CHAPTER 4

Offering Bundled Voice and Data Services

Chapter 2, "VoIP Network Architectures: H.323, SIP, and MGCP," describes a managed voice and data service architecture that uses a call agent. A call agent solution is commonly used for the small and medium-sized business (SMB) market. This solution uses an integrated access device (IAD) at the customer premises that supports voice and data using IP or Asynchronous Transfer Mode adaptation layer 2 (AAL2) transport over a T1 access link to the service provider.

This chapter continues the focus on offering managed services to SMBs. Service providers that traditionally use time-division multiplexing (TDM) access and that need to add other service offerings can bundle voice and data services over a single access link to the customer. For example, traditional T1 circuits that are offered to customers to interconnect their private branch exchange (PBX) to interexchange carriers (IXCs) can now be used instead for integrated voice and data traffic, thereby eliminating the need for multiple access links between the customer and the service provider.

Three areas are discussed to help provide an overview of bundled voice and data service architectures:

- Overview of Managed Voice and Data Services
- Managed Voice and Data Services Using AAL2
- Fundamentals of AAL2

Overview of Managed Voice and Data Services

Integrated voice and data is a new service that is offered by service providers. The architecture design to deploy managed voice and data services is based on many factors. One of these factors is the required customer premises equipment (CPE), which is based on the type of business customer.

Two general types of business customers exist. One is a business customer with fewer than 100 users, such as a doctor's office, an insurance agent's office, or a small home office. These businesses are normally single-site locations that require telephony services, Internet access, firewall, and Virtual Private Network (VPN) services. Typically, these businesses do not have older networking protocols, such as AppleTalk or IPX, and they do not have a full-time support staff to maintain their own private network. A service provider can support these services with an IAD on the customer premises, such as a Cisco 2400.

The second type of business customer is an enterprise customer. An enterprise customer has a large installed base of devices that supports many flavors of protocols, sophisticated routing designs, multiple T1s, and back-hauling needs. An enterprise customer needs a multiservice platform, such as the Cisco 2600 and 3600. Many of these enterprise businesses have their own large private networks and their own full-time staff to maintain their multiservice network. However, because of various reasons, such as fast growth and economics, many of these large customers are outsourcing some or all of their services to service providers.

Integrated Access Architectures

Traditionally, service providers offer TDM services that connect a customer's PBX to an IXC Class 4 switch, which provides long distance voice services. Many of these service providers are currently switching from using a TDM-based infrastructure to using a packet-based infrastructure, either IP or ATM. This approach allows for a more efficient method to provide voice transport and also helps to integrate voice and data services over one access link to the customer premises.

NOTE Many incumbent carriers are adding IP to their core ATM network by inserting Multiprotocol Label Switching (MPLS) technology. This change enables service providers to shift from transport service offerings to IP-based service offerings.

ADSL and T1 ATM are two types of access technologies that can be supported between the SMB and the service provider.

Other Cisco IADs

Other IADs available for SMBs are the Cisco 827-4V and the 1750. Both of these IADs support four Foreign Exchange Station (FXS) interfaces, and the 1750 also supports FX0 and E&M interfaces. The 827-4V supports a fixed configuration that includes Ethernet and DSL WAN-access only, and the 1750 supports a modular configuration that includes Fast Ethernet and multiple WAN options, such as DSL and T1 access. To help further differentiate these two IADs, the 1750 supports dual WAN interfaces for WAN backup or load sharing, hardware Triple Data Encryption Standard (3DES) encryption, and Open Shortest Path

First (OSPF) and Border Gateway Protocol (BGP) routing—these capabilities are not present in the 827-4V. The 827-4V is well suited for small offices that do not require the extra capabilities of the 1750 and are not concerned with expandability, but do require core services such as basic voice (FXS), VPN, and a firewall from their service provider.

Managed Voice and Data Services Using AAL2

AAL2, referred to as VoAAL2 in a voice network, can integrate the voice and data services offered to the customer. Alternatively, a service provider can begin with an IP-based infrastructure and build out a VoIP call agent architecture to support voice and data services to their customers, which is a more common approach today. Both technologies, VoIP and VoAAL2, offer the value of integrating voice and data while achieving efficient bandwidth use.

This section provides an AAL2 architecture that can provide trunking and integrated access services. By using AAL2, many capabilities can be obtained within the service provider's ATM network:

- Dynamically change from voice to fax demodulation
- AAL2 Type 3 cells for reliable dual tone multifrequency (DTMF) relay
- Dynamically change the compression rate to G.711 for fax calls in midcall
- Indicate end of speech burst for background noise generation during silence periods at the egress ATM switch
- Transport up to 248 voice calls with different compression schemes within one or more ATM permanent virtual circuit (PVC)

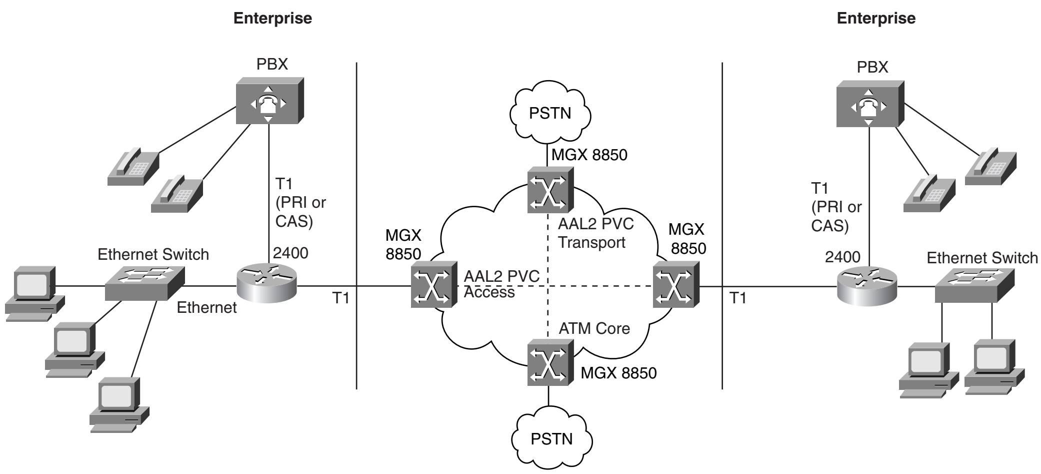
This architecture provides a Class 4 interconnect replacement, which enables an enterprise to bypass the local Tandem Switch.

NOTE A *Tandem Switch* is a switch that incumbent local exchange carrier (ILEC) networks use to route calls between COs in the same local access and transport area (LATA). These calls are referred to as intraLATA calls. Trunks at each CO are typically interconnected by a Synchronous Optical Network (SONET) ring. The Tandem Switch also connects to an IXC Tandem Office, which is called a *point of presence (POP)*. A POP houses a Class 4 switch that connects into the ILEC's Tandem Switch. The Tandem Switch aggregates interLATA traffic from multiple COs and a trunk facility. An IXC Tandem Office can have dedicated trunks to an ILEC's CO in cases where a high concentration of traffic exists between the CO and the IXC. An IXC handles interLATA traffic.

A Class 5 switch is located in an end office and a Class 4/5 switch is located in a Tandem Office. A Class 5 switch provides local services in the PSTN to the end user. A Class 5 switch provides enhanced calling features, such as call waiting and three-way calling to end users.

The architecture shown in Figure 4-1 depicts a service provider offering integrated access to business customers and trunking service between two PSTN carriers.

Figure 4-1 *Integrated Access and Trunking Service Using AAL2*



An end-to-end trunking architecture does not require a call agent. This architecture can reduce the complexity of a mesh of narrowband circuits by having only a single integrated voice and data network. IADs can support the transport of data and voice by using AAL2 and AAL5 from the customer premises to the service provider. The architecture in Figure 4-1 includes IAD 2400 at the business customer site, which terminates into a MGX8000 Voice Gateway at the service provider's network edge. The MGX8000 Voice Gateway adds packet voice capabilities to the MGX 8850 that includes VoIP, VoAAL1, and VoAAL2. The voice signaling from the enterprise customer is tunneled through the MGX8000 Voice Gateway to a Class 4 switch by using AAL2 point-to-point trunking. The data is tunneled through to an ATM switch, such as a Cisco BPX 8600.

Figure 4-1 shows AAL2 trunking services, which are indicated by the vertical dotted line. Multiple T1s or E1s with Primary Rate Interface (PRI) or channel-associated signaling (CAS) terminate from the PSTN to the MGX8000 Voice Gateway. The PSTN cloud represents another service provider offloading its voice traffic to another carrier. The integrated access service uses AAL2 PVCs between two MGX 8850s within the service provider's network. In this application, the MGX8000 Voice Gateway uses an ATM User Service Module (AUSM) card. The IAD 2400 aggregates both voice and data traffic over a T1 access line to the service provider.

CAS and PRI signaling can be supported in this architecture. The CAS information is carried in the AAL2 PVC across the network. CAS is a signaling technique that uses robbed bits within a multiframe, such as a D4 Super Frame (SF) or Extended Superframe. These robbed bits, referred to as ABCD, represent various states and transitions of a voice call. These ABCD bits are transported over the same AAL2 channel as the one used for voice because CAS does not use a separate signaling channel, such as in-band signaling; the CAS bits use AAL2 Type 3 packets as they provide CRC checks for reliability whereas voice traffic uses AAL2 Type 1 packets that are without CRC checks. An important feature that this architecture provides is *idle channel suppression*. Idle channel suppression stops sending idle channel bits that are generated from the CAS source (for example, a PBX). This mechanism results in significant amounts of bandwidth savings in the service provider's ATM network; this mechanism has no benefit in common channel signaling (CCS) configurations.

The PRI signaling channels, for example, in T1 the 24^{th} time slot and in E1 the 16^{th} time slot, are carried across the ATM network in the AAL5 PVC while the voice traffic, that is the bearer traffic, is carried by the AAL2 PVC. Thus, two different PVCs traverse the end-to-end network. One carries all the data traffic, and the other carries all the signaling traffic. D-channel information is transported across an AAL5 PVC because the signaling is in High-Level Data Link Control (HDLC) format. AAL2 does not support HDLC but AAL5 does.

NOTE CCS signaling protocols use HDLC framing, which is a link-layer protocol that provides variable length messages and supports a retransmission error correction capability to ensure 100 percent reliable data delivery. A D channel of an ISDN PRI line uses CCS protocols.

Fundamentals of AAL2

The AAL2 protocol has two layers:

- Service specific convergence sublayer (SSCS)
- Common part sublayer (CPS)

The SSCS encodes different information streams for the transport by AAL2 over a single ATM connection. The information streams might be active voice encodings, silence insertion descriptors, dialed digits, or fax. SSCS can provide error control on critical information (CAS signaling and dialed digits) by using a 10-bit CRC. This is called an AAL2 Type 3 cell. The SSCS segments the information that is being passed from a higher layer application, such as samples of voice from a digital phone into a number of units of data, and submits these units of data to the CPS for transmission. The length of the segmented data can be between one and the maximum length supported by the CPS connection, which is either 45 or 64 bytes. At the SSCS receiver, the units of data are reassembled back into the information before being passed to the higher layer application.

The second layer, the CPS, is specifically responsible for transporting end-to-end connections across the network. The format of AAL2 protocol structure is shown in Figure 4-2.

Figure 4-2 *AAL2 Protocol Structure*

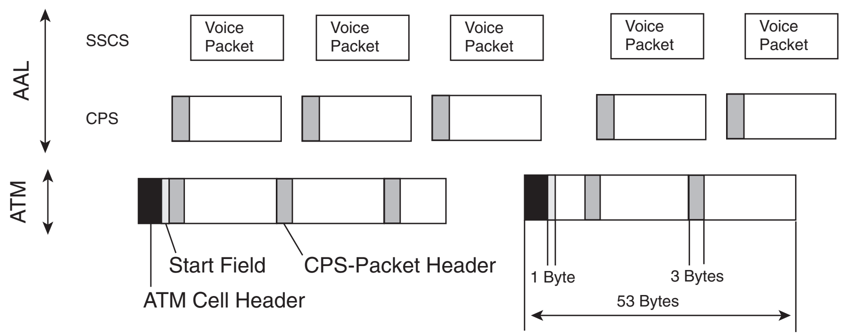
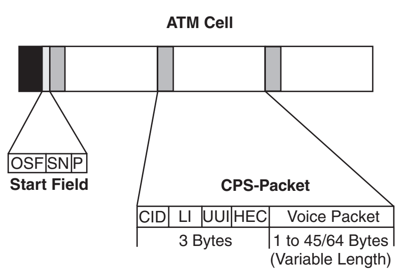


Figure 4-2 shows that AAL2 uses an additional byte of overhead for each ATM cell and an additional three bytes of overhead for each voice packet (e.g., compressed 8 kbps voice). The benefits of the AAL2 scheme are that there is no padding overhead except when there is insufficient data to complete a packet in a prespecified time interval, and the voice channels can be multiplexed over a single ATM virtual circuit.

The content of the Start field and the CPS packet are shown in Figure 4-3.

Figure 4-3 *Start Field and CPS Packet Formats*



The CPS layer enables the multiplexing of variable length voice packets of end users onto a single ATM virtual channel that is an AAL2 channel. This is accomplished through the different information fields shown in Figure 4-2. Although AAL2 with its three-byte packet header introduces some inefficiency for small packets, the improvement that is reached by having no padding more than offsets this minor inefficiency. Each of the CPS fields and the Start field are described here:

- **Start field**—Enables efficient packing of the voice packets over a single ATM virtual circuit. The Offset field is a six-bit pointer within the Start field that points to the position of the first CPS packet that follows the OSF. A sequence number protects the order of the Offset field. If a Start field parity error exists, all the CPS packets that are associated with the Start field are discarded.

- **CID (Channel ID)**—Identifies the end user, which is referred to as the SSCS entity in the International Telecommunication Union (ITU) AAL2 specifications. The CID allocates the value 1 to exchange layer management peer-to-peer procedures, such as set-up negotiations. CID enables the multiplexing of up to 248 user channels, whereas some CID values are reserved for other uses, such as peer-to-peer layer management. For example, if 8 E1s terminated on an MGX, 240 CID values would be used.
- **LI (Length Indicator)**—Identifies the length of the CPS packet. The default payload length is 45 bytes, and an optional maximum length of 64 bytes can be selected. The maximum length is channel specific.
- **UUI (User-to-User Indication)**—Provides two functions: It conveys specific information transparently between two end points (e.g., CPS or SSCS entity) and distinguishes between the different users, such as SSCS entities and layer management users.
- **HEC (Header Error Control)**—Discards the rest of the CPS packets until the next Start field. As a result, not all voice users residing on the single ATM virtual channel are affected by other end-user errors, which results in a higher end-to-end efficiency.

The CID is an important concept in AAL2. CIDs provide a binding between an endpoint and an AAL2 connection. This is the mechanism that binds the TDM traffic to the ATM traffic. For example, if a service provider needs to provision 100 DS0s between two sites for one of its enterprise customers, 100 CIDs are created across the ATM network. Furthermore, a unique coder-decoder (codec) type is assigned to each individual DS0 because the codec type is assigned to each CID through an AAL2. For example, individual customers in a multi-tenant building can each support multiple compression schemes over a single T1 access link. Each CID is configured and includes the following parameters: codec type, profile type, voice activity detection (VAD), DTFM Tones, and packet period for G.729. For example, to transmit DTMF tones transparently across the ATM PVC, DTMF must be enabled in the CID.

An AAL2 profile is a mechanism that the MGX 8850 uses to assign the compression and encoding scheme of the AAL2 trunking service. A profile is defined by a profile type, which is either an ITU standard or a custom type and a number. These profiles need to match on both ends of the network for the two end devices, such as PBXs, to interoperate. A profile is configured for each CID. For example, if the profile type is ITU and the profile number is 1, you must use G.711. In other words, the profile type and the profile number identify the compression type.

CID enables the use of subcell multiplexing, which provides many of the benefits of AAL2. If you use G.711, subcell multiplexing does not provide any value because G.711 already uses an 80-byte packet. The real advantage of subcell multiplexing is the G.729 encoding scheme. If you use G.729 with a packetization period of 30 milliseconds, three 10-byte packets of payload from one DS0 are packed into one ATM cell. Therefore, the efficiency of packing the voice sample into the ATM cell is increased threefold, and instead of 34 bytes of padding, only 14 bytes exist in the ATM cell. Assuming that VAD provides an additional 50 percent of bandwidth savings, G.729 subcell multiplexing uses approximately 6 kbps of bandwidth per DS0 channel of voice traffic. This is a significant amount of bandwidth savings.

Summary

This chapter provided an overview of managed voice and data services using ATM technology. AAL2 is an important component in providing a managed service using ATM. Because of efficient bandwidth use and the ability to transport different traffic types, AAL2 is used in trunking applications, such as interconnecting mobile wireless sites. Today, many service providers use a pure IP-based architecture to support this service.