PUBLIC SAFETY WIRELESS NETWORK PROGRAM

Saving Lives and Property Through Improved Interoperability

Software-Enabled Wireless Interoperability Assessment Report – Voice-Over-Internet Protocol Technology

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EXECUTIVE SUMMARY

The land mobile radio (LMR) industry is migrating to a new generation of control architecture based on packet-switching technology. This technology migration will likely result in a fundamental shift from the existing circuit-switched architecture to a packet-based architecture using the well-known Internet Protocol (IP). Packet networks were not designed for handling real-time data such as voice. However, as network speed and capacity have increased and new protocols have been developed, it is now practical to discuss the transmission of voice over packet networks using IP and the potential benefits and risks of leveraging this approach in the LMR environment.

This report provides the Public Safety Wireless Network (PSWN) Program with an analysis and evaluation of the state of voice-over-Internet Protocol (VoIP) technology use in the LMR environment. The report identifies issues, provides a risk/benefit analysis, and presents information on systems currently using VoIP technology.

Several technical considerations identified in this paper are critical to the quality of service (QoS) of voice information carried over packet networks. This report identifies and discusses these considerations. While the protocols that guide the implementation of VoIP service in the telephony arena have addressed these considerations, they must be addressed anew for the LMR environment. Existing protocols describe implementation of VoIP for integrated services digital network (ISDN) and local area network (LAN) conferencing, the World Wide Web (www), and other "wired" purposes. However, LMR systems exist in an environment in which effects of electromagnetic wave propagation influence voice quality and grade of service. As this report discusses, some of these considerations have been addressed in the context of LMR, and some have not, leaving a standards gap for VoIP implementation in LMR.

VoIP technology offers many potential advantages over existing circuit-switched technology as a future platform for transport in the LMR environment. These advantages include potential cost savings, simplification of networks, standardization and commonality of equipment, off-the-shelf networking equipment, built-in redundancy of packet networks, "virtual" network service over a common infrastructure, privacy between user groups, and advanced functionality not available in circuit-switched networks.

A number of potential risks are associated with applying VoIP to LMR, including technical risk, compatibility among systems from different manufacturers, and the risk of bearing development costs. There are also risks associated with not moving forward with technology, chiefly the risk of obsolescence associated with circuit-based technology.

This paper identifies and discusses key capabilities currently available in advanced LMR systems and potentially available as a result of implementing VoIP in an LMR environment. These capabilities include the ability of vendors to field a system today, system architecture approaches, voice and data integration, segregation in an integrated system, wide area capability, trunking, encryption and over-the-air rekeying (OTAR), and standards compliance. The ability of current and emergent products to provide these capabilities varies depending on the vendor.

Research identified four domestic vendors offering or planning to offer VoIP-based LMR systems: Motorola, EF Johnson, Tyco Electronics (MA/COM Open Sky), and Catalyst Communications. This report also discusses SIMOCO, a European vendor, because of its significant experience in implementing TETRA systems using VoIP. The only domestic vendor with an installed systems base in the United States is MA/COM Open Sky; however, the current product line is limited to the 800 megahertz (MHz) frequency band. Further, the Open Sky system does not and will not comply with the Association of Public-Safety Communications Officials (APCO) 25 standard on a systemwide basis. Motorola intends to offer an APCO 25 compliant system; however, it will be limited to trunking for the foreseeable future. None of the vendors surveyed can deliver an IP-based console at this time, which is a critical component in providing some IP-based services, and none of the vendors has provided a clear indication of how it intends to provide encryption and OTAR.

Several issues remain to be resolved, including the capability of systems to provide acceptable latency and class of service, scalability, reliability, security, and standards compliance. In addition, the availability of IP-based dispatch consoles remains an issue. Finally, the true cost impact of VoIP compared with circuit switching appears to depend on individual vendor architectures. Therefore, the cost impact of VoIP cannot be generalized.

In conclusion, packet switching has the potential to considerably reduce costs associated with channel bank and central switching equipment as well as costs associated with leased lines. Packet-switched transport networks using end-to-end IP protocols promise to bring advanced features and capabilities not available with circuit-switched networks. However, it is likely that dedicated networks will be required to preserve QoS in LMR systems.

Further, no standards are emerging to provide for compatibility among VoIP transport-based LMR systems, and VoIP systems currently available do not comply with the LMR standard most commonly accepted for public safety, the EIA/TIA 102 suite. To date, only one manufacturer has delivered an LMR system providing true end-to-end IP addressing—MA/COM. Motorola and EF Johnson are not scheduled to deliver systems until late 2001, at the earliest.

1. INTRODUCTION

This report provides the Public Safety Wireless Network (PSWN) Program with an assessment of Voice-over-Internet Protocol (VoIP) technology. While the world of Internet telephony (tuh-LEF-uh-ne) is already benefiting from the use of VoIP technology, it appears that applying this technology to land mobile radio (LMR) could also realize potential benefits. This report discusses VoIP technology as it would apply to LMR systems and describes how the technology might impact interoperability. More specifically, this report provides an analysis and evaluation of the state of VoIP technology use in the LMR environment. Beyond providing the PSWN Program with an assessment of the current state of the technology, it identifies issues, provides a risk/benefit analysis, and presents information on systems currently using VoIP technology.

1.1 Background

The LMR industry is migrating to a new generation of control architecture based on packet-switching technology. This technology migration will likely result in a fundamental shift from the existing circuit-switched architecture to a packet-based architecture using the well-known Internet Protocol (IP). Until recently, packet networks were not suitable for handling real-time data such as voice. However, as network speed and capacity have increased and protocols have been developed, it is now practical to discuss the transmission of voice over packet networks using IP. As LMR systems migrate from analog to standards-based digital networks and system planners envision larger networks, the concept of using packet networks with IP addressing has superseded the idea of replacing circuit-switched control of wide-area LMR repeater, base station, and control console networks. The ultimate solution is to use VoIP in a packet-switched network, replacing the complex time division multiplex (TDM) audio switches and circuit-based connections now used. Packet networking using IP has the potential to enable enhancements such as full-featured LMR data services, as well as true, end-to-end encryption.

1.2 Organization of Report

This report consists of five main sections, including this introductory section. Section 2, VoIP Technical Discussion, describes the differences between circuit- and packet-switched networks, the basics of IP transport protocol, and the basics of VoIP. Section 2 also includes an overview of applicable standards and a discussion of VoIP in the LMR environment, including vendor approaches and progress. Section 3, Survey of Major Vendor Offerings, collates the results of the vendor data-gathering effort, summarizes current vendor offerings and capabilities, and identifies and discusses issues associated with the technology. Section 4, Risk/Benefit Analysis, summarizes the advantages, disadvantages, and risks associated with VoIP for LMR systems. Section 5, Conclusions and Recommendations, summarizes conclusions and presents recommendations and next steps.

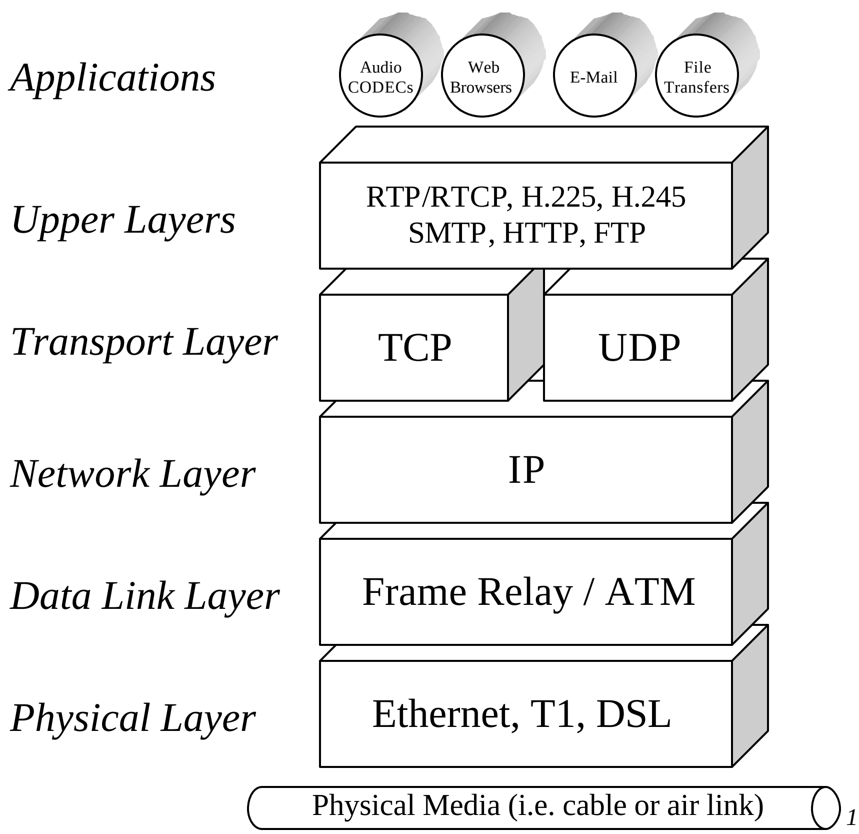
2. VOIP TECHNICAL DISCUSSION

This section provides a brief overview of the Internet Protocol (IP), compares circuit switching and packet switching, introduces VoIP protocols, and discusses approaches to applying VoIP to the LMR environment.

2.1 Basics of the Transport Control Protocol/IP

To understand many of the issues associated with VoIP, a basic understanding of the Transport Control Protocol (TCP)/IP suite is necessary. The TCP/IP suite operates in a layered fashion to provide data-networking services. The TCP/IP "stack" is based on the open systems interconnection (OSI) model. In this model, each layer communicates to its peer layer across the network and provides service to the layer above it. The World Wide Web (WWW) and Internet are both based on TCP/IP. Shown in Figure 1is a simplified view of the layered protocol stack, with TCP/IP components indicated at the appropriate layers.

Figure 1
Layered Protocol Stack



[1] Definitions for acronyms in Figure 1: RTP is the Real-Time Protocol; RTCP is the Real-Time Control Protocol; SMTP is the Simple Mail Transport Protocol; HTTP is the Hypertext Transport Protocol; FTP is the File Transfer Protocol; UDP is the User Datagram Protocol; ATM is asynchronous transfer mode; and DSL is digital subscriber line.

Briefly, the Physical and Data Link layers primarily describe the physical means of transporting the data, including the electrical characteristics of the connections and how data flow is controlled on the wires. When analyzing VoIP, the protocols operating in the Network, Transport, and Upper layers are of primary interest. The lower layers are not described in detail in this analysis.

The IP resides in the Network layer. This protocol describes how packets are addressed and how network hosts are uniquely identified. Although each packet's Data Link layer information is "re-built" with every hop through the network, the Network layer carries information (i.e., IP addresses) indicating physical location of the origin and destination—therefore, it stays intact from end to end. This is important information used in the process of routing information through the network.

Network layer protocols are designed to allow computers to communicate over the Internet. IP header information provides addressing to identify the source and destination addresses of the data to be transferred. These addresses, referred to as IP addresses, represent discrete interface points on the Internet. The information transferred to and from host computers is called the payload or datagram. IP header information provides other information used by the end points to process the data.

The primary function of the Transport layer is to control data integrity. IP is connectionless, which means no "connection" is set up for the communication—virtual or otherwise; packets are sent independent of one another and can take different routes. IP does not specify routing of the message nor does it provide for guaranteed delivery of the information on an end-to-end basis. However, TCP/IP defines two major components at the Transport layer, TCP and User Datagram Protocol (UDP). UDP is connectionless and provides no guarantee of delivery but does provide payload error detection. TCP guarantees delivery of IP packets between two points using a handshake, which is a set of signals (bits) exchanged between the receiver and sender to verify receipt of the packet. TCP provides for sequencing of packets, error detection, and retransmission of lost or late packets. Because IP does not specify routing, the network routers identify the best route on a packet-by-packet basis.

The Upper layers in the protocol stack illustrated in Figure 1 provide services describing when the job of moving data is complete and in what language the data is encoded. Section 2.4 discusses Upper layer protocols relevant to VoIP.

As stated earlier, IP is a connectionless protocol in which packets can take different paths between end points, and packets from different transmissions share all paths. This approach enables efficient allocation of network resources because packets are routed on the paths with the least congestion. Header information ensures that packets reach their intended destinations and helps reconstruct messages at the receiving end. To ensure quality of service (QoS), however, all packets should use the same path. IP headers are large (20 bytes) compared with the headers of frame relay frames (2 bytes) and asynchronous transfer mode (ATM) cells (5 bytes). Headers do not actually carry the message information, which is called the "payload" and are therefore known as "overhead." Headers add a great deal of overhead to IP traffic.

2.2 Circuit Switching Versus Packet Switching

This section describes the two major types of switching used in communications: circuit switching and packet switching.

Circuit switching is simply tying two communication lines together temporarily to complete a call. Figure 2a depicts the digital circuit-switched architecture that can be used to carry voice calls in a public or private network. To set up a call between two telephones, the switches and the intervening network must establish a route from one end of the call to the other.

In this example, it is assumed that the left side of the diagram represents the calling party or the local station, and the right side of the diagram represents the called party, or the remote station. In practice, they are interchangeable. Through the local switch, the calling station on the local loop is connected to the assigned inter-switch channel, carried over the network backbone. To complete the connection, the remote switch must connect the inter-switch channel with the local loop on which the telephone resides. The local and remote switch matrix at either end could be space division (i.e., using relays or voltage-controlled diodes) or time division (i.e., using time slots in a time-slot exchange network), while the voice backbone typically uses time slots or TDM. Each call must have its own channel, which is used for the duration of the call. Therefore, the capabilities of the switches supporting the circuit limit the number of simultaneous calls.

At this point, the reader may wonder how the switch at the remote (or called) end "knows" which inter-switch channel to connect to which local loop. The answer is that the circuit-switched architecture requires a signaling channel separate from the traffic channels to provide call setup and teardown. Switches on the voice network must be connected to the signaling channel to receive instructions for completing calls. This signaling is often referred to as common channel (inter-office) signaling. The public switched telephone network (PSTN) uses a common channel signaling scheme called Signaling System 7 (SS7).

Packet switching can be thought of as an extension of TDM. As shown in Figure 2b, voice signals are "sampled" or digitized at the originating end and converted from digital to analog signals at the receiving end. While these digital signals could be transported using circuit-switching techniques as described above, packet switching provides a more efficient alternative. In packet networks, data can be made to share simple networks—because the networks are very fast, the entire bandwidth of the network can be chopped into time slots and divided among its users. Each user could be assigned a unique time slot, as with a TDM switch. Figure 2a, however, depicts a different approach. Figure 2a indicates that if data to be transferred is packetized (i.e., divided into segments and packaged with information identifying the packet sequence, the sender, and receiver) the network can be shared more efficiently among a larger number of users.

Because packets can be sent as independent data messages, with their own addressing and routing information, physical circuits are not required, and users with no packets to send or receive will not be using bandwidth. This means that the entire bandwidth is always available to carry traffic. The greater the peak traffic, the greater the required bandwidth, which is expressed in terms of bits (of data) per second. Packet switches (along the way) read each packet's addressing and sequencing information, and identify the best available routing for the packet. At the receiving end, packets are buffered and reassembled into data streams such that it appears to the receiver that he or she is connected to the sender by a circuit. In reality, the circuit is virtual, giving rise to the term "virtual circuit" or "virtual connection."

Figure 2a
Circuit-Switched Voice Networking

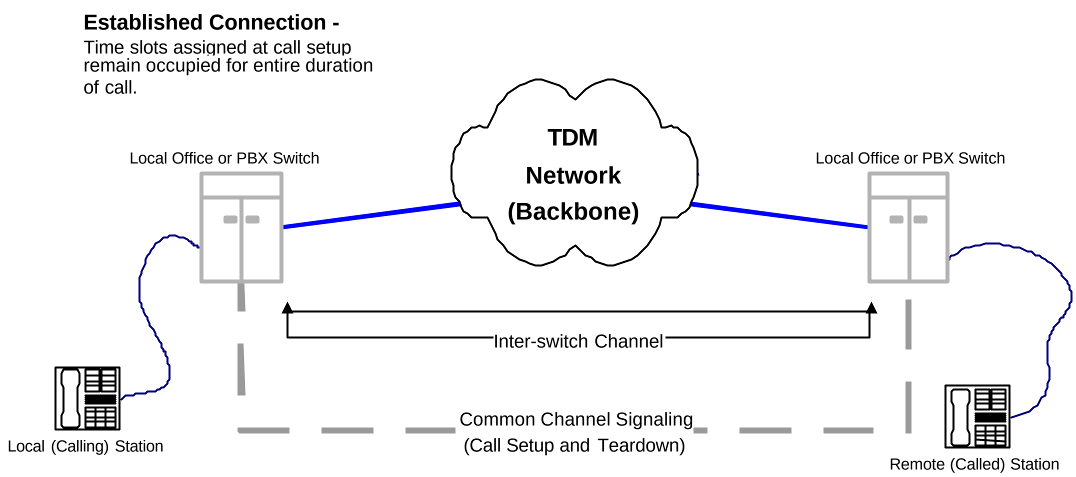
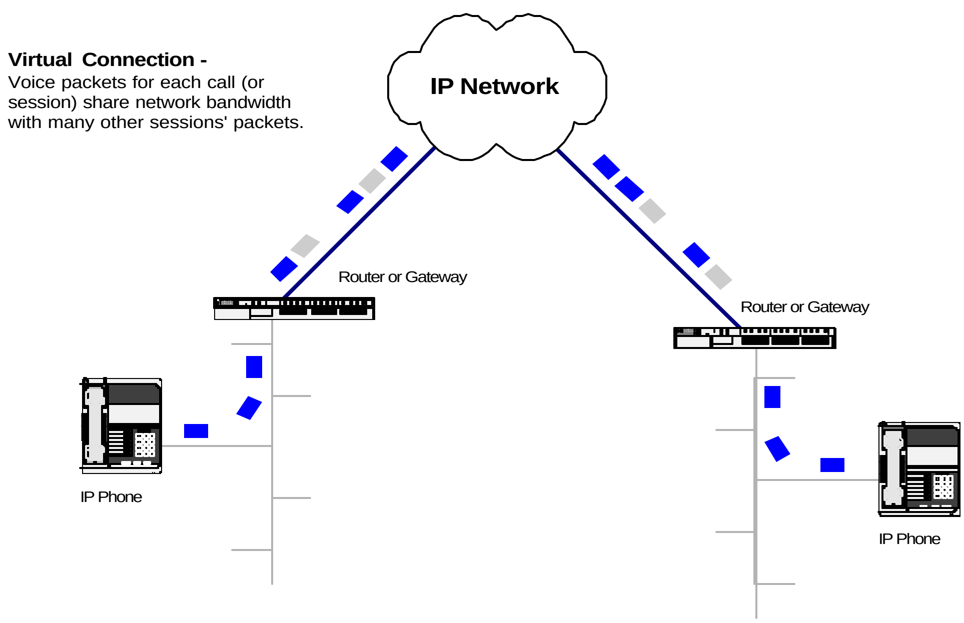


Figure 2b
Packet-Switched Voice Networking



In the packet network, redundant channels are not required to ensure delivery because data is divided into pieces with a high probability of reaching their destination in a reasonable time. Because packet data networking is a mature technology and is ubiquitous in the form of intranets and the Internet, IP networking is readily available. To mix voice and data on IP networks, however, several factors must be considered. Outlined below are some of the key considerations for implementing integrated voice and data IP networks.

- **Voice Coding**. Voice coding used in packet networks must preserve the voice waveform so that audio is reproduced with minimum discernable difference from that provided by the pulse coded modulation (PCM) encoding used in circuit-switched networks. Encoding at 64 or 32 kilobits per second (Kbps) is adequate in modern day TDM networks, and voice channels are typically encoded at one of these two bit rates. These rates, however, are impractical for transmission across crowded IP networks.

- **Signaling.** The packet-switching network does not use common signaling channels to control and manage the network. Addressing and overhead information are contained in each packet, along with the transported data, or payload. As discussed in the previous section, addressing must be included in the packets themselves, so that it is available to each node the packet passes through in the network.

  Encoding at high bit rates, however, can cause problems in IP networks. High bit rates require more or larger packets to carry voice information. Larger packets travel more slowly in a crowded network. Using smaller packets requires sending a larger number of individual packets, wasting bandwidth through constant repetition of the address information with each packet. Therefore, packet size must be managed in converged networks, depending on whether voice or data is being carried.

- **Network Congestion**. Where network congestion can be avoided and large packets are allowed, packet (i.e., IP) audio quality is comparable with that found in high bit rate (low compression) circuit-switched facilities such as a DS0 channel or a dedicated leased line.

  In a circuit-switched network there is a one-to-one relationship between calls and TDM channels. Packet switching provides a completely different scenario. A packet network carries messages from many different points to many different points over many different links. This architecture has been referred to as resembling a “web” or a “cloud.” As one moves from the edge of the web (i.e., the ingress and egress points of the packet network) inward, the communications links become larger, carrying greater bandwidth, and therefore a greater number of sessions. The deeper one looks into the web or cloud, the more sessions each link carries. Every session has a pattern of packets competing for the available bandwidth in the channel.

- **Noise and Distortion**. In a wireline network, distortion caused by noise or intermittent connections is virtually non-existent. In a packet network of the same reliability, distortion and dropouts can occur, not due to bad connections but due to lost packets within the network itself. The principal cause of packet loss is network congestion. When instantaneous network capacity is fully used by other data traffic with higher precedence or earlier arrival times, voice packets can be stored temporarily (queued) as long as their “time-to-live” parameter is not exceeded. Once time-to-live expires, the network routing devices throw the packets away. As explained above, shorter packet length can increase the probability of interleaving with other sessions. However, short packet length increases the number of packets, increasing the likelihood that packet life will expire or “time out.” When possible, assignment of high precedence in the addressing portion of voice packets can force the network routing devices to throw away other, non-voice, packets. To the extent that other sessions can tolerate such abuse, this strategy can minimize voice signal distortion.

- **Latency**. While a network can cleanly encode voice into packets and not lose them, the transmitted audio can still be unsatisfactory if overall delay is too high. For received speech to be acceptable, the end-to-end delay cannot exceed about 150 to 200 milliseconds (ms). Several processes contribute to delay, but encoding and decoding and routed network delays are by far the biggest culprits. Of the total 150 ms delay “budget,” voice coding and decoding can contribute up to 55 ms. Depending on the size of the network, transit time can add approximately 100 ms. Buffering and queuing in each router’s memory can add another 100 ms. Clearly, where network size (meaning distance between nodes) cannot be controlled, network speed (bandwidth) and routing design must be controlled.

Two other requirements are low transmission error rate and total reliability. Communications managers in the law enforcement environment expect 99.999 percent reliability (i.e., approximately 5 minutes of downtime per year). Today’s packet networks approach this level of reliability only with very careful design. Therefore networks that transport public safety communications should be carefully designed and controlled, which likely means design of special-use private networks.

Further, law enforcement demands highly secure communications, with a high level of resistance to eavesdropping or interception. Because packet networks share one “cloud” among several communities of interest and may depend on public carriers, encryption is virtually mandatory. Encryption, however, increases end-to-end delay. Encryption requires extreme care in network design, or overall delay budgets will almost certainly be exceeded.

Summarized in Table 1 are some of the more specific factors users must consider before implementing an IP packet-switching network.

Table 1
Considerations for Using IP Packet Switching

Performance Challenge	Technical Description	Underlying Causes	Mitigation Strategy
Voice Clarity	Audio frequency response and harmonic distortion must be toll quality, i.e., equivalent to the quality of a telephone call.	• End-to-end bandwidth limitations prohibit use of the 64 Kbps or 32 Kbps voice encoding that is typical of circuit-switched networks.	• Use high-compression 8 Kbps or 6.3/5.3 Kbps vocoding to minimize required voice bandwidth. • Use QoS protocols: RTP, RSVP, and other future protocols (TBD).
Voice Intelligibility	Nonlinearity distortion due to transmission errors must not affect speech intelligibility.	• Bit error rate (BER) of network media may vary from very low (for fiber optics) to relatively high (for copper pairs or radio frequencies [RF]). RF exhibits a relatively poor signal-to-noise ratio.	• Apply forward error correction.
Audio Dropouts	Lost or discarded packets cause audio dropouts.	• Network congestion and routers with insufficient memory cause this problem. • Use of short packet size increases the number of packets and overall transmission redundancy.	• Increase *precedence* in voice packets' Type of Service (TOS) field. • Increase packet length. • Ensure all routers have sufficient memory.
Variable Delay	Short-term receive packet buffering (jitter) increases delay.	• Packet routes continually vary, potentially causing packets to arrive out of order (randomly). • Packets must be queued—causing delays—to restore their correct order.	• Reduce total number of network nodes. • Increase *precedence* in voice packets' TOS field.

Performance Challenge	Technical Description	Underlying Causes	Mitigation Strategy
Overall Delay	Delays from end-to-end accumulate throughout the network.	• Combination of transmit compression, buffering of routed packets at each node, insufficient end-to-end network speed, anti-jitter receive buffering (packet queuing), voice decompression, forward error correction, and encryption, and end-to-end network delay can exceed the maximum possible to maintain voice intelligibility.	• Decrease packet length to reduce packetizing delay. • Use the higher bit rate Low-Delay Code-Excited Linear Prediction (LD-CELP) 16 Kbps vocoding (and reduce coding delay). • Simplify network (to minimize packet buffering). • Do not use forward error correction. • Eliminate encryption.
Network Reliability	End-to-end mean time between failures (MTBF) is low.	• As networks grow more complex, the number of failure points increases, lowering the MTBF figure.	• Reduce network complexity. • Apply hardware and data redundancy.
Network Security	Tactical use dictates strong security for protection of voice.	• Outside or hostile parties generally have greater access to packet networks than to circuit-switched networks.	• Apply *encryption* to voice packets.

2.3 Introduction to VoIP Protocols

The term "voice over IP" refers to the transmission of voice over a packet network using IP. During rapid expansion of the Internet in the 1990s, access to the Internet became inexpensive and ubiquitous. It is common for Internet access to be billed as a flat monthly fee for unlimited access, compared with long distance telephone access, which is charged by the minute. Because of this disparity, interest grew in saving tolls by using the Internet to carry voice information over long distances. In the most basic example, VoIP is implemented using a personal computer (PC) equipped with a headset, microphone, and an application program such as Microsoft Net Meeting.

Although VoIP was originally envisioned as a way to avoid long distance tolls, protocols have been developed by standards bodies to ensure compatibility among voice telephony users. Further, the cost of packet-switching equipment has fallen while network performance has increased rapidly. Because of these trends, service providers are emerging to provide integrated voice and data services over managed IP networks.

VoIP represents a family of protocols used to transfer voice information over packet data networks using the IP. In general, VoIP uses the UDP at the Transport layer on top of IP.

As indicated in the discussion on the concept of latency in Section 2.2, two critical factors affect speech quality in packet networks: end-to-end delay and lost or late packets.

However, because IP was originally designed and built to transmit data, it only ensures that all packets are delivered uncorrupted. IP is not concerned with the order of arrival or with latency; these issues are addressed by TCP in the Transport layer. At the time that IP was developed, networks were not capable of delivering data with the speed required for real-time applications. Today's network technology, however, provides for the possibility of sending real-time data over packet networks, if appropriate protocols are used to manage the data flow. Three parameters must be managed to provide sufficient QoS for transfer of real-time data: latency, bandwidth, and packet loss/desequencing. These parameters are optimized through enhancements in the end points and protocols.

To address these issues, VoIP uses the Real Time Protocol (RTP) at the Network layer to provide a sequence number and timestamp that can be used by an audio application to manage a buffer. This allows the network nodes and end points to handle VoIP packets so that latency and packet loss/desequencing are minimized. RTP also provides information on the payload type so that the application knows the format of the data, enabling processing without preliminary analysis.

VoIP uses the Real Time Control Protocol (RTCP), along with RTP to transfer information about quality of transmission on the network, which provides valuable information for managing bandwidth. RTCP can be thought of as the "quality control" component of the RTP. Devices called gateways act as interfaces between the IP packet network and other protocols and formats, including analog audio.

A new version of IP, known as IPv.6 has been under development for many years. It is anticipated that IPv.6 will recognize voice and data packets and route them with multiple priority levels to improve QoS for voice traffic. Major concerns, however, have surfaced relative to overlaying IPv.6 on the current version of IP, known as IPv.4. In the interim, IPv.4 has been enhanced to provide IPv.6-like functionality using the Resource Reservation Protocol (RSVP) and RTP.

2.4 VoIP Standards Overview

Several families of standards describe upper level protocols for VoIP. The four most common families are discussed in this section: H.323, Session Initiation Protocol (SIP), Media Gateway Protocol (MEGACO), and Bearer Independent Call Control (BICC). In addition, a proposed standard for LMR is discussed, FSTG/00/08/00 Project 25 Fixed Station Interface Overview and Definition—Conventional Systems. Section 2.4.5 summarizes these standards.

2.4.1 H.323

The H.323 is an International Telecommunications Union (ITU-Telecommunications) recommendation describing an architecture for operating video-conferencing systems over packet networks. H.323 is not specific to IP networks; it may also be used over IPX/SPX (Novell) or ATM networks. Currently, the second version of this protocol suite is being implemented. The protocol suite consists of a set of protocols responsible for encoding, decoding, and packetizing audio and video signals, and for providing call signaling and control, as well as capability exchange. Part of the capability information exchanged by H.323 includes unique identifiers for ITU-recognized vocoders. However, the suite of vocoders included in

H.323 are not intended for wireless use; with the exception of G.723.1, they are high bit rate and are not sufficiently robust for a wireless public safety environment.[2]

H.323 specifies gateways for connections between the packet-switched network and the switched-circuit network. The gateway performs call setup and control on both networks and provides translation between transmission formats and communication procedures.

H.323 also specifies optional gatekeepers, which have mandatory functions when they are used. These functions include address translation (i.e., from alias address or phone number to network address), admission control, bandwidth control, and zone management. Four optional functions of gatekeepers include call control signaling, call authorization, bandwidth management, and call management.

Multipoint control units (MCU) support conferencing between three or more end points. The MCU typically consists of a multipoint controller (MC) and zero or more multipoint processors (MP). The MC provides control functions such as negotiation between terminals and determination of common capabilities for processing audio or video. The MP performs the necessary processing on the media streams for conferencing (typically audio mixing and audio/video switching).

2.4.2 SIP—Session Initiation Protocol

SIP provides real-time transmission of information. It is faster than e-mail and uses a process called "forking" to send data to multiple end points simultaneously or in sequence. SIP is not designed for bulk transport applications, such as streaming media, files, or pictures, nor is it designed for asynchronous messaging applications such as e-mail.

SIP has two major architectural elements: the user agent (UA) and the network server. The UA contains two elements: the user agent client (UAC), which is responsible for issuing SIP requests, and the user agent server (UAS), which responds to requests. The most generic SIP operation involves a SIP UAC issuing a request, a SIP proxy server acting as an end-user location discovery agent, and a SIP UAS accepting the call. A successful SIP invitation consists of two requests: INVITE followed by ACK. The INVITE message contains a session description that informs the called party what types of media the caller can accept and where the media data should be sent. SIP addresses are referred to as SIP Uniform Resource Locators (SIP-URL), which are of the form: sip.user@host.domain. SIP message format is based on the Hypertext Transfer Protocol (HTTP) message format, which is text based and human readable.

SIP provides call setup and teardown functionality, and industry finds it attractive because of its simplicity, services, and cross-network communication and expansion abilities. SIP's simplicity is a significant advantage. One can create a functional SIP application by parsing only a few headers. Furthermore, because it is a text-based protocol, it is easy to debug. Ease of debugging is critical because most cable telephony devices are still under development.

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[2] "Voice-Over-Intranet Protocol for Critical Communications," by Jay Herther and Bill Haymond, *Mobile Radio Technology Magazine*, August 2001.

Finally, SIP is attractive because of its expandability. SIP can not only incorporate new features, but it has specific provisions to preserve backward compatibility.

2.4.3 MEGACO—Media Gateway Control

The MEGACO is a result of joint efforts of the Internet Engineering Task Force (IETF) and the ITU-T Study Group 16. The MEGACO protocol is used between elements of a physically decomposed multimedia gateway. There are no functional differences from a system view between a decomposed gateway, with distributed subcomponents potentially on more than one physical device, and a monolithic gateway. This protocol creates a general framework suitable for gateways, MCUs, and interactive voice response (IVR) units.

Packet network interfaces may include IP, ATM, or possibly others. The interfaces support a variety of switched-circuit network signaling systems, including tone signaling, Integrated Services Digital Network (ISDN), ISDN User Part (ISUP), ISDN Q Signaling (QSIG), and the Global System for Mobile Communications (GSM). MEGACO supports national variants of these signaling systems where applicable. All messages are in the format of Abstract Syntax Notation (ASN.1) text messages.

The MEGACO protocol provides a means to manage media gateways that convert the user traffic between telephony and packet networks. Unlike earlier attempts at defining this interface, the MEGACO standard is likely to be more widely accepted across the data (IETF) and telephony (ITU-T) worlds and is better designed to handle enhanced services. It enables multimedia applications, IVR support, conferencing solutions, wiretap, speech-to-text conversion, and many services.

2.4.4 BICC—Bearer Independent Call Control (ITU Q.765.5, Q.1901)

BICC is essentially an adaptation of ISDN and SS7 signaling to make them "bearer independent" and therefore compatible with TCP/IP transport. Basically, this standard represents the telecommunications industry's position on delivering signaling for VoIP. Two primary approaches are recommended:

- **Q.765.5**. This recommendation describes the extensions required for the transport of bearer-related information associated with BICC as defined in Q.1901. BICC is used to manage the call control instance that has been separated from the bearer control instance. BICC needs to transport bearer-related information between call control instances. The Application Transport Mechanism (APM) defined in Q.765 is used for this purpose. This recommendation specifies the APM user to support the transport of the bearer-related information for the BICC.

- **Q.1901**. This recommendation describes the adaptation of the narrowband ISUP for the support of narrowband ISDN services independent of the bearer technology and signaling message transport technology used.

2.4.5 FSTG/00/08/00 Project 25 Fixed Station Interface Overview and Definition—Conventional Systems

This technical bulletin was developed by the EF Johnson Company as a standard for a fixed station interface to the RF subsystem. The proposed standard describes an analog interface as mandatory, and two digital interfaces as standard options. Digital interfaces are defined for serial link (circuit-based) transport and for VoIP (packet-based) transport. For this report, only the VoIP interface is of interest.

The VoIP interface is designed to carry either voice or data, in the encrypted mode. The proposed standard defines the interface in terms of a multiport router that routes messages from an Ethernet interface to the common air interface (CAI) (i.e., the radio part of the fixed station) and fixed station control logic. The description of the interface implies that the P25 CAI Protocol will be "tunneled" through the interface, which means that the P25 CAI Protocol will remain intact from end point to end point and be carried in its entirety as payload over the IP network.

The standard defines the VoIP interface in terms of four of the structured protocol layers: the Physical layer, the Data Link layer, the Network layer, and the Transport layer.

- The Physical layer (layer 1) is described as meeting Institute of Electrical and Electronic Engineers (IEEE) 802.3 (10baseT) Ethernet, which provides an industry standard interface to category 5 cabling.

- The Data Link layer (layer 2) is described as meeting IEEE 802.2, which describes logical link control (LLC) process.

- The Network layer (layer 3) is described as IP, as described in Request for Comment (RFC) 791.

- The Transport layer (layer 4) is described as UDP/TCP as defined by RFC 768 and RFC 793.

- The Upper layers are described as the Dispatch Radio Voice Real Time Protocol (DRVRTP), which is defined in an appendix to the bulletin. This protocol incorporates a subset of the RTP/RTCP protocol defined in IETF RFC 1889 and has been submitted to the IETF as a proposed RFC. The bulletin also defines control messages in another appendix; these messages would be passed between the fixed station and the RF subsystem.

This proposed specification appears to provide a standard for deploying the IP to the edges of the network with a standard interface incorporated into fixed stations. It provides for the use of standard routers and networking components throughout the network and eliminates the necessity for external interfaces at the stations.

2.5 Summary of VoIP Protocols

Table 2 summarizes the standards discussed in Section 2.4. It should be noted that the first four protocols discussed are designed for Internet communications and the last one is designed for a conventional base station hardware interface. While delay can be a problem over the Internet, individual links on the Internet are usually reliable and consistent in terms of quality and reliability (typical reliability of leased lines is 99.999 percent), and bandwidth is usually not a limiting factor. This is not true of LMR systems. LMR systems for public safety use are typically designed to have an over-the-air reliability of 95 percent for a given BER ranging from 2 percent to 5 percent, depending on the tolerances of the system. LMR systems also are limited to over-the-air frequency modulation bandwidths of 12.5 kilohertz (kHz) to 25 kHz, with requirements for 6.25 kHz on the horizon. Obviously, these bandwidths present a challenge to the implementation of end-to-end VoIP. It is uncertain whether any of the protocols developed for voice telephony over the Internet would prove viable for use in the LMR environment. For this reason, EF Johnson has proposed its fixed station interface standard, and M/A-Com Private Radio Systems' Open Sky® solution uses a proprietary version of cellular digital packet data (CDPD) to provide end-to-end IP. It is expected that if a P25 standard for VoIP does not emerge in the near future, more proprietary protocols will emerge, resulting in significant interoperability issues between systems from differing manufacturers.

Table 2
VoIP Standards

Item	H.323	SIP RFC 2543	MEGACO H.248	BIC Q.765 and Q.1901	FSTG/00/08/00
Developed by or Developed for	ITU-T conferencing industry	"IP over Everything"	Circuit switch engineers	ITU standards	EF Johnson Fixed Station Interface
Applications	ISDN local area network (LAN) conferencing	I-multimedia World Wide web (WWW)	Call Agent SIP and H.323	Broadband Integrated Services Digital Networks (BISDN), Advanced Intelligent Networks (AIN) H.xxx, SIP	Conventional LMR
Network/ Transport	IP	IP	IP	"any packet"	IP
Performance	Sluggish	OK	OK	Sluggish?	Unknown
Scalability	No	Yes	No	No	Unknown
Internet and WWW fit	No	Yes	No	No	No
Open Standard		Yes			Yes—proposed P25

Item	H.323	SIP RFC 2543	MEGACO H.248	BIC Q.765 and Q.1901	FSTG/00/08/00
Major Players	Cisco, Hughes Software System, Lucent, RADVision, Trillium	3Com, Bell Laboratories, NTT, RADVision	Hughes Software System, Nortel Networks, RADVision	CooperCom, Major Service Providers	EF Johnson

2.6 Vocoders

While discussion of VoIP often emphasizes protocols, voice coders (or vocoders) play a critical role in the digital transmission of speech. The vocoder converts analog speech into a digital bit stream for transmission over digital media. At the receiving end, the vocoder converts the digital bit stream back into an analog signal. It is important to carefully choose the characteristics of a vocoder for the environment. When dealing with wireless networks and packet transport, the choice of vocoder makes the difference between intelligible and unintelligible speech. Three factors are critical in the selection of a vocoder: the bandwidth or data rate, the complexity of the vocoder, and the voice quality required.[3]

The earliest and simplest vocoders were PCM waveform coders, which simply converted speech amplitude into a binary value representing the signal level so that the speech information could then be transmitted digitally. Typically, simple PCM coders require a data rate of 64 Kbps to sample 8 bits at an 8 kHz rate. Other PCM coders apply prediction algorithms to reduce the data rate; these vocoders are often referred to as linear predictive vocoders. These prediction algorithms require the use of codebooks, increasing the complexity of the vocoder. In conclusion, because PCM coders attempt to recreate the original speech, they work well but require higher data rates.

A newer breed of vocoders is based on speech modeling. These vocoders do not attempt to recreate the original speech; rather, they model speech. The modeling coders generate parameters based on the original speech. At the receiving end, the decoders apply the parameters to the speech model and generate an approximation of the original speech. Modeling vocoders are sufficiently advanced that although the reproduced speech sounds different from the original, individual voices are recognizable. The advantage of modeling vocoders over linear predictive vocoders is that because modeling vocoders use parameters to describe speech, they effectively compress the bandwidth required to transmit speech at a given level of voice quality.

Well-designed modeling vocoders can provide high-quality speech, at low bit rates, with sufficient robustness for wireless applications. Improved multiband excitation (IMBE) and advanced multiband excitation (AMBE) vocoders, both developed by Digital Voice Systems Inc., are modeling-type low-speed vocoders that provide high-quality speech in mobile radio environments. Based on mean opinion score testing, the Telecommunications and Information Administration (TIA) selected IMBE over several linear predictive vocoders. Therefore, the IMBE vocoder is the P25 standard for digital systems. AMBE is a later version modeling

[3] Digital Voice Systems, Inc. "Voice Coding Overview".

vocoder based on IMBE. The MA/COM Open Sky system uses the AMBE vocoder, and there are plans for an improved AMBE+ vocoder that will provide high-quality audio in a smaller bandwidth.

2.7 How Would VoIP Work With LMR?

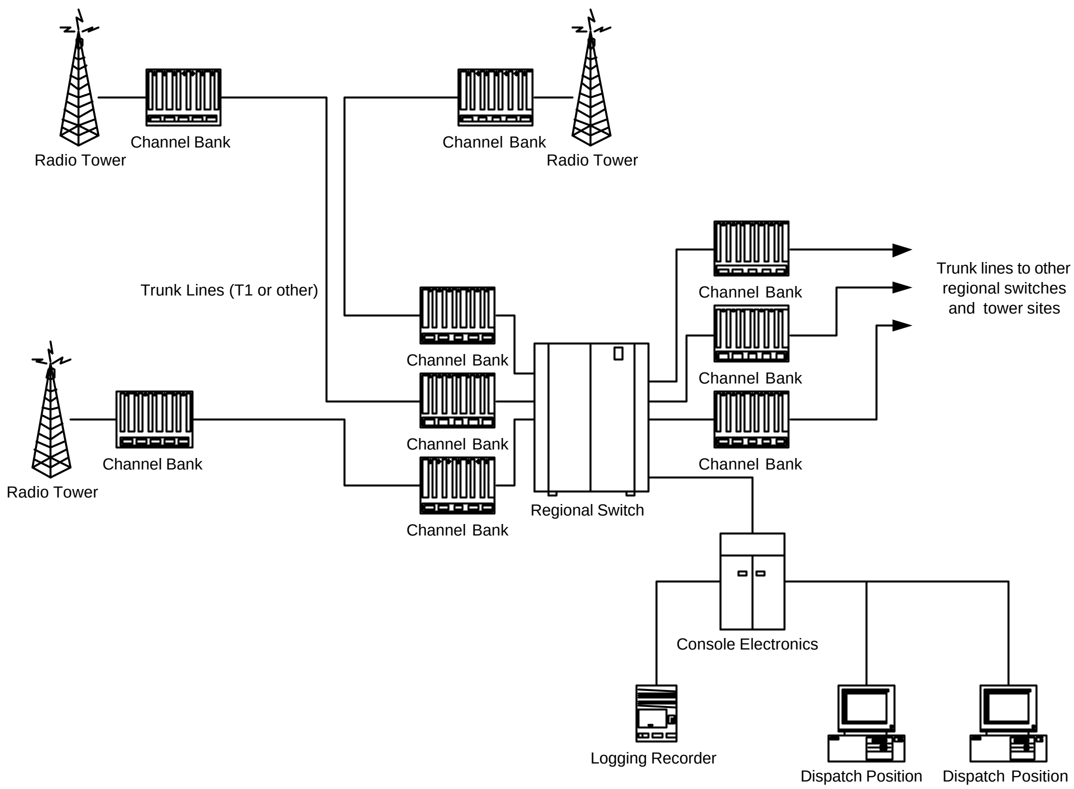
As outlined in Section 2.3, VoIP is packet based and therefore fundamentally different from circuit-based systems. The different uses of VoIP in telephony and LMR are also significant. In the telephony world, VoIP connections replace dial-up connections, which means that VoIP must not only replace the circuit-based functions of carrying voice information, it must also provide call setup and termination functions and other intelligent network functions that are handled by the common signaling channel in circuit-based networks. In the LMR world, VoIP packets carried over an IP network would replace dedicated lines that may have been connected via copper, microwave, or circuit-based digital carrier systems. In addition, LMR systems exist in an environment in which effects of electromagnetic wave propagation influence voice quality and grade of service. Finally, because LMR systems are typically used to support communications among a group of users as opposed to telephony, which typically supports communications between two users, VoIP must be able support distribution of calls to multiple addresses.

The term “voice over IP” must be considered carefully in the context of the LMR environment, because VoIP is thought of as a telephony technology. In the LMR environment, it is more accurate to think in terms of voice over packet networks, using TCP/IP protocols at the Network and Transport layers, with other protocols at the other layers. Physical and Data Link layer protocols are likely to be Ethernet or proprietary protocols. Session layer protocols are likely to be H.245 or a standard yet to emerge. It is likely that Presentation and Application layer protocols, if required, will be specific to the vendor application.

Several advantages offered by VoIP technology must be noted when considering it as a future platform for transport in the LMR environment:

- IP-based packet networks promise potential cost savings through reduction or elimination of leased line costs, and reduction or elimination of channel equipment and external interfaces. Channel banks are major cost drivers in wide area system architectures. As shown in Figure 3, channel banks are used to interconnect edge equipment, such as base stations, to regional audio switches over trunk lines, and to support connections between audio switches. In very wide area systems, the cost of channel equipment becomes a significant part of the cost because channel banks, and sometimes digital access cross connect (DACC) devices, are required at every interface between the leased or private circuits and radio equipment. At sites where central switching occurs, at least one partially loaded channel bank for each site to be connected is likely required, along with corresponding channel equipment at the remote end. With packet-switched networks, edge devices need only be connected to the network with routers providing a breakout at the site because switching is performed through packet addressing. Designing edge equipment to interface directly with the packet network eliminates not only channel equipment but also any modems and gateways that might be required to interface edge equipment with the transport network. Depending on the network configuration and size, the number of leased lines required can be reduced significantly.

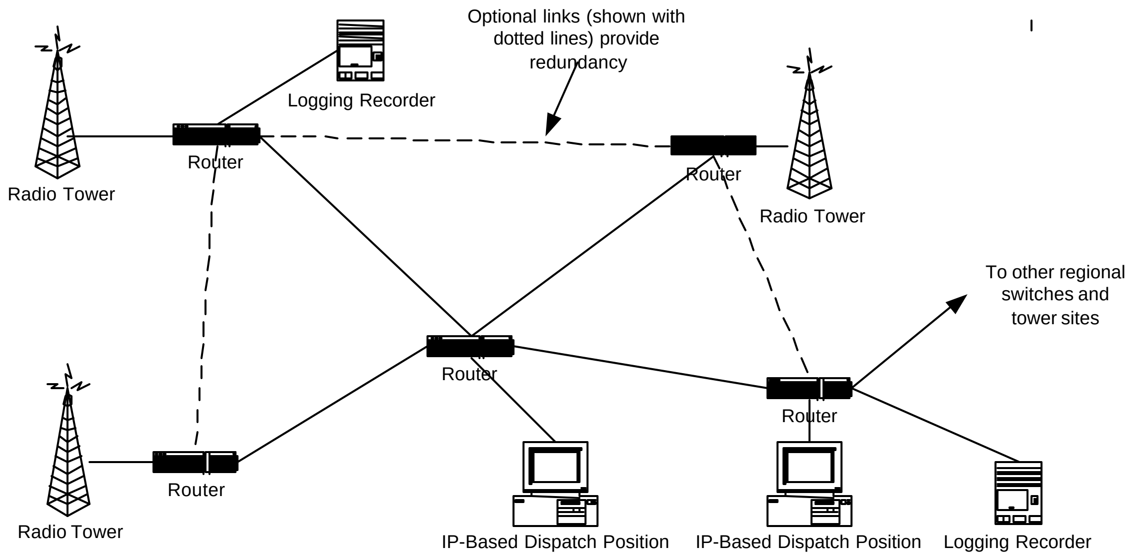
Figure 3
Simplified Circuit-Based LMR System



- Packet networks also offer simplification through standardization and commonality of equipment, augmented by the use of off-the-shelf networking equipment. Because IP is a standard protocol, IP routing equipment is available as a commercial off-the-shelf (COTS) commodity. Depending on the network architecture, channel-based networks may require consistent use of specific brands and types of equipment to maintain compatibility, especially when add-drop multiplexers (ADM) are required. Networks based on specific channel equipment put the owners at risk of equipment unavailability. On the other hand, IP-based packet routers are designed such that routers of differing manufacture can communicate with each other. In addition, while circuit-switching equipment, such as channel banks, remains expensive, packet network routers are becoming less expensive as they are more widely implemented. With packet-switched networks, each site is connected to the network through an appropriate data interface, eliminating the channel banks and replacing trunk lines with single high-speed data lines.

- Packet networks also offer redundancy through the routing capability incorporated in the IP packet structure. In packet networks, each packet carries addressing information, and proper network design provides for multiple paths between each end point. Thus, packets can be quickly and automatically routed around a bad segment with no noticeable impact on the quality of service. Figure 4 depicts the use of additional links to provide alternate routing.

Figure 4
Simplified Packet-Based LMR System With Additional Links



- Packet networks can also provide "virtual" networks over a common infrastructure, incorporating privacy between user groups. In this way, packet networks are similar to trunked radio systems in that they provide for the automatic, efficient, and private sharing of resources among a large number of users.

- Packet networks can provide switching functions normally supported by circuit-based switches. These functions can include wide area audio switching and central logging for wide area systems. Further enhancements include multimedia and data-bearer service support.

It is important to understand the difference between a true, end-to-end IP network and one that simply uses IP transport. In a true, end-to-end IP network, the system will be able to deliver end-to-end IP applications, the network will be packet switched, and most importantly not only network equipment but each piece of radio equipment will have a unique IP address. In most cases this will also mean that edge equipment, such as base stations, dispatch consoles, logging recorders, and data terminals, will interface to the transport network directly, at a digital level. In a system that uses IP for transport only, radios will not have IP addresses, and the system will not be able to support end-to-end IP services. [4]

[4] See footnote 2.

3. SURVEY OF MAJOR VENDOR OFFERINGS

3.1 Vendors

This section discusses vendors selling or developing VoIP technology for the LMR marketplace.

3.1.1 Motorola

Motorola has detailed plans for a VoIP system, referred to as “Astro 25 Integrated Voice & Data,” but has not scheduled this system for release. Motorola has been vague regarding the use of IP addressing; however, it is believed that the system will eventually use end-to-end IP addressing and will be based on the P25 CAI, control channel, and packet data service. Initially, the system will not be designed to be end to end, but rather, will require interim steps before IP can be implemented end to end. As an interim solution, Motorola plans to use its existing circuit-switching based consoles and add a gateway between the consoles and the IP packet network, as shown in Figure 5. In this architecture, IP addressing would end at the gateways, with channel banks carrying audio to the base station, the console equipment would be interfaced to the packet network through the Motorola Gold Elite Gateway (MGEG). It is anticipated that the MGEG would fill the role currently performed by digital interface units (DIU), which provide digital to analog audio conversion and encryption and decryption on a per-channel basis. In the future, Motorola plans to integrate vocoding and encryption functionality into its PC-based “Platinum Series” IP consoles to provide true, end-to-end transcoding (i.e., vocoding from analog to digital at the source end and back to analog at the reproduction end) and encryption, while eliminating the need for Ambassador Electronics Bank (AEB), central electronics bank (CEB), the MGEG, and channel banks (as shown in Figure 6). Implementation of IP-based consoles would move transcoding and encryption functionality into the console, eliminating the need for DIUs. Motorola has not scheduled the release of the integrated voice and data technology for conventional systems.

Motorola indicates that its approach, which uses COTS routers and network equipment, requires that the packet network be closed and dedicated to radio system transport. In addition, the approach uses multicast addressing for talk-group calls in multiple site systems. In this approach, voice, data, messaging, trunking control, and network management all travel over common “pipes.” Details of the IP implementation are unclear, but it appears to leverage emerging standards and tunneling (i.e., “wrapping” the higher-level protocols in IP headers) technologies to support linear simulcast and seamless communications across multiple frequency bands.

Motorola has recently indicated that it expects to have a P25-compliant, very high frequency (VHF) trunked system with 9,600 bps control channels ready for release in December 2001, and that while there is no accepted P25 standard for VoIP, the P25 9,600 bps systems will use VoIP transport.

A new device, called the Packet Data Gateway, will incorporate wireless network gateway (WNG) and packet data router (PDR) functionality. The primary function of this device

Figure 5
Motorola's Circuit-Based Migration Architecture

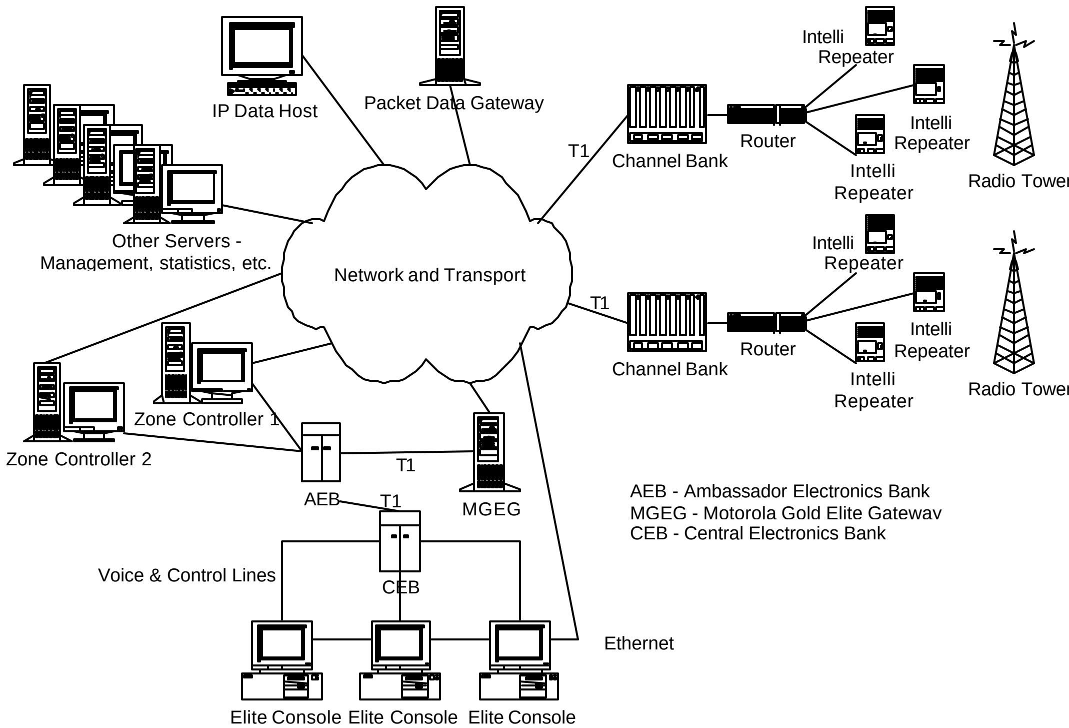
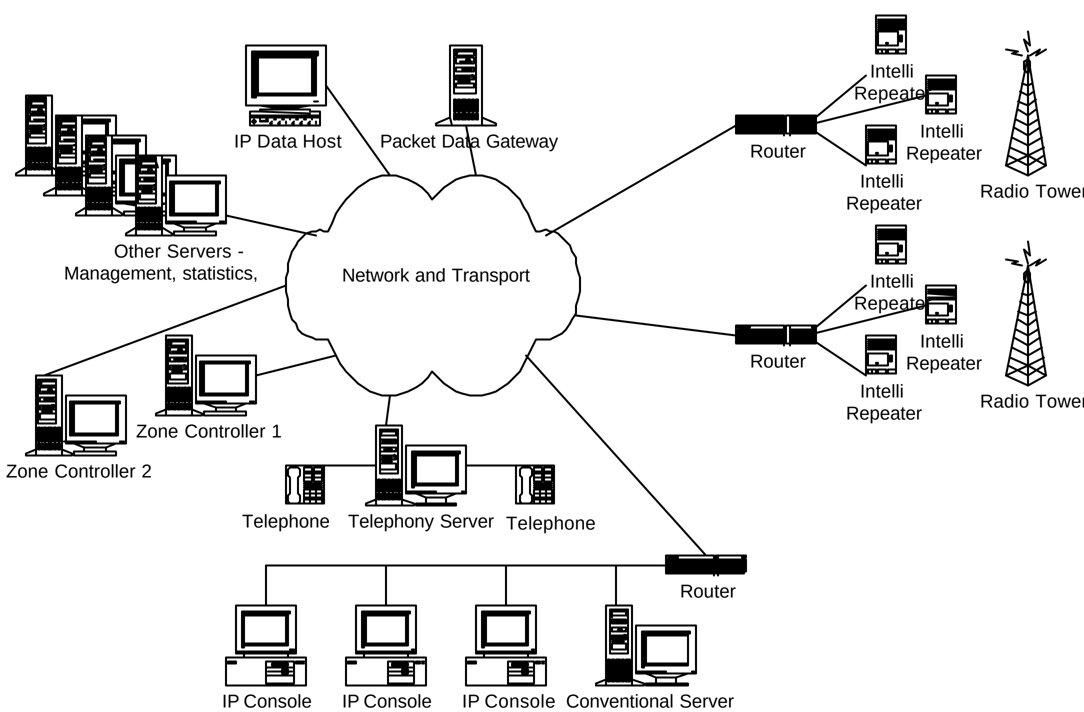


Figure 6
Motorola's IP-Based Packet-Switching Architecture



appears to be the handling of packet data and over-the-air rekeying (OTAR). Motorola has not provided any insight on its approach for supporting OTAR in the new architecture. It is anticipated that integration of the data functionality (including OTAR) with the voice network will present a challenge.

3.1.2 EF Johnson

EF Johnson was purchased by Transcrypt International, and is now a subsidiary division of Transcrypt International.

The EF Johnson approach for implementing VoIP is as follows:

- The IP domain extends to the repeater.
- Each repeater site is its own sub-network, connected via hubs or routers.
- Connections between sites use digital T1 circuits or fractional T1s.
- RTP transport is used for P25 CAI voice frames.
- IP multicast is used to provide multicast network calls.

EF Johnson is scheduling to introduce a VoIP solution supporting radio systems on the following schedule:

- Quarter 4, 2001—IP networking, conventional P25 radio systems, console access, logging capabilities, and network administration

- Quarter 2, 2002—Conventional voting implementation

- Quarter 3, 2002—P25 trunking and trunking voting.

On September 20, 2000, Transcrypt International, Inc., announced that its EF Johnson Company subsidiary had submitted a proposal to the TIA to adopt its VoIP radio networking technology as an open industry standard. The technology supports digital switching of voice and data using an IP-based infrastructure for wide area radio networks. The standards approval process is estimated to take between 6 months and 2 years to complete. At this date, it is known that at least one vendor, Motorola, has provided minor comments on the proposed standard.

On January 9, 2001, Transcrypt International, Inc. announced that its EF Johnson Company subsidiary had selected Herbst LaZar Bell (HLB) to supply the mechanical design for its new high-tier portable radio product currently under development. This radio product will offer a feature-rich solution to the local, state, and federal public safety market for EF Johnson and has an expected release date of the fourth quarter of 2001. Specifically, this product offering will support P25 conventional, P25 trunking, SMARTNET™ II / SmartZone®, and Multi-Net® protocols, and will fulfill a broad range of market requirements.

The Department of the Treasury's Integrated Treasury Network (ITN) Project Office is planning to implement a pilot project using several of EF Johnson's VoIP stations.

3.1.3 Tyco Electronics

Tyco Electronics, a division of Tyco International Inc., now has two holdings in the LMR marketplace. Their acquisition of ComNet Ericsson in early 2001 makes them the second largest manufacturer of LMR equipment, after Motorola. While ComNet has its own line of LMR products and a unique identity based on its evolution from the GE Radio Division (which once competed closely with Motorola), it appears that ComNet will be absorbed by Tyco's MA/COM Open Sky company. This move seems to be aimed at competing head-to-head with Motorola because the product lines of ComNet and MA/COM are highly complementary with the notable exception that neither ComNet nor MA/COM have shown any interest in developing a P25-compliant product line.

3.1.3.1 ComNet (formerly ComNet/Ericsson, formerly GE/Ericsson, formerly GE). ComNet has a fully developed product line of analog radio equipment for public safety, commercial, and other markets. Their product line consists chiefly of conventional analog frequency modulation (FM) equipment in VHF high band through 800 megahertz (MHz), Association of Public-Safety Officials (APCO) 16-compliant trunking systems, and commercial grade trunking systems.

ComNet, in one form or another, has been Motorola's main competitor over the years. Following the acquisition of the company by Tyco Electronics, the future of the ComNet product line is uncertain because Tyco has made it clear that ComNet will be absorbed into MA/COM.

ComNet has not, to date, shown any interest in VoIP products and does not have a P25 product line. Therefore, it is not discussed further in this analysis.

3.1.3.2 MA/COM. The Open Sky product is a TDM product that relies on time division multiple access (TDMA), uses standard 25 kHz voice channels to support 19.2 Kbps transmission rate, and supports two time slots that can be used to carry data or voice in any combination. Today Open Sky uses an AMBE vocoder operating at 3,600 bps. However, there are plans to implement the AMBE+ vocoder at a rate of 2,400 bps, which will allow the system to provide four time slots for voice and data. Open Sky claims that the four-slot system will provide audio quality comparable to the two-slot system, and that existing systems will be software upgradeable. Open Sky is only available for the 800 MHz band, and MA/COM does not currently offer a P25-compliant system; however, it claims it will offer P25 CAI compliance for talkaround by next year. While MA/Com plans to support P25 interoperability modes, it has no plans for a complete P25 product line. In addition, the acquisition of ComNet by Tyco Electronics and subsequent absorption of ComNet into M/A-COM provides M/A-COM with a wealth of technology resources with which to develop radios for the lower bands.

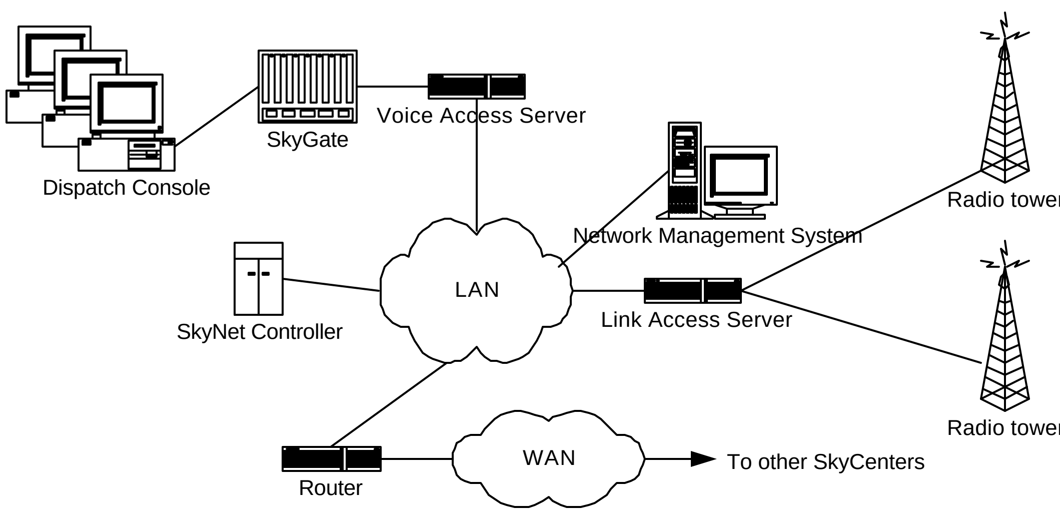
Because Open Sky is TDM, the system cannot operate in a simulcast configuration. Rather, multisite Open Sky system designs are based on the cellular concept, and frequencies are reused at geographically separated intervals. The Open Sky suite also includes a pole-mounted repeater station for use in remote areas where fill-in coverage is required and traffic densities do not require a multistation site.

Open Sky does not currently support encryption; however, the company claims it will be able to provide RSA encryption, followed by other types as software implementations.

While Open Sky manufactures its own mobile radios, portable radios, base stations, and some of its own network equipment, it does not manufacture dispatch consoles. Open Sky's approach to dispatch consoles is that it can provide a 600-ohm baseband connection to any console on a per channel basis using a dispatch gateway to interface between the transport network and the console. Of course, it could also interface the network directly to an IP-based console, but no IP consoles currently exist.

The Open Sky system also provides end-to-end IP service, i.e., each radio and network interface in the system bears a unique IP address. Packets are delivered to specific IP-addressed interfaces, which can be routers, console interfaces, or individual radio units. Further differentiating Open Sky from the other offerings is its over-the-air interface for data calls, which uses a proprietary protocol based on CDPD . It should be noted that CDPD uses TCP/IP in the Network and Transport layers. Open Sky also uses embedded trunking control; therefore, no separate control channel is required. Open Sky indicates its transport networks use COTS routers and hubs. Based on early product descriptions, base stations are tied into the main site using serial line IP (SLIP) connections and modems. As shown in Figure 7, the main site network appears to be a "collapsed" packet network used for audio switching. M/A-Com indicates its solution is capable of end-to-end IP addressing.

Figure 7
Open Sky System Architecture



Open Sky confirmed in a meeting that it is using a packet network end to end in the State of Pennsylvania. This is significant for planners of multi-agency systems, because it permits the use of an integrated network to provide virtually private service to multiple agencies. The impact of this capability is that agencies can perform dispatching of their own units regardless of dispatch location or field unit location within the network. Likewise, agencies can independently manage and perform logging on their units from anywhere in the network while restricting management, logging, and dispatching of their units by other agencies on the system.

Open Sky listed five clients thus far:

- **Federal Express.** Federal Express is Open Sky's original customer. The Federal Express Open Sky system constitutes the largest private wireless data system in the world with 20,000 Open Sky units in voice and data service nationwide. The Federal Express system is projected to grow to 40,000 units nationwide.

- **Orange County Transit Authority (OCTA), California.** The OCTA system is being installed, and initial installation is for approximately 500 buses.

- **State of Pennsylvania.** The State of Pennsylvania system will initially consist of 250 tower sites, followed by an additional 250–300 pole-mounted repeaters. Initially, the system will support as many as 20,000 users, with a guaranteed capacity of at least 150,000 users. The system has seven regional operating centers linked together over the statewide packet network.

- **Cumberland County, Pennsylvania.** Design and implementation is in process. The new system will be capable of integrating with the statewide system.

- **Lancaster County, Pennsylvania.** Design and implementation is in process. The new system will be capable of integrating with the statewide system.

3.1.4 Catalyst Communications Technologies

Based in Lynchburg, Virginia, Catalyst Communications Inc. provides VoIP products to the mobile radio industry. The company's focus is enhancing existing dispatch communications systems using its newly introduced VoIP product, IP Fleet, in conjunction with its IP Radio™ and Network Access Radio (NAR) products. IP Fleet, introduced in June 2001, provides dispatch capability for utilities, industrial users, and government agencies such as departments of transportation. Catalyst's interoperable IP Fleet concentrates on products leveraging Intel-based processors for increasing efficiency of network operators and wireless users.

IP Fleet's features include allowing PC users to change the LTR® group/system or conventional channel/set of channels on each mobile radio, thereby enabling a single radio gateway to support multiple groups and/or multiple systems, one at a time. IP Fleet also allows users to view the full name of the calling radio, which can be useful when placing or addressing a selective call. In addition, the IP Fleet display turns red to indicate that FleetSync™ radio has declared an emergency. More highlights of IP Fleet include easy maintenance of the directory of individual radios through centralized access of user database.

IP Radio allows the PC user to change the EDACS® (Enhanced Digital Access Communications System) group or conventional channel on each mobile radio. IP Radio allows customers to choose from three different vocoders, significantly reducing bandwidth consumption. It also allows system administrators to make changes to the user database, system parameters, and available groups, channels, and systems without interrupting service to end users.

Another product fielded by Catalyst is the NAR, which can interface to a variety of mobile radios, supporting both trunking and conventional operations. When different radios are paired with different servers, remote PC users can access multiple systems from a single PC. Other advanced capabilities include scanning between geographic locations, radio systems, and channels; queue management; and a user-friendly interface.

Catalyst most recently provided its IP Radio system to the City of Honolulu Police Department as an enhancement to the department's existing EDACS trunked radio system. IP Radio serves as a backup to regular police dispatch consoles, providing redundancy in the event of natural disasters and terrorist acts.

3.1.5 SIMOCO

Headquartered in Cambridge, England, SIMOCO is a major supplier of radio systems in the international marketplace, with offices in Australia, the UK, France, and Hong Kong. SIMOCO's flagship system is compliant with the European Telecommunications Standards Institute (ETSI) TETRA standard for trunked private mobile radio. The European TETRA standard uses TDM to provide full-duplex operation over wide channels, while the APCO P25 standard uses frequency division multiplexing (FDM) to provide half-duplex operation over narrow channels. Although TETRA systems are not marketed here, SIMOCO is mentioned because it was the first company to deliver a TETRA system operating over an IP network. According to company officials, SIMOCO has installed more than 40 TETRA VoIP and mobile data systems, adding credibility to the concept of using an IP network to transport voice and data signals in LMR systems.

In April of this year, SIMOCO held a product launch in Korea to announce its TETRA over IP products, and provided a live demonstration of its TETRA over IP system. In May of this year, SIMOCO held a product launch in Italy to announce its TETRA over IP products, and provided live demonstrations of voice, data, and two-way video.

Earlier this year, SIMOCO was contracted to upgrade an analog system to TETRA over IP for the urban community of Dunkerque, France. According to SIMOCO, the system supports 135 mobile units and 45 portable units, and is based on IP. SIMOCO claims that the system uses PC-based routers. The SIMOCO protocol encapsulates digitized voice in TETRA format and data in IP. SIMOCO also received an award earlier this year to provide a TETRA system for the Isle of Man in the UK. SIMOCO states that this system will use IP-based transport and a total of 21 radio sites.

3.2 Key Capabilities and Limitations

This section describes and assesses capabilities that are key to determining the viability of VoIP solutions for the public safety community. The section also discusses system capabilities affected by the switching architecture and describes how circuit and packet switching impacts these capabilities. The section concludes with a table summarizing the capabilities of each vendor currently indicating its ability to deliver a VoIP system in the U.S. marketplace.

3.2.1 Stage of Development

The stage of development of a system is of particular interest in determining the viability of any systems solution. The stages can be classified as planning, prototyping, demonstrated, beta tested, and operational. Obviously, the more mature the system solution is, the more viable that solution is.

3.2.2 System Architecture

Several variations on the VoIP system architecture should be considered. The most obvious architectural variation is that the system uses packet networking end to end. That is, the packet network extends over the air by means of tunneling IP through the lower layer protocols. In this case, each radio unit (and every other interface point on the packet network) has a unique IP address. Another variation is one in which only the transport network itself supports packet networking. In other words, only the transport network itself is packet based, and RF devices are interfaced at a digital or analog audio level through an IP gateway device. A third variation is a combination of the first two architectures. In this third case, the packet network is used for switching at a central site, and remote sites are connected via SLIP or some other serial protocol. While IP does extend end to end in this network, the packet network itself does not; therefore, tunneling is required. In this architecture, radios and routers have unique IP addresses, while other network elements may not. Only the first and third architectures support IP-based applications.

3.2.3 Voice and Data Integration

Voice and data integration refers to the capability of a system to transport voice and data over the same facilities, on an end-to-end basis. To put this in perspective, older analog systems were designed for voice transmission and later adapted to provide mobile data service. This adaptation required segregating the voice and data paths over the transport network, with the voice and data paths only converging again at the base station. Modern systems using IP over packet-based networks should be able to provide complete end-to-end integration of voice and data over the transport network and over the air. With a fully integrated voice and data system, voice and data signals would only diverge at the edge devices, where they would be reproduced by a speaker in the case of voice, or at the Application layer for data. When voice and data are fully integrated over the network, protocols are used to provide real-time delivery of voice packets, while data remains less time sensitive, as in conventional packet data networks like the Internet.

3.2.4 Integrated but Segregated

A much-anticipated advantage offered by a VoIP system is the capability to build a single system serving multiple agencies, with a level of separation previously found only in separate systems. Such a system would provide the segregation required by the varying missions of public safety agencies, while allowing for the purchase, maintenance, and operation of a single backbone.

Depending on the system architecture, varying levels of segregation can be provided. Trunked systems provide a basic level of segregation through group and individual calling. Through trunked signaling, groups of users (or pairs of users in the case of individual calls) can be assigned exclusive use of a channel for the duration of a call or conversation. These assignments preclude other users operating on the system from hearing or participating in conversations to which they are not assigned. While this functionality provides a basic level of segregation and privacy that has served multi-agency systems well for a number of years, systems employing IP transport promise additional levels of segregation. The following paragraphs describe these levels of segregation.

3.2.4.1 Dispatch Segregation. Systems using circuit-switched transport are capable of providing a degree of segregated dispatching through selection of circuits and console programming. In analog circuit-switched systems, dispatch consoles must be connected to each base station at the audio level. This means that if the console is not collocated with the base stations, circuits must be provided between the base station location and the console location. Large, centralized dispatch centers are often acceptable when multiple agencies are involved in implementing a system covering a metropolitan city or county area. These large dispatch centers can save significant costs in terms of console central electronics, circuit transport equipment (channel banks), and transport media such as leased lines and microwave. However, these large centers have two distinct disadvantages. The first disadvantage is that in geographically large systems, dispatchers may not be physically close to the users. This may not be a great problem, but it is less than optimum in large regional and nationwide systems when dispatchers have little or no local knowledge of the area for which they are responsible. The second disadvantage is that when one or more agencies use encryption, DIUs external to the console provide the encryption transcoding. The impact of this is that clear-text audio is present between the DIU and the console electronics. In addition, in consoles such as the Motorola Centracom Gold series, by using certain standard configuration options, it is possible for any operator to monitor any channel or talk group present in the console central electronics. In multi-agency systems, the potential exists for uncleared personnel to easily monitor secure communications.

Packet-based transport systems using end-to-end IP addressing facilitate the geographic separation and functional segregation of the dispatching function by making all traffic available anywhere in the system with no relation to individual circuits. This provides user agencies with the freedom to locate dispatch and logging facilities anywhere in the system where a connection to the packet network is available. As outlined in an earlier section, one characteristic of a true, end-to-end IP system is that each interface in the system has a unique IP address. If an interface has an IP address, then packets can be addressed to that interface regardless of its location in the system. The impact of this is that through IP addressing, a dispatcher located anywhere within the system can communicate with a mobile radio, portable radio, or another dispatcher regardless of location, as long as the message is properly addressed. Therefore, dispatching can be geographically distributed to suit operational needs. In addition, because IP extends end to end, encryption would also be end to end, making it difficult for unauthorized personnel to monitor secure communications without being present at the position where the secure audio was decrypted. This principle can be extended to the idea of logging. With end-to-end packet network transport, if encryption is used, each agency can separately log its own traffic, independent of geographic location, without monitoring by other agencies.

3.2.4.2 System Management. The use of packet networking also provides for easier segregation of system management functions. Similar to dispatching and logging, system management functionality becomes location independent when a packet network and IP addressing are used. Agencies can have their own geographically independent system management capabilities. Further, IP addressing allows restriction of the dispatching, logging, and management functions such that agencies can only communicate with, log, and manage units in specific IP address ranges.

3.2.5 Wide Area Capability and Architecture

The wide-area capability of a system refers to the ability of that system to provide coverage beyond that afforded by a single repeater site. Usually, the fact that the system uses more than one transmit site to provide coverage is transparent to the user. In this discussion, "transparent to the user" means that the user is not aware of the coverage limitations of specific sites, nor does the user need to make changes to the radio's channel selector to maintain communications while moving throughout the coverage area. The following paragraphs discuss several techniques that can provide extended coverage while maintaining transparency to the user.

3.2.5.1 Satellite Receiver Systems. Satellite receiver systems were originally developed to help overcome the talk-back limitations experienced by users, especially users of portable transceivers (portables). Because mobiles and portables have less power than fixed repeater stations, they are usually able to "hear" the repeater in areas where they cannot "talk." This characteristic is known as being talk-back limited. Before the development of simulcasting technology, talk-back coverage was enhanced by placing additional receivers in areas where talk-back was weak. All of the received audio was then transported to the repeater site, where the best signal was automatically selected and retransmitted. Although this technique works well to overcome the talk-back limitation of mobiles and portables, it does not extend the talk-out range of the repeater.

3.2.5.2 Simulcasting. Simulcasting, originally known as quasi-synchronous transmission, was developed to extend the talk-out range of a system beyond that of a single repeater site. In simulcasting of FM signals, the same information is transmitted from multiple sites over the same RF carrier frequency at the same time. This allows the use of a single frequency to provide coverage over an extended area, such as a city or county. For simulcasting to work with a minimum of distortion, three critical characteristics must be carefully controlled: the carrier frequencies of the transmitters must be the same and must be held very stable; the information must be launched from each transmitter at exactly the same time; and the deviation of each transmitter must be very limited. Because these parameters must be coordinated very closely, simulcast systems use highly specialized processing equipment. Simulcast systems necessarily use satellite receivers to provide talk-back that is equal to the extended talk-out coverage. Typically, all received audio in a simulcast system is transported to a master site where the best audio is voted, and the voted audio is then distributed over a specially designed transport network to each of the transmitter sites. Control of launch time and deviation is achieved through the transport network, which is continuously optimized using data from Global Positioning System (GPS) receivers at each transmit site. Atomic standards are used at each transmit site to maintain the required frequency accuracy and stability.

To minimize distortion, there are limitations on the maximum distance between simulcast sites. These distances vary, depending on the type of modulation used. It is generally accepted that for analog 25 kHz FM modulation, simulcast sites can be no more than 21 miles apart, whereas certain types of digital modulation limit the distance to 11 miles. Some systems, such as TDM systems, cannot use simulcasting as a wide area coverage technique because of the critical timing required to coordinate multiple time slots. However, Motorola's FDM approach lends itself well to simulcasting; Motorola was the originator of simulcasting technology.

3.2.5.3 Cellular. The cellular approach, which provides coverage in cellular radio systems, can also be applied to LMR systems. This approach does not use simulcasting or satellite receivers to complete calls. Instead, cellular systems rely on the concept of using low-power transmit and receive sites networked together through audio switches. Radios communicating over these systems also transmit and receive data, which is usually sub-audible. This data provides the system with information required for routing the call to one or more users. These systems rely heavily on circuit switches and processors for call routing. While cellular systems usually require at least two frequencies to complete a call, spectral efficiency is achieved because cellular systems use low-level sites, with controlled coverage. This controlled coverage facilitates the reuse of frequencies throughout the coverage area. M/A-Com's Open Sky system is an example of the cellular architecture applicable to LMR. Because Open Sky is a TDM system, it cannot use simulcast; however, because of the spectral efficiency of TDM and the ability of the system to carry embedded data, the cellular architecture works very well with the Open Sky system.

3.2.6 Trunking Capability

Trunking refers to the ability of the system to automatically share a small number of resources (channels) among a large number of users (talk groups). With trunking, user groups do not know or care what frequency they are using to complete their conversation. Similar to cellular systems, trunked systems use signaling and centralized management to administer resources and assign them to users who need them. In this way, users who need a channel are more likely to get one, and utilization of channels remains high during busy periods. This contrasts with conventional operation, in which user groups are tied to specific channels. If a conventional user's channel is busy, that user must wait until it is not busy; if a channel assigned to another user group is not busy, then it simply goes unused. Trunking technology was originally applied to the telephone system; therefore, it is no surprise that systems using the cellular architecture incorporate trunking to manage channels among users.

3.2.7 Encryption and Over-the-Air Rekeying

Before the widespread use of encrypted communications systems in public safety systems, law enforcement LMR devices employed an RF waveform consisting of a VHF or ultra high frequency (UHF) carrier that was frequency modulated by an analog signal representation of the speaker's voice. Such a transmission was highly susceptible to interception by casual eavesdroppers. To intercept and understand the communication, an eavesdropper only needed to use an inexpensive commercial scanning radio capable of demodulating FM signals at the carrier frequency of interest. When such eavesdropping became problematical for law enforcement groups, they began to use encryption to cover their communications.

A commonly used scheme of securing communications consists of digitizing the analog voice information and encrypting the digitized data using a particular algorithm known as the Data Encryption Standard (DES). Digital encryption can generally be implemented more efficiently and with less expense than analog encryption; therefore, it is usually a more desirable encryption technique. To digitize the analog voice, an analog to digital vocoder converts the analog signal representation of the speaker's voice into a digital representation. Early techniques employed a continuously variable slope delta (CVSD) modulation vocoder that digitized the analog voice information into a 12 Kbps stream of binary digits for subsequent encryption using a DES cipher feedback encryption scheme. DES can also be implemented with output feedback. The system works by the insertion of dedicated cryptographic keys.

Cryptographic keys must be managed in a secure manner to ensure their integrity. Key management is the process by which encryption keys are generated, stored, protected, transferred, loaded, used, and destroyed. It enables effective planning, implementation, and execution of security doctrine for a diverse set of user requirements. The actual system consists of a dedicated processor, or host computer, that stores the cryptographic keys. A signaling system is integrated with the key management and the RF infrastructures. Radio subscribers contain signaling modules that can request re-keys over the air through the RF infrastructure, i.e., base stations and repeaters. These requests are directed through the infrastructure encoder-decoder unit, i.e., CIU or DIU, and on to the host. This technique is called over-the-air re-keying (OTAR). In response to the re-keying request, the host selects a key, encrypts it, and causes the base station and CIU to transmit that key to the requesting subscriber unit. A series of acknowledgements or negative acknowledgements are sent back and forth. While the process might seem electronically laborious, it is very effective, permitting the re-keying of hundreds of units simultaneously in about 10 seconds.

OTAR has been around for about 10 years and is now undergoing a migration from proprietary analog technology to standards-based digital technology. It is unclear at this time how OTAR will integrate into systems using IP transport. No U.S. manufacturer has indicated how it plans to integrate OTAR capability into a VoIP system.

3.2.8 Standards Compliance

As standards help to ensure compatibility when systems are assembled using components from many different manufacturers, compliance with applicable standards is often an issue when planning a major system implementation. At this time, a suite of standards exists for narrowband digital radio systems (TIA/EIA 102). This suite of standards covers conventional and trunked digital narrowband radio systems including the CAI, but it does not address the use of VoIP as a transport for wide area systems. As outlined in Section 2.4.5, EF Johnson's proposal, entitled "FSTG/00/08/00 Project 25 Fixed Station Interface Overview and Definition—Conventional Systems," addresses a digital interface to fixed stations that would facilitate direct connection of stations to a packet network. However, while the reception of this proposal has been favorable, it has not been approved as a standard, and it does not address trunked systems. Although MA/COM has the most advanced VoIP system in terms of actual systems implemented, their approach is deliberately non-standards based because the company feels that standards such as TIA/EIA 102 impose an intolerable level of constraint on technological advancement. At this time, there is no standard for VoIP transport for LMR systems.

3.2.9 Limitations

Limitations of specific systems could include the number of users or talk groups possible in a system, number of sites in a cell, distance between sites in a cell, and number of cells. Because the information on system products is very limited, very little is known about the limitations of these systems. Motorola, for example, has not identified any system parameters and, according to Open Sky, the system limits are not yet known despite the fact that the Federal Express Open Sky system is expected to reach 40,000 units nationwide.

3.2.10 Summary Comparison of System Capabilities

The capabilities of the four systems most likely to be available in the near future are summarized in Table 3. From the available information, only one manufacturer has delivered a system in the United States using VoIP technology—MA/COM. Architectural approaches among the vendors vary. All vendors favor a dedicated transport network and agree on the need for IP addressing to extend to the edge devices (i.e., base stations, mobiles, portables, consoles all should have unique IP addresses). However, Motorola and EF Johnson appear to be taking a narrowband approach (12.5 kHz channels), while Open Sky uses wide (25 kHz) channels but provides two (four in the future) time slots per wideband channel. All vendors promise to provide voice and data integration, as well as integrated but segregated operation as described earlier. In addition, no vendor has announced an IP-based console. Motorola plans to be able to offer simulcast capability for wide area operation, while Open Sky uses a cellular approach exclusively. EF Johnson provides no information about wide area capability.

All three vendors plan to offer trunking capability, although trunking will not be available in the initial offering from EF Johnson. Motorola's plans for encryption and OTAR are not clear; however, they tend to try to conform to APCO 25, which calls for several encryption types and OTAR. It is not clear whether EF Johnson will offer encryption, and Open Sky says it will offer RSA encryption next year. However, Open Sky has not decided how to handle keying at this time. All vendors appear to be using IP multicast. Motorola plans to be compliant with TIA/EIA 102. EF Johnson plans initial compliance with TIA/EIA 102 for conventional operation and has proposed a base station interface for inclusion in the TIA/EIA suite of standards. Open Sky plans to provide a TIA/EIA 102 compliant conventional capability to provide talkaround and basic interoperability with P25 systems; however, they have no plans to develop a P25-compliant system. Limitations of each vendor's system are unknown—Motorola and EF Johnson have yet to provide details on system capacity. Open Sky claims it does not yet know the limits of its system.

Table 3
Summary of System Capabilities

System Features	Vendors		
	Motorola IV&D	**EF Johnson**	**Open Sky**
Stage of development/ production	Announced; availability planned for December 2001	Lab prototype demonstrated; initial operable system 4th quarter 2001	Several large systems implemented
Architecture	Initially IP to gateways; migration to end-to-end IP over dedicated transport	IP to edge devices; digital interface to base stations	End-to-end IP; analog console interface with gateways over dedicated transport
Spectral efficiency	Narrowband 12.5 kHz (FDMA)	Narrow band	Two time slots in a single wideband 25 kHz channel currently; in the future, four time slots with AMBE+ vocoder (TDMA)
Voice and data integration	Yes	Yes	Yes
Interoperability/ segregation	Yes, but IP console not available initially	Yes, but console approach not defined	Yes, using analog console with gateway
Wide area	Simulcast to be available	No wide area approach identified	Cellular with frequency reuse
Trunking	Only trunking will be available; conventional will not	Initial offering P25 conventional with trunking to follow	Yes
Encryption and OTAR	OTAR approach not identified	Not identified	RSA encryption to be available in 2002; OTAR approach not identified
Protocol implementation	IP multicast	IP multicast	IP multicast
Standards compliance	TIA/EIA 102	P25 compliance planned; proposed P25 standard for conventional base station interface	Conventional P25 capability for talkaround will be available in 2002
Limitations	Unknown	Unknown	Unknown

3.3 Issues

Based on this analysis, several issues are yet to be resolved before packet switching can be considered mature enough for consideration as an LMR voice transport architecture. The following sections summarize these issues.

3.3.1 Latency

LMR systems, like telephony, operate in real time. However, LMR systems often provide the primary means of supporting critical communications for law enforcement and other public safety personnel in the field. Critical LMR users demand that systems provide immediate access and imperceptible transmission delays. Standards for trunked LMR systems require system access times (from time of push-to-talk [PTT] to channel grant) of 500 ms or less. To meet these standards, a packet-switched transport would have to provide end-to-end latency of 200 ms or less consistently.

3.3.2 Class of Service

Some public safety communications can tolerate short, temporary outages, and some cannot. For many years, advanced trunking systems have provided multiple priority levels for public safety users so that critical transmissions will have a much greater chance of getting through during busy times. Advanced trunking systems can even provide for ruthless preemption of the lowest priority users to provide a channel for emergency users when the system is operating at capacity. Packet networks are similar to trunked systems in that they automatically share a fixed resource among a large number of users; therefore, some mechanism must ensure preservation of critical communications during periods of network congestion. To consider VoIP networks feasible as a transport for public safety communications, they must be able to provide multiple levels of priority to protect critical communications.

3.3.3 Scalability

The existing H.323 standard was developed for LANs, but to date no standard has been available to describe the implementation of VoIP over wide area networks. Because large, regional or national LMR networks are commonly composed of smaller networks linked by trunks for wide area calls, standards for packet data transport of LMR voice transport must be designed to provide the necessary scalability.

3.3.4 Reliability

As noted earlier, leased lines used to support public safety communications typically provide reliabilities in excess of 99.999 percent. The Internet is generally regarded as only providing 99 percent reliability when examined on a macro scale. Data networks and devices are not considered as reliable as telecommunications switches because data protocols typically include error detection and correction techniques. Further, data is not typically a real-time application like voice; therefore it is usually acceptable to design data-only systems for a higher rate of error and repetition. Packet networks are particularly good at employing embedded error detection and correction because information is broken up into smaller packets rather than a continuous stream. Error correction, however, often delays delivery of the complete message and contributes to network congestion during times of poor transmission quality. Voice traffic demands immediate delivery, whether the quality is good or marginal. Packet protocols for voice need to be designed to "cut losses and move on," rather than tying up the network while retrieving lost packets.

3.3.5 Security

IP networks do not offer the level of security provided by telephone company owned networks. The layered protocol model provides for security services at the Network layer that can be used to prevent routing of packets to areas other than those specified as destinations for the data. IP, however, does not provide this service. Encryption is usually employed in IP networks to provide end-to-end security, but in LMR networks employing end-to-end encryption, this may not be an issue.

3.3.6 Standards

As evidenced by the review of standards, VoIP, as it currently exists in the LMR environment, is a mix of standard protocols and proprietary technology. No overall standards govern VoIP when used as a transport for LMR systems, nor is there a link to existing standards such as TIA/EIA 102. In addition, it is unlikely that standards will emerge in the near future. It is more likely that a de facto standard will emerge after implementation and acceptance of a number of systems by the public safety community. Although P25 (TIA/EIA 102) provides a comprehensive suite of standards for digital narrowband communications, including voice and data, it does not address transport networks or interfacing to the edge devices (base stations and consoles). The problem with P25 is that while a great many proposals have long been accepted as standards, the entire P25 concept is at least 10 years old and therefore predates the idea of using VoIP as an LMR transport. The EF Johnson Fixed Station Overview and Definition is a very good attempt at defining a digital interface to the base stations; however, it does not address dispatching needs, and it has not been accepted as a standard. P25 Phase II, which is a project to develop a secondary standard for TDMA, is still in the very early stages of development. P25 Phase III (Project 25/34) addresses the subject of wireless transport of rate-intensive information but does not address the subject of IP transport.

3.3.7 IP Dispatch Consoles

Today, no IP-based consoles are available in the U.S. marketplace. Motorola has addressed the issue of the IP console requirement in its migration plan to integrated voice and data, but no product rollout dates have been announced. Further, no other domestic manufacturers have announced plans to produce one. While Open Sky claims to deliver an end-to-end IP system, it uses third-party circuit-based consoles and interfaces them to its system at the analog level using its SkyGate. The SkyGate provides the required AMBE vocoding at the console. Although this lack of end-to-end addressing is not a major issue for many systems where users simply need to be able to talk, it may be an issue for multi-agency systems using encryption, as described in Section 3.2.4.

3.3.8 Cost

There are strong indications that VoIP could reduce costs, particularly equipment costs in large systems, because it would eliminate channel equipment and central switching equipment. VoIP also promises savings in recurring costs because it eliminates leased lines. Equipment cost savings will depend on the level of COTS equipment used in designs. While commercially available routers are not inexpensive, purpose-built routers could be very expensive because of the level of programming involved. VoIP architectures that use COTS routers and software have the greatest potential to reduce costs. Savings achieved through the use of dedicated data circuits, rather than multiple leased lines, will depend on the density of communications provided at each network node.

4. RISK/BENEFIT ANALYSIS

4.1 Circuit Switching

Circuit switching is a proven technology that has been used for LMR transport for more than 30 years. It provides redundancy on a circuit-by-circuit basis because individual lines are used. Circuit switching provides great versatility and can support many types of existing systems. However, in some cases, separate circuits must be used when transmitting both voice and data, and some wide area circuit-switched systems require that all communications pass through at least one central hardware switch. Some wide area systems, specifically simulcast, require specialized, non-COTS transport equipment. In circuit-based systems, leased line costs can increase every time a line is added.

Circuit-switched systems often require location of dispatching consoles near the central switching equipment; in some systems the console electronics provide the switching functionality, restricting console locations and the talk groups that can be dispatched from a given console location. This architecture also requires that logging be performed close to the central switch or a remote switch, restricting the capability to log and manage systems in a distributed fashion. Finally, systems that use circuit switching usually require separate circuits for voice, data, and management. The technological risk of circuit-switched systems is low; however, there is a risk that implementing a circuit-switched system at this time would leave system managers with a system that cannot take advantage of emerging and future IP-based LMR applications.

4.2 Packet Switching

Packet-switched networks are also proven as a transport for LMR systems, although the history is much more recent. It is expected that most vendors who plan to offer VoIP systems will use mostly COTS network equipment. Packet network transport presents an opportunity to provide an unprecedented level of segregation in integrated systems through IP addressing of voice and data information including dispatch, logging, and management. Because voice, data, and management information can be carried over the same "pipes," packet networks for LMR promise greater simplicity than circuit-switched networks. VoIP also promises to provide a higher degree of interoperability for dissimilar systems by using gateways.

VoIP systems are currently only available in the United States from one supplier, MA/COM, and only in the 800 MHz frequency band. No standards govern the implementation of VoIP networks for LMR transports, and it is likely that several proprietary standards will emerge. It is also expected that to meet QoS objectives for LMR systems, packet networks supporting LMR systems must be dedicated solely for that use. Although there are systems in the field, the technical risk involved is moderate rather than low because application of the technology for LMR is fairly recent. Because it is likely that several proprietary protocols will emerge, there is a risk that system owners may become locked into a single supplier for RF and other system elements even though transport elements may be standard COTS equipment available from multiple sources. Finally, because the technology is still under development, early buyers of the technology will likely pay the bulk of system development costs. The advantages, disadvantages, and risks are summarized in Table 4.

Table 4
Summary of Advantages, Disadvantages, and Risks

	Advantages	Disadvantages	Risks
Circuit Switching	• Provides proven technology for LMR transport • Uses multiple circuits to provide protection against single point failures • Supports P25 systems	• Adds complexity in wide area systems; all audio required at a central point for processing and redistribution • Requires large amount of non-COTS equipment in some systems • Has incremental recurring line costs • Requires dispatching and logging where analog audio is present • Requires separate circuits for voice and management and reporting	• Technical risk—Low Risk that technological limitations will hinder near-term enhancement
Packet Switching	• Uses technology proven in Internet telephony and non-critical LMR applications • Uses large percentage of COTS equipment • Enables segregated operations in systems supporting multiple agencies • Provides enhanced interoperability capabilities for dissimilar systems • Can carry voice, data, and system management over the same facilities	• VoIP not addressed by any standards for LMR use • Only products currently available are in 800 MHz wideband • Requires dedicated transport network	• Technical risk—Moderate to High • Compatibility risk—Unless addressed by a standards body, evolving protocols will be largely proprietary • Financial risk that cutting-edge buyers may bear development costs

5. CONCLUSIONS

This section summarizes the conclusions reached through the analysis, provides recommendations, and identifies next steps.

5.1 Summary of Conclusions

Packet switching has the potential to save a considerably reduce costs associated with channel bank and central switching equipment as well as costs associated with leased lines.

Packet-switched transport networks using end-to-end IP protocols promise to bring an unprecedented level of segregation to integrated systems and to provide new levels of interoperability. While VoIP will not bring any new over-the-air interoperability, packet switching will provide a means for interconnecting systems on different bands using different modulation types.

Relative to QoS, it is likely that dedicated networks will be required to preserve QoS in LMR systems.

Standards are not emerging to guide the use of VoIP transport in LMR systems. The TIA/EIA 102 (P25) suite does not cover transport networks at all. However, VoIP systems may in fact meet P25, as promised by Motorola. Existing high-layer protocols are not designed for the LMR environment; they were designed for the Internet telephony and conferencing environment. Because they were not designed for LMR, they do not provide services required by LMR and they provide other services that are irrelevant to LMR. Higher level protocols for LMR use appear to be developing on a proprietary basis, leaving open the possibility that systems from different vendors will have greatly differing levels of audio quality.

To date, only one manufacturer has delivered an LMR system providing true end-to-end IP addressing—MA/COM. Motorola and EF Johnson are not scheduled to deliver systems until late 2001, at the earliest.

ACRONYMS

ADM	Add-Drop Multiplexer
AEB	Ambassador Electronics Bank
AIN	Advanced Intelligent Network
AMBE	Advanced Multiband Excitation
APCO	Association of Public-Safety Officials
APM	Application Transport Mechanism
ASN.1	Abstract Syntax Notation
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
BICC	Bearer Independent Call Control
BISDN	Broadband Integrated Services Digital Networks
bps	Bits per Second
CAI	Common Air Interface
CDPD	Cellular Digital Packet Data
CEB	Central Electronics Bank
CIU	Console Interface Unit
COTS	Commercial Off-the-Shelf
CVSD	Continuously Variable Slope Delta
DACC	Digital Access Cross Connect
DES	Data Encryption Standard
DIU	Digital Interface Unit
DRVRTP	Dispatch Radio Voice Real Time Protocol
EDACS	Enhanced Digital Access Communications System
ETSI	European Telecommunications Standards Institute
FDM	Frequency Division Multiplexing
FM	Frequency Modulation
GPS	Global Positioning System
GSM	Global System for Mobile Communications
HLB	Herbst LaZar Bell
HTTP	Hypertext Transfer Protocol
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMBE	Improved Multiband Excitation
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITN	Integrated Treasury Network
ITU	International Telecommunications Union
IVR	Interactive Voice Response
Kbps	Kilobits per Second
kHz	Kilohertz
LAN	Local Area Network
LD-CELP	Low-Delay Code-Excited Linear Prediction
LLC	Logical Link Control
LMR	Land Mobile Radio

MC	Multipoint Controller
MCU	Multipoint Control Unit
MEGACO	Media Gateway Protocol
MGEG	Motorola Gold Elite Gateway
MHz	Megahertz
MP	Multipoint Processor
ms	Millisecond
MTBF	Mean Time Between Failures
NAR	Network Access Radio
OCTA	Orange County Transit Authority
OSI	Open Systems Interconnection
OTAR	Over-the-Air Rekeying
P25	Project 25
PC	Personal Computer
PCM	Pulse Coded Modulation
PDR	Packet Data Router
PSTN	Public Switched Telephone Network
PSWN	Public Safety Wireless Network
PTT	Push-to-Talk
QoS	Quality of Service
QSIG	ISDN Q Signaling
RF	Radio Frequency
RFC	Request for Comment
RSVP	Resource Reservation Protocol
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SIP	Session Initiation Protocol
SIP-URL	SIP Uniform Resource Locator
SLIP	Serial Line IP
SS7	Signal System 7
TCP	Transport Control Protocol
TDM	Time Division Multiplex
TDMA	Time Division Multiple Access
TIA	Telecommunications and Information Administration
TOS	Type of Service
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UHF	Ultra High Frequency
VHF	Very High Frequency
VoIP	Voice over Internet Protocol
WNG	Wireless Network Gateway
WWW	World Wide Web