Multiservice Access Technologies

Multiservice networking is emerging as a strategically important issue for enterprise and public service provider infrastructures alike. The proposition of multiservice networking is the combination of all types of communications, all types of data, voice, and video over a single packet-cell-based infrastructure. The benefits of multiservice networking are reduced operational costs, higher performance, greater flexibility, integration and control, and faster new application and service deployment.

A key issue often confused in multiservice networking is the degree to which Layer 2 switching and services are mixed with Layer 3 switching and services. An intelligent multiservice network fully integrates both, taking advantage of the best of each; most multiservice offerings in the marketplace are primarily Layer 2 based, from traditional circuit switching technology suppliers.

The Importance of Voice over IP

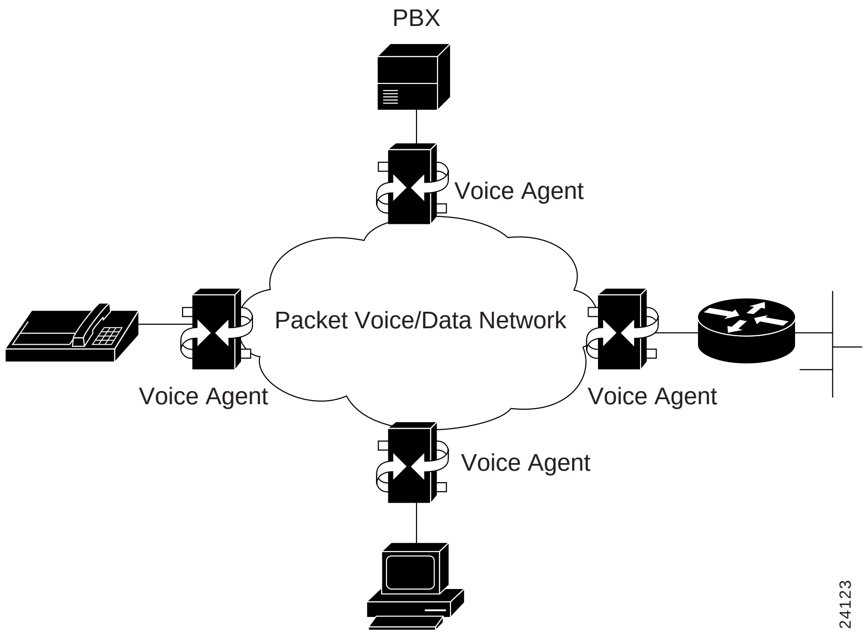
Of the key emerging technologies for data, voice, and video integration, voice over IP (Internet Protocol) is arguably very important. The most quality of service (QoS) sensitive of all traffic, voice is the true test of the engineering and quality of a network. Demand for Voice over IP is leading the movement for QoS in IP environments, and will ultimately lead to use of the Internet for fax, voice telephony, and video telephony services. Voice over IP will ultimately be a key component of the migration of telephony to the LAN infrastructure.

Significant advances in technology have been made over the past few years that enable the transmission of voice traffic over traditional public networks such as Frame Relay (Voice over Frame Relay) as well as Voice over the Internet through the efforts of the Voice over IP Forum and the Internet Engineering Task Force (IETF). Additionally, the support of Asynchronous Transfer Mode (ATM) for different traffic types and the ATM Forum's recent completion of the Voice and Telephony over ATM specification will quicken the availability of industry-standard solutions.

Packet Voice

All packet voice systems follow a common model, as shown in Figure 18-1. The packet voice transport network, which may be IP based, Frame Relay, or ATM, forms the traditional "cloud." At the edges of this network are devices or components that can be called *voice agents*. It is the mission of these devices to change the voice information from its traditional telephony form to a form suitable for packet transmission. The network then forwards the packet data to a voice agent serving the destination or called party.

Figure 18-1 This diagram displays the packet voice model.



This voice agent connection model shows that there are two issues in packet voice networking that must be explored to ensure that packet voice services meet user needs. The first issue is voice coding—how voice information is transformed into packets, and how the packets are used to re-create the voice. Another issue is the signaling associated with identifying who the calling party is trying to call and where the called party is in the network.

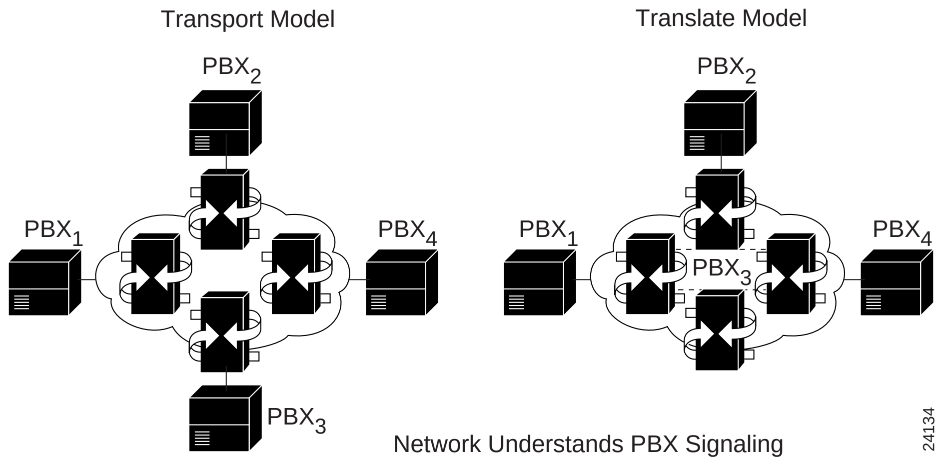
Packet Voice Transport

Integrating voice and data networks should include an evaluation of these three packet voice transport technologies:

- Voice over ATM (VoATM)
- Voice over Frame Relay (VoFR)
- Voice over IP (VoIP)

There are two basic models for integrating voice over data—transport and translate—as shown in Figure 18-2. Transport is the transparent support of voice over the existing data network. Simulation of tie lines over ATM using circuit emulation is a good example.

Figure 18-2 There are two basic models for transporting over a data network.



Translate is the translation of traditional voice functions by the data infrastructure. An example is the interpretation of voice signaling and the creation of switched virtual circuits (SVCs) within ATM. Translate networking is more complex than transport networking, and its implementation is a current topic for many of the standards committees.

Voice over ATM

The ATM Forum and the ITU have specified different classes of services to represent different possible traffic types for VoATM.

Designed primarily for voice communications, constant bit rate (CBR) and variable bit rate (VBR) classes have provisions for passing real-time traffic and are suitable for guaranteeing a certain level of service. CBR, in particular, allows the amount of bandwidth, end-to-end delay, and delay variation to be specified during the call setup.

Designed principally for bursty traffic, unspecified bit rate (UBR) and available bit rate (ABR) are more suitable for data applications. UBR, in particular, makes no guarantees about the delivery of the data traffic.

The method of transporting voice channels through an ATM network is dependent on the nature of the traffic. Different ATM adaptation types have been developed for different traffic types, each with its benefits and detriments. ATM Adaptation Layer 1 (AAL1) is the most common adaptation layer used with CBR services.

Unstructured AAL1 takes a continuous bit stream and places it within ATM cells. This is a common method of supporting a full E1 byte stream from end to end. The problem with this approach is that a full E1 may be sent, regardless of the actual number of voice channels in use. (An EI is a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 2.048 Mbps.)

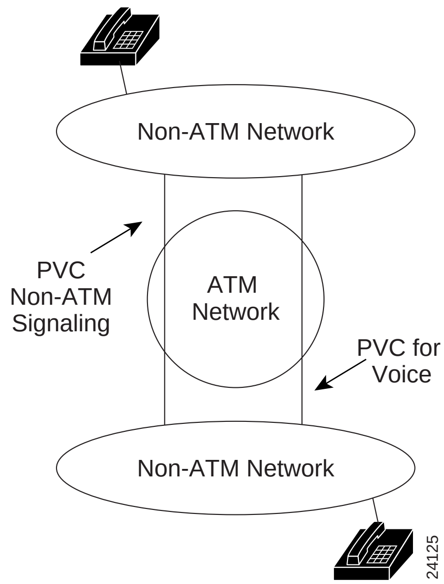
Structured AAL1 contains a pointer in the payload that allows the digital signal level 0 (DS0) structure to be maintained in subsequent cells. This allows network efficiencies to be gained by not using bandwidth for unused DS0s. (A DS0 is a framing specification used in transmitting digital signals over a single channel at 64 kbps on a T1 facility.)

The remapping option allows the ATM network to terminate structured AAL1 cells and remap DS0s to the proper destinations. This eliminates the need for permanent virtual circuits (PVCs) between every possible source/destination combination. The major difference from the above approach is that a PVC is not built across the network from edge to edge.

VoATM Signaling

Figure 18-3 describes the transport method, in which voice signaling is carried through the network transparently. PVCs are created for both signaling and voice transport. First, a signaling message is carried transparently over the signaling PVC from end station to end station. Second, coordination between the end systems allow the selection of a PVC to carry the voice communication between end stations.

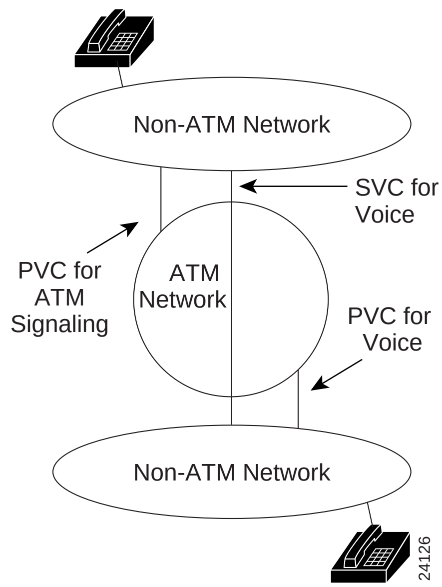
Figure 18-3 The VoATM signaling transport model describes the transport method, in which voice signaling is carried through the network transparently.



At no time is the ATM network participating in the interpretation of the signaling that takes place between end stations. However, as a value-added feature, some products are capable of understanding channel associated signaling (CAS) and can prevent the sending of empty voice cells when the end stations are on-hook.

Figure 18-4 shows the translate model. In this model, the ATM network interprets the signaling from both non-ATM and ATM network devices. PVCs are created between the end stations and the ATM network. This contrasts with the previous model, in which the PVCs are carried transparently across the network.

Figure 18-4 In the VoATM signaling translate model, the ATM network interprets the signaling from both non-ATM and ATM network devices.



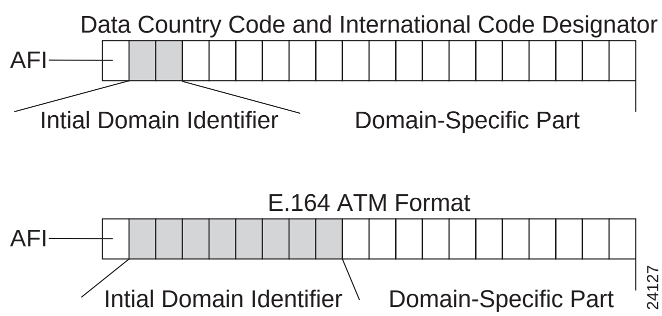
A signaling request from an end station causes the ATM network to create an SVC with the appropriate QoS to the desired end station. The creation of an SVC versus the prior establishment of PVCs is clearly more advantageous for three reasons:

- SVCs are more efficient users of bandwidth than PVCs.
- QoS for connections do not need to be constant, as with PVCs.
- The ability to switch calls within the network can lead to the elimination of the tandem private branch exchange (PBX) and potentially the edge PBX. (A PBX is a digital or analog telephone switchboard located on the subscriber premises and used to connect private and public telephone networks.)

VoATM Addressing

ATM standards support both private and public addressing schemes. Both schemes involve addresses that are 20 bytes in length (shown in Figure 18-5).

Figure 18-5 ATM supports a 20-byte addressing format.



The Authority and Format Identifier (AFI) identifies the particular addressing format employed. Three identifiers are currently specified: data country code (DCC), international code designator (ICD), and E.164. Each is administered by a standards body. The second part of the address is the initial domain identifier (IDI). This address uniquely identifies the customer's network. The E.164 scheme has a longer IDI that corresponds to the 15-digit ISDN network number. The final portion, the domain-specific part (DSP), identifies logical groupings and ATM end stations.

In a transport model you don't need to be aware of the underlying addressing used by the voice network. However, in the translate model, the ability to communicate from a non-ATM network device to an ATM network device implies a level of address mapping. Fortunately, ATM supports the E.164 addressing scheme, which is employed by telephone networks throughout the world.

VoATM Routing

ATM uses a private network-to-network interface (PNNI), a hierarchical link-state routing protocol that is scalable for global usage. In addition to determining reachability and routing within an ATM network, it is also capable of call setup.

A virtual circuit (VC) call request causes a connection with certain QoS requirements to be requested through the ATM network. The route through the network is determined by the source ATM switch based on what it determines is the best path through the network, based on the PNNI protocol and the QoS request. Each switch along the path is checked to determine whether it has the appropriate resources for the connection.

When the connection is established, voice traffic flows between end stations as if a leased line existed between the two. This specification spells out routing in private networks. Within carrier networks, the switch-to-switch protocol is B-ICI. Current research and development of integrated non-ATM and ATM routing will yield new capabilities to build translate level voice and ATM networks.

VoATM and Delay

ATM has several mechanisms for controlling delay and delay variation. The QoS capabilities of ATM allow the specific request of constant bit rate traffic with bandwidth and delay variation guarantees. The use of VC queues allows each traffic stream to be treated uniquely. Priority can be given for the transmission of voice traffic. The use of small, fixed-size cells reduces queuing delay and the delay variation associated with variable-sized packets.

Voice over Frame Relay

Voice over Frame Relay enables a network to carry live voice traffic (for example, telephone calls and faxes) over a Frame Relay network. Frame Relay is a common and inexpensive transport that is provided by most of the large telcos.

VoFR Signaling

Historically, Frame Relay call setup has been proprietary by vendor. This has meant that products from different vendors would not interoperate. Frame Relay Forum FRF.11 establishes a standard for call setup, coding types, and packet formats for VoFR, and will provide the basis for interoperability between vendors in the future.

VoFR Addressing

Address mapping is handled through static tables—dialed digits mapped to specific PVCs. How voice is routed depends on which routing protocol is chosen to establish PVCs and the hardware used in the Frame Relay network. Routing can be based on bandwidth limits, hops, delay, or some combination, but most routing implementations are based on maximizing bandwidth utilization.

The two extremes for designing a VoFR network are

- A full mesh of voice and data PVCs to minimize the number of network transit hops and maximize the ability to establish different QoS. A network designed in this fashion minimizes delay and improves voice quality, but represents the highest network cost.
- Most Frame Relay providers charge based on the number of PVCs used. To reduce costs, both data and voice segments can be configured to use the same PVC, thereby reducing the number of PVCs required. In this design, the central site switch re-routes voice calls. This design has the potential of creating a transit hop when voice needs to go from one remote to another remote office. However, it avoids the compression and decompression that occurs when using a tandem PBX.

A number of mechanisms can minimize delay and delay variation on a Frame Relay network. The presence of long data frames on a low-speed Frame Relay link can cause unacceptable delays for time-sensitive voice frames. To reduce this problem, some vendors implement smaller frame sizes to help reduce delay and delay variation. FRF.12 proposes an industry standard approach to do this, so products from different vendors will be able to interoperate and consumers will know what type of voice quality to expect.

Methods for prioritizing voice frames over data frames also help reduce delay and delay variation. This, and the use of smaller frame sizes, are vendor-specific implementations. To ensure voice quality, the committed information rate (CIR) on each PVC should be set to ensure that voice frames are not discarded. Future Frame Relay networks will provide SVC signaling for call setup, and may also allow Frame Relay DTEs to request a QoS for a call. This will enhance VoFR quality in the future.

Voice over IP

VoIP's appeal is based on its capability to facilitate voice and data convergence at an application layer. Increasingly, VoIP is being seen as the ideal last-mile solution for cable, DSL, and wireless networks because it allows service providers to bundle their offerings.

VoIP also offers service providers the ability to provision standalone local loop bypass and long distance arbitrage services. To provide a VoIP solution, signaling, routing, and addressing must be addressed.

VoIP Signaling

VoIP signaling has three distinct areas: signaling from the PBX to the router, signaling between routers, and signaling from the router to the PBX. The corporate intranet appears as a trunk line to the PBX, which signals the corporate intranet to seize a trunk. Signaling from the PBX to the intranet may be any of the common signaling methods used to seize a trunk line, such as fax expansion module (FXS) or E&M signaling. In the future, digital signaling such as common channel signaling (CCS) or Q signaling (QSIG) will become available. The PBX then forwards the dialed digits to the router in the same manner in which the digits would be forwarded to a telco switch.

Within the router the dial plan mapper maps the dialed digits to an IP address and signals a Q.931 call establishment request to the remote peer that is indicated by the IP address. Meanwhile, the control channel is used to set up the Real-Time Control Protocol (RTCP) audio streams, and the Resource Reservation Protocol (RSVP) is used to request a guaranteed QoS.

When the remote router receives the Q.931 call request, it signals a line seizure to the PBX. After the PBX acknowledges, the router forwards the dialed digits to the PBX and signals a call acknowledgment to the originating router.

In connectionless network architectures such as IP, the responsibility for session establishment and signaling is with the end stations. To successfully emulate voice services across an IP network, enhancements to the signaling stacks are required.

For example, an H.323 agent is added to the router for standards-based support of the audio and signaling streams. The Q.931 protocol is used for call establishment and teardown between H.323 agents or end stations. RTCP is used to establish the audio channels themselves. A reliable session-oriented protocol, Transmission Control Protocol (TCP), is deployed between end stations to carry the signaling channels. Real-Time Transport Protocol (RTP), which is built on top of User Datagram Protocol (UDP), is used for transport of the real-time audio stream. RTP uses UDP as a transport mechanism because it has lower delay than TCP and because actual voice traffic, unlike data traffic or signaling, tolerates low levels of loss and cannot effectively exploit retransmission.

Table 18-1 depicts the relationship between the ISO reference model and the protocols used in IP voice agents.

Table 18-1 The ISO Reference Model and H.323 Standards

ISO Protocol Layer	ITU H.323 Standard
Presentation	G.711,G.729, G.729a, G.726, G.728, G.723.1
Session	H.323, H.245, H.225, RTCP
Transport	RTP, UDP
Network	IP, RSVP, WFQ
Link	RFC 1717 (PPP/ML), Frame, ATM, X.25, public IP networks (including the Internet), circuit-switched leased-line networks

VoIP Addressing

An existing corporate intranet should have an IP addressing plan in place. To the IP numbering scheme, the voice interfaces appear as additional IP hosts, either as an extension of the existing scheme or with new IP addresses.

Translation of dial digits from the PBX to an IP host address is performed by the dial plan mapper. The destination telephone number, or some portion of the number, is mapped to the destination IP address. When the number is received from the PBX, the router compares the number to those mapped in the routing table. If a match is found, the call is routed to the IP host. After the connection is established, the corporate intranet connection is transparent to the subscriber.

VoIP Routing

One of the strengths of IP is the maturity and sophistication of its routing protocols. A modern routing protocol, such as Enhanced Interior Gateway Routing Protocol (EIGRP), is able to consider delay when calculating the best path. These are also fast converging routing protocols, which allow voice traffic to take advantage of the self-healing capabilities of IP networks. Advanced features, such as policy routing and access lists, make it possible to create highly sophisticated and secure routing schemes for voice traffic.

RSVP can be automatically invoked by VoIP gateways to ensure that voice traffic is able to use the best path through the network. This can include segments of arbitrary media, such as switched LANs or ATM networks. Some of the most interesting developments in IP routing are tag switching and other IP switching disciplines. Tag switching provides a way of extending IP routing, policy, and RSVP functionality over ATM and other high-speed transports. Another benefit of tag switching is its traffic engineering capabilities, which are needed for the efficient use of network resources. Traffic engineering can be used to shift traffic load based on different predicates, such as time of day.

VoIP and Delay

Routers and specifically IP networks offer some unique challenges in controlling delay and delay variation. Traditionally, IP traffic has been treated as "best effort," meaning that incoming IP traffic is allowed to be transmitted on a first-come, first-served basis. Packets have been variable in nature, allowing large file transfers to take advantage of the efficiency associated with larger packet sizes. These characteristics have contributed to large delays and large delay variations in packet delivery. RSVP allows network managers to reserve resources in the network by end station. The network manager can then allocate queues for different types of traffic, helping to reduce the delay and delay variation inherent in current IP networks.

The second part of supporting delay-sensitive voice traffic is to provide a means of prioritizing the traffic within the router network. RFC 1717 breaks down large packets into smaller packets at the link layer. This reduces the problems of queuing delay and delay variation by limiting the amount of time a voice packet must wait in order to gain access to the trunk.

Weighted fair queuing, or priority queuing, allows the network to put different traffic types into specific QoS queues. This is designed to prioritize the transmittal of voice traffic over data traffic. This reduces the potential of queuing delay.

Applying Packet Voice

In today's networking, there are several attractive alternatives both to conventional public telephony and to leased lines. Among the most interesting are networking technologies based on a different kind of voice transmission, called packet voice. Packet voice appears to a network as data; thus it can be transported over networks normally reserved for data, where costs are often far less than in voice networks. Packet voice uses less transmission bandwidth than conventional voice, so more can be carried on a given connection. Whereas telephony requires as much as 64,000 bits per second (bps), packet voice often needs less than 10,000 bps. For many companies, there is sufficient reserve capacity on national and international data networks to transport considerable voice traffic, making voice essentially free.

Packet voice networks can be used in two broad contexts, differentiated by geography or by the types of users to be served. The economics and technology of the network may be unaffected by these factors, but there may be legal constraints in some areas for some combinations of these two contexts, and network users or operators should be aware of them.

Telecommunications is regulated within countries by national administrations, or arms of the governments, based on local regulations. In some countries, such as the United States, there may be multiple levels of regulatory authority. In all cases, treaties define the international connection rules, rates, and so forth. It is important for any business planning to use or build a packet voice network to ensure that it is operating in conformance with all laws and regulations in all the areas the network serves. This normally requires some direct research, but the current state of the regulations can be summarized as follows:

- Within a national administration or telephony jurisdiction, it is almost always proper for a business to employ packet voice to support its own voice calling among its own sites.

  In such applications, it is normally expected that some of the calls transported on the packet voice network will have originated in the public phone network. Such outside calling over packet voice is uniformly tolerated in a regulatory sense, on the basis that the calls are from employees, customers, or suppliers and represent the company's business.

- When a packet voice connection is made between national administrations to support the activities of a single company—to connect two or more company locations in multiple countries—the application is uniformly tolerated in a regulatory sense.

  In such a situation, an outside call placed from a public phone network in one country and terminated in a company site within another via packet voice may be a technical violation of national monopolies or treaties on long-distance service. Where such a call is between company employees or between employees and suppliers or customers, such a technical violation is unlikely to attract official notice.

- When a packet voice network is used to connect public calls within a company, the packet voice provider is technically providing a local or national telephone service and is subject to regulation as such.

- When a packet voice network is used to connect public calls between countries, the packet voice provider is subject to the national regulations in the countries involved and also to any treaty provisions for international calling to which any of the countries served are signatories.

Thus, it is safe to say that companies could employ packet voice networking for any applications where traditional leased-line, PBX-to-PBX networking could be legally employed. In fact, a good model for deploying packet voice without additional concerns about regulatory matters is to duplicate an existing PBX trunk network or tie-line network using packet voice facilities.