 By George Ellis, Contributing Editor

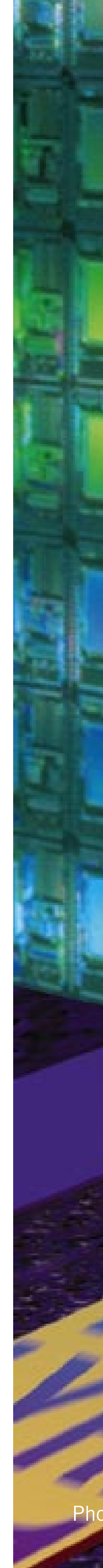
IP TELEPHONY PROMISES FREE, FEATURE-RICH TELEPHONE SERVICES, BUT QUALITY, RELIABILITY, AND SECURITY ISSUES KEEP SOME INDUSTRY EXPERTS DOUBTING. TO SUCCESSFULLY IMPLEMENT THE TECHNOLOGY, DESIGNERS MUST CONSIDER CHIPS, SOFTWARE, CODE LONGEVITY, AND VENDOR SUPPORT.

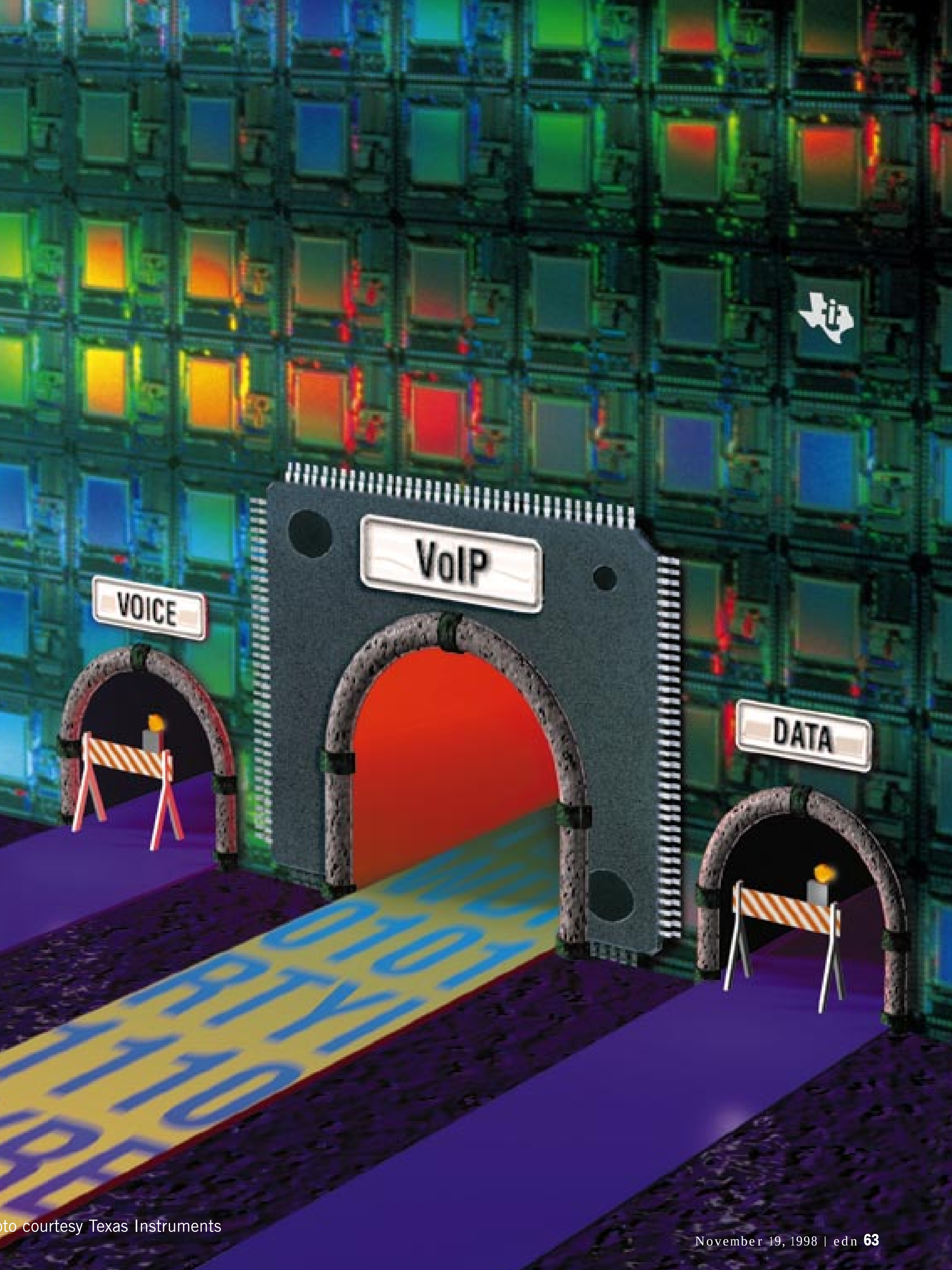
Internet Protocol: the *future route* for telephony?

INTERNET PROTOCOL (IP) telephony can yield big cost savings to both corporations and consumers. It is more efficient than the plain old telephone service (POTS) and is poised to undergo huge growth. Before that growth can occur, however, designers who want to use the technology have to vault hurdles concerning latency, quality, and security. Quality of service (QoS) is the primary problem impeding this growth. A traditional problem with this technology, QoS must improve enough to enable Internet-based services to compete with traditional telephony providers.

Many industry pundits think this scenario won't happen unless IP overcomes these problems (**references 1**, **2**, and **3**).

The term "IP telephony" covers a range of technologies, including voice-over-IP (VoIP) and fax-over-IP services, which are carried over both the Internet and private IP-based networks. IP telephony is part of packet voice,





oto courtesy Texas Instruments

which includes voice-over-asynchronous-transmission-mode (ATM) and frame-relay networks, which run faster than IP but are less common. IP telephony connects across combinations of PCs, Web-based telephones, and phones connected via public telephone lines to remote voice gateways. Because information travels in discrete packets, it doesn't need to rely on a continuously available switched circuit. Consequently, it's very bandwidth- and cost-efficient (see **sidebar** "IP telephony yields big cost savings").

WHAT ABOUT VOICE QUALILTY?

Although reasons exist to use IP telephony, QoS has been poor enough to single-handedly limit the adoption of this technology. The two most serious components of QoS, latency and lost packets, relate closely to each other. Every VoIP application involves converting speech into a series of packets, each containing about 30 msec of voice. After the speaker-side gateway receives the voice transmission, it converts it to packets and shuttles the packets onto the network. Packets traverse the network and reunite at the receiver's end. The network may lose some packets, and others arrive too late to use in the reconstructed speech. In either case, the speech plays back without these lost packets.

Processing and transmission delay all packets, and this delay causes latency in conversations. The transmission leg across the network is the longest, especially on the Internet. Lucent Technologies learned this fact in recent tests in which the company found that the largest factor affecting VoIP quality is management of the IP-network leg between gateways. Users expect latency of 250 msec or less—equivalent to the delay of a satellite link for international calls—for "toll-quality" service. Unfortunately, the Internet induces latencies that can far exceed 500 msec (**Reference 2**).

Jitter buffers also contribute to latency. Jitter is the speed variation between quickly and slowly traveling packets. The jitter buffer stores packets, allowing most of the slower packets to catch up. The less control in routing, the more jitter that results, and more jitter means a longer jitter buffer. But a longer jitter buffer introduces more latency. Too short a jitter buffer loses too many packets, causing voice quality to tumble.

AT A GLANCE

- Quality of service has been poor enough to single-handedly limit the adoption of Internet Protocol (IP) telephony.
- Latency exceeding 500 msec is a major problem that IP telephony must overcome.
- IP telephony may offer neither the reliability nor the privacy most of us take for granted with traditional telephones.
- Choosing IP-telephony DSPs depends on which algorithms the DSP can run.
- IP telephony yields the biggest savings in international services.

When the network loses a packet, VoIP products "reconstruct" it. The products cannot determine the information in the packet, but, like CD players smoothing over scratches, VoIP algorithms produce transitions that are less distracting than silence. Still, too many lost packets degrade voice quality to unacceptable levels. The maximum level of lost packets for toll-quality service is difficult to define, but 10% is a common level.

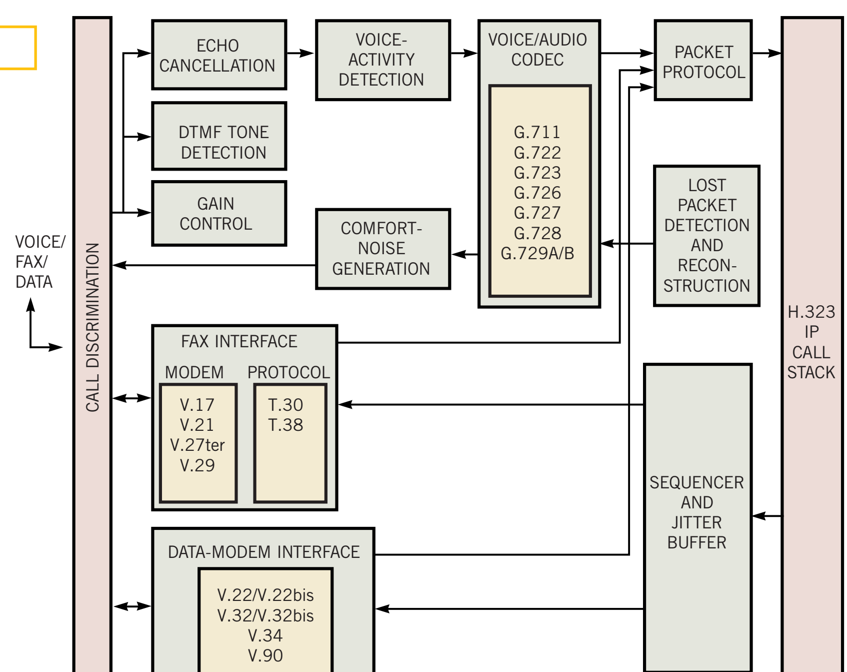
One cure for latency is to use a network that allows complete control of routing: Corporations use private intranets or Internet Telephony Service Providers (ITSPs), companies that use the IP networks to provide low-cost telephone services, usually relying on their own backbones. Private networks improve latency but at significant cost. This situation explains some of the skepticism of ID telephony: The steps corporations must take to increase QoS cut into savings.

COMPRESSION TECHNIQUES IMPROVE

Some early voice-compression techniques were so severe that the reconstructed speech sounded robotic. The G.723.1 algorithm, probably the most popular compression/decompression (codec) algorithms in the H.323 standard, seems to solve this problem (see **sidebar** "H.323: the standard in IP telephony"). G.723 can transmit voice on a thrifty 6.3-kbps stream, not counting IP overhead, and still score high in voice quality. According to Jon Louis, who heads true speech licensing in the United States for the DSP Group, G.723 scores 3.98 on the mean opinion score, an industry-accepted test of voice quality. POTS scores only 4. Experts usually consider 3.5 acceptable. Virtually no difference in voice quality exists between G.723 and

(continued on pg 68)

Figure 1



To choose a DSP for IP telephony, you need to know what algorithms the DSP can run.

IP TELEPHONY YIELDS BIG COST SAVINGS

The most common reason to use Internet Protocol (IP) telephony is to save money. Corporations can save by running intracompany voice and fax in the spare bandwidth of leased lines. Consumers can save by connecting computers to computers and by using ordinary telephones to connect to Internet-telephony service providers (ITSPs), which use IP to provide low-cost voice/fax connections through combinations of the Internet, leased lines, and the public switched-telephone network (PSTN) (**Figure A**). Using IP telephony saves money because it is more efficient than ordinary plain old telephone service (POTS) and because it avoids most of the tariffs and tolls telephone companies are subject to, especially in the monopolistic international telephone-service market. This area is changing rapidly, however, because the FCC has begun to categorize voice and fax services from ITSPs as subject to the same regulations as those from traditional telephone companies. Many experts are wary of relying on tariff avoidance because laws are changing.

IP telephony is efficient; IP voice conversations require less than 10% of the bandwidth of POTS. This situation results from two factors: First, compression techniques, such as G.723, compress the 64 kbps POTS takes to 6.23 kbps. It is true that the 6.23 kbps grows when adding the IP overhead of about 40 bytes per packet, but an overall reduction of 6-to-1 is realistic. Second, POTS requires full duplex—equivalent to 64 kbps in both directions—to support a telephone conversation. But that feature is wasteful because in conversations, only one person is speaking most of the time. Voice-over-IP (VoIP) products sense the silence to cease transmission when one party is quiet. This technique almost halves the required bandwidth. In the end, IP telephony commonly takes as little as one-twelfth the bandwidth of POTS to transmit conversations.

IP telephony yields the biggest savings in international services.

"The opportunities in the international market just dwarf domestic opportunities," says Gordon J Vanderbrug, vice president of VIP Calling (www.vipcalling.com), a Burlington, MA, company that markets carrier-grade IP voice services worldwide to telephone companies. Savings for IP-based domestic calls are modest because traditional long-distance service is already low-cost. But for companies that have multinational operations, the costs of intracompany telephone services add up. IP telephony can cut those costs. For example, one company switched from leased telephone lines to VoIP and cut costs from 50 to 17 cents a minute on calls to corporate headquarters in Japan (**Reference A**).

For consumers, the most convenient options may come through an ITSP. Whereas corporations often need to invest in equipment such as voice gateways, consumers can use ITSPs with an ordinary telephone. The ITSP provides the necessary equipment to convert the voice to data, transport the data, and convert it back to voice. Using the PSTN, consumers can connect to the ITSP with a local call. The ITSP reduces costs using the Internet with its private backbone. One of the early participants in this industry is Delta Three (www.delta3.co.il), an Israeli company that sought to serve a large Russian immigrant population by providing IP service between Israeli and Russian cities. The company relies on the public Internet and its own network to ensure high quality.

Numerous ITSPs exist in the United States. Among them, I-Link Worldwide (www.i-link.net) claims to have the largest IP-telephony network in the United States and plans next year to offer low-cost IP-telephony-based voice services to about 60% of the US population. I-Link relies on its private network rather than the Internet to control quality of service. Qwest Communications International (www.qwest.com) also offers low-cost phone-to-phone voice services to western US cities via the company's private, fiber-based IP network. IDT Corp (www.idt.net) supports Net2Phone (www.net2phone.com) software, which allows a PC to call any phone in the world. For those consumers who want maximum savings without using a computer, desktop Internet telephones support toll-free communication. For example, Aplio (www.aplio.com) provides a $200 stand-alone phone that is almost as convenient as an ordinary telephone. Users establish a call using the PSTN, and the phone then switches them to the Internet. However, both parties must own the specialized phone, and both must subscribe to an ITSP (**Reference B**).

Another cost savings for corporations is using IP telephony to reduce maintenance. Most companies have an IP-based intranet and a private-branch-exchange telephone network running side by side: Merging the two networks can ease system monitoring and software upgrades and can save costs in the long run. At the corporate level, IP telephony lets corporations integrate e-mail, fax, pagers, and voice mail allowing telecommuters and business travelers to collect and forward multiple message types during one connection (**Reference C**). Also, fax gateways support fax broadcasting, which is common for voice and e-mail systems.

For companies that use call centers for sales or support, IP telephony allows various "surf-and-call" configurations from Dialogic Corp (www.dialogic.com) and other companies. With surf and call, customers browsing a Web page on a multimedia PC can speak with a company representative through the Web page. This flexibility is important because five of six homes in the United States have only one telephone line. On the other end of the call, companies can link their sales- and phone-support systems, so that sales agents have one link to the company and the customer. Among other advantages, this feature can support remote locations for the call-center staff.

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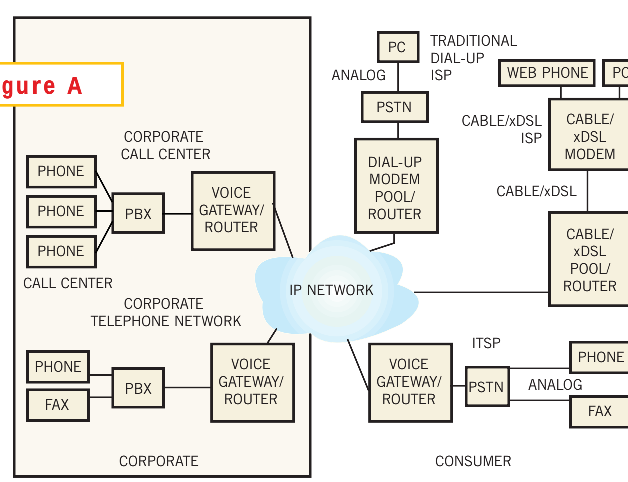
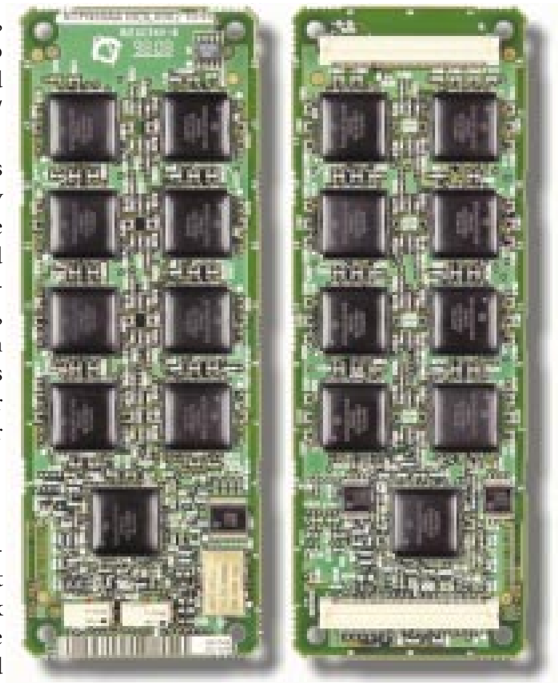
B. Bernier, Paula, "IP telephony gadgets," *Sounding Board*, August 1998 (www.soundingboardmag.com/articles/881cove2.html).

C. Jander, Mary, "One-for-all mail call," *Data Communications*, May 21, 1998, pg 78 (www.data.com).

Figure A



IP telephony yields cost savings to corporations, which can run voice and fax communications in the spare bandwidth of leased lines. Consumers can also benefit by using telephones to connect to Internet-telephony service providers.



Nortel produces this 80-channel daughtercard with nine Motorola 56307 DSPs on each side: one for the call stack and eight for telephony algorithms. You can plug 10 cards into a motherboard.

POTS, says Louis. To hear for yourself, the DSP Group lets you listen to dozens of examples of compressed speech at www.truespeech.com/samples.

In addition to QoS, two other issues arise with IP telephony: IP telephony may offer neither the reliability nor the privacy most of us take for granted with conventional telephones. IP telephony may address security for VoIP, but the solution may be encryption rather than laws enforcing privacy, as with conventional telephones. For now, privacy remains an issue for many potential customers.

QoS AND FAX

Fax escapes many QoS issues because latencies of less than 1 sec don't trouble most fax machines. Also, fax technology allows you to spool large portions of faxes at the receiving end before the fax transmits those portions to the fax machine. Thus, fax transmission is ideal for IP telephony, and many companies are interested in this technology, given that, according to one survey, about 40% of corporate America's telephone costs are for faxes (**Reference 4**). A company focusing on this market, @Fax (www.atfax.com), offers the Faxfree Server, which lets corporations send faxes over the Internet. It routes using adaptive least cost routing to find the cheapest path based on a company network. The company claims that Internet faxing can save an average of 30 cents per page.

CHOOSING THE DSP

Most IP-telephony products rely on DSPs for the heavy processing that voice codecs require. So, once you decide to build an IP-telephony product, how do you choose a DSP? First, you need to understand the algorithms the DSP can run (**Figure 1** and **Table 1**). Most chip vendors provide the software for most of these algorithms either directly or through third parties. Many protocols exist for voice, fax, and data modems.

According to Scot Robertson, marketing manager of digital modems at Analog Devices, the key to success in VoIP applications is that a product must be able to use any protocol on any port. The company's newest entry, the ADSP21-MOD970, provides six ports in any combination.

"This way, a gateway can support any combination of fax, voice, and modem on one piece of equipment, allowing full us-

FCC REGULATIONS

The FCC regulates what it terms "telecommunication services." According to the Telecommunications Act of 1996, telecommunication is the "transmission, between…points specified by the user, of information of the user's choosing, without change in the form or content of the information as sent and received." On the other hand, "information service" offers "the capability for generating, acquiring, storing, transforming, processing, retrieving, utilizing, or making information available via telecommunications." This definition lets information services escape not only FCC regulations, but also the Universal Service Fund (USF), which the FCC created to ensure local service access for low-income and rural consumers. Telecommunication service providers must give about 4% of their revenues to this fund. Those companies providing "telecommunication services" must comply with other FCC regulations, including obtaining FCC authorization for international services, filing FCC tariffs, paying regulatory fees to the FCC, and paying access charges to Local Exchange Carriers. So it's no surprise that Internet-telephony service providers (ITSPs) want to stay classified as information services.

In its April 10, 1998, report to Congress, the FCC found that, "ISPs should be considered as providing telecommunication services if they present themselves as providing voice telephony or fax transmission, permit customers to use standard telephones or fax machines, allow customers to call regular phone numbers, and transfer information without changing its form or content." (See **Reference A**.) The FCC found that several ITSPs meet this criterion. In addition, the report finds that companies that lease lines to ITSPs also provide telecommunication services and are subject to regulation. ITSPs that provide their own backbones appear exempt from most regulations, although they may later have to contribute to the USF. As a result, one of IP telephony's cost advantages—freedom from regulation—will erode in the future. And the effects may be worse for international traffic: Many countries are raising tariffs, and some countries are prohibiting IP telephony (**Reference B**).

The erosion of regulatory exemption probably won't stop IP telephony, given that many IP-telephony applications remain exempt from regulation. But for ITSPs competing in telephone-to-telephone markets on the basis of cost, regulations may prove to be a problem.

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B. Whelan, Carolyn, "Voice over IP: mixed signals," *Electronic News*, June 15, 1998, pg 53.

age of all ports", says Robertson." For network-service providers, high port usage translates into high profits because these providers can support more subscribers on the same hardware.

In most products, the voice chip does not handle the IP call stack. This fact is especially true in products for voice gateways that carry hundreds or thousands of channels. The intense computational requirements of voice mean that even a powerful DSP can handle just a few channels. The centralized nature of the call stack means that one process manages many channels, and this constraint implies that one chip processes the call stack and controls many voice-processing DSPs. You can see an interesting variation of this technology in Motorola's 56307, which can process both voice and the call stack. Nortel (www.nortel.com) produces a daughtercard that has one 56307 DSP processing the call stack and that also controls eight 56307 chips, each simultaneously handling 10 channels.

TABLE 1—IP-TELEPHONY ALGORITHMS

Algorithm	Description
Call discrimination	Most IP-telephony products are designed for more than just voice so that nonvoice protocols, such as data modems and faxes, must be recognized and relayed around voice processing.
Data modem/fax interface	Interfaces to these protocols are required for products that carry fax and data modems. The more protocols the chip supports, the better.
Echo cancellation	IP-telephony products must perform echo cancellation. The telephone company provides this service for long-distance calls, but local calls usually don't require it. Because IP telephony avoids the telephone company or relies on local connections, the telephone company's echo cancellation is usually unavailable.
Voice-activity detector and comfort-noise generation	Voice connections are almost entirely one-way. IP telephony halves the transmission bandwidth by recognizing the periods when a party is silent and ceasing transmission during those periods. The software that performs this task is called a voice-activity detector. However, people are more comfortable with a small amount of noise, dubbed "comfort noise," than they are with silence. Comfort-noise-generation algorithms perform this function.
DTMF detection/generation	Ordinary pushbutton phones use dual-tone multiple frequency (DTMF) to convert keystrokes to tones. However, audio codecs cannot reliably reconstruct these tones. Instead, the transmitting side detects these tones, encodes and relays them around the voice stream, and then regenerates them upon reception.
Voice (audio) codec	Chips need to process as many of the popular voice codecs as possible, but especially those of H.323. The protocol is fixed in cellular phones. This feature distinguishes the VoIP market from the cellular market.
Lost-packet detection and reconstruction	VoIP products must detect when packets are lost from the conversation. Then, like a CD player covering a scratch, VoIP products smooth over the missing information.
Jitter buffer and sequencing	Packets must be stored in a jitter buffer and sequenced so that the slower packets can catch up to the conversation. Poorer path control implies more jitter. Many vendors offer routers with intelligent jitter buffers that adapt to network traffic.
Packet protocol	After the voice is encoded, the data must be placed in packets that usually contain 15 to 60 msec of speech; IP overhead bytes must be added.

H.323: THE STANDARD IN IP TELEPHONY

H.323 is the omnibus specification that details many of the operations of Internet Protocol (IP) telephony and videoconferencing. The industry welcomed H.323 because it provides a route for interoperability among manufacturers' equipment. Such interoperability was not a concern when IP-telephony applications were closed systems: A corporation would purchase the equipment on both ends of the conversation. But IP telephony could not grow to its potential relying on proprietary systems. H.323 helps solve that problem.

Some of the best known parts of H.323 are the voice codecs, or voice-compression algorithms. These codecs include G.723.1, which scores high in voice quality but is still stingy with bandwidth: Its two implementations use 5.3 and 6.4 kbps, not counting IP overhead. H.323 also includes G.729A, which supports 8 and 13 kbps. Other H.323 audio codecs include G.711, G.728, G.729, and G.729B. H.323 also includes standards for video compression and many of the other functions that IP telephony and IP videoconferencing require.

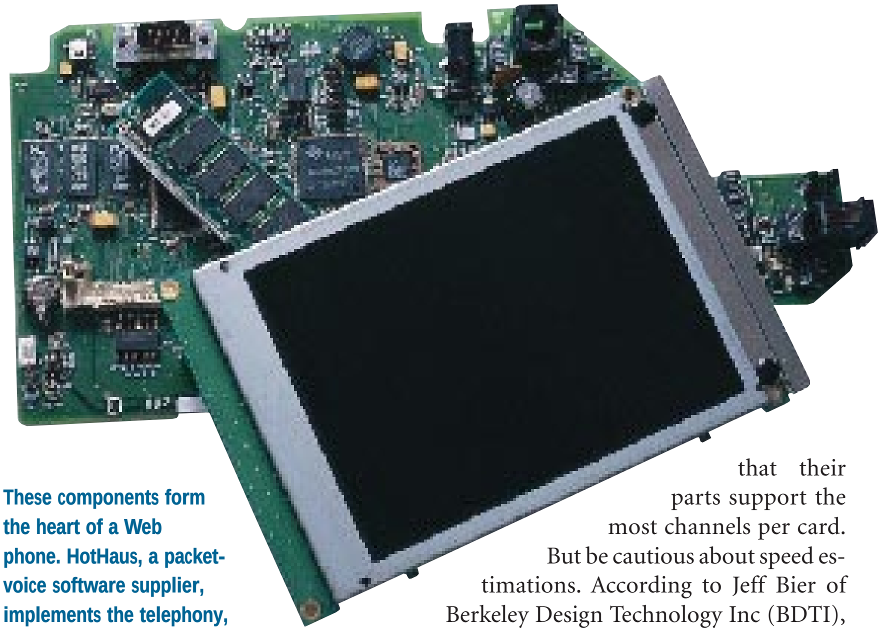
To find out more about standards for Internet telephony, contact the International Telecommunications Union at www.itu.ch, the Enterprise Computer Telephony Forum (ECTF) at www.ectf.org, the Internet Engineering Task Force at www.ietf.org, or the International Multimedia Teleconferencing Consortium at www.imtc.org.

Even in simpler products, such as Internet phones, processing the call stack differs so much from processing voice that separate chips are often required.

HIGH DENSITY: FAST, LITTLE, AND COOL

IP-telephony products fall into two broad categories: infrastructure products, such as voice gateways, which simultaneously handle many channels, and terminals, such as Web phones and videoconferencing systems, which simultaneously handle one or perhaps a few channels. High density is the goal of infrastructure products, and this density drives the main hardware features for VoIP chips: high speed, compactness, and low power. Density is important because an application could have hundreds of telephone lines entering a box that may be a small closet, according to Julie Koelsch, worldwide voice-over-packet business manager for Texas Instruments. High-speed processing supports many connections in one chip, small chips mean that you can fit many onto a card, and low-power requirements reduce heat buildup. All three work together to maximize the number of simultaneous calls a system can handle in a small space.

Another factor driving density is that customers add voice cards to existing equipment, such as routers, and, often, just a few slots may be left. Low power is critical because the power supply is fixed. Koelsch notes that power, size, and speed also drive wireless communication. That



These components form the heart of a Web phone. HotHaus, a packet-voice software supplier, implements the telephony, voice, and modem algorithms for this TI C5000-based terminal.

fact helps to explain how the VoIP market grew so rapidly from the start: The chips for early applications already existed to satisfy wireless needs. The amount of hardware a chip integrates also affects its size. Analog Devices claims that the ADSP21-MOD970 is small but that it looks even 33% smaller when you consider its integrated hardware because the chip also integrates memory.

High-speed fixed-point DSPs will emerge for infrastructure applications. Vendors compete to convince designers that their parts support the most channels per card.

But be cautious about speed estimations. According to Jeff Bier of Berkeley Design Technology Inc (BDTI), which helps manufacturers select DSPs, the most common mistake in rating chip performance is relying on vendor-estimated MIPS.

"MIPS tell you very little about a processor. We have seen a 100-MIPS processor outperform a 200-MIPS processor," says Bier. The reason for this discrepancy is that each architecture takes a different number of instructions to perform a task. For example, compare a RISC processor with a traditional DSP: DSPs with compound instructions often count multiple operations for one instruction, whereas RISC processors count each operation as an instruction.

TABLE 2—HARDWARE COMPARISON

Manufacturer[1]	Circle No.	Part	Architecture	Advertised maximum speed (MIPS)	Integrated SRAM (bits)	Power per chip (W)	Protocol types	Size (in.)	Price[3]
Analog Devices	355	ADSP21-MOD970-110	16-bit DSP	312	960k×8	0.48	Voice, data, fax	1.24×1.24	$34.67 (10,000)
DSP Group	358	CT8020 DA	DSP	40	4k×8	1.14	Voice	1.24×1.24	$14.40 (1000)
		CT8021 AB	DSP	45	6k×8	1.275	Voice	1.24×1.24	$16.90 (1000)
8×8	360	VCPex	RISC and SIMD DSP	1600	2k×32	0.6	Video, audio	0.28×0.28	$40 (100,000)
Motorola	366	56307	DSP with coprocessor	170	64k×24	0.3	Voice, data, fax	0.6×0.6	$47 (10,000)
Texas Instruments	370	TMS320-VC5420	DSP	200	400k×8	0.12	Voice, fax, data[2]	0.48×0.48	$60 (10,000)
		TMS320-C6202	VLIW DSP	2000	384k×8	2	Voice, fax, data	1.08×1.08	$140 (10,000)
ZSP	372	16402	16-bit, superscalar DSP	400	62k×16	1.2	Voice, fax, modem	0.68×0.68	$65 (10,000)

[1]For AudioCodes AC481xx-C, see TI C5000 series. Price is $70 (10,000).
[2]Not all TI third parties support all protocol types.
[3]Prices do not necessarily include software modules.

Texas Instruments' TMS320C6000 processors, often used for voice applications, provide an example of how MIPS is an unreliable measure of performance. TI rates members of the C6000 as high as 2000 MIPS, whereas its predecessor, the C5000, performs at 200 MIPS. BDTI rated these two chips and found that the TMS320-C6201, nominally rated at 1600 MIPS, scored 103 in BDTI's tests, and the TMS320VC549, nominally rated at 100 MIPS, scored 25. The C6201 scored better than the C549, but the difference was not nearly what the vendor ratings would indicate. The reason for this discrepancy is that TI counts four operations—two loads from memory, one multiply, and one accumulate—as one instruction on the C5000 but as four on the C6000.

BECAUSE PARALLELISM FOR VLIW PROCESSORS MUST BE FULLY SPECIFIED, SOME VLIW PROCESSORS CAN BE MORE DIFFICULT TO PROGRAM THAN SUPERSCALAR DSPs.

Bier doesn't disparage the C6000, however. "It's still the fastest fixed-point DSP out there," he says.

Another issue that makes speed comparisons difficult is understanding the impact of superscalar and very-long-instruction-word (VLIW) processors. Both types rely on parallelism to increase speed. VLIW uses multiple functional units and complex programming instructions that carry commands to the functional units. The TI C6000 and the 8×8 VCPex exemplify VLIW processing. Superscalar processing also uses multiple functional units, but the instruction set is dynamically scheduled, meaning that the compiler produces nonparallel but properly sequenced code and that the processor determines the parallelism. High-end desktop-PC processors are superscalar. ZSP claims that its 16402 DSP, which will enter production in the first quarter of 1999, will be the only supercalar DSP in the VoIP market.

Because parallelism for VLIW processors must be fully specified, some VLIW processors can be more difficult to program than

FOR MORE INFORMATION...

For more information on products such as those discussed in this article, circle the appropriate numbers on the Information Retrieval Service card or use *EDN*'s InfoAcess service. When you contact any of the following manufacturers directly, please let them know you read about their products in *EDN*.

Analog Devices
Norwood, MA
1-800-262-5643
www.analog.com
Circle No. 355

AudioCodes Ltd
San Jose, CA
1-408-577-0488
www.audiocodes.com
Circle No. 356

Berkeley Design Technology Inc
Berkeley, CA
1-510-665-1600
www.bdti.com
Circle No. 357

DSP Group Inc
Santa Clara, CA
1-408-986-4300
www.dspg.com
Circle No. 358

DSP Software Engineering
(software for TI DSPs)
Bedford, MA
1-781-275-3733
www.dspse.com
Circle No. 359

8×8 Inc
Santa Clara, CA
1-408-727-1885
www.8x8.com
Circle No. 360

Forward Concepts
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GAO Research & Consulting Ltd
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superscalar DSPs, and it's also harder to write VLIW compilers. And, because the job is tougher, a programmer may not always do it well enough. According to Bier, many people think they will program in C, and it almost never happens. The superscalar architecture can help solve this problem. Mikael Berner, product marketing engineer at ZSP, claims that its C compiler can produce code that is a scant 30% larger than hand-coded assembler. Bier gives high marks to the Analog Devices and Motorola architectures in general ease of programming.

So, how do you tell whether the DSP has enough processing power? Bier recommends that you find benchmarks that are meaningful for the application. The good news is that, because the VoIP market demands that vendors or their third parties provide the most computationally intensive sections of code, benchmarking is easier than in many other industries. Rather than just comparing raw-performance measures, you can compare the speed to execute the tough algorithms.

However, Bier cautions, "You are benchmarking the programming job as well as DSP itself. Variations of 20 to 30% may be insignificant." BDTI provides benchmarking information and related white papers at its Web site, www.bdti.com.

Density is not an issue for terminals such

TABLE 3—AUDIO-CODEC SOFTWARE[1]

Company	Circle No.	Part	Third party	G.711	G.722	G.723/723.1	G.726	G.727	G.728	G.729a/b
Analog Devices	355	ADSP21MOD970	Vocal Technologies	X		X	X	X		X
AudioCodes	356	AC481xxA-C		X		X	X	X	X	X
DSP Group	358	CT8020 DA				X				
		CT8021 AB		X	X	X			X	X
8×8	360	VCPex		X	X	X			X	X
Motorola	366	56307		X	X	X	X		X	X
Texas	370	C5000	HotHaus	X		X	X			X
Instruments		C5/6000	Telogy	X	X	X	X	X		X
ZSP	372	16402		X		X	X		X	X

[1] The chip vendor or vendor-recognized third party provides software.

as Internet phones. Most of today's DSPs can handle a voice connection, so speed is less of a concern. Because only one channel exists, minimizing the power is less of a concern. However, cost is an issue for terminals. Whereas vendors sell voice gateways with large financial margins, consumer products struggle as the vendors try to get users to replace $15 POTS handsets with $200 IP terminals, for example. The DSP Group focuses on the terminal market, providing the processor for the Aplio (www.aplio.com) Web Phone, Quicknet Technologies' (www.quicknet.net) Internet Phone Jack, and Selsius Systems' (www.selsius.com) Selsius-Phone.

The VCPex chip for video and audio applications from 8×8 also addresses the terminal market. The chip combines a 32-bit, 40-MHz RISC chip with a single-instruction multiple-data (SIMD) parallel DSP, which together produce 1600 MIPS. The VLIW VCPex can be challenging to program to achieve that speed. However, 8×8 offers all the software its customers need, including full turnkey video terminals, so ease of programming is less of an issue for some customers.

Beyond power, size, and speed, you must also consider whether the chip includes software from the vendor or a third party. Because voice codecs are notoriously difficult to write and test, writing your own

TABLE 4—FAX-STANDARDS SOFTWARE

Company	Circle No.	Part	Third party	V.17	V.21	V.27/V.27ter	V.29	T.30	T.38
Analog Devices	355	ADSP21MOD970	Vocal Technologies	X	X	X	X	X	X
AudioCodes	356	AC481xxA-C		X	X	X	X	Partial	X
DSP Group	358	CT8020 DA							
		CT8021 AB							
8×8	360	VCPex							
Motorola	366	56307		X	X	X	X	X	X
Texas Instruments	370	C5000	HotHaus	X	X	X	X	X	X
		C5/6000	Telogy	X	X	X	X	X	X
ZSP	372	16402		X	X	X	X	X	X

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software may make little sense. ZSP, 8×8, and DSP Group provide voice-processing software with their chips, and other vendors rely on third parties. Analog Devices provides modem and fax protocols for the ADSP21MOD970 but relies on Vocal Technologies for much of the voice-related software. TI has third-party arrangements with numerous companies, including AudioCodes, Telogy, and HotHaus.

If you decide to write your own software, you need to consider patents. According to the DSP Group's Louis, his company is one of eight independent organizations that own the patent rights to G.723, the most popular audio codec. Louis states that only these organizations and their licensees are entitled to produce products using G.723.

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Software longevity is another factor in processor selection. Some companies are careful to keep future processor families backward compatible. For example, Analog Devices claims the ADSP21MOD970 is backward-compatible with the ADSP-21MOD870. However, TI's C6000 is incompatible with the company's popular C5000. If you want to reuse software in future generations of products, choose chips for which the vendor's plans include backward compatibility.

Ease of programming is a subjective feature that depends on the tools and on the processor structure. Remember that chips belonging to popular families may have better support than those in a limited market. The TI C5000 line is a popular processor family, and TI can leverage revenue from many markets to develop the programming tools. Vendors with narrow markets may be unable to invest as much into tools. If you purchase most or all of the software, ease of programming is a small issue.

Also, consider upgrading in the field. Because IP-telephony standards are evolving, some vendors focus on helping customers build adaptable products. Be sure to evaluate the hardware and supporting software with this feature in mind (**tables 2** through **6**). The key for chip designers is to think about the total solution for the application. The solution includes the chip, the preprogrammed software, the software tools, ease of programming, code longevity, and vendor support. Choose your processor wisely because it's a decision you don't want to make twice. □

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TABLE 5—DATA-MODEM SOFTWARE

Company	Circle No.	Part	Third party	V.22/ V.22bis	V.23	V.32/ V.32bis	V.34	V.90	ISDN
Analog Devices	355	ADSP21-MOD970	Vocal Technologies	X	X	X	X	X	X
AudioCodes	356	AC481xxA-C		X	X	X			
DSP Group	358	CT8020 DA							
		CT8021 AB							
8×8	360								
Motorola	366	56307		X	X	X	X	X	X
Texas Instruments	370	C5000	HotHaus	X	X	X	X		
		C5/6000	Telogy						
ZSP	372	16402		X	X	X			

TABLE 6—OTHER IP-TELEPHONY FUNCTIONS[1]

Company	Circle No.	Part	Third party	Call discrim-ination	VAD/ CNG+	Packetize	H.323 call stack	IP Call stack
Analog Devices	355	ADSP21-MOD970	Vocal Technologies	X	X	X	X	
AudioCodes	356	AC481xxA-C		X	X	X		
DSP Group	358	CT8020 DA		X	X	X		
		CT8021 AB		X	X	X		
8×8 Inc	360			X	X	X	X	X
Motorola	366	56307		X	X	X	X	X
Texas Instruments	370	C5000	HotHaus	X	X	X	X[2]	X
		C5/6000	Telogy	X	X	X		
ZSP	372	16402		X	X	X		

[1]All products support echo-cancellation, jitter-buffer, DTMF-detection-generation, and lost-packet functions.
[2]Telogy provides H.323 call stack with a RadVision partnership (www.radvision.com).