The Evolution of the Remote Access Server (RAS) to a Universal Port-Enabled Platform

Definition

The remote access server (RAS) platform is the foundation on which Internet service providers (ISP) and other telecommunications carriers are bringing to market integrated, Internet-based versions of traditional services such as voice over IP (VoIP), fax over IP (FoIP), and data over IP. As the numbers of Internet users grows and as networking technology advances, these services are being successfully marketed as alternatives to those services offered by traditional providers. In addition to voice and fax for consumers, small offices, and telecommuters, these services include virtual private networks (VPN) for enterprises with many mobile employees or small branch offices.

RASs play a key role in the proliferation of these Internet-based services, which represent a significant expansion for the role of RAS platforms from their early use solely as data connections for individuals. Driven by these compelling market conditions, a new universal port–enabled generation of RAS equipment has emerged that provides the ability to support voice, fax, and modem services over nearly any packet-switched network.

Overview

The following tutorial discusses the evolution of the RAS to a universal port–enabled platform. A discussion of current equipment is complemented with a discussion of the next-generation RAS platform, which includes such technical particulars as channel density, power consumption, and processing capability. The concept of the universal port is presented, including explanation of architecture, implementation challenges, and features such as the DSP–based embedded software, which enables the platform's voice-, fax-, and data-over–IP functionality.

Topics

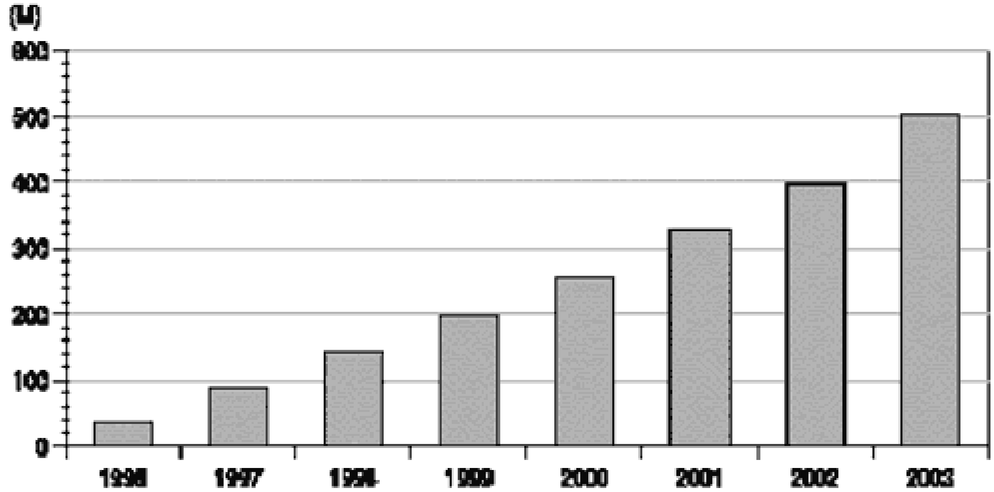
1. Introduction

> "The remote access market is expected to more than double from $3.8 billion in 1998 to $7.7 billion by 2003. Voice and data convergence creates a new category of remote access—VoIP–enabled RAS—and this segment is forecasted to realize at 81-percent compound annual growth rate (CAGR) from 1999 to 2003."
> —Brian Baldwin, International Data Corp.

There is a great deal of talk about high-speed broadband access to the home via technologies such as digital subscriber line (xDSL) and cable modems. The reality is that while these technologies are beginning to be deployed, the vast majority of us still can only obtain plain old telephony service (POTS) dialup modem access for our connection to the Internet and corporate enterprise. Thus, the dialup modem market continues to grow to support ever-increasing Internet access demands.

With the number of people worldwide accessing the Internet from home nearing 100 million and climbing fast, remote access servers (RASs) are likely to enjoy continued strong demand in the years ahead. With the global Internet explosion just dawning in many parts of the world, the figures indicate that hundreds of millions of Internet users will be counting on dialup access well into the future (see *Figure 1*).

Figure 1. Worldwide Internet Users 1996—2003



Projected number of Internet users (measured in millions)
Source: International Data, Corp., 1999

Making these numbers even more compelling for equipment providers is the emergence of VoIP. Service providers have traditionally been focused only on providing dialup modem Internet access to their customers. However, with the convergence of voice and data networks, service providers now desire to add value-added telephony services, using VoIP and fax-relay technologies as a means of generating new sources of revenue. ISPs, who may feel the particular squeeze of commodity-priced Internet service, are likely to exploit voice-over–Internet protocol (VoIP)–enabled RAS platforms to provide revenue-generating telephony services with a minimal infrastructure investment.

These services have grown to include VoIP and fax relay for consumers, home offices, and telecommuters, as well as virtual private networks (VPNs) for enterprises with many mobile employees or small branch offices. This array of services represents a significant expansion for the role of RAS platforms from their early use solely as data connections for individual users. As a result, the RAS equipment market is transitioning from modem-only to universal port (voice, fax, and modem). It should be noted that even when high-speed access services such as xDSL and cable modems become available, these services must still support dialup modem services to the user to emulate traditional POTS services fully. Service providers desire to have one set of RAS equipment that can support voice, fax, and data on a call-by-call basis. In addition, this equipment must be able to scale to new heights with regard to overall channel capacity to meet increasing Internet growth.

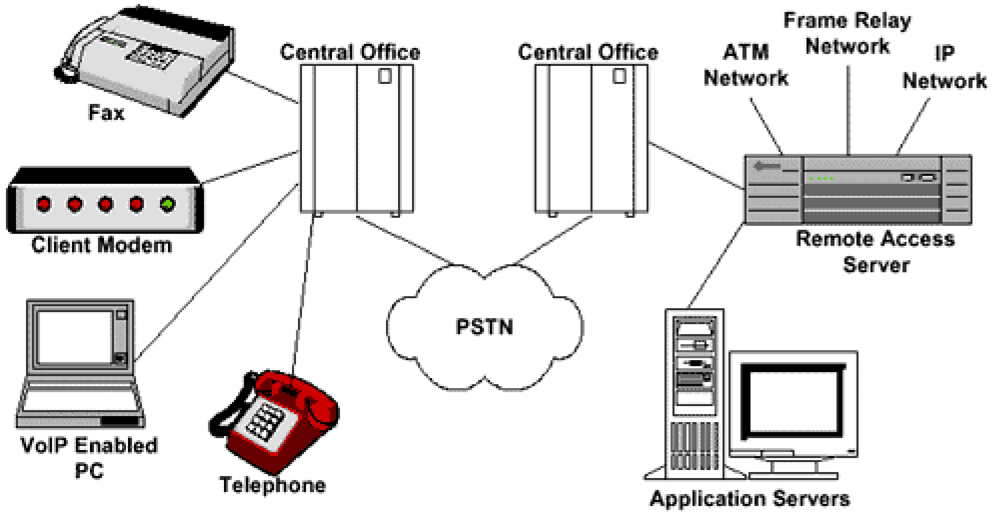
The RAS equipment market has traditionally been divided into two segments: enterprise and infrastructure. To support a remote workforce (e.g., sales offices, telecommuters, and business travelers), corporations install enterprise RAS equipment to allow their employees remote access to the corporate local-area network (LAN). With the ubiquity of the Internet and the growing acceptance of tunneling programs and firewall security, more and more corporations are addressing their remote access needs by allowing their employees to connect to the corporate LAN through the Internet via ISPs with local points of presence (PoPs). This, combined with the proliferation of home Internet access as more and more households get connected to the Internet, has created a growing demand for very high-density infrastructure RAS equipment capable of supporting thousands of connections in a single rack. The increasing scalability of RAS platforms is even enabling many service providers to wholesale Internet access and Internet telephony service to other ISPs. This low-margin, high-volume business is perfectly suited for next-generation universal ports that offer unprecedented price and performance per channel.

Enterprises are also continuing to deploy RAS systems. Though most industry analysts agree that growth in this market sector will not keep pace with growth among service providers, the increased acceptance of telecommuting, combined with the growing use of the Internet overseas by small businesses, means that the enterprise should remain a strong market for the foreseeable future.

To meet these demands, equipment manufacturers must provide next-generation RAS equipment that offers very high-density, low-power, and low-cost-per-channel solutions that are software-defined to facilitate feature updates and to provide strong network management remote debug capabilities. These platform needs are enabled by a new generation of digital signal processors (DSPs) that make it possible to run multiple channels of modem, voice, and fax in the same DSP while significantly reducing the cost and power per port and increasing the overall channel density of the solution.

Major network-equipment providers today offer a wide range of RAS systems, including some that are touted as VoIP–enabled. This equipment is in use by service providers of all sizes as well as enterprise IT departments offering remote access to employees. While technically speaking some of today's RAS platforms support VoIP, the truth is these first-generation VoIP–enabled RAS platforms are highly inefficient implementations lacking important software features and often require separate hardware modules within an equipment rack to support modem, voice, and fax services. This requires a priori allocation of hardware resources based on anticipated service needs. As a result, this equipment does not offer an optimum solution with regard to cost and features offered. *Figure 2* illustrates the use of a remote access server (RAS); also called a remote access concentrator to terminate nearly every type of fax, modem, and voice call, and prepare it for transmission over a packet-switched network.

Figure 2. Remote Access Server (RAS)



There are three significant limitations of current RAS equipment: channel density, power consumption, and processing capability. The combination of these three shortcomings will soon render older RAS equipment useless for many of the newer applications mentioned earlier, including VoIP, fax relay, VPN, and access wholesaling.

Channel Density

With space at a premium inside central offices (COs) and data centers, as well as inside the call-processing equipment itself, channel density is very important. Channel density is essentially the number of call channels that can fit into a rack of equipment.

Underscoring the importance of channel density is the typical carrier equipment-refresh cycle. RAS platform providers develop products on refresh cycles of anywhere from nine to eighteen months, and they expect channel density at least to double during each cycle. The resulting increase in capacity not only enables equipment to scale dramatically, it can lower the cost per channel by as much as fifty percent. Traditional RAS platforms offer several hundred modem channels per rack while next-generation RAS platforms will offer several thousands of universal port channels per rack.

Power Consumption

In a massively scalable environment, operating 24x7x365, small variations in power consumption can have a great impact on service providers. First, power is a utility that must be sourced. The cost is pure overhead that moves straight to the bottom line. Second, power consumption is also directly related to heat.

Sensitive electronic equipment requires a temperature-controlled environment. This could mean even greater utility costs should external cooling measures become necessary. Third, the network equipment building standard (NEBS) is a Telecordia standard required by carriers and service providers that sets limits on power consumption for a variety of safety and operational reasons.

The goal for equipment manufacturers is to offer the most number of channels in a seven-foot rack without exceeding power consumption allocated for the rack. This may be on the order of 1000 to 1500 watts for the entire rack, including common equipment. Most solutions today tend to be power-limited rather than board space–limited. As service providers evaluate next-generation RAS platforms, the power budget will shrink by about 50 percent and is expected eventually to dip down to 10 milliwatts per channel.

Processing Power

DSPs used in first-generation RAS modem implementations could only handle a single modem channel each. Density improvements were obtained by implementing multiple chip modules (MCMs), which placed multiple DSPs in a single chip, thereby increasing channel density. Newer standards and universal port features place additional demands on processing power per channel as well as program and data memory per channel. To meet the density and power objectives, more powerful and efficient DSPs are required.

2. Universal Port Overview

Universal port enables equipment providers to overcome the shortcomings of traditional RAS systems to meet the demanding requirements of ISPs, telecom carriers, and Internet telephony service providers (ITSPs). Universal port–enabled RAS equipment will support voice, fax, and modem services over packet-switched networks such as Ethernet LANs, asynchronous transfer mode (ATM) backbones, and frame-relay wide-area networks (WANs). The capabilities of universal port include the following:

- modem termination
- modem relay
- fax termination
- fax relay
- voice over packet
- signaling

- network management

Modem

Modem termination software supports dialup modem connections such as V.90, V.34bis, and various fallback rates. The modem software demodulates pulse code modulation (PCM) samples received, performs appropriate error correction and data decompression, and passes packets to the packet network. Similarly, the software processes packets received from the packet network, performing data compression, error correction, and modulation to the PCM interface. Modem termination consists of the following functionality:

Modem Pumps

- V.90 server: (28,000 bps to 56,000 bps)
- V.34/V.34+ (4,800 bps to 33,600 bps)
- V.32/V.32bis (4,800 bps to 14,400 bps)
- V.22/V.22bis (1200 bps, 2400 bps)
- Bell 212A (1200 bps)
- V.23 (75/1200 bps)
- V.21 (300 bps)
- Bell 103 (300 bps)

Modem Controller Functions

- error correction: V.42, microcom networking protocol (MNP) 2-4
- data compression: V.42bis, MNP 5
- AT command processing
- encapsulation: slip, PPP

Modem relay is employed for applications in which it is desirable to provide transparent POTS across a packet network. Two examples of modem relay include using packet networks to replace digital circuit multiplication equipment (DCME) and transporting dialup modem data through a cable modem to preserve POTS transparency to users (e.g., a household visitor using a laptop to dial into work).

Modem relay can be handled by detecting the presence of a modem and packetizing and transporting the entire 64–kbps data stream to the remote end of the packet network for retransmission using G.711 PCM. However, this is often undesirable due to the overhead incurred. Also, when G.711 coding and packetization are used for data-modem transmissions, the two modems are essentially communicating to each other and are subject to the effects of packet loss in the network. Lost data packets can cause the modems to retrain (effecting throughput) and can even result in call failures.

Furthermore, the PCM codecs do not synchronize the timing between the two endpoints, which can result in clock slips between the two modems. Modem relay mitigates the problems associated with using G.711 to send modem traffic. In modem relay, the physical layer of the modem signal is terminated locally for both ends of the call. Only the data stream is sent over the network. This leads to dramatic bandwidth savings. The modem data stream can also be transported in a redundant fashion, which allows for seamless error recovery in the event of packet loss. Currently, there are no interoperability standards for modem relay.

Fax

Fax termination supports analog Group-3 facsimile (fax) machines. Fax termination is used for fax server dial-in and dial-out services over the corporate LAN, whereby users can send and receive faxes with their personal computers (PCs). Fax termination can also be used to implement store-and-forward fax services as well as fax-to–e-mail services. Fax termination consists of the following functions:

- fax modem pumps: V.17, V.29, V.27ter, V.21
- fax termination protocols: T.30, AT commands

Fax relay provides reliable real-time fax service between two analog fax machines over a packet network. The RAS equipment at both ends of the packet network spoofs the analog fax machines such that they operate as if directly connected over a public switched telephone network (PSTN) connection. The RAS equipment performing fax relay functions must handle the effects of network delay, jitter (variable delay), and lost packets while preventing the fax machines from timing out.

Standards protocols such as T.38, FRF.11, and AAL2 exist for interoperability between equipment vendors. Proprietary techniques are often used to improve the interoperability between different fax machines that are subjected to long delay and other packet-network effects. Fax relay consists of the following functions:

- fax modem pumps: V.17, V.29, V.27ter, V.21
- fax relay protocols: T.38 [TCP/IP (transmission control protocol/Internet protocol)], AAL2 (ATM), FRF.11 (frame relay)
- fax machine spoofing protocols: proprietary

Voice

Voice-over-packet (VoP) software prepares voice samples for transmission over the packet network. There are many functions that comprise good-quality VoP software. Its subcomponents perform echo cancellation, voice compression (to conserve bandwidth), voice-activity detection, jitter removal, clock synchronization, and voice packetization. VoP consists of the following functions:

- voice codec support, including G.711 (PCM), G.722, G.723.1, G.726 (ADPCM), G.728, G.729AB, G.729E, GSM, etc.
- line echo cancellation: G.165/G.168
- voice activity detection (VAD)
- comfort noise generation (CNG)
- packet playout: delay, jitter, and lost packet compensation
- in-band tone detection and generation
- packet encapsulation: RTP (TCP/IP), FRF.11 (frame relay), AAL2 (ATM/DSL)
- value-added features such as conferencing, encryption, etc.

Signaling

RAS equipment must support signaling for call establishment, in-band signaling, and call termination. Both channel associated signaling (CAS) and common channel signaling (CCS) are employed by networks and must be supported. This includes the following:

- tone detection and generation, e.g., dual-tone multifrequency (DTMF), MF, call progress, etc.

- caller ID detection and generation
- message-based signaling support, e.g., transparent, high-level data link control (HDLC), integrated services digital network (ISDN), etc.

Network Management

Fundamental to any communications system is the ability to discover, isolate, and remedy problems as quickly as possible to minimize or eliminate the degree to which users are impacted.

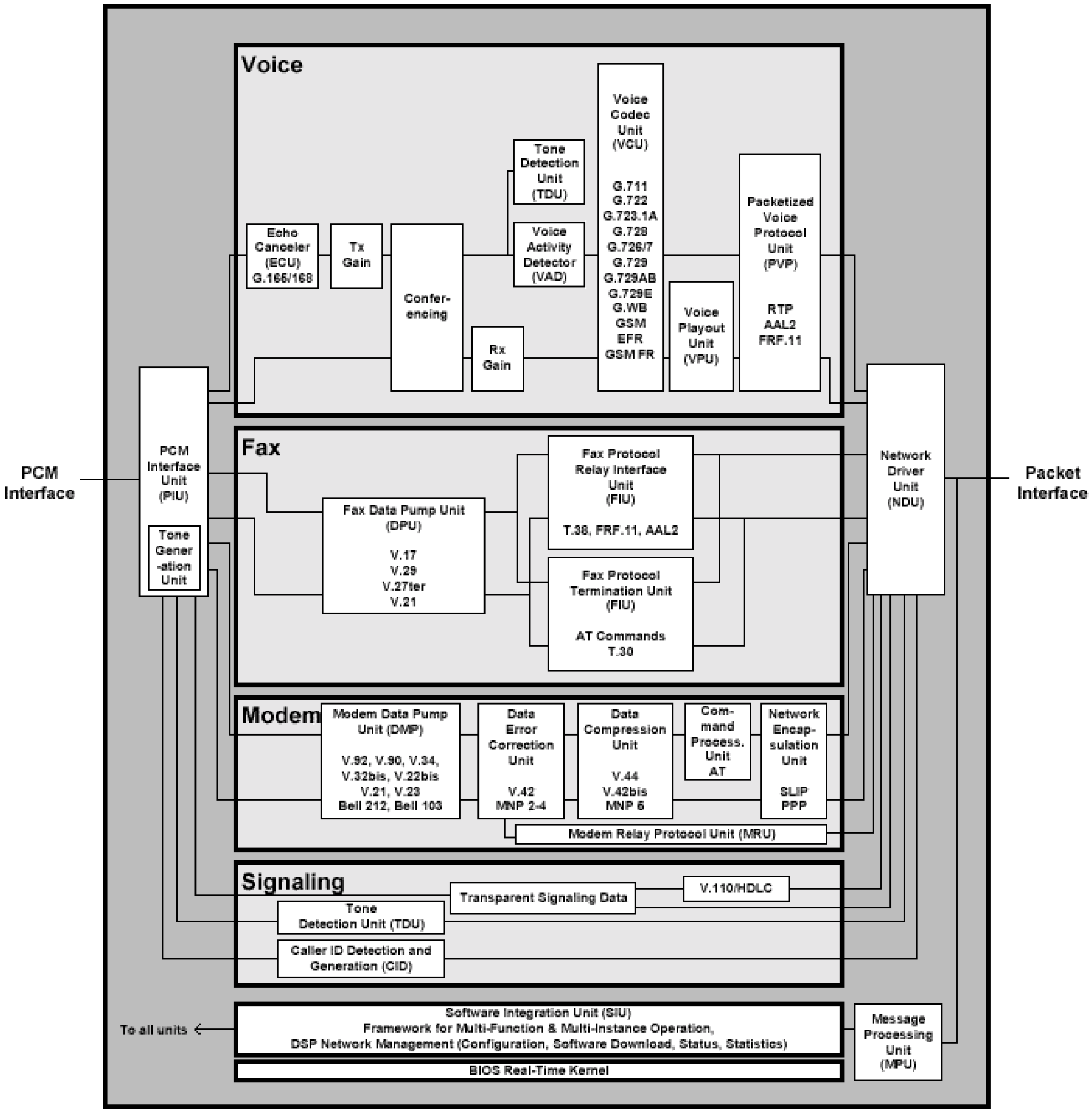
- configuration on per-channel basis, including set-able, country code-specific information
- per-channel statistics and status reporting
- per-channel, real-time trace and diagnostics

3. Universal Port Architecture

Figure 3 shows the software architecture for a universal port solution. Each box represents a software component required to implement the features for voice, fax, modem, signaling, and network management functions of the RAS. Multiple instances of each software component can exist to facilitate support of concurrent, multichannel operation. Each instance shares common program memory and has unique channel-specific data memory to maintain information regarding the state of the channel, including network management and diagnostic information.

Voice, fax, and modem have different characteristics with regard to processing (millions of instructions per second [MIPS]), delay, program, and data memory. In order for the software to operate concurrently on the same DSP, a highly efficient, real-time operating system kernel and a solid application framework are required to orchestrate the software.

Figure 3. RAS Universal Port Software Architecture



4. Universal Port Implementation Challenges

Modem

It is important for an RAS platform to support not only the latest International Telecommunications Union (ITU) standards but also older standards as a result of the wide array of legacy modems and applications that adhere to these older standards. For example, credit-card authorizations for point-of-sales (PoS) transactions use low-speed modems (e.g., V.21 and V.22). These older modems cost very little to implement in PoS terminals. Moreover, the time required to establish a connection is much shorter than with a V.90 modem because the modulation scheme is much simpler. Because credit-card authorization requires little information to be exchanged (e.g., credit-card number and the purchase amount of transaction), the lower throughput of the modem connection is not an issue. It is the call holding time of each credit-card transaction that is more important.

Considering that there are literally hundreds of brands and models of modems in use, interoperability and performance testing, often referred to as hardening, is absolutely essential to provide an effective RAS platform. The hardening process consists of functional testing, stress testing, soak and stability testing, and performance testing, including extensive live-line testing in the field. Key performance metrics for modems include connection success rate and throughput versus different network models representing real-world conditions and modem interoperability.

There are many poorly implemented dialup modems in use around the world. These modems often violate ITU standards and contain software bugs that cause misbehavior. New soft modems that use the PC's main processor to perform modem functions are notorious for bad behavior, as a result of interactions between the modem software and other applications running on the PC.

RAS equipment manufacturers do not have the luxury of telling a service provider that their customers must replace these bad modems. Instead, the RAS equipment manufacturer must develop modem software that is forgiving and can connect successfully with all modems. Thus, it is important for the modem software to be tested continually with hundreds of different modems to ensure interoperability under different line conditions. It is also important for the modem software to support built-in remote trace and debug capabilities to diagnose problems when they occur, both for development and deployment of the actual system.

Fax

Fax software hardening is similar to dialup modem hardening. Interoperability with bad fax modems and machines is a requirement. For fax relay, it is important to note that fax machines were designed for the analog PSTN and do not handle the delay and jitter conditions of packet networks very well. Therefore, the fax relay software must perform additional processing to overcome the effects of packet networks.

Delay

The delay of fax packets through a packet network causes the precise timing that is required for the fax protocol to be skewed and can result in a dropped call. Although the fax-relay protocols are able to compensate for this skewed timing to help ensure that calls are carried to completion, additional processing steps are required to reduce the risk even further. One technique is called spoofing. It emulates a continual conversation between the sender and receiver, ensuring that neither fax machine drops a call inadvertently.

Lost Packets

Lost packets can be a more serious problem for fax relay than for VoP. In a voice conversation, lost packets can be addressed by replaying last packets and by other methods of interpolation. Fax relay conversations, however, have more severe constraints on the loss of data because the fax protocol can fail if information is lost. The severity of the problem varies depending on the type of fax machine and whether error correction mode is enabled.

The fax-relay software must be tested over various network conditions, including delay, jitter, and lost packets, to make sure that the protocol software is robust and that the proprietary fax machine spoofing techniques are able to ensure successful connection between fax machines. These techniques must also provide successful page transfer (including multiple pages) and good-quality images at the receiving fax machine. It is also important to conduct interoperability testing with other vendors' implementation of the fax-relay protocol standards (e.g., T.38).

Voice

Latency

Latency causes two problems: echo and talker overlap. Echo is caused by the signal reflections of the speaker’s voice. Echo becomes a significant problem when delay is greater than 50 milliseconds. Because echo is a major quality problem, RAS equipment providers must implement echo cancellation for VoP solutions. Talker overlap becomes significant if one-way delay is greater than 250 milliseconds, so every effort must be made to minimize delay. The sources of delay in a packet network include the following:

Accumulation Delay (Also Called Algorithmic Delay)

This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. It varies from a single sample time (.125

microseconds) to many milliseconds. Standard voice coders (and their frame times) include the following:

- G.728 LD–code excited linear prediction (CELP)—2.5 milliseconds
- G.729 CS–ACELP—10 milliseconds

Processing Delay

This delay is caused by the actual process of encoding and collecting samples into a packet for transmission. The encoding delay is a function of both the processor execution time and the type of algorithm used. Often, multiple voice-coder frames will be collected in a single packet to reduce overhead. This is useful when the different voice conversations go to the same destination, as can be the case for a trunking application. For example, three frames of G.729 compressed speech, equaling 30 milliseconds of speech, may be collected and packed into a single packet. Thus, the protocol overhead is reduced by two-thirds, as one packet containing three voice frames is sent rather than three packets of one frame each. This process of encapsulating several small packets into a single larger frame is called concatenation.

Network Delay

Network delay is a function of the processing that occurs as packets are sent across a network. This delay is caused by the transmission time across the physical medium, the accumulated delays introduced by the networking equipment (routers and switches) used to transmit the voice data, and the buffers used to remove packet jitter on the receive side. These jitter buffers are used to remove the variable time between packets (jitter) created by the varying times at which each packet arrives. This delay can be a significant part of the overall delay, as it can be as high as 70 to 100 milliseconds.

Echo

One of the keys to good VoP quality is having a hardened line echo canceller that can properly cancel echo. Echo is present even in a conventional POTS network. However, it is acceptable because delay is less than 50 milliseconds, and the echo is masked by the normal side tone that every telephone generates. PSTN standards require echo cancellation if the delays exceed 50 milliseconds. Echo becomes a problem when packet networks and the PSTN are used together, because the delay in packet networks is often greater than 50 milliseconds. Due to the network topography, however, the PSTN echo cancellers do not perceive the delays from the packet network, so they do not attempt to perform echo cancellation. Thus, echo-cancellation techniques within the packet domain must be used. The ITU defines performance requirements that are required for echo cancellers. The original standard for echo cancellers was ITU Recommendation G.165. A more stringent set of requirements is provided in ITU Recommendation G.168. These standards provide a series of objective performance tests but do not describe how to implement an echo canceller, nor do they address the subjective performance of an echo canceller. A good echo canceller must have the following attributes:

- **removes echo well**—This includes removing echo at the start of a call as well as preventing any form of echo during a call.
- **handles doubletalk well**—Doubletalk occurs when both sides talk simultaneously. It can be handled by not clipping the voice at the beginning or end of a doubletalk voice spurt.
- **handles background noise well**—This includes handling high background noise and variable background noise.

Echo in the form of speech that is reflected back to the speaker is generated toward the packet network from the remote PSTN. The echo canceller compares the voice data received from the PSTN with voice data being transmitted to the packet network. The echo from the telephone network is removed by a digital filter on the transmit path into the packet network.

Jitter

The delay problem is compounded by the need to remove jitter (i.e., variable interpacket timing caused by the fact that packets do not all cross the network at the same speed). Removing jitter requires collecting packets and holding them long enough to recreate the original speech without annoying gaps in the audio. Obviously, holding the packets causes additional delay, but the quality of the speech is dramatically enhanced. The conflicting goals of minimizing delay and removing jitter have led to various schemes aimed at dynamically adapting the size of the jitter buffer to minimize its impact on latency.

Lost Packets

QoS techniques are evolving that will give priority treatment to voice in a packet network, but until these enhancements are widely deployed in the network switches and routers, voice packets will be treated exactly like data. Under peak loads and congestion, voice packets may be dropped just like data packets. The data packets, however, are not time-sensitive, and dropped packets can be recovered through a retransmission protocol such as TCP. Lost voice packets are not handled in the same manner as data because voice retransmission causes unnecessary delay. Some techniques used by VoP software to address the lost-packet problem are as follows:

Interpolation

Interpolate for lost speech packets by replaying the last packet received during the interval when the lost packet was supposed to be played out. This technique is a simple method that fills the time between noncontiguous speech frames. The time interval that is being filled is normally so small that the ear does not perceive that the information has been repeated. It works well when the incidence of lost frames is infrequent. It does not work very well if there are a number of consecutive lost packets or a burst of lost packets.

Redundancy

Send redundant information at the expense of bandwidth utilization. The basic approach replicates and sends the nth packet along with the $(n+1)$th packet. This method has the advantage of being able to correct for the lost packet exactly. However, this approach uses more bandwidth and also creates greater delay.

Voice Coder

A hybrid approach uses a lower-bit-rate voice codec to provide redundant information carried along in the $(n+1)$th packet. This reduces the problem of bandwidth consumption but does not solve the problem of delay. In addition, the voice quality may be negatively impacted by the lower resolution of the low bit-rate codec and may introduce compatibility issues if a nonstandard voice codec is used.

Indemnification

Often overlooked amidst the many technical challenges and considerations of voice, fax, and modem is the issue of patent infringement. While many people may assume this is a minor issue or simply a matter for lawyers to address, the fact is that this is a very significant business consideration for equipment providers in the IP market. There are literally thousands of patents that cover the implementation of hardware, protocols, software, and other technologies associated with packet-based communications.

It is not an exaggeration to say that it is nearly impossible to deliver a complete universal port system without confronting this issue at some point. Depending on the particular patents and the parties involved, the cost could be high enough to force a company to scrap its product plans altogether. Increasingly, equipment providers are looking to silicon and software developers to provide indemnification against patent issues that may arise regarding technology implementation. In a world in which litigation is commonplace, this is a harsh reality for technology firms that may never have faced such issues in the past.

5. Selecting an RAS Solution Partner

Another trend is the realization by equipment manufacturers that it is much harder for them to differentiate by developing the physical link layer and controller software that is required to perform the modem, fax, and voice functions required for universal port. With the widespread adaptation of V.90 ITU standard, equipment manufacturers can no longer justify the engineering costs to develop, test, and maintain this software and are instead looking to add differentiation elsewhere in the system. These manufacturers are looking for silicon vendors to provide a complete solution of silicon and hardened software along with indemnification. The silicon must be an open platform with a compelling product road map to ensure that future products will meet the ever-growing demands of the service providers. This section identifies the key considerations in selecting an RAS solution partner.

Architecture Efficiency

From a silicon perspective, the overall architecture efficiency of the solution is essential. The big three evaluation criteria from a customer perspective are as follows:

- **power per channel**—typically the most critical technology constraint for very-high-density applications
- **channel density**—e.g., square inches or millimeters per channel
- **cost per channel**—always an important issue

Availability of the Solution

Customers have been burned by new silicon and are typically not willing to design next-generation production on chips that exist on paper or appear too far in the future, because the credibility of the chip production schedule may be questionable. Samples must also be available in a timely fashion to commence board design and perform system integration test.

Availability of Quality Software

Proven software solutions must be available to meet time-to-market needs. There must also be a commitment from the solution provider to continue to add value to the software by implementing new features. The solution provider must also have a strong and dedicated customer-support infrastructure to provide support during the development, deployment, and product maintenance phases.

Flexible and Open Architecture

The silicon must be part of an overall road map that provides flexibility to the equipment manufacturer based on product needs (e.g., the ability to offer high-density, modem-only solutions as well as universal port solutions). An open silicon architecture is highly desirable to allow the customer to provide differentiated features and to make use of third-party software to speed time to market.

The software must be modular to allow different combinations of features based on customer needs. The software should also have built-in test/remote debug capabilities.

Availability of Good Development Tools

Due to the sheer wealth of new features fueled by universal port, it is important for the software to be written and maintained in a high-level language such as C. This requires the availability of good development tools, including an efficient C compiler, linker/locator, and debugger.

Product Road Map

The solution provider must provide a good silicon and software road map to allow equipment manufacturers to preserve their investments going forward. The road map should address processing performance, power consumption, cost, external interfaces, different memory configuration, and software features over time. The solution provider should be committed to supply future standards (e.g., V.92 and V.44) and provide headroom in the architecture to add as future software upgrades.

Indemnification

The solution provider must have a strong patent portfolio to provide customers with indemnification against patent issues that may arise regarding technology implementation.

6. Summary

The convergence of voice and data has resulted in a new generation of RAS equipment that provides universal port capabilities. The increasing demand for Internet access requires RAS equipment manufacturers to increase the density of their equipment dramatically. This is being facilitated by a new generation of DSPs with integrated voice-, fax-, and data-over-packet software that offer unprecedented processing power while providing very-low-power consumption.

The availability of field-proven, hardened voice-, fax-, and data-over-packet software enables equipment manufacturers to focus on higher levels of product differentiation.

7. References

1	*ITU–T Recommendation V.90, 1998*—a digital modem and analog modem pair for use on the PSTN at data signaling rates of up to 56,000 bps downstream and up to 33,600 bps upstream
2	*TIA Standard Draft North American Telephone Network Transmission Model for Evaluating PCM (V.90)* modem performance, PN 3857 Draft 4
3	*ITU–T Recommendation G.711, 1988*—PCM of voice frequencies
4	*ITU–T Recommendation V.8, 1998*—procedures for starting sessions of data transmission over the PSTN
5	*ITU–T Recommendation V.8bis, 1996*—procedures for the identification and selection of common modes of operation between data communications equipment (DCE) and between data terminal equipment (DTE) over the general switched telephone network and on leased point-to-point, telephone-type circuits
6	*ITU–T Recommendation V.34, 1996*—a modem operating at data signaling rates of up to 33,600 bps for use on the general switched telephone network and on leased point-to-point, two-wire, telephone-type circuits
7	*ITU–T Recommendation V.56bis, 1995*—network transmission model for evaluating modem performance over two-wire, voice-grade connections.
8	*ITU–T Recommendation V.32bis, 1991*—a duplex modem operating at data signaling rates of up to 14,400 bps for use on the general switched telephone network and on leased point-to-point, two-wire, telephone-type circuits
9	*ITU–T Recommendation V.32, 1993*—a family of two-wire, duplex modems operating at data signaling rates of up to 9,600 bps for use on the general switched telephone network and on leased telephone-type circuits
10	*ITU–T Recommendation V.22bis, 1988*—2,400 bps duplex modem using the frequency division technique standardized for use on the general switched telephone network and on point-to-point, two-wire, leased telephone-type circuits

11	*ITU–T Recommendation V.21, 1988*—300 bps duplex modem standardized for use in the general switched telephone network and on point-to-point, two-wire leased telephone-type circuits
12	*Bell 103* modem specification
13	*Bell 212* modem specification
14	*ITU–T Recommendation V.42, 1996*—error-correction procedures for DCEs using asynchronous-to-synchronous conversion
15	*ITU–T Recommendation V.42bis, 1990*—data compression procedures for DCE using error correction procedures
16	*ITU–T Recommendation V.25, 1996*—automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network, including procedures for disabling of echo-control devices for both manually and automatically established calls
17	*TIA PN 3857 Draft 4*—North American Telephone Network Transmission Model for evaluating PCM (V.90) modem performance, March 1998
18	*TSB 37–A, 1994*—Telephone Network Transmission Model for evaluating modem performance
19	*EIA/TIA TSB 38, 1994*—test procedure for evaluation of two-wire, 4-kHz, voiceband duplex modems
20	*ITU–T Recommendation G.722, November 1988*—7-kHz audio-coding within 64 kbps
21	*ITU–T Recommendation G.723.1, March 1996*—dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbps
22	*ITU–T Recommendation G.726, December 1990*—40, 32, 24, 16 kbps adaptive differential pulse code modulation (ADPCM)
23	*ITU–T Recommendation G.728, September 1992*—coding of speech at 16 kbps using low-delay, code-excited linear prediction.
24	*ITU–T Recommendation G.729, March 1996*—C source code and test vectors for implementation verification of the G.729 8 kbps CS–ACELP speech coder
25	*ITU–T Recommendation G.165, March 1993*—echo cancellers

26	*ITU–T Recommendation G.168, April 1997*—digital network echo cancellers
27	*ITU–T Recommendation T.30, July 1997*—procedures for document facsimile transmission in the general switched telephone network
28	*ITU–T Recommendation T.38, June 1998*—procedures for real-time, Group-3 facsimile communication over IP networks
29	*ITU–T Recommendation V.110, October 1996*—support by an ISDN of data terminal equipment with V-series-type interfaces

8. Universal Port Software Features

- physical layer analog modem protocols
  - ITU V.23 at 75/1,200 bps
  - Bell 103 at 300 bps
  - ITU V.21 at 300 bps
  - ITU V.22 at 1,200 bps
  - Bell 212A at 1,200 bps
  - ITU V.22bis at 2,400 bps
  - ITU V.32 4,800 to 9,600 bps
  - ITU V.32bis 4,800 to 14,400 bps
  - ITU V.34 2,400 to 28,800 bps
  - ITU V.34+ 2,400 to 33,600 bps
  - K56flex server 32,000 to 56,000 bps
  - ITU V.90 server 28,000 to 56,000
  - ITU V.91 server to server 28,000 to 64,000 bps
  - ITU V.92 server (future ITU standard)
  - V.2 power level over POTS

- V.25 auto call/answer
- V.8 auto startup procedure
- V.8bis auto startup procedure
- V.32bis Annex A startup procedure
- near-end echo cancellation
- modem pass-through (PCM)
- link layer and compression protocols
- V.14 asynchronous
- V.42 LAPM
- V.42bis (minimum 2K dictionary)
- MNP 2-4
- MNP 5
- modem relay
- V.44 (LZJH) compression (future ITU standard)

- command interface
  - AT command set with dial modifiers support (for modem/fax/ISDN dialout, reverse telnet, configuration, chat scripts)
  - DSP-based, in-band dial-abort and escape character detection with API support for separate notification messages
  - Hayes escape sequence
- fax protocols
  - Group III Class II
  - V.17 at 7,200; 9,600; 12,000; and 14,000 bps
  - V.27ter at 2,400 and 4,800 bps
  - V.29 at 7,200 and 9,600 bps

- T.30 fax termination (originate/answer)
- fax pass-through (PCM)
- T.38 real-time fax relay
- fax relay on FRF.11
- fax relay on AAL2
- in-band signaling
- DTMF generation/detection
- MF generation/detection
- CP tone detection for T1 CAS
- SS7 ring back tone generation
- SS7 COT
- SR-2275 Type 100 test line (DS-0 quiet test)
- SR-2275 Type 102 test line (DS-0 1kHz/1004Hz 0dBm0 test tone)
- SR-2275 Type 108 test line (DS-0 loop back)
- R2 signaling support (compelled, noncompelled, semicompelled)
- R1 signaling support (noncompelled, semicompelled)

- encapsulation
  - asynchronous PPP
  - SLIP
- digital carrier services, ISDN
  - synchronous-mode PPP (ISDN) (HDLC framed PPP)
  - HDLC framing service (required for V.120 host offloading)
  - V.110
  - 56K rate adaptation

- voice carrier services
    - G.711 PCM 64 kbps
    - G.722 wideband codec 48, 56, and 64 kbps
    - G.723.1 5.3/6.3 kbps
    - G.723.1 Annex A (silence compression)
    - G.726 ADPCM 16, 24, 32, 40 kbps
    - G.727 ADPCM
    - G.728 16 kbps
    - G.729 8 kbps
    - G.729 AB 8 kbps (Annex A and B-VAD, CNG)
    - G.729 Annex D 6.4 kbps
    - G.729 Annex E 11.8 kbps
    - GSM full-rate codec
    - GSM enhanced full rate
    - G.165/G.168 echo cancellation
    - packet playout unit (dejitter buffer, lost-packet compensation)
    - voice activity detection (VAD) silence suppression
    - comfort noise generation (CNG)
    - comfort noise level control
    - AAL2 ATM framing for voice
    - FRF.11 frame relay framing for voice
    - RTP packet encapsulation for voice
    - autoswitch to fax store-and-forward upon fax detection
    - autoswitch from G.7xx to G.711 upon fax detection
    - DTMF relay

- DTMF detection during voice mode
- configurable call progress detection parameters
- configurable voice packetization rates

- management services/other
  - API support (management, event monitoring/reporting, statistics)
  - set-able country code (per channel at run-time)
  - loopback test capabilities
  - core dump facility
  - local eye pattern support
  - remote eye pattern support
  - memory read/write support
  - trace messages

Self-Test

1. The RAS equipment market is transitioning from modem-only to universal port.
   a. true
   b. false

2. Which of the following is not a significant limitation of current RAS equipment?
   a. channel density
   b. physical distance
   c. power consumption
   d. processing power

3. Older RAS equipment will soon be rendered useless for VoIP.
   a. true

b. false

4. Interoperability standards for modem relay are now in place.

    a. true

    b. false

5. Hardening refers to ____________________.

    a. managing the network

    b. implementing the universal port

    c. testing for interoperability and performance

    d. delaying the transmission

6. Fax machines handle the delay and jitter of packet networks.

    a. true

    b. false

7. Echo becomes a significant problem when delay is greater than _________ milliseconds.

    a. 30

    b. 40

    c. 50

    d. 60

8. ________________ is a function of the processing that occurs as packets are sent across a network.

    a. network delay

    b. processing delay

    c. accumulation delay

    d. algorithmic delay

9. ______________ is variable interpacket timing caused by the fact that packets do not all cross the network at the same speed.

   a. latency

   b. echo

   c. doubletalk

   d. jitter

10. RAS silicon software must ___________________________________.

   a. be modular to allow different feature combinations

   b. have built-in test/remote debug capabilities

   c. be part of an overall road map that provides flexibility to the equipment manufacturers

   d. all of the above

Correct Answers

1. The RAS equipment market is transitioning from modem-only to universal port.

   a. true

   b. false

   See Topic 1.

2. Which of the following is not a significant limitation of current RAS equipment?

   a. channel density

   b. physical distance

   c. power consumption

   d. processing power

   See Topic 1.

3. Older RAS equipment will soon be rendered useless for VoIP.

    a. true

    b. false

    See Topic 1.

4. Interoperability standards for modem relay are now in place.

    a. true

    b. false

    See Topic 2.

5. Hardening refers to ____________________.

    a. managing the network

    b. implementing the universal port

    c. testing for interoperability and performance

    d. delaying the transmission

    See Topic 4.

6. Fax machines handle the delay and jitter of packet networks.

    a. true

    b. false

    See Topic 4.

7. Echo becomes a significant problem when delay is greater than _________ milliseconds.

    a. 30

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    d. 60

    See Topic 4.

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    See Topic 4.

10. RAS silicon software must ___________________________________.

    a. be modular to allow different feature combinations

    b. have built-in test/remote debug capabilities

    c. be part of an overall road map that provides flexibility to the equipment manufacturers

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    See Topic 5.

Glossary

ADPCM
adaptive differential pulse code modulation

CAGR
compound annual growth rate

CAS
channel associated signaling

CCS
common channel signaling

CO
central office

DCE
data communications environment

DCME
digital circuit multiplication equipment

DSL
digital subscriber line

DSP
digital signal processor

DTE
data terminal equipment

DTMF
dual-tone multifrequency

HDLC
high-level data link control

IP
Internet protocol

ISDN
integrated services digital network

ISP
Internet service provider

ITSP
Internet telephony service provider

ITU
International Telecommunications Union

LAN
local-area network

MCM
multiple chip module

MIPS
millions of instructions per second

MNP
microcom networking protocol

NEBS
network equipment building standard

PC
personal computer

PCM
pulse code modulation

PoP
point of presence

PoS
point of sales

POTS
plain old telephone service

PSTN
public switched telephone network

RAS
remote access server

TCP
transmission control protocol

VoIP
voice over Internet protocol

VoP
voice over packet

VPN
virtual private network

WAN
wide-area network