Carrier-Class, High Density Voice over Packet (VoP) Gateways

Overview

In phone-to-phone Internet protocol (IP) telephony systems—ranging from customer premises systems up to high density carrier-class systems—*gateways* handle the critical function of digitizing the speaker's voice. The growth in the voice/fax over packet (V/FoP) gateway market has led to the emergence of three distinct segments: customer premises equipment (CPE), small and medium enterprise equipment (SME), and high density carrier-class equipment. Carrier-class high density VoP gateway solutions are specifically intended for service providers deploying packet-based networks. This tutorial will focus on this new generation of high density platforms that will enable service providers to deploy a common packet-based infrastructure for voice and data services.

Topics

1. Introduction

Packet-based voice (VoP) has quickly evolved to enable a single network to deliver a new generation of business and consumer voice and data services. Carriers have announced that new network buildout will be based on packet technology that supports a converged network that carries both voice and data. According to Probe Research, the VoP market is expected to grow from 7.7 billion minutes in 2000 to 570 billion minutes by 2005.

Packet-based voice provides several advantages over circuit-switched voice. Bandwidth required for a voice call can be significantly reduced through voice activity detection (VAD) techniques and use of low bit-rate CODECs. VAD removes silence, which accounts for as much as 40 percent of the voice information that is transmitted. Low bit-rate CODECs reduce the amount of bandwidth for a voice call from 64 kbps to as little as 8 kbps.

Based on open standards, a packet-based voice and data infrastructure allows faster time to market for new services and enhancements with third-party developers offering products to service providers. Capital investment for packet-based platforms is significantly less than for circuit-switched equivalents. The savings in operational costs have been estimated at 40–50 percent by carriers based on the consolidation of platforms and network management applications.

In a few short years, packet-based voice has evolved from a technology demonstration to an integral part of next-generation networks and services. The accelerated pace at which this technology has evolved can be attributed to some key factors:

- VoP technology feasibility
- Digital signal processor (DSP)–based technology with low power consumption and scalable channel density
- Enabling transport technologies, e.g., digital subscriber line (DSL), cable
- Interoperability and standards

Early VoP applications were targeted at providing lower-cost long-distance services. One early application was transporting long-distance voice traffic between certain cities that traditionally experienced a high volume of traffic, for example, between Hong Kong and Vancouver. A second early application was carrying voice traffic between corporate facilities on their existing data networks, significantly reducing the voice traffic expenses.

It is clear that a wide range of consumer and business services will be available utilizing packet-based voice as part of bundled voice and data service offerings. VoP gateways are being implemented for a number of applications ranging from small two-port integrated access devices (IADs) for the consumer and telecommuter to very large carrier-class gateways used as Class-5 switch replacements.

Many technical barriers have been overcome to bring VoP technology to its current state. Large-scale VoP networks have become increasingly feasible as DSP–based technology has been refined to provide lower power and higher densities with manufacturers meeting the challenge of interoperability. Silicon and software technology for VoP has evolved to support hundreds of channels and will soon support OC–3 channel density on a single chip (greater than 2,000 channels). This tutorial will focus on understanding the requirements and architecture for high density VoP platforms with an emphasis on the elements required to implement VoP in carrier class applications that will meet the service providers expectations to deliver toll quality voice.

2. Meeting Service Provider Requirements with Solution Density

Service providers have a great deal to gain through VoP implementations. They envision new revenue opportunities in leveraging the open architecture of packet-based services to quickly bring to market new and enhanced services while reducing operating costs through the deployment of packet-based networks. Service providers, however, will not sacrifice the tradition of toll-quality voice for the benefit of deploying a single converged infrastructure.

The *solution density* concept can help service providers and platform developers more clearly understand and implement high density VoP. VoP solutions cannot be evaluated merely for the number of channels on a chip. From a system engineering perspective, a solution must be evaluated on how the combination of system elements deliver a complete solution with lowest power and smallest area without compromising voice quality and features. Solution density then, refers to the optimization of channel density, power, architecture, integration of system functions, and I/O with the required software-based features for the targeted application (i.e., high density, carrier-class gateways).

The solution density concept provides a framework to evaluate solutions based on the integration of all major system elements and features by considering factors beyond just the number of channels and area for any single component. The following areas are important to service providers as they move to implement packet-based voice networks:

- Quality and reliability
- Scalability
- Flexibility

Quality and Reliability

Standards for VoP quality and network reliability are the same as for traditional telephone networks. Customers expect service quality to be consistent with traditional telephone networks. Packet-based voice is transparent to the customer. High-density VoP implementations are often discussed in terms of *carrier class* or *toll quality*, which are universally associated with the high-quality voice services. These terms set the expectation for quality of service providers and more important, their customers.

Features such as tone processing, packet play-out, voice activity detection (with comfort noise generation), and echo cancellation are key in meeting quality expectations. These features require a robust and *in-service-hardened* implementation to be considered *toll quality*. The absence of any of these features will result in less than toll-quality voice service.

Scalability

The ability to scale VoP networks to large volumes of traffic is critical to service providers justifying deployment of a packet-based infrastructure. Scalability requires VoP gateways to support very high volumes of traffic without degradation of voice quality. This places increased pressure on gateway vendors to support thousands of voice channels on a single platform driving the need for a system-level evaluation of density.

In most cases, density will be limited by the power requirements for the total system. Service providers must maintain power and cooling within industry guidelines (i.e., NEBS specifies a maximum of 1275 watts for a 23-inch bay with forced air cooling). For platform developers, this important design criterion can only be met with a solution that optimizes the design for power and area while maintaining carrier-class features. Therefore, the most important performance specification is not channels per chip but power per channel. It is possible that a service provider will run out of power-budget before shelf space in a bay. This would result in underutilization of central office space driving higher operational costs. Power per channel is the key metric.

Flexibility

Flexibility includes the ability to add new services and to react to standards evolution. VoP gateway platforms based on solutions that support features beyond pulse code modulation (PCM) voice (features such as low bit-rate CODECs and fax relay) enable service providers to add services without the disruption and expense of replacing equipment. These software-based features allow platform developers to distinguish their gateways from the competition. From a solution density perspective, architectural considerations, such as sizing of DSP resources and memory, will determine a solution's level of flexibility.

Key Solution Density Criteria for High-Density Applications

Table 1 summarizes the key criteria for high density solutions from a solution density perspective.

Table 1. Solution Density Summary: Relative Importance of Architecture Considerations

RISC/DSP Processor	Integration of System Functions	Integration of System Interfaces	Power Consumption	Channel Density	Voice Features	Cost
Multicore DSP	Medium	Medium	High	High	High	High

The architecture of a high density VoP gateway is discussed in the following section with focus on power, channel density, and voice features supporting true carrier-class voice.

3. High-Density VoP Architecture

High-density VoP architectures are driven by the following critical elements:

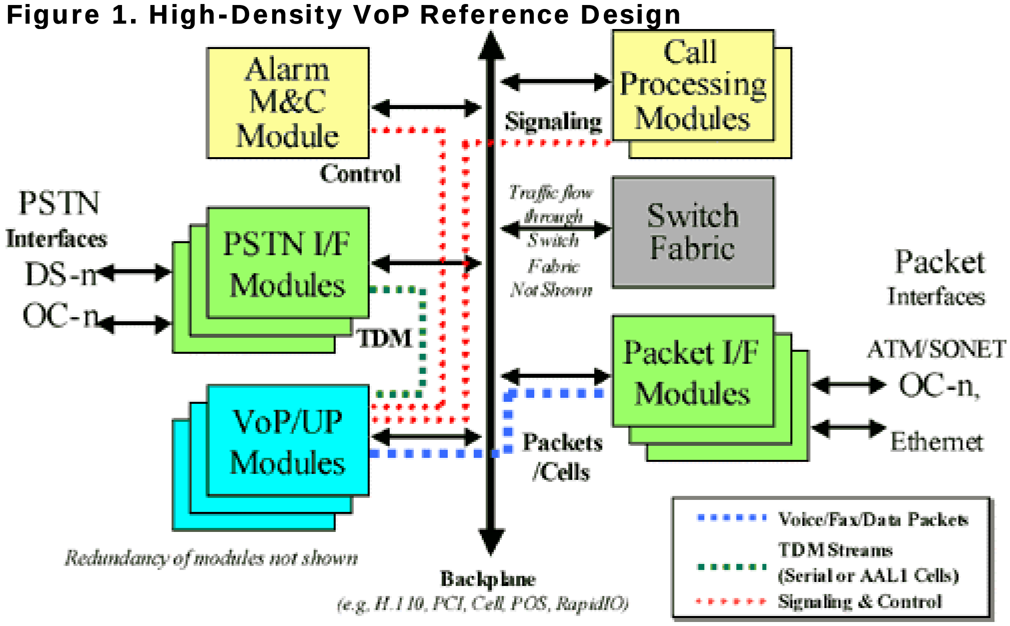
- Power per channel of the solution expressed in milli-Watts (mW) per channel
- Cost per channel of the solution that includes silicon/hardware, software, and intellectual property licensing costs
- Channel density of the solution expressed in channels per square inch
- System partitioning including packet aggregation and routing
- Software features that define the functionality of the product
- Network management capabilities to address high availability and accountability

Cost, power, and area must be evaluated on a total system basis and must be a function of the features and capabilities supported. *Figure 1* shows an example of high density design consisting of the following hardware modules:

- Alarm monitor and control (M&C) module
- Call processing modules

- Public switched telephone network (PSTN) interface modules
- Packet interface modules
- VoP/universal port (UP) modules
- Backplane interface

Figure 1. High-Density VoP Reference Design



The alarm M&C module performs the overall network management for the equipment. This includes configuration on a per-channel basis, status and statistics collection, call record reporting, and alarm processing. The call processing modules perform call establishment and call teardown for the system and performs interworking functions between the PSTN and packet network. Depending on the application and location of the equipment, the following signaling may be performed:

- PSTN telephony signaling
  - Signaling system 7 (SS7)
  - Integrated services digital network (ISDN)
  - TR08
  - TR303
- VoP network signaling
  - H.323

- Media gateway control protocol (MGCP)
- Megaco
- Session initiation protocol (SIP)
- Asynchronous transfer mode (ATM) broadband local emulation services (BLESs)

Depending on the architecture, call processing may be centralized or distributed with the VoP modules performing lower levels of the signaling protocols.

The PSTN interface modules provide the interface to the PSTN. Traditionally, PSTN interfaces for VoP consisted of T-1 (24 channels) and E-1 (30 channels). High-density VoP systems being designed today typically have multiple DS–3 (28 T-1s or 672 channels) and even multiple OC–3 (2,016 channels) PSTN interfaces as manufacturers offer equipment capable of handling in excess of 100,000 voice channels in a single rack of equipment.

The packet interface modules provide the interface to the packet-switched network. The two most prevalent networks are ATM and Internet protocol (IP). Depending on the application, VoP equipment may be ATM–centric, IP–centric, or support a hybrid of both ATM and IP voice. In many cases, it is important for the equipment to support both voice over ATM (VoATM) and voice over IP (VoP) on a per-call basis to provide interworking between ATM and the IP world. Packet interfaces include OC–n (OC–3, OC–12, etc.) optical interfaces for ATM and packet over SONET (POS) as well as multiple 100 BaseT and Gigabit Ethernet interfaces.

The switch fabric module performs the routing of cells/packets through the system. Line cards fill out the appropriate header information that is used by the switch fabric to direct cells/packets to the appropriate line card/external interface.

As shown in *Figure 2*, the VoP modules consist of a “farm” of DSPs that perform the actual conversion of the voice streams between the PSTN and packet worlds. In the PSTN–to-packet network direction, the VoP modules receive 64 kbps data streams from the PSTN interface modules and output packets or cells to the packet interface modules. Similarly, in the packet network to PSTN direction, the VoP modules receive packets or cells from the packet interface modules and output 64 kbps streams to the PSTN interface modules. The DSPs are controlled by a “host” processor that is responsible for configuration and software download of the DSPs as well as assisting in call establishment and termination and other network management functions.

In order to concentrate a large number of VoP channels, aggregation logic is required. This logic performs the following functions:

- Aggregates packet streams from multiple DSPs to the backplane/packet network interface

- Routes incoming packets from the backplane/packet network interface to the appropriate DSP

- Provides a standard interface to the backplane/packet network interface

- Filters network management and call setup/teardown information to a host processor

There are many different backplane interfaces that are used in systems such as these. Most typical are PCI and cell bus variants as well as POS. Time division multiplexing (TDM) samples from the PSTN can be relayed over an H.110 TDM bus or the PCM samples can be encapsulated in ATM cells to be sent over the same cell bus that is used for packet traffic.

Figure 2. VoP High Density Module

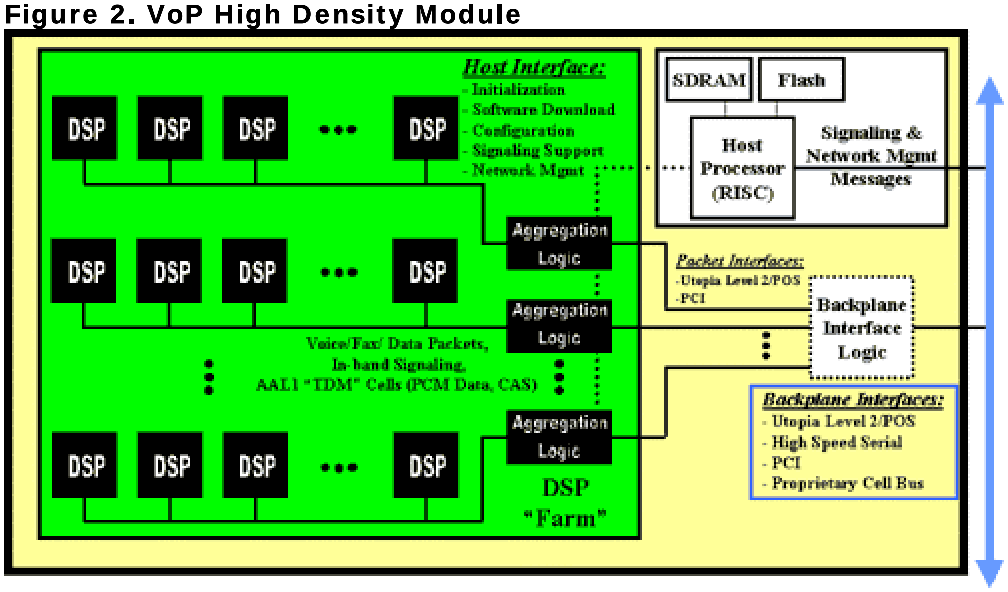
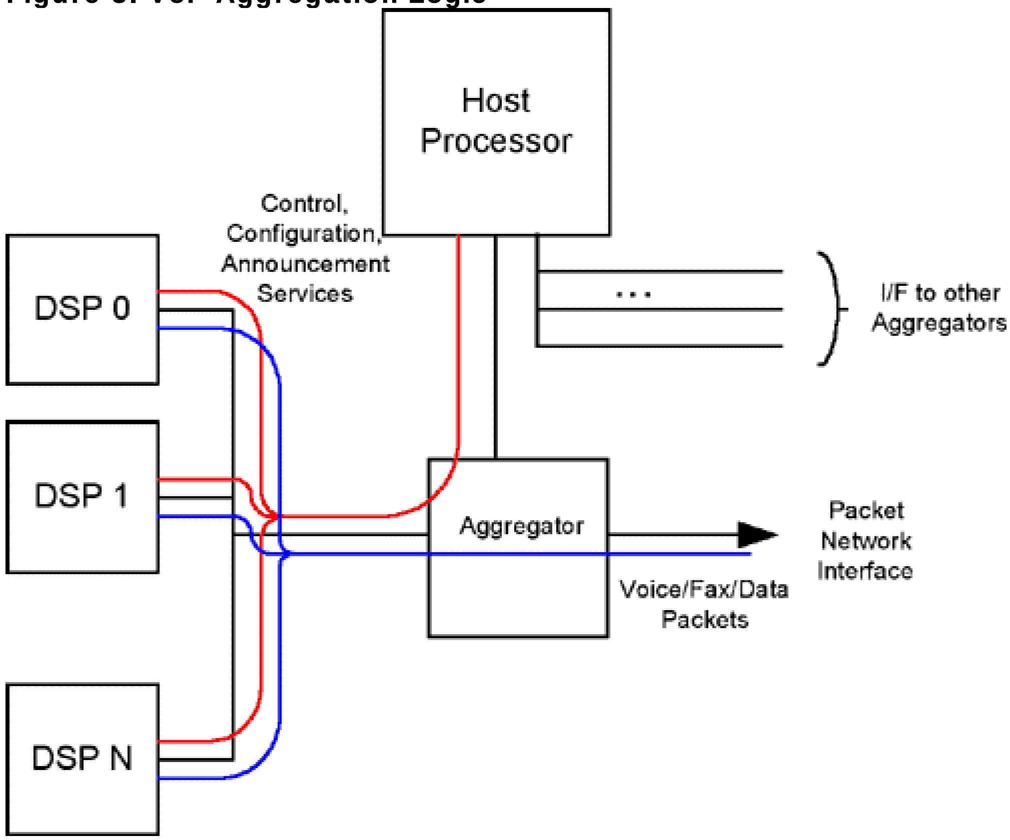


Figure 3. VoP Aggregation Logic



Software is a critical ingredient of high-quality VoP systems. There are many features that must be implemented for carrier-class systems. The most important software features include the following:

- Echo cancellation
- Voice compression
- Packet play-out software
- Tone processing
- Fax and modem support
- Packetization
- Signaling support
- Network management

Echo Cancellation

One of the keys to high-quality VoP is having a hardened line echo canceller that can properly cancel echo, which is present even in a conventional POTS network. In this type of network, echo is acceptable because delay is less than 50 milliseconds, and the echo is masked by the normal side tone that every telephone generates. Echo becomes a problem in packet networks because the delay is almost always greater than 50 milliseconds, thus requiring echo-cancellation techniques as part of the VoP solution. The ITU defines echo cancellation performance requirements. 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:

- It removes echo well. This includes removing echo at the start of a call as well as preventing any form of echo during a call.
- It handles double talk (both sides talk simultaneously) well. This includes not clipping the voice at the beginning or end of a double-talk voice spurt.
- It handles background noise well. This includes handling high background noise and variable background noise.
- It exceeds G.165/G.168 and provides support for future (more stringent) ITU EC standards such as G.168-2002.
- It is field proven. Note: Compliance with G.165/G.168 alone is no guarantee that the EC will work properly in real-life situations.
- It provides fast convergence time, low residual echo (depth of convergence), reliable detection of double-talk without divergence or clipping, and handles background noise and narrowband signals well.
- It supports up to 128 millisecond tail (often specified for carrier-class solutions) including support for multiple reflections over the entire 128 msec tail.
- It is capable of dynamically tracking echo path changes and is required for conferencing, call transfers, permanent off-hook connections, and to support redundancy.

- It behaves properly in the presence of 4-wire connection and low hybrid attenuation.

- It has built-in configurability and instrumentation.

Voice Compression

G.711 PCM (64 Kbps) is being most widely deployed for very high density VoP solutions. This is enabled by having “bandwidth to burn” in the infrastructure and the fact that PCM requires less processing power per channel than for low bit-rate (LBR) compressed voice results in higher densities at a lower cost from an equipment perspective. LBR compressed voice is required for broadband access equipment where bandwidth to the infrastructure is constrained, such as the following:

- Derived voiceover–cable modem and fixed wireless applications where there is a shared medium from the subscriber device to the head-end

- Derived voice channels over DSL applications where the subscriber line is a long distance away from the CO

- Mobile-to-mobile cellular calls where voice is already compressed, and it is undesirable to perform transcoding (degrades voice quality and adds delay)

- Emerging Internet voice portal market

- Internet audio content is stored and/or transmitted as compressed data compatible with PC programs

- Wideband voice compression where new algorithms fulfill the promise of packet voice quality better than the PSTN

The following are important considerations for voice compression:

- Passes all ITU test vectors

- Configurable packetization for maximum flexibility

- Proprietary VAD required for voice CODECs that do not support an integral VAD

- Adaptive comfort noise generation (CNG) in conjunction with VAD

- No degradation when all channels are active

Packet Play-Out

Packet play-out addresses the effects of network impairments on the voice. Impairments include lost packets, and delay of packets, including variable delay that distorts the timing sequence of the original voice. Therefore, it is essential for high-quality VoP to have packet play-out algorithms. These algorithms should do the following:

- Compensate for packet loss, delay, and jitter
- Be adaptive for lowest delay
- Reside in DSP for system scalability
- Be highly configurable and provide comprehensive network management statistics

Tone Processing

Tone processing is essential for call setup and termination as well as handling in call user functions such as accessing voice mail, making credit card calls, etc. The following are key elements for high density systems:

- Reliable tone detection (no false detects, no failure to detect)
- Early detection to minimize delay and to prevent in-band tone leakage that can lead to false tone generation at the remote end
- Different detection requirements based on network application and system architecture: dial digits, fax detection, modem detection, and call progress tones
- Support for bidirectional tone detection and generation in cases where the customer premises equipment (CPE) does not perform these functions

Fax and Modem Support

One of the key features of a VoP System for carrier-class applications is to fully emulate the PSTN, e.g., Class-5 switch replacement. Therefore, it is essential that these systems handle fax and modem in addition to voice. Fax and modem can be handled by a technique known as “PCM up-speed” where the VoP system, upon detection of fax or modem, forwards all data for a given channel as a 64 kbps transparent PCM stream between the two end points. While this works well for networks with no packet loss or excessive delay/jitter, networks (particularly VoP networks) experience occasional packet loss (less than 1 percent) that can cause the modems to retrain or even call failure. In cases where PCM up-speed is inadequate, the system must support fax and modem relay.

Fax relay provides reliable real-time fax service between two analog fax machines over a packet network. The VoP equipment at both ends of the packet network spoofs the analog fax machines such that they operate as if directly connected over a PSTN connection. The VoP 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 and AAL–2 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), AAL–2 (ATM)
- Fax machine spoofing protocols: proprietary

Currently, there are no interoperability standards for modem relay, but standards are being proposed. Modem relay consists of the following functions:

- Modem pumps: e.g., V.90, V.34, etc.
- Modem relay protocol: negotiation, flow control, error control

Packetization

The following packet encapsulation is performed in the DSPs to facilitate scalability and flexibility:

- VoP (RTP/RTCP)
- AAL–2
- AAL–1xN (for videoconferencing streams)

These should be supported on a per-channel basis to support hybrid ATM/IP networking equipment. Another important feature is something known as network channel switching. This is the ability to route from packet network to packet network. The routing can include the ability to transcode the voice payload and/or change encapsulation format, as in the following examples:

- VoP (RTP) <–> AAL–2

- G.726 <–> G.729AB

Signaling

Signaling support is an essential element of the DSP software. Features include the following:

- Full tone detection and generation capabilities: DTMF, MF R1/R2, SS7 COT, call progress tones, bidirectional tone processing
- Channel associated signaling (CAS) support: CAS bit processing
- Common channel signaling (CCS) support: HDLC, MTP1 (SS7)
- CAS and CCS support in DSP to off-load the host for higher scalability
- Play-out of service announcements (TDM or packet network direction)

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 capabilities
- Bellcore test line support for diagnostics
- Redundancy support

Figure 4. High-Density VoP Software Architecture

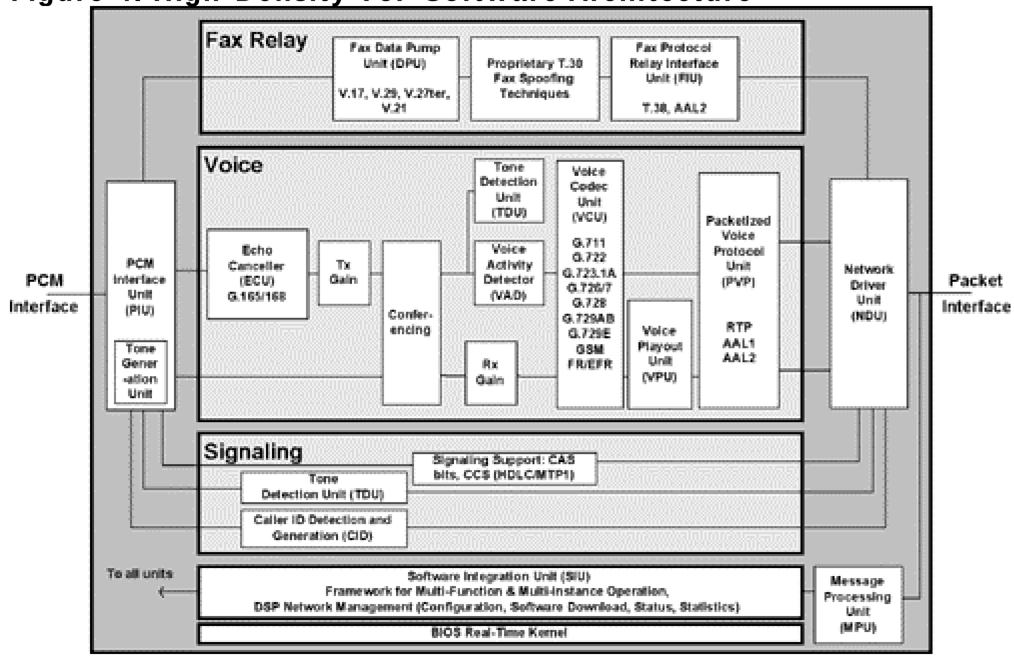


Figure 4 shows the architecture for high density VoP software. The software should be designed to minimize delay and maximize scalability. This includes the following:

- Efficient and adaptive algorithms
- Header encapsulation in VoP device, (RTP, AAL–1, AAL–2) on a per-channel basis
- Low-latency implementation
- Features to off-load “host” processor to drive overall channel density

4. Conclusion

Solutions based on the solution density concept that offer high density and low power and provide full-featured performance will enable a new generation of high density, carrier-class gateway platforms. This new generation of platforms will enable service providers to deploy a common, packet-based infrastructure for voice and data services enabling a new generation of services for businesses and consumers.

Self-Test

1. Which of the following is an advantage that packet-based voice provides over circuit-switched voice?

    a. bandwidth can be significantly reduced through VAD

    b. faster time to market for new services

    c. capital investment for packet-based platforms is significantly less than circuit-switched equivalent

    d. all of the above

2. Early VoP applications were targeted at providing lower-cost data services.

    a. true

    b. false

3. VoP gateways are being implemented for a number of applications ranging from small two-port integrated access devices (IADs) for the consumer and telecommuter to very large carrier-class gateways used as Class-5 switch replacements.

    a. true

    b. false

4. Standards for VoP quality and network reliability are the same as for traditional telephone networks.

    a. true

    b. false

5. The most important performance specification is not ____________ but ____________.

    a. power per channel but channels per chip

    b. channels per chip but power per channel

6. The _____________ module performs the overall network management for the equipment.

    a. alarm monitor & control

b. call processing

c. PSTN interface

d. switch fabric

e. VoP

7. The _____________ module performs the routing of cells/packets through the system.

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8. The _____________ modules consist of a “farm” of DSPs that perform the actual conversion of the voice streams between the PSTN and packet worlds.

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9. ____________ becomes a problem in packet networks because the delay is almost always greater than 50 milliseconds.

a. Tone

b. Packetization

c. Echo

d. Support

10. Fax and modem can be handled by a technique known as “PCM up-speed” where the VoP system, upon detection of fax or modem, forwards all data for

a given channel as a 64 kbps transparent PCM stream between the two end points.

a. true

b. false

Correct Answers

1. Which of the following is an advantage that packet-based voice provides over circuit-switched voice?

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   See Topic 1.

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   b. false

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4. Standards for VoP quality and network reliability are the same as for traditional telephone networks.

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b. false

See Topic 2.

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Glossary

ATM
asynchronous transfer mode

BLES
broadband local emulation service

CNG
comfort noise generation

DSL
digital subscriber line

DSP
digital signal processor

IAD
integrated access devices

IP
Internet protocol

ISDN
integrated services digital network

M&C
monitor and control

MGCP
media gateway control protocol

mW
milli-Watt

OC
optical carrier

PCM
pulse code modulation

POS
packet over SONET

PSTN
public switched telephone network

RISC
reduced instruction set computing
SIP
session initiation protocol
SONET
synchronous optical network

SS7
signaling system 7

TDM
time division multiplexing

UP
universal port

VAD
voice activity detection

VoATM
voice over ATM

VoIP
voice over IP

VoP
voice over packet