PUBLIC SAFETY WINS
WIRELESS INTEROPERABILITY NATIONAL STRATEGY

Technical Solutions

Voice-Over Internet Protocol

THIS VOICE-OVER INTERNET PROTOCOL (VoIP) SOLUTION ANALYSIS HIGHLIGHTS THE FOLLOWING

- Technical background
- Technical description and conceptual drawings
- Appropriate uses
- Advantages and disadvantages
- Costs
- Spectrum requirements
- Management issues
- Security and standards issues
- Additional Information

THE LAND MOBILE RADIO (LMR) INDUSTRY MAY BE MIGRATING TO A NEW GENERATION OF CONTROL ARCHITECTURE BASED ON PACKET-SWITCHING TECHNOLOGY

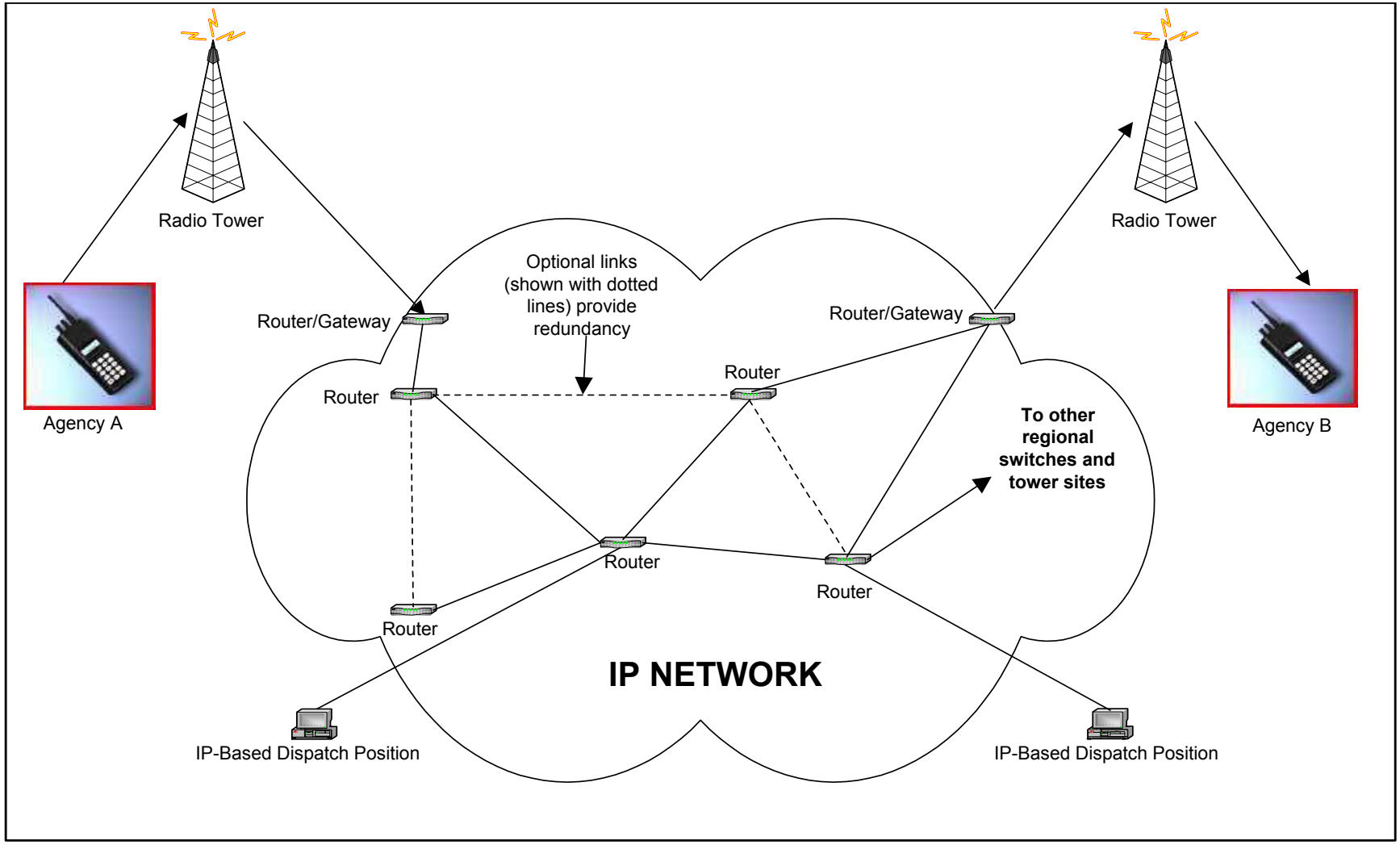
- Traditionally, LMR systems use analog, circuit-switched architecture
- Interest in VoIP originally grew as a result of the perceived cost savings that could potentially be realized by using the Internet rather than long distance telephone service to make long distance calls
  - Internet access is billed as a flat monthly fee for unlimited access while long distance service is charged by the minute
- As network performance has increased and the cost of packet-switching equipment has decreased, it is now practical to consider the VoIP solution for public safety wireless communications

VoIP TECHNOLOGY CAN BE IMPLEMENTED TO ACHIEVE INTEROPERABILITY

- In general, VoIP refers to the method of sending voice (converted into a digital stream) in discrete packets using IP, rather than in the traditional circuit-committed protocols of the public switched telephone network
- The Internet is the means by which data is sent from one computer to another
- Each computer (or host) on the Internet has at least one IP address that identifies the origin and destination of the data being transferred
  - Hosts can be assigned IP addresses in two ways
    - Static IP address—This address will never change and is preferred for VoIP access
    - Dynamic IP address—The server automatically assigns an address each time the host logs in
- Transmitted data is divided into packets each of which contains the sender's and the recipient's address
- Each message is divided into a number of discrete packets that can often be sent via different routes (typically the route with the least congestion), and can arrive at different times in a different order than originally sent
- IP simply delivers the message and relies on the Transmission Control Protocol (TCP) to reorder the packets in the proper sequence
- Gateways, which perform call setup and control, act as interfaces between the IP packet network and other protocols and formats, including analog audio

Voice-Over Internet Protocol Solution…Conceptual Drawing…

THE DRAWING BELOW ILLUSTRATES THE FUNCTIONAL CONCEPT OF VoIP



VoIP MAY BE A VIABLE SOLUTION WHEN THE PUBLIC SAFETY AGENCIES THAT NEED TO INTEROPERATE USE—

- Proprietary trunked systems from different manufacturers
- Proprietary trunked systems from the same manufacturer with the newer system not backward compatible with the older system
- The same or different frequency bands (for either trunked or conventional communications)
- Different operational modes (e.g., digital versus analog)
- Different technologies (e.g., conventional versus trunked)

THE VoIP SOLUTION HAS SEVERAL ADVANTAGES

- IP-based packet networks reduce the need for dedicated wireline voice circuit connections, channel equipment, and external interfaces
- Because IP is a widely used standard protocol, IP routing equipment and network infrastructure is available as a cost effective, scalable commercial off-the-shelf (COTS) commodity, offering economies of scale, and interface assurance
- Packet networks provide “virtual networks” over a common infrastructure, incorporating privacy between user groups
- VoIP allows for the complete integration of voice and data on an end-to-end basis over a common platform

THE VoIP SOLUTION ALSO HAS SEVERAL DISADVANTAGES

- VoIP for LMR systems is an emerging technology
  - Specific standards have not been adopted, resulting in the installation of equipment with proprietary protocols
  - There are no current standards for Quality of Service (QoS) for VoIP which yields unpredictable voice quality
  - Information on system products is limited
  - System limitations are unknown
- Error correction delays the delivery of the complete message and contributes to network congestion
- Detailed memoranda of understanding (MOU) and interlocal agreements are required. Creating and carrying out these MOUs can sometimes be a very lengthy, costly process requiring significant legal input
- Packet networks are more susceptible to unauthorized use and malicious attacks due to inherent vulnerabilities

THE VoIP SOLUTION PRESENTS SEVERAL COST CONSIDERATIONS

- Equipment costs will vary
  - Costly channel and switching equipment are no longer necessary and will be replaced by less costly and more scalable routing equipment
  - Dependent on the amount of COTS equipment and software used in system designs
  - Unique purpose-built routers could become very expensive because of the level of programming involved
- Ongoing or recurring costs are relatively modest
  - Savings achieved by using multiplexed data circuits, rather than dedicated leased voice lines, will depend on the density of communications provided at each network node
  - As new systems are identified to provide interoperability, the interlocal agreements or MOUs will have to be updated or new ones created

THE VoIP SOLUTION REQUIRES NO ADDITIONAL SPECTRUM

- The use of VoIP as a transport protocol is independent of the frequency
- The use of VoIP as a user access technology could have a large effect on spectrum requirements
  - VoIP allows over subscription of a frequency which will allow several users to share a frequency at the same time
  - By having a total IP access network, technologies such as Time Division Duplex (TDD) may be used which allows the transmit location and the user to use the same frequency

THE VoIP SOLUTION REQUIRES SIGNIFICANT MANAGEMENT BY THE PUBLIC SAFETY AGENCIES INVOLVED

- Significant coordination is required between VoIP-enabled systems and traditional systems
  - A gateway must be installed in the traditional system
  - Routing tables in the gateway must be kept up-to-date
- Development of agreements that govern agency cooperation is essential; however, it is a time-consuming process
  - Detailed MOUs or interlocal agreements are necessary so that all agencies understand their roles in the partnership
  - Formal interoperability procedures are needed to ensure that users follow established procedures on each other's systems
- System managers must have a working knowledge of data networking and must manage the network traffic to maintain QoS standards
- VoIP technology will be seamless to the end user; therefore additional training may not be required
- VoIP must provide the segregation required by the various missions of public safety agencies, while allowing for the purchase, maintenance, and operation of a single backbone

VoIP TECHNOLOGY PRESENTS SEVERAL SECURITY AND STANDARDS ISSUES

- Outside or hostile parties generally have greater access to packet networks than to circuit-switched networks
- Encryption is typically employed in IP networks to provide end-to-end security; however, this may not be an issue in LMR networks that employ end-to-end encryption via subscriber and base station equipment
- The Project 25 or Telecommunications Industry Association (TIA)/Electronic Industries Alliance (EIA) 102 suite of standards does not address the use of VoIP as a transport for LMR systems
- Several manufacturers of VoIP equipment have been reluctant to adopt standards and have deliberately produced proprietary solutions

PUBLIC SAFETY AGENCIES SHOULD ADDRESS SEVERAL ADDITIONAL ISSUES WHEN CONSIDERING THE IMPLEMENTATION OF THE VoIP SOLUTION

- Integrating voice and data
- Addressing specific challenges faced when implementing an IP packet-switching network
- Exploiting the best practices and lessons learned from other agencies that have implemented VoIP

SEVERAL KEY CONSIDERATIONS MUST BE ADDRESSED WHEN IMPLEMENTING VOICE AND DATA ON IP NETWORKS

Key Considerations for Implementing Voice and Data on IP Networks	
Considerations	**Description**
Network Congestion	• This challenge is the principle cause of distortion and dropouts • When network congestion can be avoided, IP audio quality is comparable with DS0 channels or dedicated leased lines • Unlike the traditional one-to-one relationship between calls and time division multiplexing (TDM) channels, packets carry messages from many different points to many different points over many different links
Noise and Distortion	• Distortion and dropouts can occur due to lost packets within the network • If the network is at full capacity and the “time-to-live” parameter is exceeded, routing devices discard packets • The number of lost packets can be minimized by assigning high precedence in the addressing portion of the voice packet, thereby overriding non-voice packets and reducing distortion
Latency	• Transmitted audio can be unsatisfactory if overall delay is too long • For received speech to be acceptable, end-to-end delay must not exceed 150 to 200 milliseconds • Several processes contribute to delay, among them are— – Encoding – Decoding – Transmit time – Buffering – Queuing • If network size cannot be controlled, network speed (bandwidth) and routing design must be controlled and optimized
Signaling	• Rather than using common channel signaling, packet-switching networks use addressing and overhead to carry call setup and routing information along with the transported data • Packet size must be managed in order to efficiently carry voice over the network
Voice Coding	• This technique is used in digital voice transmission to preserve the voice waveform with minimum discernable difference from the original signal • Traditional encoding bit rates (32 or 64 kilobits [Kbps] per second) are impractical for transmission across IP networks and in particular over wireless networks

IN SUMMARY, SEVERAL SYSTEM PERFORMANCE CHALLENGES SHOULD BE CONSIDERED AND EVALUATED BEFORE IMPLEMENTING A PACKET-SWITCHING NETWORK

Considerations for Using IP Packet Switching			
Performance Challenge	**Technical Description**	**Underlying Causes of Poor Performance**	**Mitigation Strategy**
Voice Clarity	Audio frequency response and harmonic distortion must approach 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 quality of service protocols: Real Time Protocol (RTP) and other future protocols
Voice Intelligibility	Nonlinear distortion due to transmission errors must not affect speech intelligibility	• The bit error rate 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 contribute to 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 continuously vary, potentially causing packets to arrive out of sequence • Packets must be queued—the process of resequencing introduces delays—to restore their correct order	• Reduce total number of network nodes • Increase precedence in voice packets' TOS field

IN SUMMARY, SEVERAL SYSTEM PERFORMANCE FACTORS THAT USERS SHOULD CONSIDER AND EVALUATE BEFORE IMPLEMENTING A PACKET-SWITCHING NETWORK (CONTINUED)

Considerations for Using IP Packet Switching			
Performance Challenge	**Technical Description**	**Underlying Causes of Poor Performance**	**Mitigation Strategy**
Overall Delay	Delays from end to end accumulate throughout the network	• The 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, encryption, and end-to-end network delay can exceed the maximum tolerances to maintain voice intelligibility	• Decrease packet length to reduce packetizing delay • Use the higher bit rate Low-Delay Code-Excited Linear Prediction 16 Kbps vocoding (to reduce coding delay) • Simplify network (to minimize packet buffering) • Do not use forward error correction • Eliminate any encryption (to reduce overhead)
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

WHILE USE HAS BEEN LIMITED IN THE PUBLIC SAFETY ENVIRONMENT, SEVERAL PRIVATE ENTITIES AND GOVERNMENT AGENCIES HAVE IMPLEMENTED OR ARE IN THE PROCESS OF IMPLEMENTING THE VoIP SOLUTION

- Currently, Federal Express operates the largest private wireless data system in the world. The system consists of 20,000 units that provide voice and data service nationwide. The Federal Express VoIP system is projected to grow to approximately 40,000 units

- The Commonwealth 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 will have seven regional operating centers linked together over the statewide packet network

- Lancaster and Cumberland counties in Pennsylvania are in the process of implementing a VoIP system that will be capable of integrating with the statewide system

- The Orange County Transit Authority in California is currently installing a system that will initially support approximately 500 buses

- Many European private entities and government agencies achieve interoperability by using TETRA VoIP implementations