducing your large infrastructure in some far-off land.

Media servers typically have the responsibility of processing voice. Intelligent media gateways can perform this task as well. They can conference calls, play prompts, record audio and collect digits. Although it might seem like a useful but optional feature, it is really very important. The ability to process voice directly allows a VoIP application to be housed in a single device, without sending the call back into the core for processing. From a technical perspective, this reduces the bandwidth required between the remote sites and the central facility. From an operations standpoint, you can clearly tie the application to a certain geography or a facility.

Once you have one, what do you do with it?

You can use an intelligent media gateway any place you could use a first-generation gateway. There's no need to change any other network element. The benefit is that you have a cost effective, scalable, and robust platform. Over time, you may need to add intelligence around your network, and then you will have the option of distributing that intelligence to wherever it needs to go. A fantastic example is that as your business grows you decide to go to a distributed network model to increase stability and performance. With the right equipment selection today, your network is future proofed.

Another common use is to extend legacy equipment. Many rural local exchange carriers use older TDM switches to provide telephony service. They often find that their options are quite limited when they wish to go to VoIP. They can replace their entire infrastructure (AKA the "forklift strategy"), or they can set up a parallel infrastructure (AKA the "double your op-ex strategy"). Using an intelligent media gateway, they can add a simple VoIP service and place it next to their legacy switch. Because it's scalable, they can start it small and grow it with revenue. Because it handles all protocols, their existing TDM customers can take advantage of the new service, just as if they were VoIP based.

Just as minicomputers empowered smaller groups inside large corporations, intelligent media gateways will do the same inside large service providers. Since they don't require a complete infrastructure, and they are scalable, they provide the perfect platform for smaller, niche service offerings. Just like minicomputers, they can be used to overcome issues of performance, geography, politics, or legacy.

What do you look for? What should you avoid?

When it comes to purchasing intelligent media gateways, there are a growing number of vendors to choose from. Each has some variation in their offerings, but all provide the basic functionality in their platform. Here are the questions you want to ask:

1. Openness: Is the device based on open standards? Remember that openness refers not only to call control standards, but to hardware ones as well. Pick a solution that uses common technologies such as CompactPCI, so that you don't limit yourself from selecting other third-party technology components such as ASR or line cards.

2. Interoperability: How much interoperation has the device been through? Could you drop it in to replace a standard first-generation gateway?

3. Application Support: What applications are available now? Are they in use and will they generate immediate revenue? How do you create new ones? What partners are available for custom development?

4. Traction: How many are in the field? What large companies or organizations have standardized on the device? What industry awards has it received?

5. Carrier Class: Is it fault tolerant? Scalable? What are the management interfaces? IT

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VoIP: New Technology Versus Legacy Tax Policy

With the advent of Voice over IP (VoIP) (define - news - alert), the landscape of telephone communications is changing as dramatically as it did upon the advent of automated switching and cost-effective long-distance technologies. Until recently, the basic technology to make a telephone call had remained the same, but VoIP is taking hold and changing the way the world uses telephony.

When VoIP services first became commercially available, initial criticism of VoIP was aimed at its inconsistent, dubious quality, but today, the emergence of high-speed broadband services has significantly diminished these arguments. In fact, now enterprises are embracing VoIP services and on-premise IP telephony equipment, with some estimates saying that VoIP could completely supplant traditional voice services in less than 15 years. The implications of this forecasted revolution are far-reaching. Given that it is happening in an environment that is still dominated by the heavily regulated telephone services industry, these changes are bringing about unprecedented battles on the regulatory and tax fronts.

A New Era In Regulation And Tax

Currently under discussion is how this revolutionary technology should be treated for regulatory and tax purposes. The regulatory classification of these services could affect any or even all of the following:

- Whether or not state and local taxes are collected, and how;
- Applicability of local exchange carrier access fees;
- Universal service fund contributions;
- Availability of 911 emergency services;
- Law enforcement capabilities under the Commission on Accreditation for Law Enforcement Agencies (CALEA);
- Requirements, or lack thereof, for regulatory oversight and reporting.

In general, the U.S. Congress has been reluctant to impose tax burdens on anything remotely associated with the Internet. To date, it has prevented heavy regulation and taxation in an effort to avoid curtailing the robust, technological growth of online business, which might discourage investment in the Internet and related concepts. The tech market "bubble" still haunts recent memory; therefore, any potential slowing of Internet proliferation by raising the costs of use is deemed unacceptable.

So far, telecommunications service buyers, largely commercial enterprises and public users, have shouldered the financial burden of driving continued growth of Internet traffic and striving toward making telephone and Internet access universally available. Consumers, from both business and personal perspectives, pay taxes to support public Internet access and universal telephone availability, generally in the form of telecommunications taxes and surcharges — particularly charges related to the Federal Universal Service Fund. This fund, combined with existing state and local levies on telecommunications, can drive the tax burden to exceed 25 percent of the cost of service.

To date, the Internet Tax Freedom Act and its offspring (collectively ITFA) have been the major tax legislative guide regarding federal intent related to the Internet. Reaffirming its desire to squelch taxation of Internet access, Congress passed a retroactive Internet tax moratorium which was signed into law by President Bush in December 2004.

A significant battle has been raging at the Federal Communications Commission (FCC) where the issue is how, when, how much, and even if VoIP services should be regulated. If the FCC advocates limited regulatory oversight for all VoIP services, or even

By Jim Nason

advocate complete independence from oversight by classifying VoIP as an "information service," state taxing bodies could struggle to retain the litany of telecommunications taxes, fees, and charges.

Many tax administrators argue that most state and local tax telecommunication laws are written and enforced comfortably outside the relative reach of both Internet immunity considerations and regulatory classification. However, all parties tend to agree that the lack of clear tax guidance magnifies the potential tax issues.

The Impact Of Transaction Tax "Nexus"

Even if taxation of VoIP services could withstand the FCC's broad-reaching regulatory interpretations, taxing jurisdictions must still operate within the guidelines of long-standing judicial decisions, tax law, and supporting regulations.

Many feel that VoIP taxation will be tested against the beleaguered and aged concepts surrounding "nexus," which is roughly defined as the level of contact an entity must have with a jurisdiction before it is subject to the tax rules of that jurisdiction. As a result, nexus, rather than regulatory or even ITFA-like requirements, is likely to be the area around which the most significant skirmishes are fought. Nexus rules have, at their most basic level, dictated that companies must collect tax in jurisdictions where it has property or payroll (employees). The technology itself could allow VoIP providers to benefit from these rules. Lacking most of the requirements to operate a bulky, capital-intensive, circuit-switched infrastructure, a VoIP provider could limit its tax "footprint" to a small number of jurisdictions, reducing or altogether avoiding taxation. As a result, taxing jurisdictions, unable to reach VoIP providers due to nexus considerations, could be forced to forfeit this lucrative revenue source.

Ironically, this potentially dramatic shrinking of a traditionally stable tax base may go far beyond creating new pressures on state and local taxing jurisdictions, but also threaten the very mechanisms that fund universal service and expand public access to the Internet. Consumer advocates argue that since high-speed Internet access is usually required to use VoIP services, people unable to afford broadband access could be the only ones to bear the burden associated with universal services funds.

The U.S. government is already inundated by refund requests and litigation relating to the federal excise tax (FET) on communications, because it failed to keep pace with changing technologies. Although not a direct comparison, this tax controversy is indicative of what can be expected when tax law is not clear and up-to-date. Here, the U.S. tax code roughly holds that long-distance communications services are only subject to FET if they are measured by "time and distance." In today's world of plans that allow unlimited calling throughout the country, this wording has outlived its viability. The IRS argues that the intent of Congress was to tax long distance communications, but the problem is, that's not what the law states. The matter is now in the hands of District, Circuit, and Claims Courts throughout the country.

The Cascade Effect

Far more than just a problem suffered by the telecommunications-buying consumer, the lack of clarity by various taxing bodies has a cascade effect on the communications companies. Today, the term, "communications company," is

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not limited to the big carriers, but now includes wireless, wireline, telecommunications resellers, VoIP, and Internet access providers throughout the marketplace. This means these entities, too, must literally predict taxation rules that are yet to be written. They must interpret wide-ranging, conflicting laws and regulations across all the states and federal taxing bodies. In addition, potentially conflicting regulatory classification and broad-impact federal legislation could make meaningful compliance untenable, at best, for VoIP service providers.

There's a good reason for state and federal taxing bodies to be concerned about a loss of revenues with the lack of VoIP taxation. After all, telecommunications is already one of the most comprehensively taxed services in the United States. In fact, aside from "sin" taxes (on products like tobacco and alcohol) and fuel taxes, it is difficult to find an industry levy with a higher burden or more complicated tax regime than the taxes, fees, and surcharges on telecommunications and related services. Taxes on telecommunications bring tens of billions of dollars into federal, state and local coffers, financing not only typical general fund expenditures, but also many programs, including universal service, deaf and disabled services, 911 services, and drug prevention.

VoIP Delivers Technology, Yet Wreaks Havoc On Taxes

Despite all the clear technological advances of VoIP, telecommunications tax departments are challenged with applying complex antiquated tax law to evolving VoIP product offerings and billing requirements. As a result, the telecommunications industry must constantly walk the tightrope between applying aggressive tax policy, which means charging customers less tax on services and thereby suffering the wrath of a government auditor's alternative interpretation, or applying conservative tax policy ("when in doubt as to taxability, collect tax from customers") and facing possible customer class-action suits.

While VoIP may rewrite our understanding of technology in the communications industry, it isn't without a price. Its complexity just may compromise a formidable and somewhat stable tax base. If alternative methodologies are not implemented or existing legal interpretations reinforced, the uncertainties surrounding the questions of whether and how to apply tax could also lead to consumer and provider confusion, resulting in inconsistent billing amongst the players in the communications space and limiting new competition entering the market. With so many undecided factors, one thing is true: VoIP will fundamentally change the telecommunications landscape by altering the network used to complete the call. **IT**

After all, telecommunications is already one of the most comprehensively taxed services in the United States.

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Small Business:

"What's All This I Hear About Return On Investment?"

Are there savings to be had for small to medium size businesses (SMBs) in moving to IP telephony, and if so, where are they? Fortunately there are many areas of potential savings, and hence cost justification for smaller companies to move to IP-PBX (define - news - alert) phone systems. Some are more quantifiable than others, but it is now clear why smaller companies are beginning to move rapidly to this exciting technology.

For the smaller business, their phone system has traditionally been viewed simply as a depreciable asset that was a necessity for doing business. Larger businesses have for some time been cost justifying (at least partially) their phone systems with complex formulas eliciting savings from things like "moves, adds and changes" (MACs) all the way to more nebulous areas such as savings from "more efficient use of IT manpower" and a "consolidated network." But the "big hitter" savings from MACs that larger organizations realize just don't translate well into smaller businesses.

The focus of this writing is on the IP-PBX phone system, but there are clearly benefits for some smaller businesses today who want to move their long distance to VoIP, or Internet long distance calls. If companies are using large amounts of five-cents-per-minute long-distance, then they should certainly consider trying one of the Internet services. The overall quality of these services continues to improve; and with growing adoption in the residential market, VoIP long-distance is now making serious inroads into the business realm.

The potential SMB savings by replacing an existing phone system with an IP-based PBX breaks down into two general areas:

- Hard dollar savings which are clearly measurable.
- Soft dollar savings which are more difficult to quantify, yet can have a huge impact on their overall business.

As one might expect, many smaller businesses will reap their greatest payback from the soft savings area. Here are some specifics:

Hard Savings

While not the biggest payback for small businesses, companies should look at and measure their potential hard savings of moving to an IP-PBX phone system. There often are significant savings to be attained in this area.

VoIP – Intra-company long-distance: If companies have more than one location, they can connect the phone systems in each office together over the

By Paul Vance

Internet and eliminate any long-distance charges associated with those calls. This goes for teleworkers too! Of course, they must net out any incremental cost associated with improving their inter-office Internet connectivity, but this is usually far outweighed by the long-distance savings and the centralized control and call detail reporting.

Moves, adds, and changes (MACs): Although the savings aren't as significant as with large businesses, MAC savings are still relevant and measurable. Instead of expensive visits by the phone system provider to make changes, with an IP-PBX system, the customer manages their own MACs, typically by the IT manager or possibly the office manager. Large businesses often use numbers in the range of $100 per change and .9 changes per year per employee. For smaller businesses, the cost is more like $50 per change and .6-.8 changes per year per employee.

Conference Bridging: Most IP-PBXs provide for multiparty conference calls (more than three-way calling) which can eliminate the need for any third-party bridging service.

Soft Savings

Soft savings are where the meat of the benefit is for SMBs. Some would argue that several of the items listed below are "hard savings" in that you can come up with a metric to determine a value to the business. And companies are strongly encouraged to do just that. But since the specific values will vary dramatically from company to company, they are shown here in the "soft" category.

Voice Mail Communications

Visual Voice Mail — With many IP-PBXs, users have the ability to visually see all their voice mail messages on their desktop computer. This means they no longer have to listen to all the messages sequentially, just to get to the important one they are looking for. They can also annotate the message so they never need to listen to the message a second time, hence increasing their productivity.

Notification — Users can get immediate notification on their PC, cell phone, pager or Blackberry, not only that they have a new voice mail message, but who the call is from with the Caller ID data.

Companies should look at and measure their potential hard savings of moving to an IP-PBX phone system.

This results in a less intrusive interruption to the employee (especially when traveling), yet makes them more available to customers and fellow employees alike.

Forwarding — There is now a wide array of voice mail forwarding options allowing the user to annotate (with voice or text) a voice mail and send it on to others as a voice mail, or an e-mail sound file. Again, this is a soft savings, but there is clear value in terms of better customer service and better intracompany communications.

Call Forwarding

Call forwarding (and follow-me calling) is a standard feature on many of today's IP-PBXs. This powerful tool simply lets the user direct calls going to their office extension to a different location. One user of this feature recently said: "It has changed the way I do business. My customers and prospects can now reach me at their convenience." Particularly useful for sales, marketing and customer service personnel, it gives their customers easy, but controlled access to them.

Enterprise Instant Messaging

This allows for employees within the business to quickly communicate with each other. It works without the interruption of a phone call, yet is more urgent than an e-mail. By providing this feature only to users on the company LAN / WAN, it eliminates the worry of introducing global IM to the office with the inherent temptation to "chat" with family and friends. The savings is measured in terms of better internal communications (i.e., "Mr. Williams is on line four, do you want me to ask him to hold till you finish your call?")

Computer Telephony Integration

CTI has historically been considered to be the ability to dial a phone number from a computer application, such as Microsoft Outlook, and to get a callerID based "screen pop" with an inbound call. Depending on the customer's operation, this can offer a significant savings. The inbound screen pop in particular can save a customer a minute or two per inbound call by eliminating the customer database search. And the caller gets better customer service since the employee can immediately have their account record in front of them.

Automatic Call Distribution

For departmental use and for inbound call centers, ACD can be an effective tool. By routing calls to a department or group of operators rather than an individual, businesses can provide quicker service for their customers, and their operators spend less time routing and re-routing calls.

Interactive Voice Response

While normally more difficult to quantify, the IVR systems of IP-PBXs carry huge potential savings. From the simplest feature of say, offering directions to the office (i.e., "press four for directions"), to providing their customers with 24 X 7 access to their account information, the company benefits from better customer service and higher employee productivity.

Teleworker Integration

With most IP-PBXs, telecommuters as well as road warriors can be completely and transparently integrated into the office phone system. This means that calling their normal extension will ring their phone at their home office or their soft phone on their PC in their hotel room (if they are logged in). And, the teleworker and employees in the office can all "see" the status and availability of each other on the presence management screen on their computer. Company managers also have immediate and complete visibility into their employees' telephone activity. This technology now makes the overall concept of teleworkers feasible for small businesses, and has the added potential benefit of extending the business' hours of operation.

Integration of Calls With Customer Database

By doing some back-end database work, IP-PBX users now have the ability to automatically integrate their telephone activity with customers directly into their customer database. Businesses are doing such things as putting the call detail record of each call, along with a link to any recordings of the call, into their primary customer database as an activity. Some companies that record their calls are putting the actual recordings directly into their customer database. By putting this key data element (the phone call) into their customer relationship management system, smaller companies enjoy a huge new area for IP-PBX payback opportunity.

Reporting

With most IP-PBXs, the call detail records are readily available to the company's management. Of course management can observe the live calls at any time, including who is on the phone with what customer. They can get their normal reports on a regular basis, but they also now have the capability for quick ad-hoc information. For example, from a Web browser they can request to see all of the calls made today by any employee or group of employees. Of course the value of this management tool will vary from company to company, but many report getting higher productivity when their employees know all of their phone calls are visible to management.

There is a wide range of technology areas where an IP-PBX can pay for itself. Many companies will see one of the biggest payoffs to be the integration of the IP-PBX with their customer database system. Then look to voice mail management, branch office and teleworker integration, and interactive voice response as areas of strong payback potential.

A great many of the features and savings areas described above have never before been available to smaller businesses. Small businesses can benefit from this technology in other ways, including reduced telephone tag, easy use of formerly complex features (e.g., conference call), increased employee mobility, higher employee productivity, better information for faster decision making. When added up, there is a powerful return on investment to many smaller companies. IT

FacetCorp is a privately held company that develops business communications and networking solutions with an emphasis on premier customer service. For more information, please visit the company online at http://www.facetcorp.com.

Sample Small Business IP-PBX Return On Investment

Cost Item	Calculation*	Specific Value	Annual Savings*
VoIP — Intra-company Long Distance	$0.05/minute * minutes per year	2 teleworkers at 4 hours/month	$ 288.00
Moves, Adds & Changes	$50/change * .75 changes /year * # employees	25 employees	$ 937.50
Conference Bridging	$0.10/minute * minutes per year	4 connection hours per month	$ 288.00
Voice Mail Communications			
-Visual Voice Mail	1 hr/month * $30/hour * # employees	25 employees	$ 750.00
-Notification		Better customer service**	$ 500.00
-Forwarding		Better customer service**	$ 500.00
Call Forwarding		Better customer service**	$ 500.00
Enterprise Instant Messaging	1 min/day * 21 day/mo * 25 people * $30/hr	25 employees at 1 min/day	$ 3,150.00
Computer Telephony Integration	15 secs/inbound call * # of inbound calls	100 inbound calls/day	$ 3,150.00
Automatic Call Distribution		Better service, balanced work**	$ 500.00
Interactive Voice Response		Better customer service**	$ 500.00
Teleworker Integration		Better internal & external communications**	$ 500.00
Integration of Calls with Customer Database		Better customer records**	$ 2,000.00
Reporting Benefits		Better management**	$ 250.00
Total Annual Savings			$ 13,813.50
Approximate IP-PBX Cost			$ 20,000.00
Approximate Payback in Years			1.45
* Assumptions:			
-Number of Employees	25		
-Average Cost per Employee Hour	$30		

** All companies will not get benefit from all the cost items above.
Companies should try to assign a perceived value for their business for each of these areas.
The numbers used here are estimates for a typical small business.

Deploying IP PBXs With An ITSP

With Internet Telephony Service Providers (ITSPs) spending large sums of money on advertising and growing at exponential rates, they are quickly becoming household names. Businesses are also continuing to replace analog and digital PBXs with IP PBXs (define - news - alert) due to their increased functionality. The deployment of an IP PBX in combination with an ITSP is a natural combination that offers numerous benefits to businesses, ultimately increasing their efficiency and lowering costs.

The demand for IP PBXs and the services provided by ITSPs will continue to increase due to the need of businesses to minimize long-distance and operating costs, and provide their employees with communications tools to make them more productive in and out of the office. The combination of an IP PBX with an ITSP is a formula that businesses are learning helps them to achieve these goals.

Lowering Costs

The key benefit businesses will realize with an IP PBX and service from an ITSP is the reduction in costs paid to their traditional telephone carrier. Before the advent of IP PBXs, all long distance calls would be sent to a traditional carrier over dedicated circuits and through large and expensive central office switches. The number of lines a small business required was proportional to the number of employees. In addition, the cost of moves, adds, and changes on the local PBX was substantial.

Today an IP PBX gives businesses the option of managing their own system through graphical user interfaces with PC-like wizards. The same intuition used to install and operate office software on a PC can be used by common business people to manage their own moves, adds, and changes on the local IP PBX.

Traditional long-distance (LD) carriers are becoming obsolete. IP PBXs can make calls to other compatible IP PBXs and VoIP telephones free of charge. For LD calls back to the traditional PSTN, an ITSP can deliver the call at a fraction of the cost of a traditional LD carrier. The effective per-minute rates offered by ITSPs are much lower than traditional carriers.

Although they may still provide the Internet service, the traditional telephone carrier is starting to lose out on the voice lines into the business location. The number of business lines to each location can be dramatically reduced due to the ITSP's unique ability to deliver multiple virtual lines over a single broadband pipe. In many cases, the ITSP will not limit the number of simultaneous calls allowed on a single account, effectively offering the business as many lines as they have bandwidth to handle. This, combined with burstable Internet service, completely changes the way business plan for their telecom requirements. No longer will a business need to order lines based on the number of employees, but instead will just need to ensure their broadband connection has sufficient overhead to handle peak calling periods. It is still common to have a limited number of FXO business lines connected to the local exchange carrier to provide lifeline services and to serve as an emergency backup.

How It Works

The IP PBX provides many features to end users locally ensuring local communications and feature availability no matter what is taking place in the WAN. Some of the features offered locally include voicemail, auto attendant, conferencing, forwarding, company-wide dialing plans, and advanced features such as call relay, only available on IP PBXs.

SIP Call relay is an example of a feature not available on traditional PBXs. It allows a call to originate on one network and through the use of two-stage dialing, terminate on a different type of network. For example, a traveling businessperson could call into a local IP PBX by making a local PSTN call from a cell phone and then relay onto the VoIP network to call anywhere in the world in a more cost-effective manner. In this case, the businessperson has then completed a VoIP

By Brandon Weilbacher

A business' operation costs can be reduced substantially with an IP PBX.

call from their cell phone.

The IP PBX or legacy PBX equipped with an external VoIP gateway can register itself with the ITSP. In most cases a local DID PSTN number will be provided by the ITSP for incoming calls. Often this number can be an existing number ported over to the ITSP. In addition to providing multiple virtual lines for incoming calls, routing rules can be established on the IP PBX to send all outbound LD calls to the ITSP. Local outbound calls and emergency (911) calls can be sent over the LEC provided lines.

When calls come into the assigned DID number, the ITSP serves as the gateway from the PSTN and terminates the call as VoIP to the IP PBX, where the call can then be routed to a specific phone or an auto attendant. As long as the bandwidth is available, many simultaneous calls can come from the ITSP over a single broadband connection. This effectively provides multiple lines and minimizes the requirements for physical lines from the LEC.

For outgoing calls from the IP PBX there are many possible paths that can be configured depending on cost and reliability requirements. Calls can travel directly to a peer IP PBX without using the ITSP, they can traverse the ITSP network, or they can be delivered over the PSTN. It is generally recommended that all outbound LD traffic be sent by default to the ITSP and fall back to the PSTN in case of a network failure. Local calls and emergency calls should be sent out to the PSTN over the dedicated lines. These lines still provide network power and should serve as lifelines in the case of local power failures. Some IP PBXs are able to deliver the network power through the IP PBX to a limited number of phones at the customer premise.

Key Benefits

Less Expensive: Although LD costs have fallen over the years, they still represent a significant expense for businesses. By installing IP PBX systems, substantial savings will be realized. Not only will companies benefit from lower LD costs, but dependence on the LEC for costly lines will also be reduced. In addition, a business' operation costs can be reduced substantially with an IP PBX. The initial cost of a traditional telephone system accounts for only about half of the cost of ownership. The other half is moves, adds, and changes, which become much simpler with an IP PBX. With traditional telephone systems, moves required a technician on site that may charge for three to five hours of labor to physically change ports in the wiring closet and/or reconfigure the extension on the PBX. With an IP PBX, these moves are as simple as moving a computer. The phone can be unplugged by the user and plugged in at the new location without any changes on the PBX or wiring closet eliminating the cost of the technician.

Expansions will also be less costly. Most IP PBXs are designed with simple-to-use GUIs instead of the proprietary interfaces available on many traditional PBXs. With minimal training, the phone system can easily be managed by the same person managing the data network. In addition, expansions no longer require expensive and proprietary PBX line cards. Low-cost data switches, already in place for the data network, are used as the physical interface between the phone and the IP PBX switching matrix.

Virtual Local Numbers: An ITSP can establish local DID numbers in multiple markets that all terminate as VoIP to a centralized IP PBX. Of course these DID numbers can terminate as VoIP to IP PBXs at branch offices locations consistent with the area code, but they do not have to. A business can now create a virtual local presence in multiple markets and countries with local numbers that terminate to an IP PBX that may be on the other side of the world.

Free On-net Calls: Many ITSPs do not charge for calls that do not leave the ITSP's network. For example, a business may register their IP PBXs at each of their sites with the ITSP. Since all the IP PBXs are on-net, intra-company calls are free. This can be a significant cost reduction.

Productivity Gains: One important benefit may not quantifiable. IP PBX systems offer many new features that increase the efficiency and productivity of every employee on the system. One important benefit is the dissolution of distance now possible with these modern communications systems. Whether an employee is in the office, in a car with a cell phone, or using a soft phone client on a laptop in a hotel room, features are available to simultaneously ring these non-traditional extensions, or hunt different locations to find the person. If an employee does not want to be found, they can set preferences to send calls to voicemail and send voicemails and faxes to their e-mail inbox. All this can be managed with simple GUI interfaces even the most novice computer user can manipulate. Other users can utilize features such as call relay to make international calls from wherever they are instead of waiting until they get back to the office. Communications are key to successful businesses and IP PBXs are making communications more efficient.

Both the IP PBX and the ITSP are hot trends in the Internet telephony market. Putting them together makes logical and financial sense for businesses. As this trend continues, more and more IP PBX manufacturers and ITSPs will be announcing partnerships and interoperability. IT

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Defining VoIP:

Improving Business Performance

One of the great exercises that exist within the enterprise communications industry is the relentless search for VoIP's "killer application." For a majority of manufacturers, customers, distributors, analysts and media, this quest has inspired significant discussion and deep reflection. Trying to identify one compelling business rationale that would convince any reasonable businessperson to invest in this technology has become the telecom equivalent of searching for the Holy Grail. Interestingly, however, the market's perception of VoIP is much more granular than those expressed by the pundits. Customers are continually defining the value of VoIP on their own terms, through a number of tools that address real business needs. As an industry, we need to pay close attention to what our customers are saying if we are to capitalize on the promise VoIP technology offers to the enterprise.

The reality is that there is no singular, one-size-fits-all benefit for VoIP communications. There is no one killer application that is driving adoption rates, nor should one exist. Instead, VoIP serves many different functions for a multitude of business purposes. VoIP is best defined as a "killer enabler," a technology that delivers real, tangible applications that can positively impact a customer's most fundamental and specific challenges, from increasing revenue, to improving operations, to controlling costs.

Addressing these core business issues is driving a growing number of customers to seriously look at VoIP as a means to improve their productivity and efficiency.

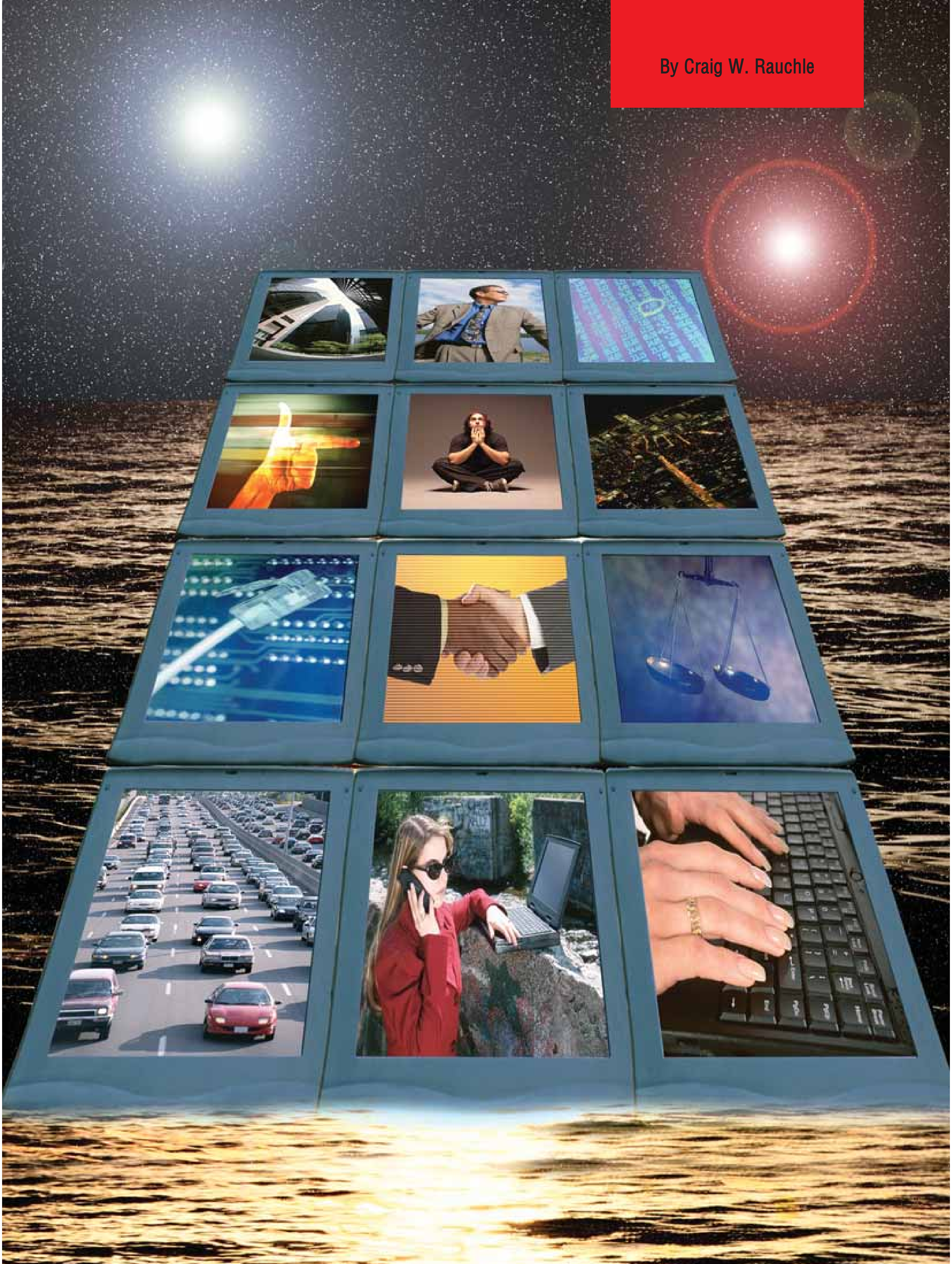
They've learned that VoIP makes the most sense when it is used to transport emerging applications that meet business challenges. It is a powerful technology that can deliver robust productivity tools to every employee and stakeholder throughout the enterprise, regardless of geography and infrastructure. The market understands where the technology fits in, and as a result, VoIP-enabled applications are being seen in a new light that are re-energizing the workplace. Some customers see VoIP-enabled applications as a cost-effective means for employees around the globe to communicate efficiently. Other businesses use the technology as a conduit to improve operational functions like managing inventory and improving workflow. And for other users, VoIP can help maximize revenue through the delivery of powerful applications that improve sales results.

In the mind of the customer, these are all compelling reasons to adopt VoIP. Truth be told, VoIP technology serves a vital, yet undeniably secondary role in terms of delivering a viable solution to the enterprise. It's the backbone the customer doesn't see, and in all actuality, doesn't really care about. Business owners vote with their pocketbooks, and they are clearly choosing VoIP for its impactful applications, and not for its commoditized transport capabilities.

In some respects, the maturation process of VoIP in the enterprise market is analogous to the commercial adoption of electricity. Even after Benjamin Franklin's profound experiments in 1752 unearthed many of the physical properties of electricity, the concept of harnessing electrical power in a form that would be beneficial to an individual or business evaded most people. In fact, some 120 years lapsed between Franklin's discovery and Thomas Edison's grand invention, the light bulb, the first real practical application of electricity. Once Edison illustrated how electricity could serve the general population, the world was transformed. Perhaps on a smaller scale, VoIP-enabled productivity tools offer their own compelling rationale for businesses to disrupt the way they conduct their operations. Our industry's mission is to illustrate how these solutions best serve the needs of our customers, on their terms.

VoIP-enabled applications that have received a great deal of coverage, like presence, collaboration, instant messaging, and mobility, are perfect examples of solutions that are generating excitement among the enterprise. While these tools all leverage the converged infrastructure, the real benefit lies in their unique ability to positively impact a series of business challenges.

For example, manufacturing companies are using presence and unified communications as a mechanism to initiate contact with their colleagues, vendors, By Craig W. Rauchle



and customers to track parts, confirm orders, and discuss operational issues with other facilities around the globe. By "seeing" the current status of these individuals throughout the enterprise network, users can quickly initiate voice calls and other communications as soon as their colleagues become available. In addition, many companies use presence as a viable tool to control and prioritize incoming communications. We're seeing sales people leverage presence to ensure that the call from a hot prospect is immediately routed to their current location, while relying on the application to direct other callers to team members for personal assistance, or even voice mail. In terms of generating revenue and streamlining efficiency, the value delivered by presence is gaining more and more attention from end users.

Collaborative applications are another example of VoIP-enabled solutions that will drive VoIP solution sales. One of the key reasons is the growing number of multi-location businesses that require efficient workflow and seamless internal communications, even though their facilities may be spread out halfway around the globe. VoIP-enabled collaboration tools are imperative to help companies address these mission-critical challenges.

In a VoIP context, collaboration takes many forms and performs a number of functions that serve the enterprise. As entrepreneurs, many of us have set up our organizations to leverage the unique attributes that certain locations offer. It's not uncommon for a company to have its headquarters in an area like downtown Chicago, a manufacturing plant in a country such as China, sales offices in cities like London and San Francisco, and a back-office location off shore, Barbados for example. Maintaining all these sites in different parts of the globe, in a number of varying time zones is a challenge for any business. And making sure these locations can talk to one another — and get work done — is a Herculean task altogether. Fortunately, we've seen several instances where VoIP-enabled collaboration solutions have helped customers make remarkable progress in improving operations.

One such collaborative tool is document sharing. Many companies are using this technology for a variety of purposes, like creating and editing sales presentations, reviewing schematics and other technological documents, and developing advertising copy, with their colleagues throughout the globe. VoIP-enabled document sharing means greater efficiency by enabling users to access these materials anywhere within the enterprise infrastructure, in real time. The technology allows for team members to retrieve their critical documents at airport hot spots, or just about anyplace that has a high-speed data connection. The barriers of time and distance are eliminated in the VoIP age, and the ability to use the technology to collaborate on documents wherever and whenever one chooses infinitely improves workflow and communication.

Conferencing is the cousin of the data-driven document sharing application, and that too is a collaboration solution that has been generating a great deal of interest. In the VoIP world, conferencing delivers much more than just connectivity between remote parties. The technology enables users to participate in events using myriad devices such as PDAs and other wireless devices, desktops, and even softphones, in nearly any location. The IP network's greatest attribute, delivering robust communications in mobile environments, is at its most profound in terms of conferencing. Companies that are deploying these applications are achieving better results in improving workflow, increasing sales activity, and streamlining communications.

VoIP is a powerful technology that can deliver robust productivity tools to every employee and stakeholder throughout the enterprise.

In addition, the emergence of instant messaging adds significantly to the overall value of VoIP communications. As a one-way vehicle, businesses use IM in a number of facets, like identifying the status and immediately contacting team members, and for initiating private communications with colleagues to enhance collaborative sessions. Instant messaging is just another example of VoIP-enabled applications that are gaining attention in the market.

As we continue to see the market express more interest in VoIP communications, it is important to come to the realization of what is driving this growth. VoIP is not a blanket technology that means all things to all people. There is no one killer application that will leave customers breathless, clamoring to integrate it into their infrastructures.

Instead, VoIP can best be defined as a "great enabler," delivering applications like presence, collaboration, messaging and mobility that are, by virtually all accounts, improving business performance in ways that were unimaginable just a few years ago.

And for businesses, that's the best possible reason to invest in VoIP. IT

Craig Rauchle is co-chief executive officer and chief operating officer for Inter-Tel, Inc. For more information, please visit http://www.inter-tel.com.

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CEO Spotlight

Michael Skubisz
Chief Technology Officer
Pannaway Technologies, Inc.

In the CEO Spotlight section in *Internet Telephony*® magazine, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Michael Skubisz, Chief Technology Officer at Pannaway Technologies, Inc.

GG: What is Pannaway's mission?

MS: Pannaway's mission has remained consistent since the company's inception: to be the premier access provider for the Incumbent Local Exchange Carrier (ILEC) community and to provide them with cutting-edge IP access technology and support that empowers them to increase services revenue, reduce operational costs, and compete more effectively against local cable and wireless companies.

Our company helps ILECs improve their success and increase profitability through the deployment of new and emerging IP-based services. We are heavily focused on providing quantifiable return on investment and value to our ILEC customers by delivering a robust suite of fully integrated IP access products enabling true "Triple Play" and beyond service delivery — and at a per-port cost that is unmatched in the industry.

GG: What is your vision for Pannaway and how is the company positioned in the next-generation telecom market?

MS: The ILECs that we speak with, almost exclusively, are targeting IP (either over copper or fiber) as their network transport for the delivery of high-revenue services like telephony, high-speed Internet as well as Video on Demand (VoD) and High Definition Television (HDTV). Additionally, technology analysts and other industry experts predict that IP will be the primary transport architecture used by carriers to deliver services in the foreseeable future.

Moving forward, we feel that our company is well positioned to take advantage of the increased interest in IP by delivering solutions to ILECs that are advanced, reliable, and above all else, cost-effective. Because we've built our architecture on IP as well as emerging technologies like SIP, and ADSL2+ Pannaway has the ability to move very quickly and will continue to stay on the leading-edge of the industry with regards to developing faster and smarter transport solutions. In 2005, we will challenge and break down access performance and feature barriers ensuring our ILEC customers are always prepared to support the next "hot" application.

GG: Describe some of the services Pannaway provides that sets you apart from your competition.

MS: VoIP might be the most sought-after technology in 2004 but many industry pundits agree there are still issues that need to be addressed before the technology is ready for primetime delivery. For instance, current VoIP offerings preclude customers from making E-911 calls during power or Internet outages.

We quickly recognized that the majority of our ILEC customers would not implement VoIP if it meant foregoing this precious life saving feature making VoIP suitable for secondary lines only. Working very closely with existing customers such as Oklahoma-based Cross Telephone, we architected our solution to deliver the industry's first and only VoIP offering that is fully Lifeline POTS capable — and with support for advanced SIP-based calling features like distinctive ring tones and conditional call forwarding. In the case of a power failure the Pannaway network continues to deliver phone service and E-911 calling capability to the subscriber without interruption. And, the caller doesn't lose any of the advanced calling features offered by VoIP.

This life-critical capability in conjunction with truly converged IP-based "Triple Play" service integration including IP Video, HDTV, Video-on-Demand (VoD), and bi-directional high-speed Internet access provides ILECs with the ability to offer their customers new emerging services with increased monthly subscriber income upwards of 200 percent.

GG: Now that it appears that growth and opportunity are the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

MS: From an ILEC's standpoint, VoIP by itself probably won't be a major revenue generator in the near future. While there are significant administrative, operational, and cost savings benefits to deploying VoIP from a telco's point of

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view, the technology on its own merit probably won't warrant exorbitant infrastructure upgrade expenses.

Emerging services like IP Video, HDTV, VoD, and online gaming will be the primary revenue drivers for telcos moving forward and will warrant the purchase of end-to-end IP transport infrastructures — of which VoIP will be an inherent component. That said, the development and proliferation of IP Video services will drive VoIP deployments because it makes the economic model work for ILECs. Also, the availability and security of digital video content will play a significant role in the advancement of IP Video services which in turn will drive VoIP deployments.

Another area that I think needs to be watched very closely is the regulatory landscape. Policy makers must refrain from applying traditional telecoms regulation to IP-based data and voice communications. VoIP is not just another flavor of telephone service. It's a new frontier in communications for individuals and businesses alike, and it requires forward-thinking regulatory approaches. If regulators subject this new technology to legacy telecom regulation, consumers and business users will miss out on the new services, increased choices and better prices that VoIP can deliver.

GG: What makes Pannaway's services unique and how can a client benefit from using them?

MS: Today, Pannaway provides the only "facilities-based" transport solution for the delivery of "Triple Play" services. Unlike current VoIP offerings from companies such as Vonage and AT&T, the Pannaway solution has been designed to deliver advanced IP services using the incumbent carriers existing physical plant. This approach removes any issues surrounding Enhanced 911 and Lifeline power requirements allowing carriers to deliver IP-based services without requiring their customers to change the way they use their phones or televisions.

This transparent introduction of IP technologies significantly strengthens an ILEC's position to roll out new revenue-generating services to subscribers. Deployment time is lessoned, price per port is reduced, and transport performance is substantial enough to support new and emerging applications.

GG: Describe your view of the future of the IP telephony industry.

MS: In the very near future, we will see a merging of traditional telephony features with Web-based technologies and the television. This convergence will introduce for the first time, a practical and widespread video telephone service. Also, users will have the ability to rent movies, make video calls to friends and family, retrieve voicemail and video mail, as well as send and receive traditional e-mail though their digital media entertainment system (formally known as the television).

Home telephony will become highly personalized incorporating many of the elements found on today's cellular phones including distinctive ring tones, follow me technology, and personal directories. Additionally, I strongly believe that 'toll calling' will go away completely. The new telephony model will be much more like the Internet providing any-to-any connectivity and "all-you-can-eat" calling plans — regardless of geographic location.

ILECs are in a position to capitalize on and even drive these emerging trends today and at Pannaway, we will continue to build advanced IP features and performance capabilities that will enable them do just that. IT

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