# *Beyond Dial Tone: Opportunities for Value in IP Telephony*

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*Abstract:*

*Though far from mainstream, IP telephony has progressed from its pioneering stages with a flood of technologies, products and service offerings. The market is now showing signs of maturing. This paper provides potential implementors with a model for potential IP telephony applications, and how the various technical approaches fit. Through an understanding of the applications and available technologies, users will be able to select and deploy solutions that provide tangible benefits.*

**INTRODUCTION**

Internet Protocol (IP) telephony was born commercially in early 1995 when VocalTec first introduced its Internet Phone software. This allowed two Internet-connected individuals anywhere in the world to have a live conversation using the Internet and their multimedia PCs. Internet Phone bore numerous limitations vis-à-vis traditional telephony—not the least of which was that the call participants had to prearrange their Internet call via e-mail or a standard telephone call. However, in spite of the constraints, the concept of Internet telephony captured the imaginations of those who saw the opportunity for low-cost long-distance calling as well as voice-based community-building on the web.

Since that time, IP telephony has become the focus of many technology vendors and service providers. While the market models have matured considerably, the benefits sought by users still tend to fall into two major categories:

- reduce the cost of voice and fax calls through toll bypass
- enable new applications through the combination of telephony, computer and data functions, using IP networks as the common communications platform

In spite of the flood of product and service announcements, actual implementation of IP telephony is only just beginning to take place at the enterprise level. This lag is natural for any new technology. After all, MIS and telecom managers' first priority is to provide reliable services and applications to their user communities. Now, the IP telephony marketplace has matured to the point where enterprise and carrier technology implementors can begin to assess the opportunities and potential benefits of IP telephony for their organizations.

## TECHNOLOGY BASICS

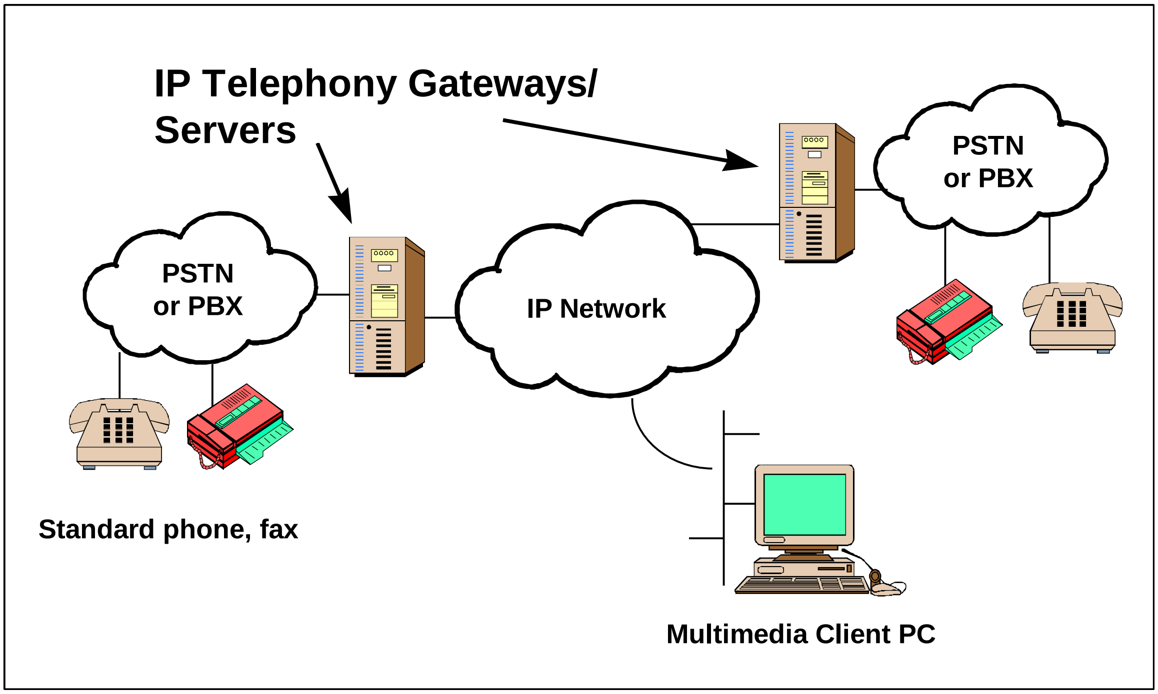
While this paper is not focused on the technical issues, it is useful to have a familiarity with the technical underpinnings of IP telephony as a foundation for understanding its applications and potential.

IP telephony can be defined simply as the transmission of voice and fax over IP data networks. Originating and terminating devices may be traditional telephones and fax machines, multimedia PCs, or a new class of 'Internet aware' fax machines and telephones. Within this paper, most of the discussion will focus on applications that have traditional phones and fax machines at one or both ends of the call. This emphasis is based on the pervasiveness of the PSTN and the corresponding requirement for most IP telephony solutions to interact with it.

### *IP Telephony Gateways and Servers*

Standard PSTN phones and fax machines connect to IP networks via IP telephony gateways and servers (see figure below). These devices perform some or all of the following functions:

- *voice/fax compression* - using any of several proprietary or standard coding algorithms, such as G.723.1, G.729a, T.38 or fax relay.
- *packetization* - formatting the compressed data into IP packets which contain routing and sequence information.
- *call routing* - transmitting the call to a remote gateway or server that is closest to the destination phone/fax number.
- *quality management* - a variety of techniques including buffering, interleaving and bandwidth reservation that compensate for delay, packet loss and congestion that may occur in router networks.
- *fax connection management* - sophisticated IP fax gateways use functions such as spoofing to support the tight synchronization of fax-machine-to-fax-machine call setup protocols that might otherwise fail due to network delays.
- *interactive voice response (IVR)* - necessary for some applications, prompting callers to enter PINs or credit card numbers priorto the real time session.
- *applications interface* - allows the server to support value-added applications, for example web-accessed call centers.




***IP telephony gateways and servers link traditional phone devices to IP networks.***

Both gateways and servers occupy the same positions within the network topology. The term *gateway* refers to devices that largely provide a connectivity function, while *servers* perform some value-added application in addition to providing connectivity.

***Technical Challenges***

Sending voice and fax over IP presents a number of technical challenges. Most of these relate in some way to preserving the quality of experience to which PSTN users have become accustomed over many years.

Delay is the biggest single concern for network managers trying to maintain good voice quality. Delay is introduced at a number of points in the network, including compression/decompression, buffering, and routing. Toll-quality voice service requires delay not greater than about 250 milliseconds. Delays that exceed this threshold become noticeable and annoying to callers. The public Internet is subject to highly variable delay conditions, and as a result is not considered viable for quality voice transmissions. Delay and other challenges, such as congestion and packet loss, must be addressed through proper network design and capacity planning.

***Technical Advantages of IP Telephony***

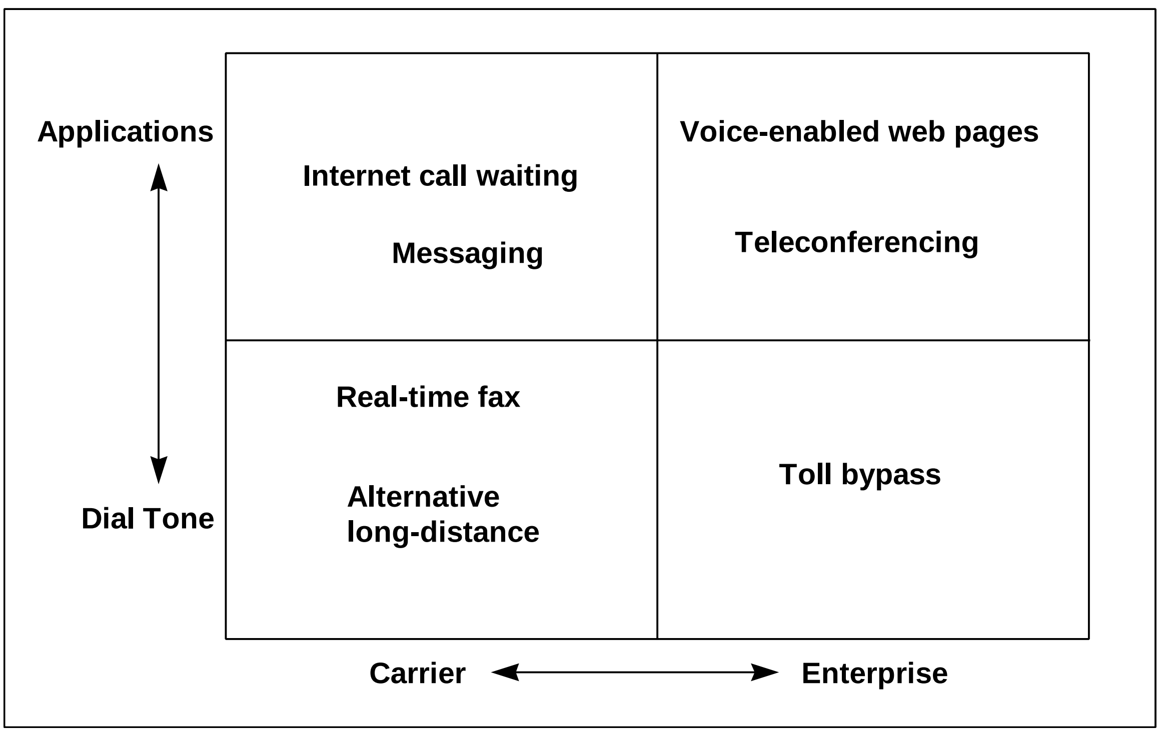
Technical benefits are achieved on several levels by transmitting calls over IP networks. From a utilization standpoint, packet-based IP networks are more efficient than circuit-switched networks which allocate a full end-to-end circuit for the duration of a call. IP telephony also saves network resources through compression, reducing 64 kbit/s PCM streams down to typically 5-8 kbit/s for voice or 14.4 kbit/s for fax. Further savings are made through silence suppression. It is estimated that up to 50% of a voice conversation is silence; circuit-switched networks cannot reallocate silent intervals to other calls, but packet switched networks can. These bandwidth savings are somewhat offset by IP protocol overheads, but they are still significant.

At an application level, IP telephony brings the benefit that it treats voice as a *data type*, alongside text, graphics, video, etc. This is the basis for value-added applications, which will be discussed later.

**EVALUATING OPPORTUNTIES FOR IP TELEPHONY IMPLEMENTATION**
A plethora of products and services are now offered under the heading of IP telephony. One way to bring order to the confusion is to map each solution along two dimensions:

- the core value of the solution, going from simple dial tone up through value-added applications
- the venue for implementation - i.e. the carrier vs. the enterprise

The figure below illustrates this categorization graphically, with one of the dimensions on each axis, and representative solutions in each of the four quadrants.




***IP Telephony Solutions Classified by Venue and Value-Add***

By classifying solutions within this model, we gain a better understanding of their value to the user, and of the requirements of the system being deployed. This will become evident as we take a look at each quadrant and its solutions.

***Carrier-Based Dial-Tone Solutions***

This set of solutions, shown in the lower left-hand corner of the matrix, essentially strives to replicate existing PSTN voice and/or fax services at lower cost by using IP as a transport medium. These services may take the form of consumer dialing plans and calling-cards. They may also be offered to enterprises as an IP voice backbone, similar to circuit-switched virtual private networks that have existed for some years.

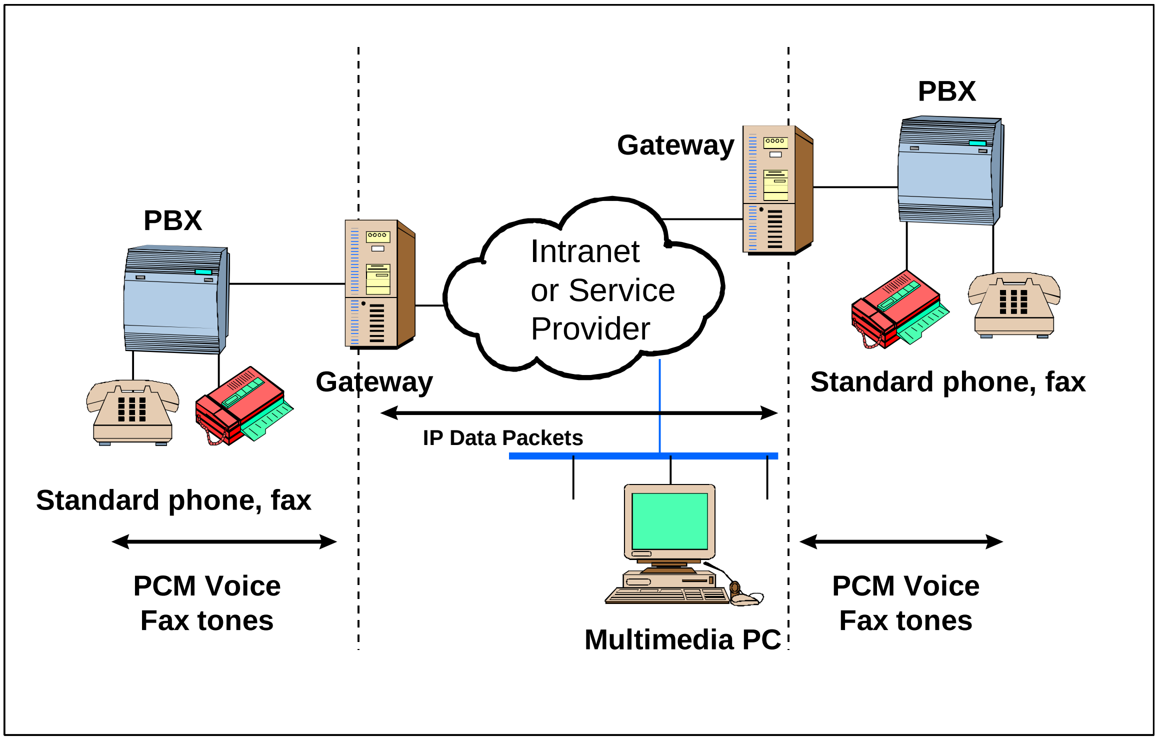
From a user perspective, the main requirements of carrier-based dial-tone IP solutions is that they be virtually indistinguishable from the PSTN services they are replacing in terms of voice quality. Consumers may be willing to tolerate minor inconveniences such as two-stage dialing if they perceive that they are obtaining significant price reductions; enterprise users will expect transparent dialing.

From a system perspective, carrier-based IP dial tone solutions need to be large scale, 1) so that they can have termination points close to the majority of call destinations and 2) so they can attract enough traffic to gain economies of scale sufficient to achieve low cost. IP telephony gateways used in these solutions must offer high density—hundreds or even thousands of ports per system—at the lowest possible per-port cost. These gateways need not support fancy user applications, but they do require real-time network monitoring tools to ensure service availability, and sophisticated billing systems to support the business.

Requirements for fax-only services differ from those for voice in a couple of important ways. Fax-only gateways can be less dense because fax calling volumes are less than voice volumes. Also, networks that support only fax can tolerate significantly higher delays—up to 3 seconds or more—thanks to spoofing technologies employed on intelligent fax boards such as the Brooktrout TR114, and utilized in carrier class gateways such as those from Clarent Corporation and Inter-Tel.

### *Enterprise Dial-Tone Solutions*

Enterprise solutions seek to save toll charges by routing long-distance calls over dedicated 'data' lines between company offices. Even external calls may be routed over the company network to the network location nearest the call-destination, saving toll charges on all but the "last mile". In terms of quality and user experience, the requirements of these solutions are similar to those of carrier-based systems. Additionally, these systems must support existing PBX-based functions, such as call-forwarding and conference-calling. Enterprise solutions tend to be much smaller in scale, requiring no more than hundreds of ports at all but the largest sites, and as few as one or two ports at small branch offices.

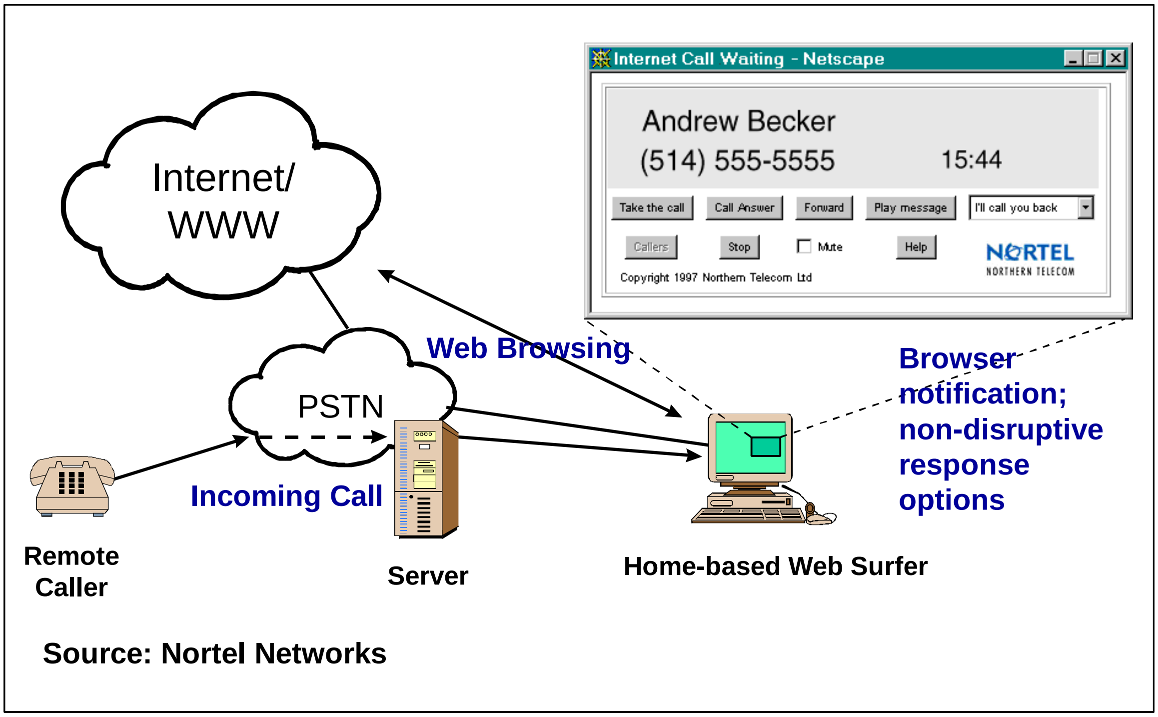



***Enterprise Dial-Tone Solution: Saving Toll Charges By Transporting Voice and Fax Over the Corporate Data Network***

### *Carrier-Based Value-Added Solutions*

A number of applications are now being introduced by service providers in which the main benefit is not connectivity per se, but some value-added function related to the connection. A good example is Internet call-waiting, developed by companies such as Nortel. Internet call waiting allows home-based web surfers to act upon incoming calls without interrupting their Internet connection. A window pops up in the browser, notifying of the incoming call and providing several options for dealing with it. Another example of carrier-based value-added solutions is messaging. A number of services, developed by companies such as Voice and Data Systems, exist that allow users to access voice-mail, e-mail and faxes through a single desktop or telephony interface.

Systems for carrier-based value-added solutions are generally smaller in scale than those for dial-tone solutions. While they may need to handle a large number of subscribers, only a subset of subscribers will be using the value-added function at any given time. Because of the smaller scale and higher intrinsic value-add, these solutions are not as cost-sensitive on a per-port basis. They do need to be very flexible, however, to support rapid development and modification of applications. Carrier-based value-added systems may need to support more computing functions, such as database lookups or message storage and retrieval. Finally, these systems must incorporate billing and accounting functions that feed the appropriate service usage data into the carrier’s mainline billing systems.




***Carrier-Based Value-Added Solution: Internet Call Waiting***

### *Enterprise Value-Added Solutions*

IP telephony provides a number of opportunities for value-added communications within the enterprise, or between an enterprise and its customers. One example is call-enabled web pages, variously called 'voice-button' or 'push-to-talk', as developed by Nortel and eFusion. This application allows visitors to a company web site to select a button on the web page which automatically establishes a call into a pre-programmed corporate location, such as the ordering department or customer service. Depending on the implementation, remote callers may speak through their multimedia PC or through their traditional desktop phone.

Another enterprise value-added solution is teleconferencing, by which geographically separate employees can hold an online meeting, using voice, data and possibly video to replicate as well as possible the face-to-face meeting experience. Teleconferencing is not new, but ubiquitous IP networks and OS-independent browsers remove barriers to widespread adoption.

Yet another class of enterprise applications use the flexibility of LAN servers to act as programmable call-switching and routing nodes. For example, in an application developed by InVADE, IP telephony and wireless LAN technologies are combined to support facility-wide wireless phone service within environments such as entertainment centers and warehouses where staff are continually moving about.

System requirements for enterprise value-added solutions are similar to those for carrier value-added solutions, though they may be smaller in scale. Applications that interface with the PSTN will require gateways that support the chosen application. Because the value is in the application, flexibility and rapid development are at a premium. Depending on the application, enterprises may be willing to forego so-called 'carrier class' attributes such as hot-swap and power redundancy, but enterprise systems still need to operate reliably and consume minimum possible MIS resources.

### *Little Things Mean A Lot*

It is always tempting to paint grand future visions, but small new applications are often the ones that take off, especially if they efficiently address a particular need. For example, Internet call waiting not only solves the problem of busy phone lines, it also provides an ongoing revenue opportunity for service providers. Likewise, push-to-talk technologies may in future become the basis for intranet-enabled company call directories. Employees could be searched by name or department, and then phoned with the push of a button.

It's still too early in the market to say which applications will catch on and which ones will never go beyond the trade show floor. However, it's important to keep in mind that some of the more mundane-sounding applications can end up being the most popular, because they fill a specific need and they do not require a major change to work behavior or business practices.

## CONSIDERATIONS FOR IMPLEMENTATION

Organizations that choose to implement IP telephony applications face a number of choices in terms of technologies, standards and business practices. In this section we will examine two of the major choices: open vs. embedded systems, and standards-based vs. proprietary systems.

### *Comparing Open and Embedded Systems*

Many of the early entrants to the IP telephony gateway and server market have based their products on open systems. These are PC or workstation chassis with computer telephony boards that perform key functions such as voice compression and line interfaces. Some products, such as the Brooktrout TR2001, combine multiple functions on a single board. By using open systems, these vendors have been able to take advantage of hardware available on the open market while focusing their own development efforts on application software, hardware integration, and network implementation.

In parallel, many voice and data networking vendors have focused on adding IP telephony gateway capability to their existing network devices—PBXs, routers and frame relay access devices (FRADs). These vendors are looking to leverage their existing product architectures and installed bases. Both approaches have their merits, depending on the existing network environment and the intended applications.

Embedded systems may offer significant cost advantages, assuming that the IP telephony function can be added as a board or firmware upgrade to existing equipment. If this is not the case, then a new box must be purchased regardless. Embedded systems may also have a cost advantage and very low and very high port counts. The overhead cost of the PC motherboard and chassis must be spread over at least eight ports, preferably 24, to provide a competitive per-port system cost. At the very high end (DS3 and above) open-standard form factors such as ISA and PCI cards do not support high port counts as efficiently as do larger, proprietary forms. The emerging compact PCI form factor increases the port scale at which open systems are economically competitive.

Open systems have the chief advantage of supporting value-added applications. These systems are based on popular operating systems such as Windows NT and UNIX. Not only can applications be more easily developed on these operating systems, but enterprises and service providers are not reliant upon the equipment vendor for the applications; applications can be developed internally or procured in the open market. Likewise, open systems also take advantage of the competitive market for computer telephony hardware.

Open system devices are most cost competitive in the mid-range scale—from one to eight T1/E1 spans, or 24 to 240 ports per chassis.

Many network managers consider a PC to be insufficiently reliable for use in network applications, particularly voice telephony, where users expect uninterrupted service. However, when properly configured for telecom environments, open systems can be extremely reliable. Many vendors offer industrial-class rack-mounted PCs, with features such as built-in redundancy. Compact PCI offers telecom rack-style form factors with higher-grade physical connectors and support for features such as hot-swappability, power redundancy, and extensive cooling.

| Capability | Open Systems | Embedded Systems |
|---|---|---|
| Add-on to existing infrastructure | - | X |
| Dial-tone platform | X | X |
| Applications Platform | X | - |
| Flexibility | X | - |
| Economic Scale (ports) | | |
| very high | CPCI | X |
| mid-range | X | - |
| very low | - | X |

***Open vs. Embedded Systems: Pros and Cons***

### *Standards-Based vs. Proprietary Systems*

Enterprises and service providers considering IP telephony implementation are also confronted with competing standards as well as proprietary systems. As with the open systems vs. embedded systems debate, there is no single right answer; the choice depends on the application.

The major benefit of standards is interoperability. The best-known standard for IP telephony is ITU-H.323. H.323 is a far-reaching umbrella standard which includes coding/compression algorithms for voice, data and video, as well as call-establishment and switching functions, and recently incorporation of T.38 real-time fax. It's important to recognize that H.323 was originally developed for multimedia teleconferencing over IP; it is part of a family of H.32x standards which includes conferencing over PSTN (H.324) and ISDN (H.320). As such, H.323 defines more than is required for many voice over IP applications. The chief advantage of this is that H.323 provides room for growth from early single-media applications to multimedia applications in the future. Microsoft and Netscape are currently supporting H.323 in their multimedia browser extensions.

The disadvantage of H.323 is that its call-establishment methods—because they incorporate such rich application support—may be inefficient for certain high-scale voice/fax applications. The H.323 standards groups have work to address this with new ‘fast setup’ versions of the call establishment protocols in a the latest H.323 Version 2. It remains to be seen whether these improvements will support carrier-scale environments in which tens of thousands or even millions of calls must be set up and routed simultaneously.

There are also alternate proposals for standards aimed at providing more efficient, scalable call establishment methods, in addition to H.323 V2. The Internet Engineering Task Force (IETF) has proposed the Session Initiation Protocol, or SIP, and Bellcore has proposed its Simple Gateway Control Protocol, or SGCP. It is too early to tell whether these will take hold in the marketplace, but both of them address a requirement that needs to be filled.

At the media coding level, the compression methods called within H.323 are gaining widespread adoption across the board. These include G.711, G.723.1 and G.729A for voice, and H.263 for video. Even those products that shun H.323 for call establishment generally support at least one of the G.-series vocoders. These compression algorithms are already supported by multiple vendors, with implementations on digital signal processors as well as microprocessors. For example, G.723.1 speech compression/decompression can run on a DSP in a high-density multi-port server, but it also runs on a single-session client software on a Pentium-based desktop PC.

For provisioning of IP-based dial tone, it is not necessary to use the upper layers of the H.323 standard, and alternative approaches, including proprietary ones, may be advantageous as discussed above. In the long run some standard will be desirable to allow networks to inter-work with one another, and to ensure an open market for equipment. For provisioning of value-added applications, H.323 is important, because it takes advantage of existing desktop H.323 compliant client software to support the application. H.323 is with us to stay; there is already a lot of compliant software and a lot of industry energy to evolve the standard to support more applications. But alternative standards will no doubt also gain critical mass for carrier-scale dial-tone applications.

| Standard | Organization | Status | Function |
|---|---|---|---|
| H.323 | ITU | Approved | Network call control<br>Multimedia conferencing |
| G.711 | ITU | Approved | 64 kbit/s PCM voice coding |
| G.723.1 | ITU | Approved | 5.3 and 6.4 kbit/s voice coding |
| G.729A | ITU and Frame | Approved | 8 kbit/s voice coding |

| | | | |
|---|---|---|---|
| | Relay Forum | | |
| SIP | IETF | Proposed | Network call control |
| SGCP | Bellcore | Proposed | Network call control |
| T.37 | ITU | Approved | Store-and-Forward Fax |
| T.38 | ITU | Approved | Real Time Fax |

***Key Standards for IP Telephony***

## CONCLUSIONS

The IP telephony marketplace has matured considerably in the past year. It is no longer simply a technological phenomenon; real products are available that target specific solutions for value-add. Enterprises and service providers, armed with an understanding of what they need to achieve, can make intelligent choices among the products and technologies that are available.

## BROOKTROUT IN IP TELEPHONY

Brooktrout is an important provider of technology platforms for development of IP telephony systems and applications. Here is an overview of the company's current IP voice and fax product offerings:

### *TR2001 IP Telephony Platform*

The Brooktrout TR2001 is a platform for developers of IP telephony gateways and application servers. The TR2001 contains DSP resources and on-board T1, E1, ISDN and Ethernet interfaces to support up to 60 simultaneous voice/fax calls in a single PCI slot. TR2001 firmware complies with ITU H.323, G.723.1 and G.729a standards for IP telephony call control and voice coding. TR2001 developer kits provide full API support for voice/fax compression, data network call control, voice network call control and interactive voice response (IVR), with drivers for Windows NT and UNIX.

### *TR114 with Real Time Fax API*

The TR114 is Brooktrout's award-winning Universal Port intelligent fax/voice platform. The Real Time Fax API, an extension to the Brooktrout API, allows TR114 developers to create real-time fax applications on IP networks, providing tools that ensure reliable transmission even under significant network delays. The TR114 is available in a wide variety of configurations and is approved for use in over 30 countries worldwide.

### NT/FaxRouter™ SDK

The NT/FaxRouter SDK is network fax routing and management software that includes Brooktrout's network fax transfer protocol (RSAP), fax mailbox system, telephone number to IP address service, SNMP MIB and an interface to the Brooktrout CNMS configuration network management software. It allows developers to implement complete store-and-forward fax over IP systems based on TR114 Series hardware for up to 48 ports per system

### IP/FaxRouter™

The IP/FaxRouter is an embedded low-density peripheral for system vendors and service providers building store-and-forward fax networks based on the NT/FaxRouter SDK. The IP/FaxRouter reduces the costs of sending faxes by routing faxes over the Internet and companies' private networks. Brooktrout's IP/FaxRouter API enables developers to integrate the IP/FaxRouter into their fax applications to make those applications capable of using low-cost routing over the Internet.

For further information, visit Brooktrout’s web site at http://www.brooktrout.com.