2009 SCTE Cable-Tec Expo®

Business Services – The Suite of Services Required
by Small, Medium Businesses (SMBs) and Enterprise Customers

PacketCable™ Architecture for Business Voice Services

Glenn Russell
Director, Business Services
CableLabs

David Hancock
Project Director, Signaling Architecture
CableLabs

Sandeep Sharma
Senior Architect, PacketCable
CableLabs

Abstract

This paper provides a brief overview of how business voice services can be implemented in PacketCable networks. In particular, we will focus on two basic business voice service models: 1) network hosted or managed voice services, and 2) interconnection to business private branch exchange (PBX) systems using Internet Protocol-based trunking.

CableLabs®, working with cable operators and equipment manufacturers, has defined a standards-based architecture for delivering enhanced voice services to SMBs (small and medium-sized businesses) and Enterprises. Based on the Session Initiation Protocol (SIP), the PacketCable™ 2.0 Business Services architecture is transport agnostic and can be used to provide a wide range of enhanced voice features and services over coax or fiber-based networks.

The paper introduces two new application profiles for PacketCable 2.0 – Business SIP Services and SIP Enterprise Connect, and describes how they support the voice services models required by cable operators.

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1 INTRODUCTION

The cable industry has achieved considerable success deploying residential voice services based on the PacketCable VoIP architecture. The technology platform used for delivering residential voice service is readily extensible and, with some architectural modifications, is capable of supporting the voice communication needs of millions of small and medium-sized businesses as well as enterprise customers in North America.

Insight Research estimates that over the next five years, cable will capture nearly ten million small business lines from the Incumbent Local Exchange Carriers (ILECs) – a $4.6 billion revenue opportunity. One of the greatest opportunities is serving small and medium-sized businesses (SMBs) using the existing hybrid fiber coax (HFC) broadband service delivery platform.

Increased awareness of broadband and Voice over Internet Protocol (VoIP) has made small businesses more open to purchase voice from broadband service providers. Furthermore, SMBs are eager for an alternative to those offered by traditional telecom service providers. With success in broadband and now residential telephony services, cable is seen as a reliable business telecommunications alternative to Local Exchange Carriers (LECs) and Internet Service Providers (ISPs).

1.1 PacketCable 1.5 – Basic Voice for Small Businesses

Figure 1 shows a common network implementation for providing voice line service to small businesses via the PacketCable 1.5-based architecture [PKT-TR-ARCH1.5]. Developed for residential voice services, PacketCable 1.5 has a number of feature extensions that can be utilized to provide better voice communication services to business customers. Equipment manufacturers recently developed a range of Multi-Line Terminal Adapters (ML E-MTAs) that are well suited to service business environments. ML E-MTAs can provide up to 24 analog lines from a single device.

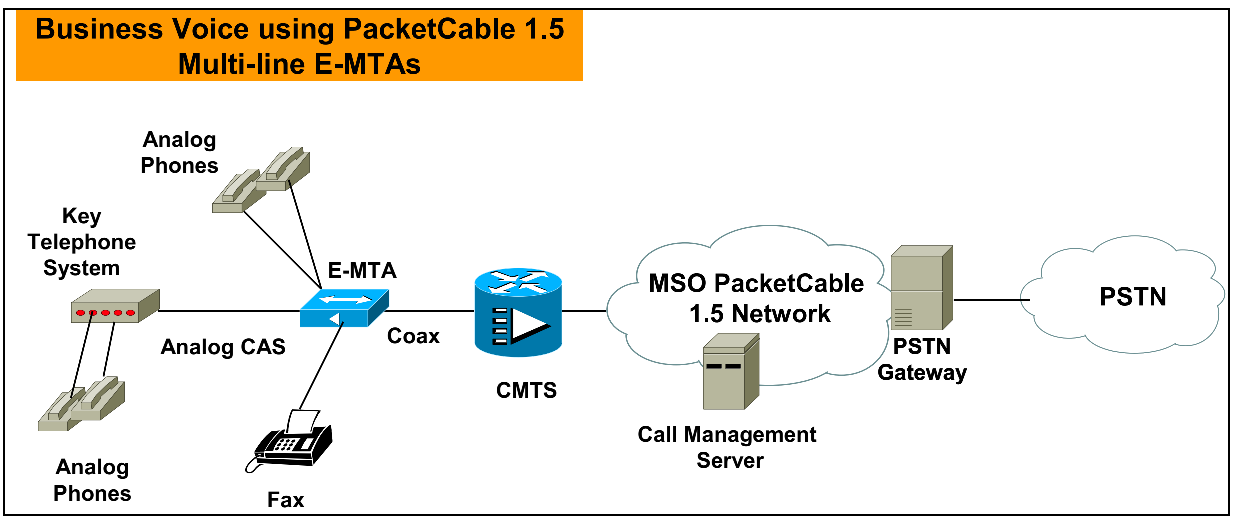

Figure 1 – Multi-Line Voice - PacketCable 1.5 Architecture

Multi-line E-MTAs can serve the voice needs of many small businesses since they provide a standard analog phone line interface to business devices such as business telephones, facsimile machines, data modems, security/alarm systems, point-of-sale terminal devices such as credit card machines, and automated teller machines (ATMs). Additionally, E-MTAs that implement the PacketCable 1.5 "Analog Trunking for PBX" specification provide multi-line services to Key Telephone Systems (KTS) and traditional time division multiplexing (TDM) PBXs via a standard analog trunking interface.

Additional voice line subscriber features necessary for a small business environment can be implemented either on the Call Management server or via a separate business telephony feature server. Such business line features might include call transfer, consultation hold, and multi-line hunting.

1.2 PacketCable 2.0 – Enhanced Business Voice Services

Cable operators are increasingly interested in serving a wider range of endpoint devices. Today's business environment is likely to include VoIP phones, software voice clients on a personal computer, mobile smartphones, and multimedia conference terminals, all of which are typically connected to Internet Protocol-based communication networks.

In this environment, SIP has become the de-facto communication protocol for establishing voice and multimedia communications. An increasing number of Customer Premise Equipment (CPE) and endpoint device manufacturers are building SIP-based voice products designed specifically for business and enterprise environments. Leading business telephony application server vendors and IP-PBX manufacturers also embrace SIP as a framework for voice and multimedia service delivery.

In recognition of these industry trends, the PacketCable 2.0 network architecture was designed to provide a range of communication services to SIP-based clients. To serve business customers, including SMBs and Enterprises, two application service models have been defined:

Business SIP Services (BSS) – Network Hosted Business Line Services: In this model, the cable operator controls all call signaling and feature implementation. Telephony application servers on the service provider network 'host' the voice services that are delivered to client devices (i.e., telephone sets) on the customer's business network. This service model is attractive for small businesses that require basic voice services and who do not wish to own or manage an on-site PBX system.

SIP Enterprise Connect (SIP-EC) – PBX Interconnection: In this model, most voice features are provided locally by the PBX system located on the customer premise network. The cable service provider manages the features that provide interconnection between the on-site PBX system and the public telephone network. Connectivity and features between the cable service provider network are delivered via a 'trunk' – a point-to-point communication link capable of simultaneously supporting multiple connections over a single access facility, such as a broadband cable network. This model is used to support businesses whose communication requirements demand an on-site PBX system. They typically require greater capacity and feature functionality than can be provided by the service provider network.

Service and feature requirements needed to support these two application models are described in the next section.

2 SERVICE REQUIREMENTS

2.1 Hosted Multi-Line Voice – Business SIP Services

PacketCable Business SIP Services (BSS) specifies a multi-line voice service similar to a business Centrex or hosted IP-PBX model. Non-facilities-based hosted solutions provide several benefits to business customers. Specifically, they are able to get advanced business features typically available only on enterprise PBX systems. Since the service is hosted by the cable operator, businesses save costs by eliminating management and operational expenses of running a PBX system.

Figure 2 illustrates how hosted business voice line services can be delivered to a customer in a PacketCable 2.0 network.

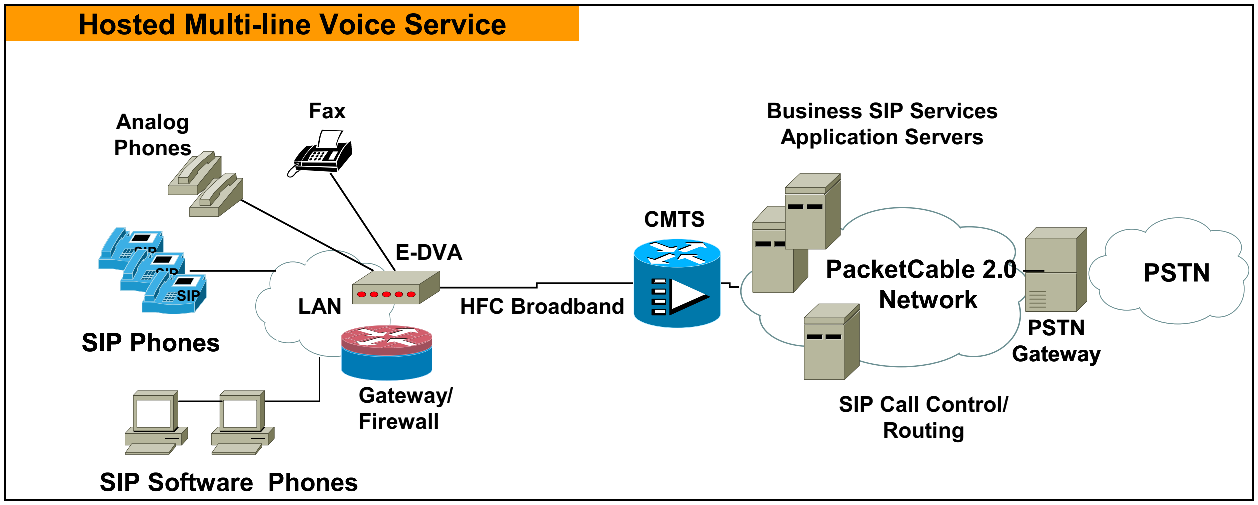

Figure 2 – Hosted Multi-Line Voice Business SIP Services

The PacketCable 2.0 Residential SIP Telephony specification (RST) defines a complete architecture for delivering residential voice services using the IP Multimedia System (IMS) architecture. RST defined the term "Embedded Digital Voice Adapter" to describe a PacketCable 2.0 SIP voice client embedded within a cable modem. As shown in the figure above, a SIP-based Embedded Digital Voice Adapter (E-DVA) replaces the PacketCable 1.x E-MTA as the gateway element at the business premise. Functionally equivalent to an E-MTA, the E-DVA provides the analog interfaces necessary to connect to analog business telephone sets, fax machines, and other terminal devices needed by the business customer. In the service provider network, SIP-based telephony application servers 'host' and provide signaling and control of all telephony features delivered to devices on the customer network.

This BSS architecture is also capable of providing voice services to Internet Protocol-based telephone sets that are connected to the customer Local Area Network (LAN) via an Ethernet interface. This hosted service scenario does not rely on embedded devices such as E-DVA or E-MTAs. IP-phones that support PacketCable 2.0 are capable of receiving call signaling and media directly from the telephony application servers in the cable service provider network. Hosted solutions can be delivered over both Data Over Cable Service Interface Specifications (DOCSIS®) and non-DOCSIS (e.g., Fiber/Ethernet) based networks. A network address translation (NAT) and firewall enabled gateway device between the clients and the network can be an important component of a hosted solution.

2.1.1 Call Features Supported by BSS

BSS includes all of the residential features listed below in Section 4.3. The strength of the BSS architecture is its support for an extended list of business-oriented features needed by larger enterprise customers. This includes individual business line features such as: automated attendant, direct inward dialing, direct outward dialing, extension dialing, alternate number, intercom, shared call appearance, sequential and simultaneous ring, call jump, and remote office.

Additionally, group features are essential to many business environments. Some of the more prominent group features supported include: account codes, authorization codes, hunt groups, call park features, call pickup features, outbound call restriction, charge number service and loudspeaker paging. In addition, BSS also covers web portal services, third party call control features, e-mail, address book, and directory integration.

2.2 PBX Interconnection – SIP Enterprise Connect (SIP-EC)

Over 60% of businesses today are estimated to have a local PBX or Key Telephone System on their premises to provide internal telephone services. Many of these businesses have replaced or plan to replace their internal telephone systems with PBXs based on VoIP technology. While some business PBXs may chose to be connected to the public switched telephone network (PSTN) via dedicated, private line circuits, it is more efficient for service providers to provide trunking services to businesses over an Internet Protocol-based network.

Cable's hybrid-fiber coax (HFC) network infrastructure currently passes to close to millions of small, medium, and large sized businesses, many of whom will be using IP-enabled PBX systems. Modest extensions to cable's existing VoIP service delivery architecture will enable cable operators to provide IP-based interconnection services to enterprise customers who have migrated to IP-based technologies.

PacketCable's SIP Enterprise Connect (SIP-EC) defines how the Session Initiation Protocol (SIP) is used to establish voice communications services over an IP-network between an IP-PBX and the service provider network. This communication service is commonly known as a "SIP Trunk." It provides interconnection services that are functionally equivalent to traditional telecommunications trunk services such as Integrated Services Digital Network Primary Rate Interface (PRI). As shown in Figure 3, a SIP Trunk is used to transport multiple voice communication sessions between the enterprise PBX and the PSTN.

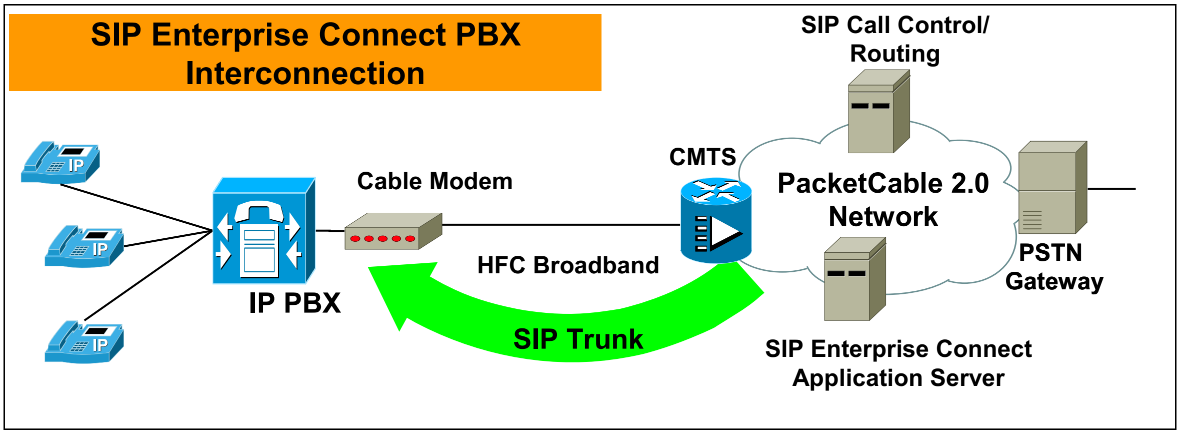

Figure 3 – SIP Trunking over Cable Broadband Network

The Service Provider network provides the SIP Trunk that terminates at an IP-PBX on the customer's network. Actual deployment scenarios may include intermediate network gateway devices (e.g., Internet Access Devices, Session Border Controllers) that physically terminate the media or signaling elements prior to delivery of the service to the IP-PBX.

2.2.1 Benefits of SIP Trunking

A SIP Trunk, delivered over unified access networks, provides the following benefits over a traditional TDM trunk delivered over dedicated facilities.

- Cost savings to the customer by eliminating expensive trunking gateways and transcoding systems used to mediate between VoIP and TDM networks**.**
- Delivery of multiple services (voice and data) over a unified access network. This will reduce the dependency on single-purpose dedicated hardware solutions that are unique to a single service type and single signaling format.
- Increased flexibility and speed of service activation and provisioning by utilizing dynamic, software-based signaling solutions based on standard IP technology.
- Ability to dynamically set up and tear down call sessions per application need will improve network resource efficiency by freeing up network bandwidth when call volumes are low.
- Ability to avoid call blocking situations by being able to burst beyond the contracted bandwidth during periods of high usage.

2.2.2 Features Supported by SIP Enterprise Connect

SIP-EC defines how to transport voice communications sessions (media and signaling traffic) between a Private telephone switching system elements in the service provider network. The IP-PBX may be managed by the cable service provider, managed by the customer, or the management may be shared between the customer and the cable service provider. In general, IP-PBX users receive all their features from the IP-PBX. However, certain call features may be provided to the IP-PBX users by the service provider network.

Voice calling features supported include:

- Basic voice calls
  - Direct inward and outward dialing (DID/DOD)
  - Calling name and number delivery and blocking
  - Early media
- Voice (G.711 codec) and FAX (T.38 FAX-relay)
  - Regulatory features
  - Emergency calls
  - Operator services
- Features provided by the SIP-PBX for calls established with the service provider
  - Multi-party features (hold, conference, transfer)
  - Call Forwarding
  - Voice Mail
- Features provided by the Service Provider network on behalf of the SIP-PBX
  - Call forwarding (when SIP-PBX is offline)
  - Hosted Voice Mail

3 PACKETCABLE 2.0 REFERENCE ARCHITECTURE

The PacketCable 2.0 architecture provides a rich and modular platform upon which a variety of IP communication services may be built for a diverse set of clients. As shown in Figure 4, the architecture decouples application control from the application-independent support functions. The PacketCable 2.0 base network architecture provides all the support functions, such as authentication, authorization, and routing. The application logic resides in Application Servers and clients that plug into standard interfaces to utilize the support services of the base architecture.

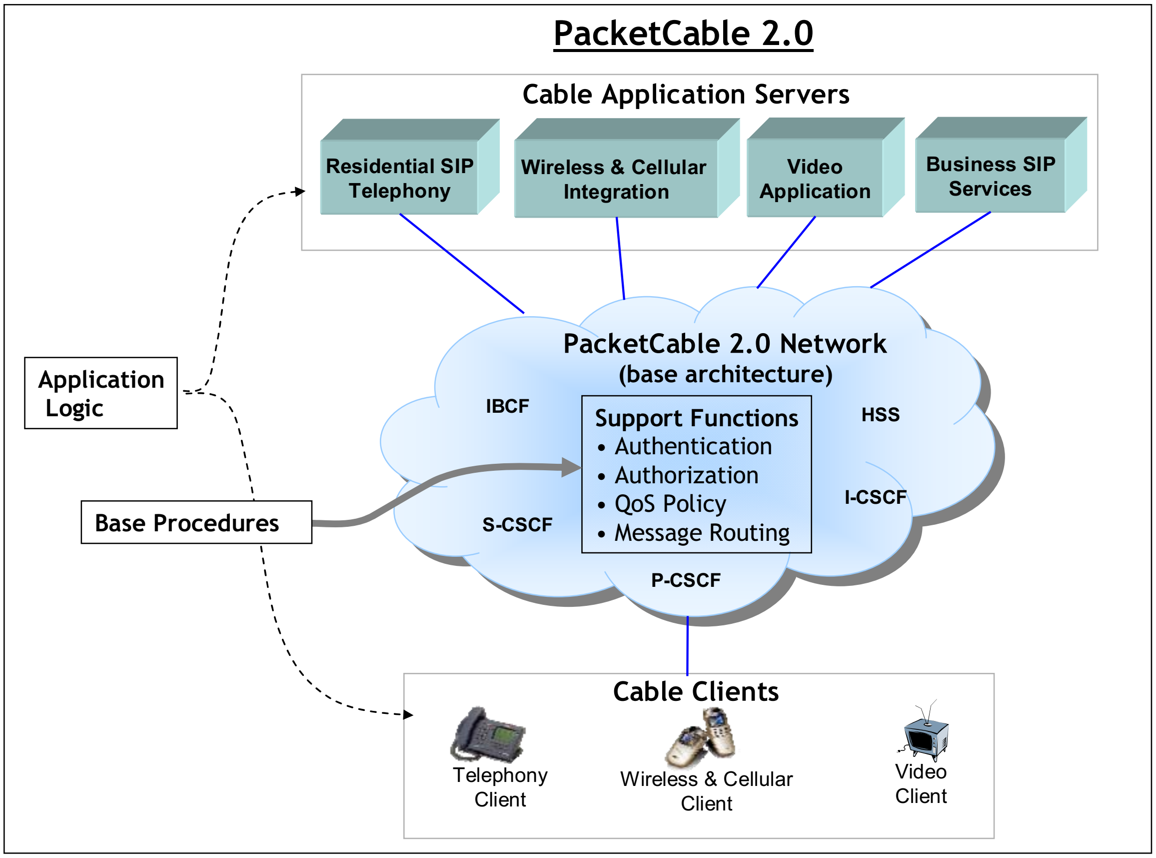

Figure 4 – PacketCable 2.0 Framework

The architecture is divided into several logical areas or functional groupings. These functional groupings, along with various elements within the groupings, are described in detail in the PacketCable 2.0 Architecture Technical Report.

The base PacketCable 2.0 technical specifications specify the detailed technical requirements on the components in support of the functional requirements. Some of these functional requirements are in the areas of routing, security, authentication, accounting, and registration. The base PacketCable 2.0 specifications leverage the IP Multimedia System (IMS) developed by the 3rd Group Partnership Project (3GPP) specifications for the signaling requirements between the various components. The suite of base PacketCable 2.0 specifications address cable requirements in certain areas such as accounting, codec and media, electronic surveillance, provisioning, and Quality of Service.

The application profiles leverage the base PacketCable 2.0 functionality and define service specific requirements on the corresponding clients and ASs to implement the service logic. The application profiles relevant to business voice services are introduced in the next section.

4 PACKETCABLE 2.0 APPLICATION PROFILES

Application Profiles leverage the generic functions of the base PacketCable 2.0 architecture to provide specific features and services to residential and business users. PacketCable has developed specifications for three different applications; Residential SIP Telephony (RST), Business SIP Services (BSS), and SIP Enterprise Connect (SIP-EC). RST provides residential phone service to the home, while BSS and SIP-EC provide business telephony services to enterprise customers. Figure 5 shows how these applications are supported by the PacketCable 2.0 network architecture.

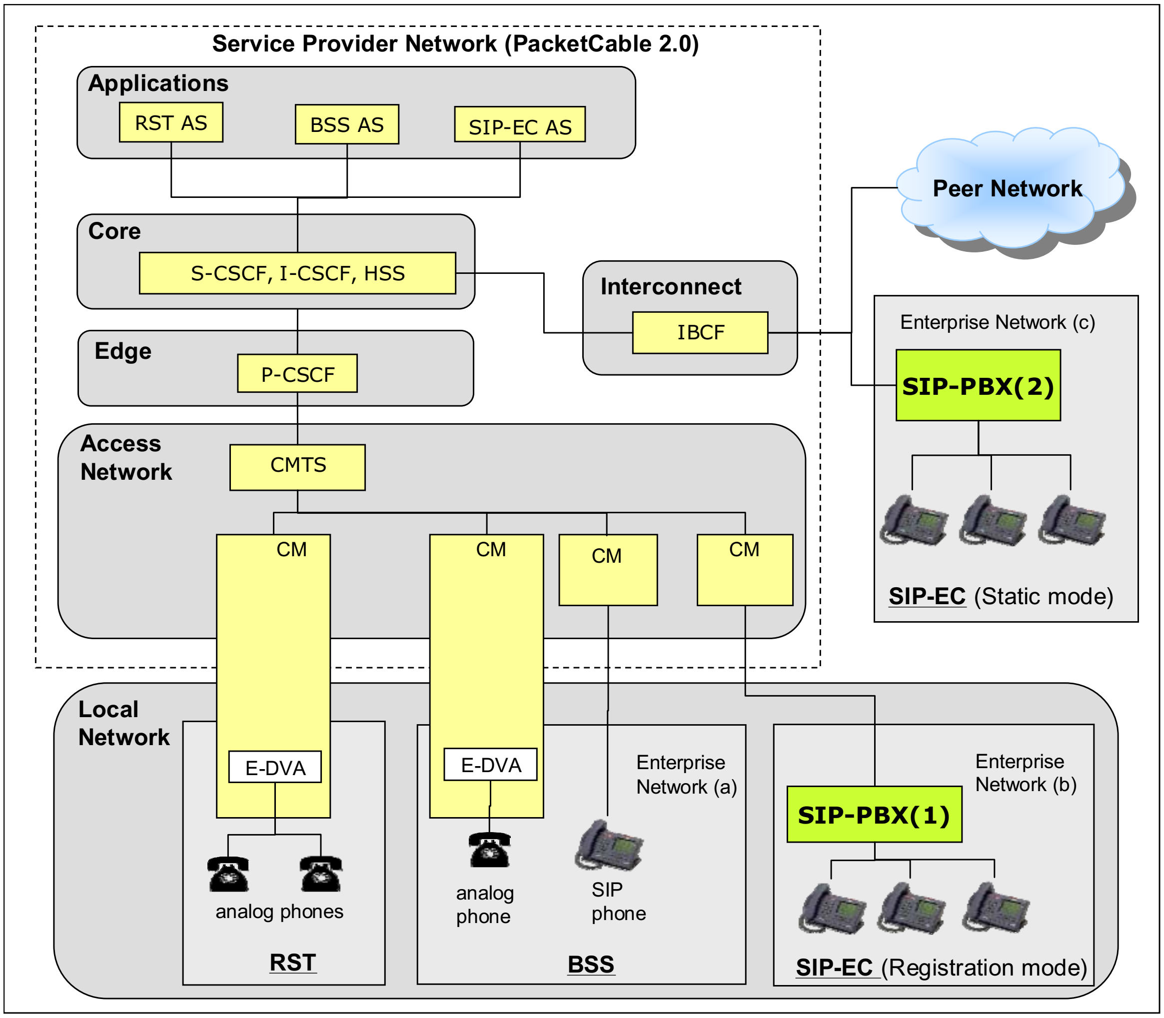

Figure 5 – PacketCable 2.0 Application Architecture

The local network provides the IP attachment point for the client devices (e.g., analog phone, SIP phone). The local and access networks together provide an IP pipe to carry SIP messages between the client and the Proxy Call Session Control Function (P-CSCF). The P-CSCF acts as the client's SIP entry point into the PacketCable 2.0 network, exchanging all SIP messages between the client and the core. The core provides the non-application-specific support functions, such as authentication, authorization, routing, and basic session establishment. The core also invokes ASs located in the Applications function, as needed to provide additional features and services to the user. Application control resides in the Application Servers (AS) and clients. The Interconnect function provides access to other networks such as peer VoIP networks.

Figure 5 shows the ASs and the client devices required to support the RST, BSS, and SIP-EC applications. These components contain the application logic that rides on top of the generic base architecture to provide application-specific services to the PacketCable 2.0 users. The RST AS and the E-DVA located in the home network provide traditional analog phone service to residential users. The BSS AS and the E-DVA and SIP phones located in enterprise network (a) provide business services to the enterprise (a) users. The SIP-EC AS and SIP-PBX(1) located in enterprise network (b) provide business services to enterprise (b) users.

Figure 5 also shows SIP-PBX(2) located in enterprise network (c). In this case the SIP-PBX connects to the PacketCable 2.0 network via the Interconnect Border Control Function (IBCF). The PacketCable 2.0 network views SIP-PBX(2) as a peer VoIP network.

4.1 Residential SIP Telephony (RST)

The PacketCable Residential SIP Telephony effort is intended to replicate many of the residential features currently available on the public switched telephone network (PSTN), except that the features operate using a PacketCable 2.0-based IP network using SIP for VoIP sessions. To implement these features, the RST user equipment (UE) and RST AS need to implement feature-specific requirements that are defined in the RST specifications. These components are introduced below. The PacketCable RST specifications leverage the base PacketCable 2.0 architecture and specifications outlined above.

RST User Equipment (UE): The functional element subscribers use to attach to the PacketCable network. UEs may take any number of forms, including clients embedded into cable modems, clients embedded into set-top boxes, clients embedded into SIP Phones, and software clients on PCs or laptops. Ues support several interfaces to the PacketCable network:

- Session Initiation Protocol (SIP): For signaling the creation, modification, and destruction of sessions.
- Media: For carrying audio or other media traffic, and related statistics.
- NAT Firewall Traversal: For permitting Residential SIP Telephony features to operate through NAT Firewall devices.
- Security: For securing interfaces.

The RST E-DVA is a RST UE that provides residential features to traditional analog phone devices.

RST AS: The Residential SIP Telephony AS is a logical network component that implements some aspects of RST features. The PacketCable architecture provides a framework for triggering SIP messages to the application server based on the service profile of a user. Network-based features are completely implemented on the RST AS.

Feature groups defined in RST are listed below, and are defined in more detail in [PKT-SP-RSTF].

- Basic voice calling
- Caller ID delivery & blocking
- Call forwarding
- Call blocking
- Multi-party features (call waiting, hold, transfer, and three-way calling.)
- Miscellaneous features: including distinctive alerting, speed dialing, and customer-originated call trace features
- Emergency services
- Operator services: including busy-line verification and emergency operator interruption capabilities

RST Specifications

Residential SIP Telephony consists of the following suite of specifications:

- PacketCable Residential SIP Telephony Feature Specification [PKT-SP-RSTF]. The RST Feature Specification that defines the SIP signaling requirements on the RST UE and RST AS to implement the residential features.
- PacketCable Residential SIP Telephony Accounting Specification [PKT-SP-RST-ACCT]. This specification defines the collection of usage data needed to support Accounting of Residential SIP Telephony Features.
- PacketCable Residential SIP Telephony E-DVA Specification [PKT-SP-RST-E-DVA]. This specification defines the embedded Digital Voice Adaptor (E-DVA) requirements for the analog interface and for powering of the E-DVA.
- PacketCable Residential SIP Telephony Provisioning Specifications [PKT-SP-RST-UE-PROV] [PKT-SP-RST-EUE-PROV]. These specifications define the data model and the protocol mechanisms used for provisioning and configuration of the RST UE.

4.2 Business SIP Services (BSS)

The BSS architecture supports analog phones via the E-DVA as described above for RST. BSS also supports SIP phones, where the SIP User Agent and application logic is located outside the cable modem (CM) in the SIP phone itself. The PacketCable BSS specifications leverage the base PacketCable 2.0 architecture and specifications outlined above. Since many of the features in scope for BSS overlap with the residential set, BSS specifications point to the corresponding RST specifications for those requirements. BSS specific feature requirements are implemented on the BSS UE and the BSS AS.

BSS UE: the functional element at the customer premise that attaches to the PacketCable network. Ues may take any number of forms, including clients embedded into cable modems, clients embedded into set-top boxes, clients embedded into SIP Phones, and software clients on PCs or laptops.

BSS AS: The Business SIP Services Application Server is a logical network component that implements some aspects of BSS features.

A high level description of the features can be found in the BSS Technical Report.

4.2.1 BSS Specifications

The Business SIP Services suite of specifications is based on the technical report and consists of the following specifications.

- PacketCable Business SIP Services Feature Specification [PKT-SP-BSSF]. This specification defines requirements on the UE and ASs to implement Business SIP Services features. Additionally, this specification defines the signaling requirements for feature activation, deactivation and feature execution. The interactions between various features are also defined.

  In addition to the feature specific requirements, the feature specification also addresses a number of related areas such as:

  - Marking of packets for quality of service
  - User interface capabilities needed to support 'black phones', advanced SIP phones and, softphones
  - Security considerations
  - NAT and firewall considerations
  - Considerations for adapting a residential E-DVA for a small business environment

- PacketCable Business SIP Services Provisioning Specification [PKT-SP-BSS-PROV]. This specification defines the requirements for provisioning a Business SIP Services UE.

4.3 SIP Enterprise Connect (SIP-EC)

SIP-EC provides network access to SIP-based PBXs located in the enterprise. SIP-EC differs from BSS primarily in that enterprise user features are controlled by the SIP-PBX itself, and not the PacketCable 2.0 network. The primary role of the PacketCable 2.0 network is to provide connectivity between the SIP-PBX and the global network. The PacketCable 2.0 network may host limited services such as incoming call handling when the SIP-PBX is offline, or network-based voice-message service. These network-based services are provided by the SIP-EC AS.

4.3.1 Modes of Operation

SIP-EC supports two modes of operation; the Registration mode, and the Static mode. These modes differ primarily in the way the SIP-PBX delivers its SIP signaling address to the PacketCable 2.0 network.

In the Registration mode, the SIP-PBX conveys its SIP signaling address to the service provider network using the SIP registration procedure. In effect, the SIP-PBX registers with the SP network, just like a directly hosted SIP endpoint would register. In the Static mode, the PacketCable 2.0 network is either configured with the SIP-PBX signaling address or discovers the address using Domain Name Service (DNS). The Registration and Static modes also differ in SIP-PBX interconnection point to the PacketCable 2.0 network. A SIP-PBX operating in the Registration mode is located in the Local Network, and gains access to the PacketCable 2.0 network via the Access network (see SIP-PBX(1) in Figure 5). A SIP-PBX operating in the Static mode appears as a peer network to the home service provider network, with the IBCF acting as the entry point to the PacketCable 2.0 network (e.g., SIP-PBX(2) in Figure 5).

The reason for defining two modes is that each mode targets a different market segment. The Registration mode lends itself to small-to-medium sized enterprises, where the SIP-PBX typically obtains a dynamic IP address from DHCP and conveys it to the service provider network using the SIP registration procedure. The Static mode is often used for larger enterprises, where the size of the enterprise warrants more explicit provisioning of connection and service information within the service provider network.

4.3.2 Authentication and Security

SIP signaling messages between the enterprise and service provider Network are secured using Transport Layer Security (TLS). The operator may choose to disable TLS, for example, when the underlying transport technology provides adequate security.

Authentication of the service provider by the enterprise is supported using TLS authentication procedures. Authentication of the enterprise by the service provider network is supported using SIP Digest authentication in the Registration mode, and TLS authentication in the Static mode.

4.3.3 SIP Enterprise Connect Specifications

The specifications for SIP-EC are currently under development, with a planned release in late 2009 or early 2010. The overall objective of this effort is to align with existing procedures and practices specified within other PacketCable and industry specifications, as follows:

- The SIP-PBX requirements are specified in the “SIPconnect 1.1 Technical Recommendation” currently being developed by the SIP Forum.
- Enhancements to the base PacketCable 2.0 procedures required to support SIP-EC will be specified in an updated version of the base PacketCable 2.0 specifications.
- The PacketCable 2.0 SIP-EC AS requirements are primarily specified in the PacketCable Business SIP Services Feature Specification.
- A new specification called the “PacketCable SIP Enterprise Connect Specification” will contain normative references to the above documents, plus specify any additional requirements for the SIP-EC AS that are not covered by the Business SIP Services Feature Specification. This is depicted in Figure 6.

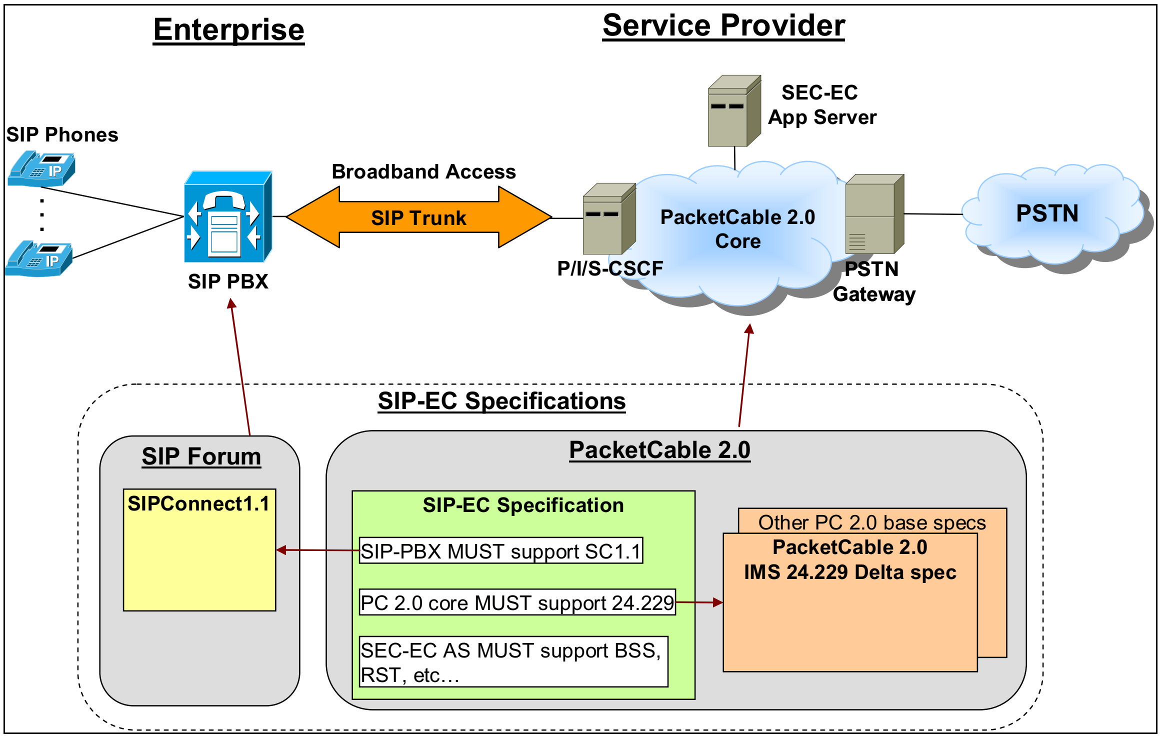

Figure 6 – SIP Enterprise Connect Specifications

5 CONCLUSION

PacketCable provides a wide range of options for delivering voice services to business customers. Home office and small businesses needing basic voice line services are well served by solutions based on the PacketCable 1.5 architecture. Businesses that are larger in size or that demand support for a wider range of calling features are effectively served by PacketCable 2.0 Business SIP Services and SIP Enterprise Connect.

6 ACKNOWLEDGEMENTS

The authors gratefully acknowledge the significant contributions made by CableLabs member companies. Additionally, we thank the equipment manufacturers who have actively contributed to the development of the PacketCable 2.0 specifications for Business Services.

7 REFERENCES

All PacketCable specifications and technical reports referenced in this paper can be found at: http://www.cablelabs.com/specifications/.

Specific Business Services Specifications and Technical Reports referenced in this paper are listed below.

PacketCable Architecture Framework Technical Report	[PKT-TR-ARCH1.5]
PacketCable Analog Trunking for PBX Specification	[PKT-SP-ATPBX1.5]
PacketCable RST Accounting Specification	[PKT-SP-RST-ACCT]
PacketCable RST E-DVA Specification	[PKT-SP-RST-E-DVA]

PacketCable Residential SIP Telephony (RST) Feature Specification	[PKT-SP-RSTF]
PacketCable RST E-UE Provisioning Specification	[PKT-SP-RST-EUE-PROV]
PacketCable RST UE Provisioning Specification	[PKT-SP-RST-UE-PROV]
PacketCable RST Feature Definition Technical Report	[PKT-TR-RST]
PacketCable Business SIP Services (BSS) Feature Specification	[PKT-SP-BSSF]
PacketCable BSS Provisioning Specification	[PKT-SP-BSS-PROV]
PacketCable BSS Feature Description Technical Report	[PKT-TR-BSS]

8 ABBREVIATIONS AND ACRONYMS

This technical paper uses the following abbreviations and acronyms:

AS	Application Server
ATM	Automatic Teller Machine
BGCF	Border Gateway Control Function
BSS	Business SIP Services
CAS	Channel Associated Signaling
CLEC	Competitive Local Exchange Carrier
CM	Cable Modem
CMTS	Cable Modem Termination System
CPE	Customer Premise Equipment
CSCF	Call/Session Control Function
DOCSIS	Data Over Cable Service Interface Specification
DNS	Domain Name Service
DID	Direct Inward Dialing
DOD	Direct Outward Dialing
DTMF	Dual Tone Multi Frequency
DVA	Digital Voice Adapter
E-MTA	Embedded Multimedia Terminal Adapter
E-DVA	Embedded Digital Voice Adapter
HFC	Hybrid-Fiber Coax
HSS	Home Subscriber System
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
ILEC	Incumbent Local Exchange Carrier
IBCF	Interconnect Border Control Function

IP	Internet Protocol
IP PBX	Internet Protocol Private Branch Exchange
ISDN	Integrated Services Digital Network
KTS	Key Telephone System
ISP	Internet Service Provider
LAN	Local Area Network
MTA	Multimedia Terminal Adapter
NAT	Network Address Translation
OSS	Operational Support System
PDA	Personal Digital Assistant
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RST	Residential SIP Telephony
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SIP-EC	SIP Enterprise Connect
SMB	Small to Medium Business
TCP	Transport Control Protocol
TDM	Time Division Multiplexing
UDP	User Datagram Protocol
UE	User Equipment
UI	User Interface
VoIP	Voice over Internet Protocol