

Implications of DOCSIS-QoS on Cable Broadband Service

Submitted by

Chris Rohrer
Interdisciplinary Telecommunications Program
University of Colorado at Boulder

and

Douglas C. Sicker
Department of Computer Science
and
Interdisciplinary Telecommunications Program
University of Colorado at Boulder

August 31, 2004

Abstract

As we move services, such as voice, onto Internet Protocol (IP) based networks, the Quality of Service (QoS) that these networks can provide becomes an increasingly important issue. Unlike the Public Switched Telephone Network (PSTN), packet switched networks are designed on the simple premise of best-effort, meaning that the network does not provide a guarantee of delivering the data or maintaining well-paced streams. Voice and other real-time services require certain levels of quality to maintain satisfactory communications and most carriers have added some type of QoS capability to their networks, including cable modem, DSL, WLAN and satellite providers.

The ability to make use of these QoS services may become a significant issue. For example, on broadband cable access networks, the lack of access by third-party VoIP entrants to such QoS capabilities may result in an inability for these providers to offer such realtime (or QoS sensitive) services at a desired level of quality. They would need to rely on best-effort handling, which could result in an increased possibility of quality impairment especially on shared access facilities. To address this issue, facilities based carriers would need to provide access to the data link layer Quality of Service mechanisms used for managing packet flows. Clearly, such efforts would have an impact on the design, operation and management of a network, however, there is good reason to believe that even though all users stand to benefit from a higher quality level, facilities-based carriers would consider such measures if for no other reason than to facilitate third party business opportunities.

In this paper, we describe the significance of this issue, provide evidence for its existence, indicate the consequences and propose a possible course of action. While this problem applies across all access networks, we look specifically at cable networks. And while this paper suggests specific solutions for this issue, we do not necessarily advocate regulatory mandates for its implementation.

Table of Contents

I.	Introduction.....	3
II.	Cable Broadband and Telephony Services	3
	A. MSO High Speed Data access	4
	B. MSO circuit switched cable telephony	5
	C. MSO packet switched cable telephony	6
	D. Third-party VoIP.....	8
	E. Internet Backbone and HFC Broadband Access.....	9
	F. Quality of Service Definitions	10
III.	DOCSIS and PacketCable Operations	12
	A. The CMTS and DOCSIS	12
	B. PacketCable.....	13
	1. E-MTA.....	14
	2. S-MTA	15
IV.	Third-party SIP VoIP Operations	16
V.	The DOCSIS-QoS Problem	17
	A. Only an E-MTA has access to DOCSIS QoS	18
	B. The RSVP DOCSIS-QoS control protocol is not used.....	19
	C. Only an E-MTA has priority access to E911/911 emergency service during congestion	20
VI.	Course of Action	22
	A. Third-party VoIP providers use broadband cable operator’s PacketCable E-MTAs and service	23
	B. Third-party VoIP providers use the PacketCable Multimedia architecture and protocol specification.....	24
VII.	Discussion Items	25
	A. Telephone service quality expectations	25
	B. PacketCable Multimedia.....	26
	C. Section 706 relevance	27
VIII.	Conclusion	29

I. Introduction

Considerable advances in the areas of telephony and Internet access over broadband cable TV systems have been made recently but problems remain in moving forward to solutions that will work well for these and other IP-enabled applications operating over the Internet. In this paper, part of a broader research effort concerning issues in VoIP, we indicate what some of these problems are, show that they are impediments and suggest solutions. These range from proposing the adoption of new service quality enhancing mechanisms in the access network to modification to FCC oversight of broadband service proliferation.

Section II provides background for cable-based services and gives some relevant definitions and context. Section III covers DOCSIS and PacketCable 1.x operations as a telephony platform. Section IV briefly addresses third-party operation of VoIP using SIP techniques. Section V presents problems with current and planned cable broadband access regarding the use of Quality of Service features. Section VI gives some workable alternatives for moving toward a solution. Section VII presents some important points for discussion and follow-on activities that are closely related to the problems and solutions covered. Section VIII concludes with a reiteration of major points.

II. Cable Broadband and Telephony Services

This section introduces a high level logical progression of cable broadband service types and motivations. High speed data services and several forms of cable

telephony are covered as well as definitions used for network components and Quality of Service.

A. MSO High Speed Data access

Cable TV system operators, also known as MSOs,¹ have been offering IP² (Internet Protocol) broadband³ or high speed data access to the Internet in addition to television signals on their HFC (Hybrid Fiber/Coax)⁴ systems since the mid to late 1990s. Although initially only proprietary solutions were available, these MSOs now typically deploy equipment based on the widely adopted DOCSIS⁵ standard from CableLabs⁶ for subscriber cable modems⁷ (CMs) and associated cable system headend⁸ equipment

¹ MSO means Multiple System Operator. It is a loosely applied term that may refer to an operator of multiple cable TV systems or to an operator of a single system.

² IP means Internet Protocol described in RFC 791 (and subsequently updated in other RFCs), September, 1981, and is the IETF standard for packet switched networks. It forms part of the well-known TCP/IP suite of communications protocols used on the Internet and elsewhere. It is available at <http://www.ietf.org/rfc/rfc0791.txt>.

³ The term “broadband” typically refers to high speed Internet access over cable networks, DSL facilities and other platforms. The term is used less precisely in this paper. The current US government definition says that 200 kbps in both upstream and downstream directions is the threshold. *See In re Inquiry Concerning the Deployment of Advanced Telecommunications Capability to All Americans in a Reasonable and Timely Fashion, and Possible Steps to Accelerate Such Deployment Pursuant to Section 706 of the Telecommunications Act of 1996*, CC Docket No. 98-146, Second Report, FCC 00-290 (rel. Aug. 21, 2000) at § 8.

⁴ HFC refers to Hybrid Fiber/Coax distribution plant. It is made up of both optical fiber and metallic coaxial cable components.

⁵ DOCSIS is the *Data Over Cable System Interface Specification* produced as a suite of documents by CableLabs. The Radio Frequency Interface specification is SP-RFIV1.1-I10-030730 available at <http://www.cablemodem.com>. There are currently three versions: 1.0, 1.1 and 2.0. Version 1.0 supports only cable modem operation. Versions 1.1 and 2.0 support PacketCable 1.x telephony as well as other applications. CableLabs has a testing and certification program for CMTSs and CMs.

⁶ CableLabs is Cable Television Laboratories, Inc., in Louisville, Colorado. It is a research organization funded by member companies in the cable TV industry. CableLabs® is copyrighted by Cable Television Laboratories. *See* www.cablelabs.com for further information.

⁷ Cable modems are communications interface devices (analogous to telephony dialup modems) that provide subscriber access to a broadband cable access network.

⁸ A headend is a signal distribution location in a cable TV system. A cable TV system headend houses television transmission equipment including modulators, amplifiers, optical/electrical converters and equipment for services such as high speed Internet access and telephone service. Such a location might serve an entire city or a smaller portion and generally has its electrical power backed up with generators and batteries.

known as cable modem termination systems (CMTSs)⁹. Access speeds vary, but a typical system will provide at least 1 Mbps service. The best-effort¹⁰ support provided is appropriate for bursty, irregular data transmission such as web surfing, file sharing and other applications without time sensitivity. Approximately 18 million CMs have been installed in the US alone as of 2003.¹¹ Most MSOs provide cable modem service on their access networks.

B. MSO circuit switched cable telephony

In initiatives separate from cable modem access some MSOs have deployed proprietary circuit switched telephony services over their HFC plant as facilities-based competitive local exchange carriers (CLECs)¹² since the Telecommunications Act of 1996 (“Telecom Act”)¹³ allowed entry into this line of business. Although a digital technology, there is no DOCSIS or packet switched or Internet component in the architecture and it is assigned its own bandwidth completely separate from what is assigned to cable modem operation on the HFC plant.¹⁴ It is intended to be fully equivalent to the service available from incumbent copper loop local exchange carriers

⁹ CMTS means Cable Modem Termination System and is the headend device that controls a population of DOCSIS modems on an HFC node.

¹⁰ The term “best-effort” refers to the typically available mode of operation on packet switched networks using the IP protocol. Best-effort implies that the network merely forwards a packet of data to a destination without any sort of guarantee of arrival and no explicit indication as to whether it arrived. This is often also called “datagram” service. Additionally, and of considerable consequence to voice services and other realtime applications such as videoconferencing, there is no attention paid to maintaining any sort of time or ordering relationship from source to destination. Quality of Service mechanisms address these requirements.

¹¹ See <http://www.cablemodem.com> for background information.

¹² CLEC means Competitive Local Exchange Carrier. These are telephone carriers that compete with the incumbent LEC or ILEC. This is part of the Public Switched Telephone Network (PSTN).

¹³ Telecommunications Act of 1996, Pub. L. No. 104-104, 110 Stat. 56 (codified in various sections of the Communications Act of 1934, 47 U.S.C.).

¹⁴ Headend equipment consists of an HDT or Host Digital Terminal supporting several hundred subscriber NIDs. This then connects to traditional circuit switched transmission components and from there to a telephony switch which provides dial tone and other telephony services.

(LECs)¹⁵. This includes fulfilling subscriber expectations of PSTN-equivalent primary line¹⁶ service with high availability¹⁷ and sound quality, assignment of a conventional telephone number, access to E911/911¹⁸ services and comparable operation when using dialup modem, fax and TDD¹⁹ devices. Service is delivered to homes via a coaxial cable feed to a network interface device (NID)²⁰ installed on the outside of the residence and uses the same inside wiring and telephone sets as with the LEC service. MSOs have installed approximately 2 million lines²¹ to date in the US using this technology.

C. MSO packet switched cable telephony

Interest in a packet switched replacement for this digital circuit switched service technology has varied over the last several years but has increased recently. There is now great motivation to migrate from the obsolescent circuit switched telephony technology platform to packet mode architectures that converge voice and data operations on a single integrated access network platform. Increased efficiencies based on bandwidth conservation and hardware reduction are major drivers for this. Expressly for the cable TV community, CableLabs developed the PacketCable²² 1.x system architecture and suite

¹⁵ LEC means Local Exchange Carrier. Services offered can include speed dial, CallerID, three-way calling, voicemail and many more.

¹⁶ Primary line is a term that connotes operation as the primary or possibly only means of telephone communications a subscriber might have. It implies the highest levels of quality and availability.

¹⁷ Availability for telephone service from the PSTN is generally specified as five-nines or 99.999% uptime. This design level corresponds to a down time of approximately 5 minutes per year.

¹⁸ E911/911 refers to emergency services accessible in a local area.

¹⁹ TDD means Telecommunications Device for the Deaf. Sometimes called TTY, it is a protocol used between user terminals that are effectively low speed electronic typewriters that communicate with signals analogous to those used by dialup modems.

²⁰ NID means Network Interface Device and is an electronic device that converts signals on the coax cable to signals appropriate for use with a user's telephone.

²¹ See <http://zdnet.com.com/2100-1103-1008962.html> "Time Warner Cable dials in phone service," May 22, 2003.

²² PacketCable1.x is an initiative of CableLabs begun in the late 1990s. It defines a full architecture and suite of specifications to support PSTN-equivalent telephony service operations independent of the existing PSTN. PacketCable™ is a trademark of CableLabs. For documentation, see <http://www.packetcable.com>.

of telephony oriented, packet switched communication protocols with an accompanying product testing and certification service. PacketCable 1.x, as an application, makes use of the underlying DOCSIS base and is designed to operate over MSO HFC systems in very much the same way as the existing circuit switched service from the perspective of providing a primary line service equivalent to PSTN levels of high quality and availability, assignment of a conventional telephone number, access to E911/911 services and comparable operation when using dialup modem, fax and TDD devices. Its architecture is generally referred to as NCS (Network-based Call Signaling)²³ and the interface box attached to the side of a subscriber's residence (or, alternatively, located indoors) is called an MTA (Multimedia Terminal Adapter).²⁴ To be fully workable, the next generation of DOCSIS (version 1.1) is also specified as a requirement and many MSOs have performed the upgrade to their equipment in anticipation of supporting PacketCable1.x systems. PacketCable 1.x equipment implementations with the DOCSIS 1.1 base, which provides so-called DOCSIS-QoS (Quality of Service) capabilities to handle the real time demands of voice traffic, are currently in limited trial deployment on several cable systems.²⁵ On these systems, new PacketCable 1.x telephony services will be offered along with broadband cable modem Internet access. PacketCable 1.x systems are designed to provide one form of carrier grade (PSTN quality and availability) service

²³ NCS means Network-based Call Signaling and is a network centric control signaling technique. The term is used to differentiate it from an alternative PacketCable1.x technique called DCS or Distributed Control Signaling. Only NCS is currently in use.

²⁴ The MTA or Media Terminal Adapter performs functions equivalent to the circuit switched platform's NID. Like the NID, it is available in weatherproof enclosures for outdoor mounting. Indoor models are also available and look very much like a cable modem. MTAs typically can have more than one telephone line connection and can have Ethernet connectors as well for high speed data access applications.

²⁵ Cox Communications, Cablevision Systems and Comcast have announced their intentions to operate VoIP trials in 2004. Comcast will go to Indianapolis, Springfield, MA, and Hartford, CT, in 2004. See <http://www.voip-news.com/mso.html> for further details. (last visited August 29, 2004)

offering of what is generally referred to as Voice over IP or VoIP and will be called PacketCable or NCS VoIP in this paper.

D. Third-party VoIP

Independent of PacketCable 1.x, other individuals and groups, without carrier grade telephony service necessarily in mind,²⁶ have developed techniques to send voice transmissions over IP networks of varying technology with varying degrees of success over the last several years. Some use software executing on a user's computer to send and receive packetized speech with the use of a microphone and headphones or speakers. Such a user's computer is connected to a cable modem on an MSO's broadband access network in the present context. Users find out about each other and their immediate availability on web based servers and can set up connections with other users typically using a protocol called SIP²⁷ (Session Initiation Protocol). Net2Phone²⁸ and Free World Dialup²⁹ are examples of this mode of operation and there are some based on services such as AOL³⁰, Microsoft³¹ and others. Making connections is typically cost-free and sound quality can be good but, especially on cable broadband access networks, is subject

²⁶ See http://techupdate.zdnet.com/techupdate/stories/main/Why_Vonage_Just_Fad.html "Why Vonage is just a fad," May 19, 2004. (Last visited August 28, 2004.)

²⁷ SIP means Session Initiation Protocol and is now considered the leading protocol for VoIP call signaling. It can be found at www.ietf.org/rfc/rfc3261.txt.

²⁸ Net2Phone provides a VoIP service that operates on a computer. See www.net2phone.com for further information. (Last visited August 29, 2004.)

²⁹ Free World Dialup or FWD is a VoIP service provided by pulver.com. See www.pulver.com for further details. (Last visited August 29, 2004.)

³⁰ AOL means America Online, Inc. It offers a proprietary, free browser-based service called Instant Messenger which can be used to communicate between end users at computers with either text, voice or video. See www.aol.com for further details. (Last visited August 29, 2004.)

³¹ Microsoft offers a proprietary, free browser-based service called MSN Messenger. It can be used to communicate between end users at computers with either text, voice or video. See www.msn.com for further details. (Last visited August 29, 2004.)

to the effects of random packet loss and other congestion related mechanisms based on best-effort packet handling.

A related but much more recent type of entry into this arena is similar to the computer-based voice services just described but is more complex. A subscriber acquires an interface box (often called an Analog Telephony Adapter or ATA) from a third-party provider (such as AT&T CallVantage,³² Vonage³³ or others), plugs it into an existing cable modem, then plugs a typical telephone into the box and gets a form of telephone service that uses SIP (or similar) call control mechanisms³⁴ and best-effort packet handling, as opposed to PacketCable 1.x mechanisms as on a cable system. A conventional telephone number is assigned to the subscription and it can be called from any telephone on the PSTN making it superficially very similar to LEC telephone service. These modes of operation, with and without a separate interface device, are also generally referred to as VoIP and will be called SIP VoIP in this paper.

E. Internet Backbone and HFC Broadband Access

For the purposes of this paper, we simplistically divide the Internet into two major parts: the backbone network that supplies long haul connectivity and the access networks or “last mile” facilities supplied by MSOs, LECs, wireless providers and others. Except to include them for contrast and completeness, we will not discuss the operation of backbone networks in detail.

³² AT&T provides IP-enabled telephony service over broadband access. See <http://www.usa.att.com/callvantage/index.jsp?> for more information. (Last visited August 30, 2004.)

³³ Vonage provides IP-enabled telephony service over broadband access. See <http://www.vonage.com>. (Last visited August 29, 2004.)

³⁴ Such services are designed to operate over any broadband access technology.

In a cable broadband architecture for a small city-sized system, the access network is made up of the headend and its equipment, the HFC plant, consisting of many segments of bi-directional³⁵ fiber and coax cable and active components such as amplifiers serving residential neighborhoods of typically 1000 to 1500 homes passed. It is not uncommon for a system to have a hundred such segments. The data traffic on these segments, also called fiber nodes,³⁶ is controlled by a CMTS that uses DOCSIS procedures for managing the possibly several hundred attached subscriber cable modems. The important operational characteristic is that the downstream direction of transmission (CMTS to subscriber) is independent of the upstream direction of transmission (subscriber to CMTS). Due to well-entrenched HFC bandplans,³⁷ or bandwidth allocation decisions, the downstream direction typically has enough bandwidth to easily handle the data needs of all the simultaneously active subscribers but the upstream direction has a very limited amount of bandwidth that must be shared on a competitive basis among all cable modems with data to send. Each HFC node can be considered apart from the others in a system such that each is its own bandwidth commons. It is the bandwidth contention and its effects that are fundamental to the arguments in this paper.

F. Quality of Service Definitions

In modern telecommunications, especially when packet mode operation is involved, the term “Quality of Service” or QoS is confusingly used to mean more than

³⁵ Modern HFC outside plant is designed to carry signal flows in both the downstream or outbound direction from the headend to subscribers and the upstream or inbound direction from subscribers to the headend.

³⁶ A fiber optic node is a piece of outside plant equipment that performs a conversion between fiber optic transmission and electrical transmission on coaxial cable segments.

³⁷ The use of the very commonly seen, highly asymmetric sub-split arrangement where only about 30-40 MHz of usable upstream bandwidth is available is based on the position of TV channel 2 which begins at 52 MHz in the downstream direction. This limited upstream bandwidth must also be shared with any other services using upstream communications.

one quality-oriented concept, depending on the exact context. In the general case regarding VoIP, it is taken to mean a constellation of telephony service factors or metrics including service availability, customer service response time, sound quality, dial tone delay, E911/911 availability and other somewhat qualitative concepts. The term “telephony-QoS” will be used to mean these broader telephony service-oriented features and capabilities.

In packet switched or IP operation, however, Quality of Service refers narrowly and specifically to the ability of packet handling equipment to deliver streams of packets with engineered levels of packet loss, delay, delay variation and, possibly, prioritization. Delay variation is extremely important to a real-time, streaming application (such as voice) where packets are sent on a regularly recurring basis and must arrive in that same time-sensitive pattern. The term “IP-QoS” will be used to refer to the packet flow controlling mechanisms in packet handling equipment and has applicability in all parts of a network.

Another term used extensively in this paper, “DOCSIS-QoS,” refers to the capabilities in the DOCSIS 1.1 transport mechanisms that underlie packet transmission only on broadband cable access networks. It is these mechanisms dealing with bandwidth allocation, delay variation and prioritization that allow the well-regulated flow of voice packets interleaved with other arbitrary data packets in the downstream direction as well as the upstream TDMA³⁸ (Time Division Multiple Access) on the HFC plant between subscribers and the headend. Both IP-QoS and DOCSIS-QoS have an impact on aspects of telephony-QoS.

³⁸ Time Division Multiple Access is a technique for management of the use of a communications medium by multiple users. It can support various disciplines for allocating access equitably and predictably or operations in contention mode.

DQoS or Dynamic Quality of Service refers to a specific mechanism within the PacketCable 1.x specification used to control DOCSIS-QoS. It is a term used in this paper, but is not in general use. PQoS or Proxied Quality of Service refers to a specific mechanism within the PacketCable MultiMedia specification. It is intended to be used to control DOCSIS-QoS. As with DQoS, it is not in general use.

III.DOCSIS and PacketCable Operations

This section gives some deeper details on the technology of broadband cable access networks and components used for telephony in the PacketCable 1.x model.

A. The CMTS and DOCSIS

The broadband access equipment used on an HFC access network³⁹ consists of a single “master” controller, situated in a cable system headend and potentially hundreds of “slave” devices at subscriber locations. This is repeated for each segment or node fed by a headend. The slave devices are the cable modems (CMs) and the master controller is called the cable modem termination system or CMTS. The CMTS and CMs are built according to the cable industry’s standardized DOCSIS specification developed at Cable Labs. The DOCSIS protocol has a position at the data link layer (layer 2 of the OSI communications model) and at the physical layer (OSI layer 1) and describes all of the details of operation necessary to support high speed data transmission over the miles of a fiber and coaxial cable plant. From a data networking perspective, this is somewhat like

³⁹ Cable access networks are generally built in such a way that most of the available bandwidth is used in the so-called “forward” or “downstream” direction and carries dozens of TV channels, as well as high speed data signals and telephone calls, to subscribers. They also have a small portion of bandwidth devoted to carrying signals in the “reverse” or “upstream” direction. This is used for various purposes such as for the management of network devices, pay per view movie selection, high speed data such as web surfing commands and telephone calls. The upstream bandwidth capacity is a scarce resource.

a collection of large Ethernet LAN segments but with some very important security and traffic handling distinctions to allow operation as a shared public access network.⁴⁰ A system built from a DOCSIS 1.0 CMTS and DOCSIS 1.0 CMs provides best-effort access only and is appropriate for web browsing, etc. A system built from a DOCSIS 1.1 CMTS and DOCSIS 1.1 CMs additionally contains Quality of Service mechanisms (called DOCSIS-QoS in this context) used to manage the flow of streams of packets requiring different priorities and transmission rates. A DOCSIS 1.1 system can operate simultaneously in best-effort and QoS modes so that it can support traditional web surfing traffic from users with CMs as well as stream oriented traffic such as PacketCable 1.x telephony users with MTAs.

A CMTS that incorporates DOCSIS 1.1 mechanisms and is used for combined voice and data operation will typically have the bandwidth capacity it manages allocated in two separate pools, one for general or best-effort use, and the other for QoS traffic. The portion designated for PacketCable 1.x telephony makes use of DOCSIS-QoS to ensure a managed flow of voice packets for each individual telephone call.

B. PacketCable

The CableLabs PacketCable1.x initiative defines an architecture and a suite of specifications for implementing PSTN-quality telephony as well as video and multimedia applications over broadband cable access networks although only telephony has received any meaningful development to date. The architecture includes components required to construct a packet based telephony service analogous in operation to the circuit switched

⁴⁰ LAN stands for local area network, a term used to describe a collection of computers, wiring and network equipment. Ethernet is an example of a LAN technology.

model and so includes mechanisms for softswitch-based call management,⁴¹ high quality speech, security, record keeping, support for the Communications Assistance to Law Enforcement Act (CALEA)⁴² and other features. Its configuration includes the use of the DOCSIS 1.1 communication specification and the use of DOCSIS-QoS mechanisms in support of real-time voice streams requiring differential treatment from other data traffic handled by a best-effort approach. The major goal of the initiative is to provide a framework for building a broadband cable-based, packet-mode telephony service equivalent to what the PSTN offers today.

1. E-MTA

The subscriber location equipment used to provide PacketCable 1.x telephony service is known as a Multimedia Terminal Adapter or MTA. The main function and challenge for an MTA is to provide a reliable telephony experience and to operate over an asynchronous, lossy packet network of unpredictable characteristics in conjunction with a distant MTA to provide an audio channel that mimics a PSTN-quality circuit.⁴³ The use of DOCSIS-QoS mechanisms improves an MTA's ability to do this.

The E-MTA (Embedded MTA) consists of MTA hardware and software embedded in the same chassis with a DOCSIS 1.1 CM. The two sub-units are tightly coupled internally from both a hardware and software perspective in such a way that a

⁴¹ A softswitch is a call management server and occupies a position in packet-mode telephony analogous to a circuit switch in the conventional telephony architecture.

⁴² CALEA means Communications Assistance to Law Enforcement Act. It refers to the requirement to aid law enforcement in performing legal wiretapping. *See* 47 U.S.C. §§ 1001-1021 (2000). *Also see* 16 F.C.C.R 17,397 (2001) for the applicability of CALEA to packet switched communications, where it is discussed that the way in which this will operate remains unclear.

⁴³ There are several mechanisms an MTA sender or receiver can use to give the impression that a high quality circuit is in use. These include receive packet buffering, packet loss concealment, interpolation and others beyond the scope of this paper. *See* Hardy, W., "VoIP Service Quality: Measuring and Evaluating Packet-Switched Voice," McGraw-Hill 2003 ("Hardy 2003") at various places.

subscriber has no access to the interface between them. The MTA sub-unit can communicate with the CM sub-unit using the full range of mechanisms as described in the DOCSIS 1.1 specification. Most notably, this includes the capability of selecting DOCSIS-QoS parameter settings⁴⁴ that will support a real time voice packet flow. This is accomplished by means of a PacketCable 1.x mechanism called dynamic QoS or DQoS. The two mechanisms operating together support the selection of the amount of guaranteed bandwidth needed for a telephone call as well as the amount of tolerable packet delay and jitter.⁴⁵ The E-MTA has a standard telephone jack where a standard telephone set can be connected by a subscriber and in many cases also has a 10BaseT or USB jack where a subscriber can connect a computer for a high speed data service. A current cable system operator business model for offering PacketCable 1.x VoIP is to provide the E-MTA as part of the telephony service agreement. An E-MTA is registered to operate on the system that provided it and cannot be used on another system.

2. S-MTA

The PacketCable 1.x S-MTA (Stand-alone MTA) is designed to be separate from a DOCSIS 1.1 cable modem and to connect to one in the same way a personal computer would, that is, by plugging into its 10BaseT or USB interface. According to the PacketCable DQoS specifications, an S-MTA communicates DOCSIS-QoS requests

⁴⁴ The DOCSIS 1.1 UGS or Unsolicited Grant Service is the mechanism put into place to support time sensitive flows.

⁴⁵ See Hardy 2003. An important packet telephony design principle that can be extended to the handling of other media is that of maximizing the evenness of generation of packets at the sender so as to minimize the processing required at the receiver. Buffering at the receiver to compensate for delayed or lost packets adds unwanted delay to the end-to-end operation of a real time communications service. Additionally, it is advantageous, from an end-to-end sound quality perspective, to produce a voice packet stream that is as well-formed and well-paced as possible since other equipment in a call path will act to degrade the time-integrity of the stream cumulatively.

using the RSVP⁴⁶ protocol. The existence of this alternate mode of operation for requesting DOCSIS-QoS has potential implications for the operation of third-party SIP VoIP and for other applications requiring its availability. This technique appears, however, not to have been adopted by the PacketCable 1.x community and may be deprecated.⁴⁷

IV. Third-party SIP VoIP Operations

The implementations of the current third-party SIP VoIP entrants operate in widely differing ways and at varying levels of overall expected service quality. A PSTN level of telephony speech quality may not even be envisioned by all entities claiming to be telephony service providers positioning themselves as alternatives for existing PSTN service. Given this, some SIP VoIP entrants may not be at all concerned about having access only to best-effort operation.

However, the creation and operation by a SIP VoIP entrant of a PSTN level telephony offering as an IP-enabled service available via cable broadband access brings with it several issues. Since best-effort packet mode operation on such networks has inherent characteristics that make it potentially unsuitable for high quality speech transmission without some use of mitigating techniques⁴⁸ in the subscriber interface devices, there may be a need or desire for DOCSIS-QoS access.

⁴⁶ RSVP is the Resource Reservation protocol and is described in the IETF document RFC 2205 available at the Internet Engineering Task Force's website at <http://www.ietf.org>.

⁴⁷ See Evans, D.R., "Digital Telephony Over Cable: The PacketCable Network," 2001, Addison-Wesley at 291 (explaining the use of DQoS and its use in E-MTAs for DOCSIS-QoS control as well as the unlikely use of RSVP as an alternate control mechanism.)

⁴⁸ These include packet loss concealment, jitter buffers and others, all of which impose delay on the voice packet stream. See <http://global.mci.com/us/enterprise/insight/whitepapers/voice/> for a clear explanation of the issues of operating voice over a packet switched substrate. (Last visited August 25, 2004.)

The Analog Terminal Adapter used as the telephony subscriber interface equipment in some SIP VoIP systems occupies a position architecturally equivalent to the S-MTA of PacketCable 1.x. The whole SIP VoIP model of access platform independence and portability of the ATA⁴⁹ to any Internet access port is a fundamental differentiator from the PacketCable 1.x model.

IP-enabled services, however, cannot make use of the QoS voice traffic handling capabilities that the PacketCable 1.x plus DOCSIS 1.1 platform provides and therefore are subject to potentially lesser levels of transmission or service quality. Note that this is also true in the general case for other arbitrary IP-enabled services.

V. The DOCSIS-QoS Problem

Telephony-QoS depends on the layers of functionality below it. If the lower layer mechanisms are not implemented or are disabled or unavailable, the upper layers will feel the effects and users will experience the results. There are concerns with the current specification and implementation of DOCSIS-QoS on broadband cable access networks that they can fully support only E-MTA operation and not S-MTA operation. The E-MTA is designed to make use of the DOCSIS-QoS mechanisms necessary for setting up and managing a continuous stream of packets containing voice samples as they are produced.

⁴⁹ An ATA user can connect the unit anywhere in the world where there is an Ethernet port to an Internet access provider. The subscriber's telephone number and features move with the unit.

We present three service affecting items concerning access to service quality mechanisms and E911/911⁵⁰ availability, which indicate that cable broadband access networks are not presently suitable for use by third-party devices or services requiring Quality of Service packet handling.

A. Only an E-MTA has access to DOCSIS QoS

First item:

This excerpt from the PacketCable DQoS specification describing the interface between the MTA and the CM highlights a major problem for any user not using an E-MTA who wants to make use of the data link layer QoS mechanisms in DOCSIS:

This interface⁵¹ is only defined for the embedded MTA. The interface decomposes into three sub-interfaces:

- Control: used to manage DOCSIS service-flows and their associated QoS traffic parameters and classification rules.
- Synchronization: used to synchronize packetization and scheduling for minimizing latency and jitter.
- Transport: used to process packets in the media stream and perform appropriate per-packet QoS processing.

This interface is conceptually defined in Appendix E of the DOCSIS RFI specification. *For standalone MTAs no instance of this interface is defined.*⁵²

⁵⁰ See Hatfield, D., "A Report on Technical and Operational Issues Impacting the Provision of Wireless Enhanced 911 Services," 2002 at 41 (clarifying concerns regarding commonality in the wireless broadband and wired broadband realms with respect to dealing with VoIP and its use of SIP as an interface protocol and the complexities of dealing with the mobility of end stations.) available at http://gullfoss2.fcc.gov/prod/ecfs/retrieve.cgi?native_or_pdf=pdf&id_document=6513296239.

⁵¹ The interface described here refers to the messaging protocol between an MTA and a CM.

⁵² See PKT-SP-DQOS-I08-040113 § 2.4. This document is available at www.packetcable.com. Emphasis added.

The last sentence states that even though the required functionality to make use of DOCSIS-QoS is defined in DOCSIS 1.1, it can be accessed only by a PacketCable E-MTA (that is, a unit built as a combined MTA and CM). An S-MTA or any other application device such as a SIP VoIP ATA connecting to a CM has no way to invoke DOCSIS-QoS mechanisms.

A VoIP call's packet stream should be managed in some way on all segments of its end-to-end traversal across multiple networks although error mitigation techniques at the end points might improve quality in some cases. As with all forms of network technology, the local or last mile segments have the greatest susceptibility to impairment, and thus have the greatest need for such bandwidth and delivery management to be available and used. Further, the worst bandwidth scarcity location on a cable system is in the upstream part of the access network. Relying on network best-effort performance characteristics to stay the same under all load conditions and as average usage grows is unrealistic.⁵³

This limitation disadvantages third-party SIP VoIP service providers in particular since only the incumbent makes use of E-MTAs. This situation disallows access to levels of packet handling mechanisms available to the incumbent that have a direct impact on speech quality.

B. The RSVP DOCSIS-QoS control protocol is not used

Second item:

⁵³ See Bricklin, D., "Why We Don't Need QoS: Trains, Cars, and Internet Quality of Service" at <http://www.bricklin.com/qos.htm> for a position against the need to use QoS to ensure quality. While this may be true in limited cases, if the situation is one of multiple, independent, concatenated network paths between a sender and receiver, there is no guarantee that all equipment will not be experiencing congestion at the same time for packets along such paths.

Even though there is an alternative mechanism defined in the PacketCable DQoS specification based on the Reservation Protocol (RSVP) that would allow an S-MTA or other application device to control DOCSIS-QoS mechanisms, it has not been implemented on any equipment. As a result there are no CMs, CMTSs or MTAs being built to support the use of this mechanism.

This limitation is an impairment to third-party service providers since it could be used to support access to DOCSIS-QoS.

C. Only an E-MTA has priority access to E911/911 emergency service during congestion

Third item:

A typical CMTS is designed to allow allocation of separate pools of usable bandwidth to best-effort use and to real-time flows. Additionally, the bandwidth allocated to real-time flows can be further managed to a fine degree to support emergency calls in the event of congestion. An S-MTA or other third-party application device restricted to using only the best-effort allocation will not be able to use any of the controlled real-time flow prioritization mechanisms. In the following excerpt from the PacketCable DQoS specification, admission control policy refers to how bandwidth for telephone calls is managed:

The CMTS MUST implement at least two admission control policies, one for normal voice communications and one for emergency communications.⁵⁴

⁵⁴ See PKT-SP-DQOS-I08-040113 § 5.1.4. This document is available at www.packetcable.com.

Having a requirement for at least two admission control policies indicates that such a prioritization mechanism will be available in any CMTS and will, in fact, be accessible when an emergency call is attempted. The text goes on to say:

The admission control policy MAY also specify whether a new session of that type may “borrow” from lower priority classes or should preempt an existing session of some other type to satisfy the admission control policy settings.⁵⁵

This indicates that under heavily loaded conditions an emergency call can preempt an existing normal call and be assigned its bandwidth. This cannot happen with third-party SIP VoIP or similar devices using a best-effort packet service since there is no concept of prioritization within the best-effort bandwidth pool. There is no knowledge in the CMTS that a call is even in progress or being attempted since the packets are in no way differentiated from any other data packets. Thus, if the best-effort allocation is congested, it may not even be possible to make a call or to have enough bandwidth to carry on an intelligible conversation.

Denial of access to these call prioritization mechanisms due to denial of access to QoS mechanisms is an impairment to third-party service providers since it denies access to the mechanisms that incumbents have access to for controlling E911/911 call prioritization.

⁵⁵ *Id.*

VI. Course of Action

In March, 2004, the FCC released a Notice of Proposed Rulemaking (NPRM)⁵⁶ for IP-Enabled Services and announced the formation of several so-called Solution Summits to address the important operational social issues surrounding the use of the Internet for telephony services. These include the operation of E911/911, access for the disabled, support for the needs of the Communications Assistance to Law Enforcement (CALEA) and support for the Universal Service Fund (USF). New VoIP entrants with interconnection to the existing PSTN are expected to operate with these requirements. It could be argued that if it is the FCC's expectation that such third-party SIP VoIP providers are to be subject to Telecom Act Title I, Title II or even to more recent IP-Enabled Services oversight, then those entities should be given the same rights, privileges and use of technical mechanisms given to other participants and entrants including incumbent cable providers offering telephony service.

The most workable solution regarding non-discriminatory DOCSIS-QoS access should be found to support this. There is no obvious way to use telephony-QoS, an application layer concept, or a contractual service level agreement between a broadband cable operator and a third-party SIP VoIP provider to compensate for the lack of an enabling QoS function at a lower layer. The only way to address this problem is to establish a mechanism for third-party access to data link layer DOCSIS-QoS. There are several possible ways to address this problem. Each differs in how the cable operator and the third-party VoIP company would work together to offer services and all are not

⁵⁶ See FCC 04-28 WC Docket No. 04-36. *In re* IP-Enabled Services, hereinafter referred to as "IP-Enabled NPRM")

identified here. While other solutions may exist, here we present two very different approaches:

A. Third-party VoIP providers use broadband cable operator's PacketCable E-MTAs and service

Using a cable operator's E-MTAs, third-party VoIP providers' customers would have a telephony user experience equivalent to that of the service from the incumbent cable operator offering telephony service. There could be several variants to this alternative, and they would require a third-party provider to in some way base a service on a broadband cable operator's PacketCable 1.x VoIP switching equipment.

Additionally, it would imply a service resale option or a possible implementation of a conventional telephony-like unbundled network element regime in the broadband cable operator space analogous to what is described in the Telecom Act, Section 251.

Depending on how the rules were designed, advantages might include: 1) ease of entry and deployment for a new entrant, 2) efficiency of incumbent network plant use, and 3) solution to all QoS-related access issues raised.

The disadvantages to this line of thinking are many. They include: 1) This would most likely open a long term line of inquiry and rule making and run counter to the principle of less regulation in the IP-enabled services space, not more. 2) Since, unlike the Ethernet-connected S-MTA (or, similarly, SIP VoIP ATA) concept where such units are plugged into arbitrary existing installed cable modems, E-MTAs are not so flexible in their operation; they are not Ethernet devices on their network side. Requiring their use by third-party providers would destroy the attractive mobility and flexibility of the services currently offered by such third-party providers. 3) The complications and

restrictions on business models with respect to rules for competition, pricing, operations, management, billing and more would be prohibitive. 4) Since single purpose devices (E-MTAs) would be used, such an approach would be restricted to only the VoIP application and would not address the more general multimedia services that also require access to DOCSIS-QoS mechanisms for proper operation at high quality levels. It is a narrow solution at best.

This set of possibilities may be of interest to some to pursue, but we make no further exploration in this paper.

B. Third-party VoIP providers use the PacketCable Multimedia architecture and protocol specification

The CableLabs specification, PacketCable Multimedia⁵⁷ (PCMM), is very new but promises to allow a leap in functionality and flexibility for both SIP VoIP and other multimedia applications that need the bandwidth on demand and Quality of Service mode of operation. This next stage in the CableLabs PacketCable initiative uses an additional set of headend equipment beyond the standard PacketCable 1.x devices and a technique called proxied QoS (PQoS) to allow a user to set the speed and DOCSIS-QoS settings as required at any time while connected to a standard DOCSIS 1.1 cable modem.

Advantages to this approach include: 1) It supports SIP VoIP as well as general packet speed and delivery sensitive multimedia applications. 2) User equipment and applications need not be telephony oriented and need not use an incumbent's installed PacketCable 1.x switching equipment. 3) SIP devices such as S-MTAs or SIP VoIP ATAs can be readily accommodated with no modification. 4) CALEA requirements for

⁵⁷ See PKT-TR-MM-ARCH-V01-030627 available at <http://www.packetcable.com>.

SIP VoIP calls could potentially also be met since the proxy server sets up circuits on the CMTS in the same way as with PacketCable 1.x calls. 5) PCMM is an easily scalable architecture that provides a migration path for existing PacketCable 1.x deployments that can coexist with SIP VoIP and other services.

The disadvantages include: 1) This is a new specification and only limited vendor devices with no track record are available at this time. 2) Interoperability testing is happening, but no certification program is in place at CableLabs. 3) Third-party service providers must contract with an MSO operating PCMM in order to have DOCSIS-QoS invoked for their subscribers using that application.

VII. Discussion Items

The research for this paper turned up more questions than answers. This section contains some items for further research and discussion for their technical as well as policy implications.

A. Telephone service quality expectations

Historically, telephone service has been held up to the highest standards with descriptive terms such as PSTN or toll quality, carrier grade, five nines availability and others. Looking at the level of quality designed into the PacketCable 1.x version of VoIP as a PSTN equivalent and then at what might be expected from a third-party SIP-VoIP entrant we wonder if some telephony user expectations have, in fact, been lowered. From a policy implications perspective, we wonder if there can be a relaxation or resetting of

quality thresholds for telephony in general or only if a reasonable trade-off such as price or mobility applies.⁵⁸

B. PacketCable Multimedia

According to the technical documentation and some trade press coverage,⁵⁹ the new PacketCable Multimedia architecture from CableLabs is the right thing to use to take cable broadband access networks forward. PacketCable 1.x covers only telephony whereas PCMM provides mechanisms to support any application that requires Quality of Service packet handling and bandwidth on demand for long or short time periods. Although things will change later during the course of evaluation, in PCMM's early phases, it will work with client devices, for example, game consoles, with no change at all to those devices. All the intelligence to set QoS values will be in new devices such as the PCMM policy server that performs the proxied⁶⁰ QoS function on behalf of these devices. Since these applications related servers are in the network and an application operator must contract with the network to enable proper operation for that service, this could be seen as a step in the direction of making the access networks more and more application aware and less and less transparent. Although access network operators stand to gain tremendously with such an arrangement since many applications will ultimately be controlled this way, it is unclear if this is a prudent direction in which to take the Internet.

⁵⁸ There is evidence that consumers are willing to trade service quality for price and mobility (e.g., wireless service); however, there remains the question of the consumer recognizing and understanding this trade-off, which is unlikely in many VoIP scenarios.

⁵⁹ See "10 Hottest Technologies for 2004 PacketCable Multimedia: Beyond Black Phones" <http://www.telecommagazine.com/default.asp?journalid=3&func=articles&page=0404t08&year=2004&month=4&srchexpr=pcmm#hls1>, April 2004. (Last visited August 28, 2004.)

⁶⁰ The use of the term "proxied" indicates that QoS is being controlled by a device other than the one that would ordinarily do it. Internet applications make widespread use of proxy servers for various needs.

C. Section 706 relevance

In looking at the issues surrounding access to DOCSIS QoS mechanisms on cable broadband access networks, other related areas of interest arise. Finding technical issues with specifications, such as those supporting PacketCable 1.x, that need to be addressed is one aspect of such an analysis. There is also a policy and process aspect that bears scrutiny. Given the national interest in broadband access and spreading such capabilities to all Americans, as reflected in Section 706 of the Telecom Act, it is interesting to see the focus of the periodic reviews of the state of broadband in the nation. Four cycles of Notice of Inquiry,⁶¹ comment and Report⁶² have occurred since 1996 and the major finding each time was that broadband access proliferation, as measured by access network speeds greater than 200 kbps on platforms such as cable, DSL, satellite and wireless, was on track.

⁶¹ For example, *See In re Inquiry Concerning the Deployment of Advanced Telecommunications Capability to All Americans in a Reasonable and Timely Fashion, and Possible Steps to Accelerate Such Deployment Pursuant to Section 706 of the Telecommunications Act of 1996*, CC Docket No. 98-146, Notice of Inquiry, FCC 00-57 (rel. Feb. 18, 2000).

⁶² For example, *See In re Inquiry Concerning the Deployment of Advanced Telecommunications Capability to All Americans in a Reasonable and Timely Fashion, and Possible Steps to Accelerate Such Deployment Pursuant to Section 706 of the Telecommunications Act of 1996*, CC Docket No. 98-146, Second Report, FCC 00-290 (rel. Aug. 21, 2000).

Given the language of Section 706⁶³ (“The Commission...shall encourage the deployment...of advanced telecommunications capability”, which is defined as enabling “users to originate and receive high-quality voice, data, graphics, and video telecommunications...”), it appears reasonable for the FCC to monitor the development of QoS and similar attributes of the network. The ability of broadband access networks to support the high quality, advanced services as explicitly stated in Section 706 will depend substantially on such advancements.

There is another matter to consider when assessing broadband network quality capabilities on a national scale. For example, with respect to voice services, where access network quality affects what is sent as well as what is received, a network

⁶³ The principal section of the 1996 Telecom Act concerning advanced telecommunications capability is Section 706, Pub. L. 104-104, Title VII, § 706, Feb. 8, 1996, 110 Stat.153, reproduced in the notes under 47 U.S.C. § 157. It provides:

SEC. 706. ADVANCED TELECOMMUNICATIONS INCENTIVES.

- (a) IN GENERAL-The Commission and each State commission with regulatory jurisdiction over telecommunications services shall encourage the deployment on a reasonable and timely basis of advanced telecommunications capability to all Americans (including, in particular, elementary and secondary schools and classrooms) by utilizing, in a manner consistent with the public interest, convenience, and necessity, price cap regulation, regulatory forbearance, measures that promote competition in the local telecommunications market, or other regulating methods that remove barriers to infrastructure investment.
- (b) INQUIRY- The Commission shall, within 30 months after the date of enactment of this Act, and regularly thereafter, initiate a notice of inquiry concerning the availability of advanced telecommunications capability to all Americans (including, in particular, elementary and secondary schools and classrooms) and shall complete the inquiry within 180 days after its initiation. In the inquiry, the Commission shall determine whether advanced telecommunications capability is being deployed to all Americans in a reasonable and timely fashion. If the Commission’s determination is negative, it shall take immediate action to accelerate deployment of such capability by removing barriers to infrastructure investment and by promoting competition in the telecommunications market.
- (c) DEFINITIONS- For purposes of this subsection:
 - (1) ADVANCED TELECOMMUNICATIONS CAPABILITY.—The term “advanced telecommunications capability” is defined, without regard to any transmission media or technology, as high-speed, switched, broadband telecommunications capability that enables users to originate and receive high-quality voice, data, graphics, and video telecommunications using any technology.

externality⁶⁴ component applies since the better the quality of sound a user can produce on one end the better the experience of the user at the remote end will be and vice versa.

Moving forward, an expansion of the Section 706 reporting process to include several indicators, including QoS capabilities and their effects on the quality of the evolving national communications infrastructure, is warranted.

VIII. Conclusion

We have shown the origin of and motivation for the use of Quality of Service mechanisms within the PacketCable 1.x VoIP PSTN-equivalent architecture and why QoS might be of great importance to SIP VoIP and other applications. There are potential speech quality and E911/911 service related problems for current third-party, non-facilities-based SIP VoIP providers using broadband cable access networks. The lack of non-discriminatory access for third-party devices is due to the implemented features of the current version of PacketCable 1.x broadband cable access equipment. It does not permit third-party control of DOCSIS-QoS settings. The result is third-party VoIP calls that make use of best-effort packet handling as is used for other data packets giving an increased possibility of quality impairment. We have described approaches to solving this problem. We further suggest that since new IP-enabled SIP VoIP entrants will be subject to some sort of oversight regarding traditional telephony obligations they should have access to the appropriate network mechanisms (DOCSIS-QoS) to allow their services to be of as high a quality as the incumbent's.

⁶⁴ A network externality refers to a situation where by simply having, for example, more users on a communications system, the overall value of the system is enhanced for all users.

Because of the success of VoIP companies such as Vonage, Skype,⁶⁵ pulver.com's Free World Dialup⁶⁶ and others, it has become more timely to solve issues of implementation of E911/911, sound quality, access for the disabled, and law enforcement. For broadband cable systems, the mechanism with the most promise to address these challenges is the next architecture and specification from the CableLabs PacketCable Multimedia initiative. It includes operation of SIP VoIP and other application devices, and extends to a larger, more general scope of support for bandwidth on demand and multimedia applications.

We indicate a plausible avenue for improving the oversight of the proliferation of broadband access networks so that they will be evaluated not only for their raw speed but also their ability to support the high quality, advanced services for which they were intended. This could correspond with an evaluation of a network's Quality of Service capabilities and could be done as part of an extension to the current Section 706 inquiry and reporting process.

Even though the immediate issues revolve around telephony as the problem to be solved, there is a general opportunity here that should not be missed in terms of including the best architectural principles to cover where convergence of voice, data, multimedia and more is headed and offering guidance where appropriate.

⁶⁵ Skype provides a PC-based telephony service over broadband access and calls are cost-free. See <http://www.skype.com/> for more information. (Last visited August 29, 2004.)

⁶⁶ Pulver.com provides a PC-based telephony service over broadband access and calls are cost-free. See <http://www.pulver.com/> for more information. (Last visited August 29, 2004.)