

**Implementing Voice Over IP Telephony  
in 2-1-1 Call Centers**

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## **Implementing Voice Over IP Telephony in 2-1-1 Call Centers**

### **Introduction**

As data traffic has continued to increase at a much faster pace than voice/telephone traffic, there has been growing interest in consolidating voice and data networks. While previous emphasis has been on transporting data over voice networks and using modems to convert digital data to analog phone signals, the current emphasis is to provide voice services over data networks. These networks are optimized for carrying data and are an important step in phasing out the existing voice networks. Since it makes use of the ubiquitous IP protocol, Voice over Internet Protocol (VoIP) has become a leading technology used in implementing voice services over data networks. Businesses are increasingly switching over to VoIP systems. Frost and Sullivan, a marketing research firm, recently estimated the annual growth rate for IP-enabled telephone equipment to be 132% from 1997-2002.<sup>1</sup> Additionally, it is estimated that 70% of Fortune 1000 companies use VoIP in their network structure.<sup>2</sup>

VoIP can provide a number of benefits to 2-1-1 providers. Consolidating both data and voice networks provides potential cost savings to agencies. Communicating over data rather than voice system cuts down on long distance charges between agencies. This infrastructure also allows smaller agencies with limited operating hours to “roll over” their services to larger call centers that have expanded hours. Additionally, VoIP can provide an infrastructure that allows for a statewide integration of resources and databases. Finally, a VoIP system provides transparency to callers. Allowing call centers

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<sup>1</sup> <http://www.voipwatch.com>

<sup>2</sup> *ibid.*

to share databases provides a platform to share resources across agencies, such as multi-lingual staff members who can answer calls from geographic locations across the state.

This paper will provide an overview of the technical details of the VoIP system. Currently, the only state to implement a VoIP system for 2-1-1 is Texas. The next section will highlight how the Texas system works. Finally, we provide a list of resources for 2-1-1 providers to learn more about how VoIP might benefit their agencies.

## **Background on VoIP**

VoIP is defined as the use of IP-based data networks to make telephone calls and to send and receive facsimiles with an acceptable quality of service (QoS).<sup>3</sup> VoIP applications transmit voice data through implementation of the International Telecommunication Union (ITU) protocol specification H.323, the protocol for transmitting multimedia data across a packet switched network.<sup>4</sup> This protocol is an umbrella protocol for VoIP, specifying components and procedures that provide multimedia communication services. There are many other protocols and standards that H.323 calls upon when transferring voice over an IP network. These other protocols handle specific VoIP related tasks such as:

- call signaling (H.225)
- transmitting media streams (H.245)

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<sup>3</sup> Cisco systems develops a number of Quality of Service (QoS) technologies and provides a comprehensive definition: “Quality of Service (QoS) refers to the “capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. The primary goal of QoS is to provide priority to dedicated bandwidth, control jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics” (QoS technologies provide the elemental building blocks that will be used for future business applications in campus, WAN, and service provider networks” ([http://www.cisco.com/univercd/cc/td/doc/cisintwk/ito\\_doc/qos.htm#wp1024961](http://www.cisco.com/univercd/cc/td/doc/cisintwk/ito_doc/qos.htm#wp1024961)).

<sup>4</sup> The International Telecommunication Union (ITU) was established in 1865 to manage the first international telegraphs. Since then, ITU’s goal is to integrate new technologies into telecommunications networks throughout the world.

- data transfer (T.120)
- multiple voice encoding and decoding standards (G.711, G.726, G.723.1).

In addition, H.323 implements several protocols specified by the Internet Engineering Task Force (IETF).<sup>5</sup> These IETF protocols have significant responsibilities. For example, Real-Time Transport Protocol (RTP) is the Internet standard protocol for the transport of real-time audio/video data and is used in coordination with UDP to transport voice data packets over a network. UDP, or User Datagram Protocol, is a transport level protocol similar to the more commonly used TCP. UDP does not provide the same kind of flow and congestion controls as TCP, however it is faster and is commonly used in conjunction with Real-Time Transport for multimedia streaming applications. Resource Reservation Protocol (RSVP), another IETF protocol, is used to reserve network resources for VoIP calls, thus ensuring a high enough QoS.

### **Benefits of VoIP**

VoIP services aim to provide a superior cost/benefit to the public switched telephone network (PSTN). For example, VoIP technology utilizes relatively inexpensive bandwidth to transport phone calls over IP networks. Therefore, VoIP can potentially offer significant savings on long distance and international calling. Sending faxes over VoIP networks is another promising application of the technology. In addition, VoIP leverages the convergence of voice and data networks by offering new types of integrated voice/data applications not available using PSTN. For example, integrated messaging, which allows users to retrieve or leave voice mail over the Internet is one possible application of VoIP technology. Other possible applications include multimedia

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<sup>5</sup> The Internet Engineering Taskforce (IETF) is an international community of network designers, operators, and researchers concerned with the evolution of Internet architecture and the smooth operation of the Internet. For more information on IETF, see <http://www.ietf.org>

teleconferencing over the Internet and so-called “click to talk” web sites. Multimedia teleconferencing would allow people to communicate in real time with audio and video while possibly manipulating data such as documents and presentations. “Click to talk” websites are E-commerce sites where customers can talk to sales or customer service representatives at a call center by clicking on a phone icon.<sup>6</sup>

### **Components of VoIP Network**

VoIP networks are traditional IP networks with the expanded capability of carrying packetized voice in addition to traditional data packets. To facilitate the transfer of voice data, VoIP networks contain additional components such as gateways, gatekeepers, multipoint control units, and CODECs. These components provide essential VoIP services such as encoding/decoding of calls, call setup, routing calls, logging calls, QoS control, and support for conference calls. The following section will outline components of VoIP networks including terminals, gateways, gatekeepers, multipoint control units, and CODECs.

#### ***Terminals***

Terminals are the endpoints on a local area network (LAN) from which users make and receive VoIP phone calls. Terminals must be able to support real-time, two-way audio communication with another entity using VoIP. Optionally, terminals can support video and data. In other words, a VoIP terminal must be able to encode and decode phone conversations and possibly be able to encode and decode video and data packets. VoIP phones, similar to traditional handsets, are an obvious example of

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<sup>6</sup> CrystalVoice, a company offering VoIP technologies and services, has an example of a “click to talk” website set up at: <http://www.crystalvoice.com/ACME/index.htm>

terminals, however terminals can be multimedia computers or even personal digital assistants (PDAs). Using PDAs and computers as terminals allows for the design of easy-to-use graphical user interfaces (GUIs). GUI's take full advantage of a computer's mouse and keyboard and allow for easier user control of a computer than the twelve-button dialpads allow for phones.

### ***Gateways***

The Terminals in a VoIP system connect to a Gateway. Gateways are the points where local area VoIP networks interface with larger IP networks and circuit switched telephone networks. Due to its position in the network architecture, gateways must provide several services including translating between network formats, call setup and call clearing, and call routing. To translate between network formats gateways encode incoming calls from the PSTN network into voice packets for the VoIP network and vice versa. Call setup and call clearing are very similar to call signaling in traditional telephony. Call setup is the procedure initiated when a user picks up a phone. Call setup allocates the necessary resources needed for a phone call and provides the user with a dial tone. Once a dial tone has been provided and a user has entered a telephone number a gateway routes the call to the appropriate location in the network, which is either another IP network or a PSTN network. Call clearing is similar to call setup and is initiated when a user hangs up his or her phone, signaling the gateway to free up network resources allocated by the call setup procedure.

### ***GateKeepers***

Gatekeepers are optional when deploying a VoIP system. However, they provide many convenient services such as routing, QoS control, call control signaling, and call authorization. QoS control is one of the more important functions of gatekeepers. By allocating a minimum amount of bandwidth for a phone call, QoS control assures a certain level of quality for that call. QoS control can be configured in a variety of ways. For example, inter-office calls could have a lower minimum bandwidth requirement than a customer call. Gatekeepers also handle call authorization, by determining whether users have permissions to make certain kinds of calls, such as international long distance calls. Many of the other services provided by gatekeepers, such as call control signaling, including call setup and clearing, and routing are handled by gateways if gatekeepers are not present in the network.

### ***Multipoint Control Unit***

As mentioned earlier, VoIP technology can provide multimedia teleconferencing over the Internet. Multipoint control units (MCUs) are the network components that provide this kind of service in a VoIP system. Multipoint control units support communication between three or more endpoints. MCUs can be placed in different parts of the network. For example, an MCU can be built into a stand-alone device such as a PC or phone, or alternatively can be part of a gateway or gatekeeper. MCUs facilitate teleconferencing by handling call control signaling and by receiving voice/data streams, processing them, and returning them to endpoints participating in the teleconference.

## ***CODECs***

Coder/Decoders, or CODECs as they are commonly called, are located at the terminal level, and are used to convert between analog and digital data formats. For example, a coder would take a message spoken into a terminal and convert it into packets of data suitable for an IP network and a corresponding decoder would convert the packet back into an analog signal understandable by a user. CODECs are also used for compressing and decompressing packets, saving bandwidth.

## **Anatomy of a VoIP Call**

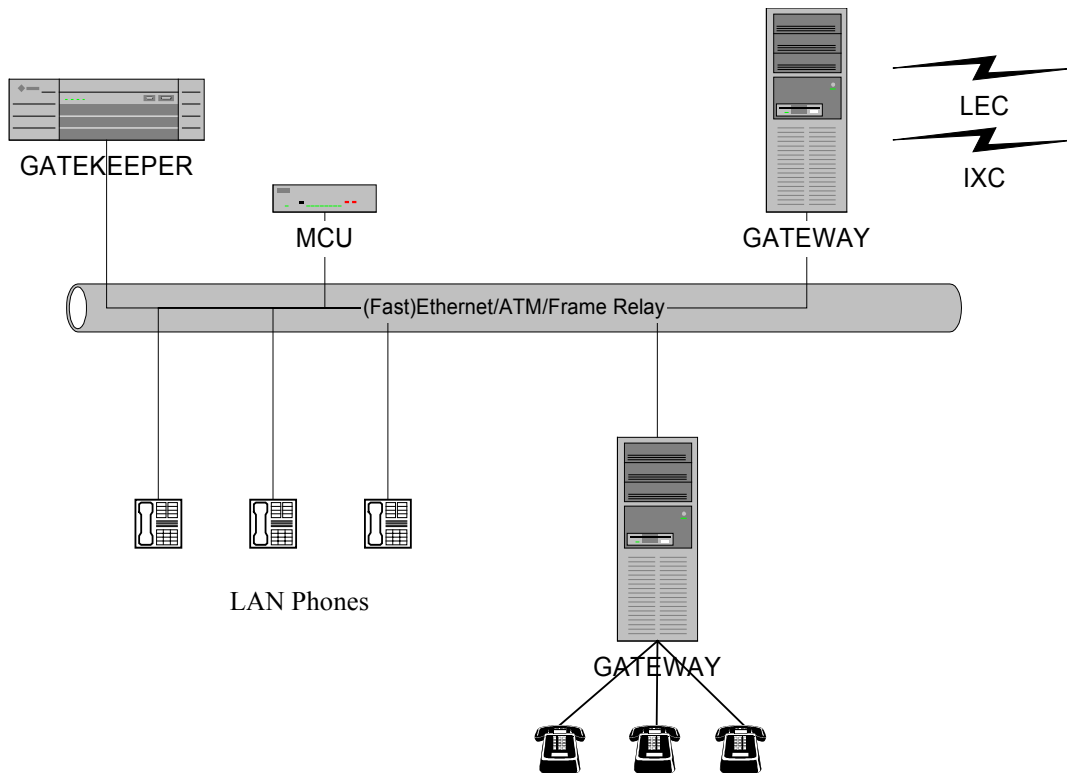
A VoIP phone call is initiated when a user picks up a handset signaling an off-hook condition to a VoIP Gateway, as opposed to signaling the local exchange carrier (LEC). The signaling application then issues a dial tone and waits for the user to enter a telephone number. After a telephone number that can be matched to a destination has been entered it is mapped to the network's VoIP Gateway. This mapping is either over a LAN or is a local phone call. After the gatekeeper application authorizes the call, the Gateway then either signals the destination telephone or forwards the call to a private branch exchange, which handles completing the call.<sup>7</sup> If the IP host establishes the call it initiates the H.323 session protocol, opening a transmission channel and a reception channel using RTP/UDP/IP. At this point many VoIP applications are allocated network resources by IP hosts using Resource Reservation Protocol (RSVP), insuring a high enough quality of service. CODECs are used, at the terminals, to digitally encode and decode messages to reduce the amount of bandwidth used and to packetize the voice signal. These CODECs are standardized using coding schemes designated by the ITU. If

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<sup>7</sup> A private branch exchange, or PBX, is a private telephone network used within a business. In addition to PBXs, many companies will have a number of outside lines used for making calls external to the company.

the VoIP call involves more than two parties, the encoded messages are sent to the Multipoint Control Unit, which processes them and sends them to the other participants in the conference call. When either side of a two party call hangs up the connections are closed, the session ends and is logged by the Gateway or Gatekeeper. Figure 1 illustrates the components of the VoIP network structure.

**Figure 1. VoIP structure**



Source: Sethi, Vish, Jignesh Dixit, and Benjamin Barr. "IP Telephony, Voice Over IP (VoIP)." Found at [http://www.stuart.iit.edu/courses/tcm517/winter2001/classfiles/121801/Vishvash%20Sethi%20voip\\_combined.ppt](http://www.stuart.iit.edu/courses/tcm517/winter2001/classfiles/121801/Vishvash%20Sethi%20voip_combined.ppt)

## **Challenges and Issues Facing VoIP**

There are several challenges in implementing a Voice over IP system. For example, VoIP must provide comparable voice quality to that of traditional phone networks (so called “toll quality voice”) to gain acceptance by users. Similarly, the telephone calling process should be transparent. In other words, users should not need to know what kind of technology is being used to implement telephone service in order to make a phone call. Organizations seeking to implement VoIP must also have intranets capable of supporting VoIP and their ISP should be able to provide and guarantee a high enough quality of service for voice traffic.

### **Case Study: Texas**

In the Fall of 2002, the Texas I&R Network became the first 2-1-1 provider to launch a VoIP System. The Texas I&R Network is a public private partnership that includes statewide agencies such as the Texas Health and Human Services Commission, the United Ways of Texas, and the Texas Alliance for Information and Referral, as well as community based organizations and local nonprofits. The goal of the Texas 2-1-1 system was to provide both a cost-effective and efficient statewide system. Texas covers a large geographical area with twenty-five Area Information Centers (AIC). Each of these centers had their own PBXs and separate phone numbers.

The Texas I&R Network worked with Cisco Systems to design an infrastructure that would integrate the twenty-five AICs. Three primary features were incorporated into the network design including: shared resources among the AICs, 24-hour year-round availability, and affordability. Many of the AICs were not stand alone centers, but instead

existed within other agencies. Thus, Cisco integrated new VoIP technology with the AICs' existing PBX's, integrating with their LANs and enabling them to use existing technology when possible.

For the statewide 2-1-1 system to be successful, each of the 25 AICs had to be able to communicate and share resources, providing transparency to callers. While each AIC had access to its own centralized database of local and state resources, databases were not integrated, limiting the ability to share resources. In each of the AICs, calls were delivered over a 1-800-line that incurred per-minute fees. Additionally, the long-distance charges for centers to transfer calls to each other and share information was estimated to cost one million dollars a year.<sup>8</sup>

The other goal in the network design was to make 2-1-1 services available 24 hours a day. The strategy was to have calls roll over from smaller centers that closed its operating hours in the evening to larger centers that provided continuous service. A VoIP system allowed the centers to share data and work more efficiently so that callers did not know the call center was in a different area.

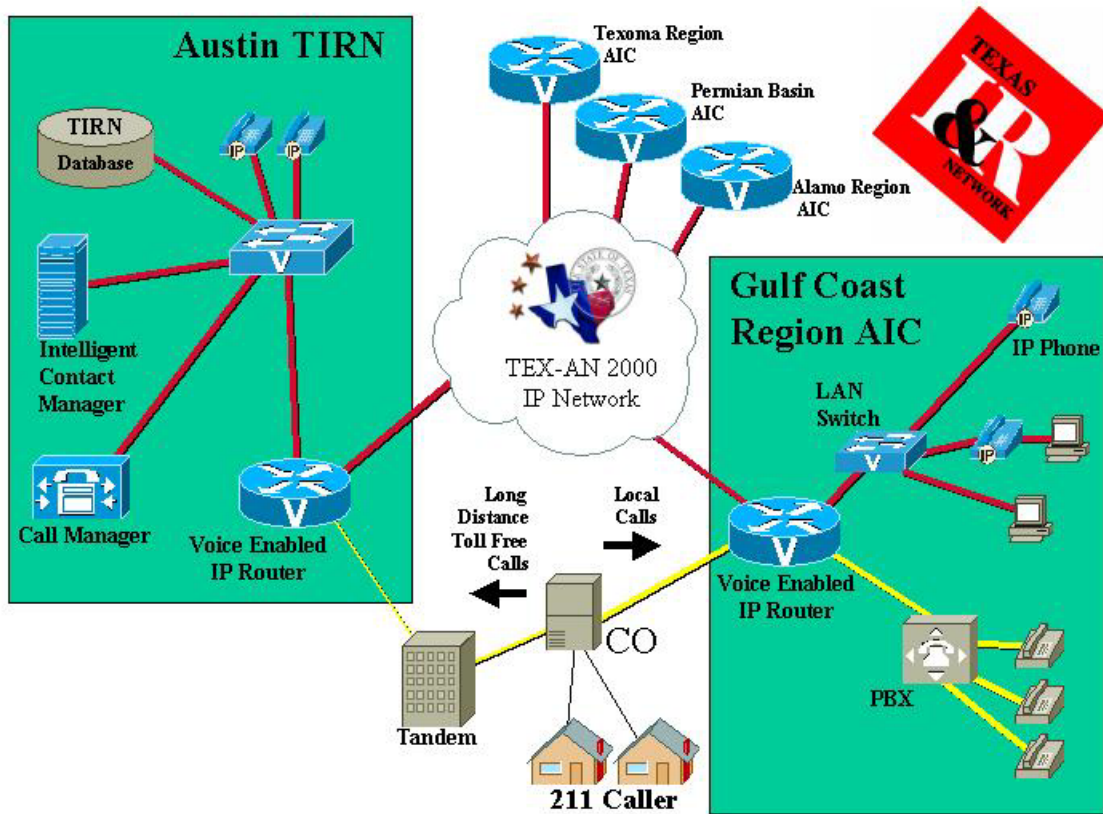
The VoIP system allowed Texas I&R Network to create a networked virtual organization in the form of a statewide virtual call center. The Network had to follow a number of steps in order to implement this system. First, the Network worked with the Public Utilities Commission (PUC) to get local telephone companies to implement the 2-1-1 dialing switching statewide and to negotiate reasonable switching rates. Next, the agency worked with Cisco systems to facilitate site surveys, develop an installation and maintenance plan, and work toward network design. In turn, Cisco also developed an e-

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<sup>8</sup>[http://business.cisco.com/prod/tree.taf%3Fasset\\_id=103214&ID=48296&ListID=44692&public\\_view=true&kbns=1.html](http://business.cisco.com/prod/tree.taf%3Fasset_id=103214&ID=48296&ListID=44692&public_view=true&kbns=1.html)

learning proposal for training call center staff and agreed to conduct staff training. Figure 2 outlines the components of the Texas VoIP system.

**Figure 2. TX 2-1-1 VoIP Network**



- Key:**
- AIC: Area Information Center
  - CO: Central Office
  - LAN: Local Area Network
  - PBX: Private Branch Exchange
  - TIRN: Texas Information and Referral Network
  - TEX-AN 2000: centralized telecommunications system for Texas

The following features were included in the VoIP system:

- Integrated email/voicemail
- Call monitoring
- Call recording
- Internet Access

- XML phone messaging
- All traditional phone features

Additionally, the VoIP system allowed AICs to share specialized skill sets, such as bilingual services. One of the biggest advantages of this system was that it allowed all call centers to have the same capabilities regardless of their size and local resources.

The VoIP initiative began in March 2002, with implementation starting in early October. In December 2002, fourteen of the twenty-five centers were live. These fourteen AICs covered approximately 83 percent of the Texas population.<sup>9</sup> In February 2003, 74,000 calls were answered without a problem.

The Texas 2-1-1 system has been a success. When the Columbia shuttle disaster occurred in February 2003, the VoIP structure allowed every call center in the area to be alerted that there may be debris in the area and that they should inform the public of how to deal accordingly. Additionally, the 2-1-1 system has become a centralized site for disseminating information for social services and is prepared to assist in homeland security. Judy Windler, Director of the Texas A&R Network, anticipated a savings of \$400,000/yr, and additional cost savings by reducing the need for full-time, multilingual staff at all call centers (because operators can transfer calls to other centers) and eliminating incoming toll and outgoing long-distance charges.<sup>10</sup> Additionally, call routing is transparent to the callers.

### **Other VoIP Applications**

Commercial businesses increasingly are switching to a VoIP system because of its advantages, such as integrating email and voicemail, “click and talk” websites, and

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<sup>9</sup>ibid.

<sup>10</sup> ibid.

economic savings through convergence.<sup>11</sup> Higher education institutions are also opting for VoIP infrastructure as a way to facilitate communication across several locations. For example, Brigham Young University, located in Provo, Utah, recently installed about 6,000 phones across their campus.<sup>12</sup> Additionally, Florida International University switched to a VoIP system, citing innovative features, superior voice quality and efficiency, and emergency response as driving the change.<sup>13</sup>

This next section highlights the implementation of VoIP from the University Corporation for Atmospheric Research (UCAR). This case study illuminates both the challenges and advantages of implementing VoIP.

### ***NCAR VoIP Case Study***

The University Corporation for Atmospheric Research (UCAR) is a nonprofit organization formed in 1959 by a number of research institutions with doctoral programs in the atmospheric and related sciences.<sup>14</sup> UCAR carries out its mission through two programs, the National Center for Atmospheric Research (NCAR) and the UCAR Office of Programs (UOP). The entire organization is located in five facilities in Boulder, Colorado, including the Mesa Laboratory, Foothills campus, Center Green Campus, the Jefferson County Airport Research Aviation Facility, and the Marshall Field Site.

The UCAR telephone system previously included five separate PBX systems with over 2,500 active ports. This PBX system was scheduled for replacement and updates. As a result, the organization had to decide whether to stick with a traditional telephone

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<sup>11</sup> For example, Dow Chemical has about 40,000 IP phones. (Personal communication, Jeff Custard, July 17, 2003)

<sup>12</sup> *ibid.*

<sup>13</sup> For more information on Florida International University's VoIP deployment, see [http://newsroom.cisco.com/dlls/video\\_audio\\_archive/?1300](http://newsroom.cisco.com/dlls/video_audio_archive/?1300)

<sup>14</sup> Information for the NCAR case study was provided by Jeff Custard, UCAR/NCAR/SCD/Network Engineering and Telecommunications Section and the website <http://www.scd.ucar.edu/nets/intro/staff/jcustard/>

system or to opt for an updated Voice over IP (VoIP) system. Around January 2001, members of the Network Engineering and Telecommunications Section (NETS) IP Telephony (IPT) team began meeting with the organization's advisory board and Information Technology Council (ITC) to discuss the benefits of VoIP for this organization. VoIP presented a number of advantages over the existing PBX system. A VoIP system seemed most logical because the facility exists across several locations, and UCAR has a diverse IT environment with many of the major divisions providing their own IT support services. Examples include separate email servers, separate PC support staff, and different client and server operating system environments. UCAR also requires many moves, adds, and changes when various groups change locations throughout the facility. The VoIP system makes these changes much easier to accomplish.

The NETS IPT team emphasized the technical advantages of a VoIP system. First, since VoIP used the network backbone, the agency would save costs by utilizing the existing infrastructure. The VoIP system also offered increased system redundancy and failover above and beyond the PBX system's capabilities. Additionally, cost savings would extend to ongoing costs through allowing ease of future moves, adds, and changes. Each site would be able to back up another site for main call routing functions. Redundancy would allow more flexibility in the organization's operation. Redundancy examples start at the server level where RAID disk arrays, redundant power supplies, and so on provide increased system reliability. Uninterruptible Power Supplies (UPS) also provide for system uptime in the event of power outages. All the facility's networks have dual uplinks, dual-power supplies, which increase overall system reliability and availability. The team also emphasized the standards upon which VoIP is based, including H.323 and the IP protocol and potential future system integration with other

protocols like the Session Initiation Protocol (SIP). These standards would allow for future upgrades and choice in the software solutions and vendors. Finally, with a VoIP system, UCAR would switch to a single wiring closet (i.e., the same wiring infrastructure can be used for both voice and data connectivity), have a choice of phone power, and voice quality guarantees.

Network engineers designed a phased roll-out of the VoIP system in order to provide transparency to end users. One of the challenges for UCAR was the variation in existing hardware in place at the various facilities. Designers started with an initial “sandbox,” which was a testbed for initial set up and basic component interaction testing. This sandbox included about twenty-five phones with a variety of system and user configurations. By 2002, the Marshall Field site was fully operational with the VoIP system. Complete organizational conversion from the old PBX system to the new IPT system will be completed by the end of calendar year 2003. The final deployment will include 4 Cisco Call Managers, 1 Cisco Unity Voice Mail Server, 1 Cisco Emergency Responder (CER) server, several other “supporting services” servers (e.g., call authorization, billing, etc.) and about 1,800 IP phones.

One of the current challenges for NCAR is the ability for users to be able to dial 9-1-1 from the IP-enabled phones. To this end, they have installed Cisco Emergency Responder (CER) which provides the location of the IP phones. The FCC ruling on E 9-1-1 required that organizations be able to locate the position of the caller.

## **Conclusion**

VoIP is emerging as a cost effective alternative to conventional telephone networks, and it offers considerable opportunities for cost savings and enhanced resource sharing across distributed operations. The Texas 2-1-1 system is intended to act as a

model of how other states can implement a statewide 2-1-1 VoIP system. Future plans for Texas providers include the possibility of acting as a host center for other states by providing 2-1-1 services at a cost efficient rate.

In conclusion, organizations wishing to switch over to a VoIP system may consider the following in their technical plans<sup>15</sup>:

1. Make your testbed as realistic as possible. Test everything you hope to deploy. Scale it as much as you can.
2. Spend a lot of time up front working through your Quality of Service (QoS) plan. Start with the high level service definitions, then work through high levels of detail down to the individual device setting level (e.g., switch port settings, etc.).
3. When working on cost models, don't forget to include licensing for all components (e.g., do specific software packages require individual phone licenses?).
4. What is your existing level of system administration support? Ensure you will have adequate staff to deal with the system administration load that will be introduced by the VoIP system you select.
5. Work on your disaster recovery plan as early as possible. VoIP offers many advantages in this area. Consider backup strategies, physical separation of various system elements, and so on.
6. Carefully consider your deployment options. Would a phased conversion or a "forklift upgrade" make more sense?
7. Do not forget legacy application support issues (this might include things like existing analog line support; paging systems, recording options, etc.).
8. Be sure to include ongoing support and maintenance costs for all your hardware and software components. This is a crucial and often not inexpensive category. Unless your maintenance covers this, you may need to consider having various spare components on site--factor these into your budget as well.

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<sup>15</sup> These considerations were provided through personal communication with Jeff Custard, July 31, 2003.

## **Additional Resources**

<http://research.ihets.org/research/iptel/articles/ihetspart1.jsp>

<http://www.cisco.com/go/avvid>

<http://www.internet2.edu/>

<http://www.iptnetworkers.com/>

<http://forums.cisco.com/eforum/servlet/NetProf?page=main>

<http://www.scd.ucar.edu/nets/docs/reports/trips/2002/jc-2002-07-08-cisco-net-workers-trip-report.htm>

<http://www.scd.ucar.edu/nets/docs/reports/trips/2002/jc-200204-i2-voip-mtg-trip-report.htm>

<http://www.erlang.com/>

<http://www.nwfusion.com/research/voip.html>

[http://www.cis.ohio-state.edu/~jain/refs/ref\\_voip.htm](http://www.cis.ohio-state.edu/~jain/refs/ref_voip.htm)

<http://www.protocols.com/voip.htm>