Voice over Cable (VoCable)

Definition

Cable operators are busy optimizing the existing bandwidth in their networks to deliver high-speed Internet access and are turning their efforts to delivering integrated Internet and voice service over the same cable spectrum. Voice (and fax) service over these cable networks is referred to in this tutorial as cable-based Internet protocol (IP) telephony. This technology, targeted primarily at the home, poses many of the same challenges faced by telecom carriers as they work to deliver to enterprise voice and fax over other types of packet networks such as asynchronous transfer mode (ATM) and frame relay.

Overview

This tutorial discusses the challenges of providing cable-based IP telephony and provides discussion of quality of service (QoS) challenges and the effort to develop and implement interoperability standards. An overview of an embedded software architecture is presented, and a system is described for sending voice, fax image, data, and signaling information over the cable network. Benefits to designers and manufacturers of this embedded approach are lower cost of goods sold, faster time to market, and lower development costs.

Topics

1. Introduction

Cable-based IP telephony holds the promise of simplified and consolidated communication services provided by a single carrier at a lower total cost than consumers currently pay to separate Internet, television, and telephony service providers. Cable operators have already worked through the technical challenges of providing Internet service and are optimizing the existing bandwidth in their cable plants to deliver high-speed Internet access. Now, cable operators have turned their efforts to the delivery of integrated Internet and voice service using that same cable spectrum.

Cable-based IP telephony falls under the broad umbrella of voice over IP (VoIP), meaning that many of the challenges facing cable operators are the same challenges that telecom carriers face as they work to deliver voice over ATM (VoATM) and frame-relay networks. However, ATM and frame-relay services are targeted primarily at the enterprise, a decision driven by economics and the need for service providers to recoup their initial investments in a reasonable amount of time. Cable, on the other hand, is targeted primarily at the home. Unlike most businesses, the overwhelming majority of homes in the United States is passed by cable, reducing the required up-front infrastructure investment significantly.

Cable is not without competition in the consumer market, for digital subscriber line (xDSL) has emerged as the leading alternative to broadband cable. However, cable operators are well positioned to capitalize on the convergence trend if they are able to overcome the remaining technical hurdles and deliver telephony service that is comparable to the public switched telephone system.

2. Making Cable a Viable Telephony Medium

For almost a century, Americans have taken for granted the nearly unfailing service provided by the public telephone network, often referred to as plain old telephone service (POTS). If cable is to emerge as a legitimate alternative, many technical issues must be addressed. Perhaps the most fundamental of these is the evolution of the nation's cable infrastructure from a one-way, broadcast medium to a two-way, personal communications medium.

Half-Duplex versus Full-Duplex Cable Infrastructure

Cable was first introduced in the United States in the late 1950s. For the next 30 years, nearly every mile of buried cable was half duplex and therefore capable of broadband transmission in the downstream direction (i.e., from the head end to the subscriber, but not in the upstream direction). Communication from the subscriber back to the head end was possible only by means of a telephone line.

This makes half-duplex lines cumbersome even for premium TV services, such as pay-per-view, that require upstream communication. It also makes half-duplex lines extremely inconvenient for Internet service, as a result of the fact that outbound e-mail messages and hypertext transfer protocol (HTTP) requests must be sent via the phone. Furthermore, it renders half-duplex lines completely useless for voice, as such service requires packets to be sent up- and downstream continuously.

In recent years, cable operators have heavily investigated possibilities for upgrading their buried cable from half to full duplex as a necessary first step toward capitalizing on the demand for integrated data and voice services. While upstream transmissions still are not as fast as downstream (typically 1.5 to 3 Mbps downstream and 500 kbps to 2.5 Mbps upstream), full-duplex lines offer sufficient throughput to support cable-based IP telephony. As cable operators compete for subscribers with xDSL providers, the speed with which cable operators replace older lines with full-duplex lines will be critical to their ultimate success.

Telephony Service across a Broadcast Media

Unlike POTS, which was developed from the outset as a point-to-point communication technology, cable networks were originally designed to broadcast one signal to many recipients. There was no concept of dedicated circuits, and there was no need to parcel out bandwidth to individual subscribers. To enable cable-based IP telephony, modifications must be made to the way bandwidth is allocated and packets are delivered. This must be done without using the bulk of the cable spectrum, because most of the bandwidth will continue to be used for TV broadcasts.

Direct Connect

Callers must be able to send and receive only their own voice packets, and these packets must be given priority over data packets to ensure that callers experience smooth, uninterrupted conversations. The first step in this process was addressed by the Data over Cable Service Interface Specification (DOCSIS). DOCSIS established universal ground rules for the transmission of packets across cable networks, ensuring that packets will not be routed incorrectly.

DOCSIS was later enhanced (in version 1.1) with QoS and security features necessary for voice communication. DOCSIS 1.1 also enables the prioritization of packet traffic. This allows cable operators to give certain packets (i.e., voice) the right of way and allows other traffic to be sent with a best-effort priority, as determined by bandwidth availability. However, even this second-generation DOCSIS standard was not intended to address all of the technical issues associated with cable-based voice service.

To fill in the gaps left by DOCSIS, CableLabs® created the packet cable specification known as the network-based call signaling (NCS) protocol for signaling voice calls over cable networks. [1] NCS leverages the existing media gateway control protocol (MGCP), and the protocol is thus sometimes known as MGCP NCS. NCS uses network-based call agents to negotiate cable-based IP telephony calls. Call agents, which will be discussed later in the tutorial, ensure that voice packets traverse the network and are audible only at the two conversation end points.

Security

While POTS is considered an extremely secure service, cable-based IP telephony is not. Much like cellular telephony, cable-based conversations are susceptible to illegal wire tapping and inadvertent chat conditions. To address this untenable situation, DOCSIS and NCS support multiple security services.

NCS currently supports the secure version of IP (IPsec) authentication specification. Adequate protection of telephony connections can be achieved if the telephony gateway accepts only packets that have been authenticated by IPsec. DOCSIS supports an encapsulation protocol for encrypting packet data across the cable network. The encapsulation protocol defines the frame format for carrying encrypted packet data, the set of supported data-encryption and authentication algorithms, and rules for applying the cryptographic algorithms to packet data.

DOCSIS currently employs the cipher block chaining (CBC) mode of the U.S. data encryption standard (DES) to encrypt packet data. The protocols are extensible, can support multiple encryption algorithms, and will, in all likelihood, be extended to support the new advanced encryption standard (AES) once it is in place.

Power Consumption

As most people know, traditional telephones draw all the power they need from POTS lines. Because the public phone system has evolved to such a reliable state and is essentially immune from the effects of power outages, it is exceptionally rare that service is lost. Electrical utilities in most areas do not, however, offer this degree of unfailing reliability. Therefore, head-end and customer-premises

[1] CableLabs is a research and development consortium of cable operators from across North America and South America. The organization's guiding concept is interoperability as cable networks evolve from a regional to a national scope, and the industry branches into Internet and telephony services.

cable equipment that relies solely on the local electric company for power puts users at risk of losing phone service should a power outage occur.

To address this issue, "lifeline service" requirements are being implemented across the country that require IP phones, such as those that connect to cable lines, to provide at least four hours of battery backup. To meet this requirement, equipment manufacturers must develop phones that can be powered by as little as three watts. A key to achieving this is a telephony chipset that minimizes idle processing cycles and offers sufficient onboard memory to handle all signal processing.

The ideal cable-based, IP–telephony system is typically built with a reduced instruction set computing (RISC) microprocessor to handle the signaling functions and digit collection. The necessary telephony peripherals, such as a local-area network (LAN) controller and universal serial bus (USB), are on a single chip to conserve power, and dedicated hardware should be used for the cable-communications protocol. Several megabytes of high-speed random access memory (RAM) are needed for signal processing, and the same amount of nonvolatile memory is needed to store the telephony application. The nonvolatile memory should be electrically reprogrammable, like a FLASH memory, to enable on-line software updates.

A high-performance, low-power digital signal processor (DSP) is needed to support the analog functionality (e.g., codecs), noise reduction, and echo cancellation. A programmable DSP can greatly reduce application-development time for solution providers. Texas Instruments' TMS320C54x DSP is one such chip.

Billing

Telephony billing is an extremely complex process. Most cable TV customers receive the same bill each month. Aside from pay-per-view requests, there is no need to meter or monitor customer usage. Telephone billing is quite different. A typical bill includes recurring monthly service fees, international and long-distance charges that vary based on time and day, and premium services, such as *69 and directory assistance, that are billed on a per-use basis.

To enable timely, accurate billing, call agents or broadband telephony interfaces (BTI) must collect all relevant usage data. The BTI is the cable equivalent to the phone box that is outside every home. In the absence of a BTI, cable-based IP telephony can also be delivered using voice-enabled cable modems inside a customer’s home. If the call agents collect the billing data, the BTI or cable modem need not be involved. Otherwise, the software inside the BTI or cable modem must provide application programming interfaces (APIs) so that the billing system can access the relevant data. Depending on each cable operator’s implementation, the data may be contained in standard management information base (MIB) files or in unique files set up specifically for telephony metering.

3. Challenges of Providing Toll-Quality Telephony Service on an IP Network

For cable operators, choosing which standard to support and preparing their infrastructures to support voice is only the beginning of the technological obstacle course. What remains is the QoS challenge inherent in all VoIP implementations. Among the most significant QoS hurdles are transmission latency, echo, jitter, and lost packets. These QoS factors are relatively harmless for data transmissions but must be dealt with aggressively to provide acceptable voice quality.

Latency

Latency, or delay, 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 significant quality problem, equipment providers must implement echo cancellation. 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 VoIP implementation 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 long distance (LD)–code excited linear prediction (CELP)—2.5 milliseconds
- G.729a, b, e convergence sublayer (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. For example, three frames of G.729 code words, equaling 30 milliseconds of speech, may be collected and packed into a single packet. 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 physical medium and the protocols used to transmit the voice data as well as by the buffers used to remove packet jitter on the receive side. The jitter buffers add additional delay that is used to smooth the 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.

Polling Delay

Cable-based IP telephony creates an additional latency that other packet networks do not because of the way head-end systems collect packets from callers. The head end polls the BTI at each customer location. Because the head end does not maintain a continuous connection with each BTI, there is a transmission delay while voice packets wait for the next poll. Therefore, it is important that cable-based IP–telephony equipment minimize this delay by anticipating when the next poll will arrive (a process called grant synchronization), so that the packets are queued up and ready to go.

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. Echo becomes a problem in VoIP networks because the delay is almost always greater than 50 milliseconds. Thus, echo-cancellation techniques must be used. The International Telecommunication Union (ITU) standards G.165 and G.168 define performance requirements for echo cancellers.

Echo is generated toward the packet network from the telephone network. The echo canceller compares the voice data received from the packet network 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—a 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 allow the slowest packets to arrive and be played in the correct sequence. This causes significant delay. The conflicting goals of minimizing delay and removing jitter have led to various schemes aimed at optimizing the size of the jitter buffer to minimize its impact on latency.

A common approach in cable-based IP telephony is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is then used to adjust the jitter buffer to target a specific late-packet ratio.

Lost Packets

In today's IP networks, voice frames are treated exactly like data. Under peak loads and congestion, voice frames will be dropped at the same rate as data frames. The data frames, however, are not time-sensitive, and dropped packets can be corrected through retransmission. Lost voice packets cannot be handled in the same manner. Some techniques used by VoIP software to address this problem include the following:

- **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 scheme is a simple method that fills the time between noncontiguous speech frames. It works well when the incidence of lost frames is infrequent. It does not, however, 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-bandwidth voice coder 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.

4. Fax Offers Additional QoS Challenges for Cable Networks

The challenges of implementing fax-over-cable networks are similar to those of voice. The two most significant issues are timing and lost packets. 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 the loss of the call. The fax-over-packet protocol compensates for the skewed timing of messages so calls are not dropped, and the accuracy of faxed images is not compromised.

Lost packets can be an even more serious problem for IP fax systems than for IP voice systems. In a VoIP application, the loss of packets can be addressed by replaying last packets and using other methods of interpolation. Fax-over–IP applications, 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 or not error correction mode is enabled.

CableLabs is working on specific fax services that will be added to NCS to standardize the implementation of fax-over-cable networks, and there is currently an optional fax relay service in the NCS protocol.

5. International Standards

The European Cable Modem Consortium has for some time been promoting and developing cable-modem products based on the U.S.–developed multimedia cable-network system (MCNS), synonymous with DOCSIS. Its members are U.S. firms and a few European proponents of MCNS, including 3Com, Broadcom, Cisco Systems, Dassault, General Instrument, Motorola, Pace, and Thomson. MCNS is the de-facto standard in the United States, and the equipment manufacturers that support it are now looking to Europe to leverage economies of scale.

At the same time, Alcatel, Cocom, DiviCom, Hughes Network Systems, Nokia, Sagem, Simac, Thomson Broadcast Systems, and Thomson Multimedia have teamed up to create the digital broadcast video (DVB)/Digital Audio Video Council (DAVIC) Interoperability Consortium. DVB and DAVIC technology is the incumbent European standard for digital set-top boxes and is starting to gain momentum for cable modems.

The ITU has accepted DOCSIS as a cable-modem standard, called ITU J.112. In April of 1999, CableLabs issued DOCSIS 1.1, adding key enhancements to support IP telephony and other constant–bit-rate services. The newer standard is backward-compatible, enabling DOCSIS 1.0 and 1.1 cable modems to operate in the same spectrum on the same network. CableLabs is now considering a third-generation DOCSIS standard, tentatively known as DOCSIS 1.2, which focuses on increasing upstream transmission capacity and reliability.

In an aggressive move to accelerate the acceptance of DOCSIS internationally, a European version of DOCSIS, named EuroDOCSIS, is now being pushed.

EuroDOCSIS is essentially the same as DOCSIS apart from the physical layer, which is DVB–compliant.

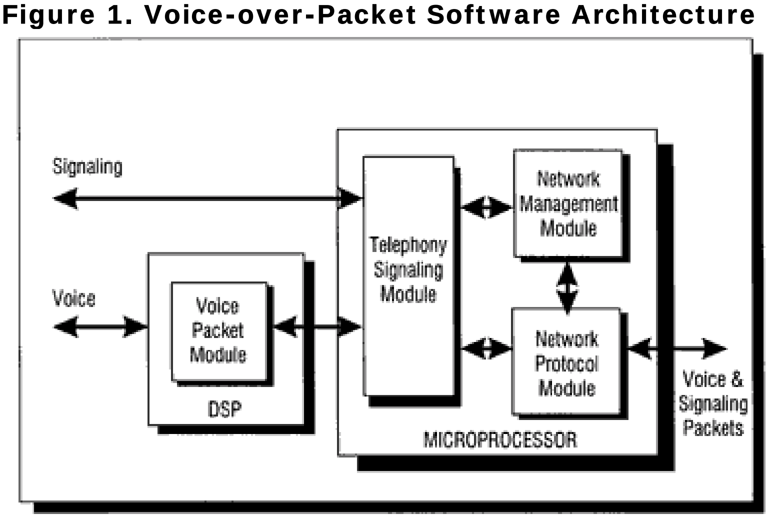
EuroCableLabs (ECL), operating under the direction of the European Cable Communications Association (ECCA), has championed a DVB/DAVIC–based EuroModem as an alternative to DOCSIS. DVB cable modems meet the preference of some European operators for a standard that better fits their set-top architectures. There is also a desire to support homegrown European products, rather than importing solutions from American suppliers.

The DVB 2.0 specification has been formally adopted by the European Telecommunications Standards Institute (ETSI) as ETS 300800. The standard describes the out-of-band and in-band transmission options applicable to interactive set-top boxes and cable modems respectively, enabling the deployment of interactive television, data, and voice services over a common platform. ETS 300800 has also been selected by DAVIC to be the DAVIC 1.5 specification for cable modems.

6. Embedded Software Architecture

Texas Instruments and its subsidiary, Telogy Networks, have developed one implementation of embedded VoIP software for cable-based IP telephony. The software supports cable modems and BTIs (up to four ports), as well as the telephony gateway (up to several thousand ports), at the cable head end. The software supports MGCP as well as the session initiation protocol (SIP). The basic purpose of the two protocols, which is to process packetized voice traffic, is the same. However, the software supports both because standards bodies are divided as to the relative merits of each. As shown in *Figure 1*, this software interfaces to both streams of information from the telephony network and converts them to a single stream of packets transmitted to the packet network.

Figure 1. Voice-over-Packet Software Architecture



MGCP is a centralized call-processing system in which the intelligence resides primarily at the head end. The cable modems and BTIs are similar to dumb clients, and the system relies on call agents to negotiate the call through the network. SIP is a distributed system in which the intelligence resides in the BTI, and the head end is mainly a gateway to the public telephone network.

The greatest benefit of MGCP implementations is simplified, efficient management and administration. Fault detection and isolation are typically limited to the head end. Furthermore, there is no need to distribute software upgrades and patches to customers, and, thus, there is also no concern about software-version synchronization among all BTIs.

The supporters of SIP, on the other hand, argue that it is a more scalable and reliable system. The case for scalability relies on the fact that the head end, acting mainly as a gateway, is unlikely to bottleneck subscriber capacity. Traffic load, rather than processing time, would be the only potential bottleneck. Supporters also claim that SIP is more reliable, because a SIP–based network architecture does not have a single point of failure.

The MGCP– and SIP–compliant architecture processes voice packets similarly using either protocol. The software is broken down into two parts: the DSP and microprocessor components. The DSP processes voice data and passes voice packets to the microprocessor with generic voice headers. The microprocessor component is responsible for moving voice packets and adapting the generic voice headers to the NCS protocol. The microprocessor also processes signaling information and converts it from a telephony signaling protocol to IP.

This partitioning of functionality between the DSP and microprocessor provides a clean interface between the generic processing functions (such as compression, echo cancellation, and voice-activity detection) and the application-specific signaling and protocol processing.

DSP Component (or Voice-Processing Module)

This software prepares voice samples for transmission over the packet network. Its components perform echo cancellation, voice compression (to conserve cable bandwidth), voice-activity detection, jitter removal, clock synchronization, and voice packetization. This unique software, along with TI's programmable DSP technology, provides a comprehensive yet flexible foundation that allows equipment providers to shave months off of typical development schedules, resulting in tremendous cost savings and a critical time-to-market boost.

Microprocessor Component

In NCS–based products, the microprocessor component handles detection of various events, reporting of events to call agents, dual-tone multifrequency (DTMF) digit collection and reporting, application of signals, and forwarding of audio packets. The microprocessor component in NCS–based products is comprised of the following software modules:

- XGCP signaling module (XGCM)
- digit collection module (DCM)
- DSP interface module (DIM)

DSP Component

PCM Interface

This interface receives pulse code modulation (PCM) samples from the digital interface and forwards them to the appropriate DSP software modules for processing. It also forwards processed PCM samples received from DSP software modules to the digital interface and performs continuous resampling of output samples to avoid sample slips.

Tone Generator

This generates DTMF tones and call-progress tones under command of the host (e.g., telephone, modem, private branch exchange [PBX], or telephone switch). It supports U.S. and international tones.

Echo Canceller

This performs G.165– and G.168–compliant echo cancellation on sampled, full-duplex voice signals. It has a programmable range of tail lengths.

Voice Activation Detector

This monitors the received signal for voice activity. When no activity is detected for a specific period of time, the software informs the IP. This prevents the encoder output from being transported across the network when there is silence so as to save bandwidth. This software also measures the idle noise characteristics of the telephony interface. It reports this information to the IP in order to relay this information to the head end for noise generation when no voice is present.

Tone Detector

This detects the reception of DTMF tones and performs voice and fax discrimination. Detected tones are reported to the host so that the appropriate speech or fax functions are activated.

Voice Codec Software

This software compresses the voice data for transmission over the packet network. It is capable of numerous compression ratios through its modular architecture. A compression ratio of 8 to 1 is achievable with the G.729 voice codec.

Fax Software

This software performs a fax relay function by demodulating PCM data, extracting the relevant information, and packing the fax-line scan data into frames for transmission.

Voice Playout Unit

This buffers voice packets received from the packet network and sends them to the voice codec for playout. The following features are supported:

- first in, first out (FIFO) buffer that stores voice codewords before playout to remove timing jitter from the incoming packet sequence

- continuous-phase resampler that removes timing-frequency offset without causing packet slips or loss of data

- timing-jitter measurement, which allows adaptive control of FIFO delay

The voice packetization protocols use a sequence number field in the transmit-packet stream to maintain temporal integrity of voice during playout. Using this approach, the transmitter inserts the contents of a free-running, modulo-16 packet counter into each transmitted packet, allowing the receiver to detect lost packets and to reproduce silent intervals during playout.

Packet Voice Protocol

This encapsulates compressed voice and fax data for end-to-end transmission over a backbone network between two ports.

Control Interface Software

This coordinates the exchange of monitor and control information between the DSP and host via a mailbox mechanism. Information exchanged includes software downline load, configuration data, and status reporting.

Real-Time Portability Environment

This provides the operating environment for the software residing on the DSP. It also provides synchronization functions, task management, memory management, and timer management.

Unsolicited Grant Service (UGS)

Cable networks are asymmetric; i.e., the downstream data received is streaming while the upstream data transmitted is either transmitted on a collision time frame or must get a time slot or grant. Because requesting a grant can cause significant delay, UGS ensures that cable modems will be contacted at regular intervals without having to make separate requests. The concatenation process mentioned earlier can lighten UGS requirements and increase the efficient use of bandwidth

UGS with Activity Detection (UGS–AD)

Upon detection of voice inactivity, UGS–AD enables network resources to be diverted to other cable modems and data flows, maximizing the efficiency of all data transmissions.

Microprocessor Component

DIM

The DIM is responsible for the interface to the discussed DSP software. The microprocessor communicates with the DSP through a shared memory arrangement, the mechanics of which are hidden by the DIM. The DIM shields the rest of the microprocessor software from the complexities of the DSP interface.

XGCM

This module is responsible for providing MGCP embedded client functionality. It parses and processes each message received from an MGCP call agent. It reports detected events to the call agent, generates signals requested by the call agent, reports detected DTMF digits, and sets up connections requested by the call agent. This module is also responsible for forwarding audio packets received from the DSP to the packet network interface and forwarding audio packets received from the packet network interface to the DSP.

DCM

This module is responsible for processing dialed digits received from the XGCP module. It accumulates all the dialed digits and matches them against the digit map. It reports the results along with the accumulated digits to the XGCP module.

Network Management Module

This module is responsible for providing the management interface to configure and maintain the other modules of the software. A sample module is provided, but the customer may replace the sample with a custom module. A proprietary voice packet MIB is supported because no standard MIB exists.

7. Fault Analysis

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. The software described in this tutorial offers a rich set of diagnostic features to accomplish this.

Loop-Back Capability

PCM loop-back is used for diagnosing problems related to the telephony side. Packet-send loop-back diagnoses problems related to DSP software performance. Packet-receive loop-back diagnoses problems related to the packet network.

Signal-Level Measurement

The software described in this tutorial can perform signal-level measurements on the telephony-side interface for a particular channel. It reports instant and mean values for signal power in both directions.

Packet Network Statistics

The software generates an extensive set of performance statistics for each channel. The statistics include number of transmitted and received packets, minimum and maximum packet interarrival times (i.e., jitter measurement), number of invalid packet headers, and number of lost packets. In addition, the voice playout unit reports number of lost voice frames, number of repeated frames, number of idle frames, number of dropped frames, and average packet jitter.

PCM Sample Trace

The DSP software can provide 10 milliseconds of PCM samples (approximately 80 samples).

Memory Trace

The DSP software can provide 40 memory values, starting from the requested location. Memory locations can be from data memory or program memory.

Fax Processing Debug

The software described in this tutorial provides a stream of debug information, tracing the performance of fax operations.

Echo Canceller Statistics

The software offers a detailed set of echo-canceller debug statistics.

8. Summary

With the merging of telecom carriers, cable operators, and Internet service providers (ISPs), most experts agree that convergence is not merely a trend but an inevitability. The potential cost savings, consolidated billing, streamlined network management, and overall convenience are too compelling for service providers and consumers to ignore. With buried cable passing hundreds of millions of homes worldwide, it is logical to assume that cable will be front and center as convergence becomes mainstream.

The technical challenges will be overcome as innovation and experience combine to provide cable-based IP–telephony solutions that are equals of the public telephone system. The software architecture outlined in this paper has been field-tested and is designed to provide equipment manufacturers with a repeatable, core starting point to help them develop unique, value-added telephony solutions and bring those solutions to market as quickly and cost-effectively as possible.

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Self-Test

1. ATM and frame-relay services are targeted primarily at the home.

    a. true

    b. false

2. Cable service is targeted primarily at the enterprise.

    a. true

    b. false

3. Which of the following is the leading alternative to broadband cable?

    a. broadband wireless

    b. DSL

    c. fiber

    d. satellite

4. If buried cable is half duplex, it is capable of broadband transmission in the ___________ direction.

    a. upstream

    b. downstream

5. Full-duplex lines do not offer sufficient throughput to support cable-based IP telephony.

    a. true

    b. false

6. ____________ established universal ground rules for the transmission of packets across cable networks.

    a. DOCSIS

    b. MGCP

    c. DAVIC

    d. MCNS

7. While POTS is considered an extremely secure service, cable-based IP telephony is not.

    a. true

    b. false

8. Which of the following is present even in a conventional POTS network?

    a. processing delay

    b. jitter

    c. lost packets

    d. echo

9. The embedded software for cable-based IP telephony described in this tutorial supports MGCP and SIP.

    a. true

    b. false

10. Which of the following is used for diagnosing problems related to the telephony side of the network?

    a. signal-level measurement

    b. loop-back capability

    c. packet network statistics

    d. PCM sample trace

Correct Answers

1. ATM and frame-relay services are targeted primarily at the home.

    a. true

    b. **false**

    See Topic 1.

2. Cable service is targeted primarily at the enterprise.

    a. true

    b. **false**

    See Topic 1.

3. Which of the following is the leading alternative to broadband cable?

    a. broadband wireless

    b. DSL

    c. fiber

    d. satellite

    See Topic 1.

4. If buried cable is half duplex, it is capable of broadband transmission in the ___________ direction.

    a. upstream

    b. **downstream**

    See Topic 2.

5. Full-duplex lines do not offer sufficient throughput to support cable-based IP telephony.

    a. true

    b. false

    See Topic 2.

6. ____________ established universal ground rules for the transmission of packets across cable networks.

    a. DOCSIS

    b. MGCP

    c. DAVIC

    d. MCNS

See Topic 2.

7. While POTS is considered an extremely secure service, cable-based IP telephony is not.

    a. true

    b. false

    See Topic 2.

8. Which of the following is present even in a conventional POTS network?

    a. processing delay

    b. jitter

    c. lost packets

    d. **echo**

    See Topic 3.

9. The embedded software for cable-based IP telephony described in this tutorial supports MGCP and SIP.

    a. **true**

    b. false

    See Topic 6.

10. Which of the following is used for diagnosing problems related to the telephony side of the network?

    a. signal-level measurement

    b. loop-back capability

    c. packet network statistics

    d. PCM sample trace

    See Topic 7.

Glossary

AES
advanced encryption standard

API
application programming interface

ATM
asynchronous transfer mode

BTI
broadband telephony interface

CBC
cipher block chaining

CELP
code excited linear prediction

CS
convergence sublayer

DAVIC
Digital Audio Video Council

DBV
digital broadcast video

DCM
digit collection module

DES
data encryption standard

DIM
DSP interface module

DOCSIS
data over cable service interface specification

DSL
digital subscriber line

DSP
digital signal processor

DTMF
dual-tone multifrequency

ECCA
European Cable Communications Association

ECL
EuroCableLabs

ETSI
European Telecommunications Standards Institute

FIFO
first in, first out

HTTP
hypertext transfer protocol

IP
Internet protocol

ISP
Internet service provider

ITU
International Telecommunications Union

LAN
local-area network

LD
long distance

MCNS
multimedia cable-network system

MGCP
media gateway control protocol

MIB
management information base

NCS
network-based call signaling

PBX
private branch exchange

PCM
pulse code modulation

POTS
plain old telephone service

QoS
quality of service

RAM
random access memory

RISC
reduced instruction set computing

SIP
session initiation protocol

UGS
unsolicited grant service

UGS-AD
UGS with activity detection

USB
universal serial bus

VoATM
voice over ATM

VoIP
voice over Internet protocol

XGCM
XGCP signaling module