Voice over IP

A Primer for Resellers

Content

Voice over IP Primer

Introduction

Voice over the Internet Protocol (VOIP) is one of the hottest topics for multi-location businesses of any size because of its promise to transmit voice over data networks free of charge. Instead of paying the phone company to route long distance calls between offices, any IP data network can route the calls over a private intranet or the public Internet and save thousands of dollars annually.

The ideal customer for this technology is a business with multiple locations. We've found that typically 25-40% of a company's long distance bill is spent on intra-office communication. Recouping this portion of the long distance bill can add up to significant savings.

Many companies have invested thousands of dollars in their voice and data networks. Multi-Tech's approach to Voice over IP is to protect that investment by tying both the voice and data network together with our MultiVOIP gateway. MultiVOIP is a point-to-point, or point-to-multipoint, IP gateway. It integrates seamlessly into the data network and operates alongside existing PBXs, or other phone equipment, to simply extend voice capabilities to remote locations. The voice traffic essentially "rides for free" on top of the data network, using the data infrastructure and hardware already in place.

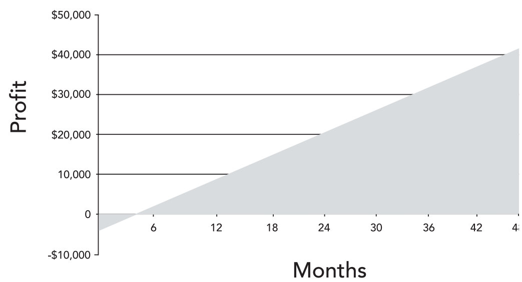
With MultiVOIP, a small investment will pay for itself within the first six months to one year and then start paying the company back. In our company example, which we will refer to throughout this primer, we have three locations that utilize a MultiVOIP solution: Minneapolis-corporate, Los Angeles-branch office and London-branch office. At headquarters, they have 60 employees and 16 phone lines. At the Los Angeles office there are 12 employees and 5 phone lines. And, in London, there are 8 employees and 4 phone lines. In the following ROI analysis, you can see that even with low long distance rates, MultiVOIP will return the company's investment within 4 months. Beyond that, the company will start to profit from the solution.

Voice over IP Return on Investment

Return on investment within six months.

Locations	MultiVOIP Cost	Long Distance Cost/Minute	Minutes/ Line/Day	MultiVOIP Payback
Corporate Site/ Minneapolis	$1,999 MVP410 (4 lines)	$0.04	90	139 days
Branch Site/ Los Angeles	$1,099 MVP210 (2 lines)	$0.06	60	153 days
Branch Site/ London	$1,099 MVP210 (2 lines)	$0.08	60	115 days

Same customer making money with MultiVOIP.



While VOIP is a compelling technology for a business to embrace, the challenge is in bringing together the two very different worlds. This primer will serve to educate you on the voice network, the data network and the new converged network utilizing our MultiVOIP gateway. In addition, it will cover many of the frequently asked questions that a data communications manager and/or telecommunications manager may have with the new technology. Finally, we will put it all together and discuss how to configure a MultiVOIP Voice over IP network.

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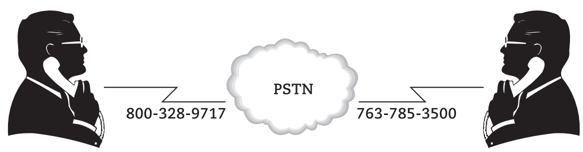
In order to understand how voice communications would be treated in a data network, let's first take a look at the typical topologies of voice and data networks running separately.

Understanding Voice Networks

Public Switched Telephone Network

For the past 100 years, people have relied on the Public Switched Telephone Network (PSTN), otherwise known as POTs (Plain Old Telephone Service), for voice communication. Although it is very reliable, it utilizes a very basic and inefficient method for making a connection called circuit-switching. To illustrate a circuit-switched connection, the following is the process that takes place when you make a phone call. First, you dial the number of the party you wish to talk to. The call is then routed through the switch at your local central office (CO) to the party you are calling, opening the circuit. Depending on location, the call may be routed through multiple CO connections opening a circuit through each one. During the call, the routed line is dedicated to the two parties. This means no other information can travel over the line, even though there is plenty of bandwidth available.



Private Branch Exchanges

In a corporation, voice communication has traditionally been handled by proprietary platforms called private branch exchanges (PBXs). A PBX is essentially a switch used to connect a number of phones (extensions) to each other and to one or more outside phone lines. To illustrate how the PBX works, when a user picks up a phone (extension) a PBX dial tone will be heard. At this point, the user can dial any other extension on the PBX. To reach an outside line, the user typically dials a "9" (or presses a pre-programmed button) to access the PSTN network.

A PBX was originally designed to save the cost of requiring a line for each user to the telephone company's central office (CO). In effect, the PBX acts like a mini-CO, owned and operated by the corporation.

A limitation to the traditional PBX is that it is a location-centric platform. The networking options to extend voice communications to other remote locations (e.g. branch offices, sister companies, satellite offices, telecommuters, etc.) are few and can be costly.

One option, if the remote office is large enough, is to add another PBX at the remote site, and set up a private network between the two with leased lines (tie lines) purchased from the phone company. To make an outgoing call to the remote office, the user would dial an "8" (or press a pre-programmed button) to access the tie line and then dial the remote office extension. Tie lines, however, are expensive because they add extra monthly phone charges. And, the telecommunications manager may be faced with the challenge of tying together two dissimilar proprietary PBX systems that were not designed to be networked together.

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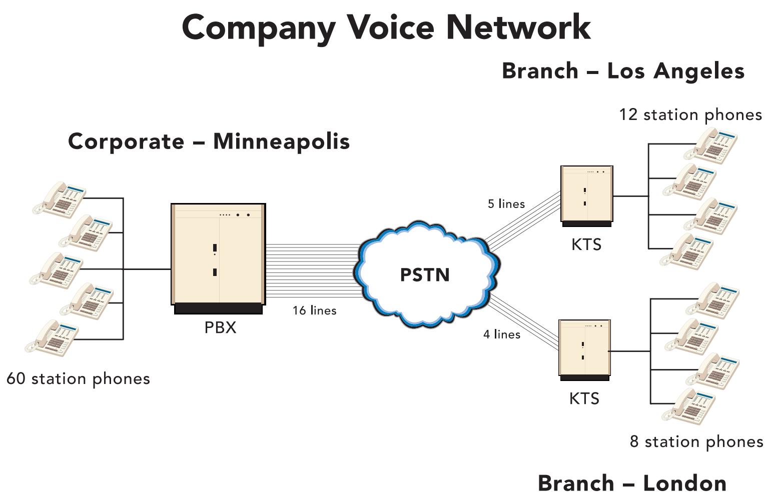
Another option is to provide the remote office with a key telephone system. A key system is a lower priced, reduced functionality version of the headquarters PBX. Because it is a scaled down system, it isn't designed to be networked with other phone systems. Therefore, calling a remote office is like calling a separate company.

Using our company example, the following diagram maps out their existing voice network.



The Challenge - Limited Networking Solutions

With limited networking solutions, remote office workers often feel handicapped by the difficulty of communicating with the rest of the organization. And, telecommunications managers are challenged with creating more efficient and cost-effective voice communications in an environment that wasn't designed for networking.

Understanding Data Networks

The Internet Protocol

The Internet Protocol (IP) was designed specifically for the Internet to act as the first truly universal networking language. It is like a postal carrier — its job is to faithfully transport packages (or packets) from anyone to anyone over any type of physical connection.

How a Typical IP Data Network Works

An IP data network is a highly distributed networking environment in which clients access information stored in servers throughout the network. These servers can be anything from giant mainframes to small departmental file servers running on PCs. An IP data network utilizes packet-switched connections, routers and IP addresses to communicate with the different networked devices.

Packet-Switched Connections

IP data, whether in the form of a Web page, a downloaded file or an e-mail message, travels over a system known as a packet-switched network. The sending computer chops data into small packets, with an address on each one telling the network where to send them. When the receiving computer gets the packets, it reassembles them into the original data.

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Routers and IP Addresses

A router is an advanced networking component that determines the route that IP packets of data will take. It has two separate, but related, jobs:

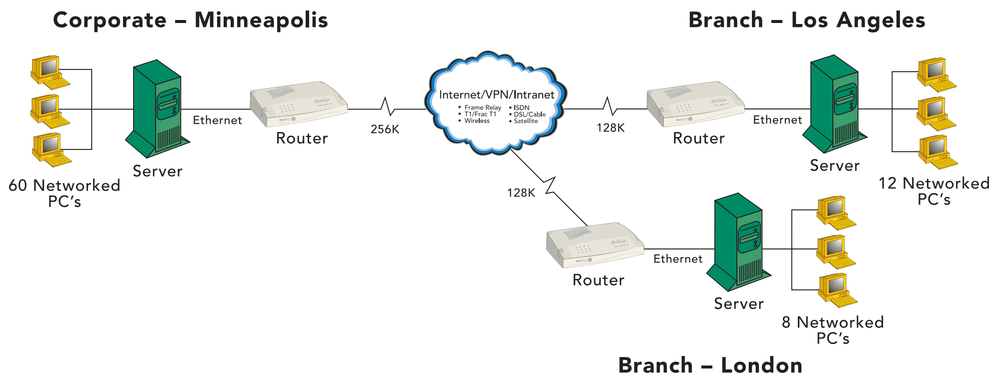
- It ensures that information doesn't go where it's not needed. This is crucial for keeping large volumes of data from clogging the connection.
- It makes sure that information does make it to the intended destination.

In performing these two jobs, a router is extremely useful in dealing with two separate computer networks. It joins the two networks, passing information from one to the other. It also protects the networks from one another, preventing the traffic, on one, from unnecessarily spilling over to the other. Regardless of how many networks are attached, the basic operation and function of the router remains the same. Since the Internet is one huge network made up of tens of thousands of smaller networks, its use of routers is an absolute necessity.

In order to route data through a network, routers need a way to locate each other. Therefore, every device on the Internet has a unique identifying number, called an IP Address. A typical IP address looks like this: 200.2.9.1. An Internet Service Provider (ISP), or network administrator, permanently or dynamically assigns an IP address to a network device.

Using our company example, the following diagram maps out their existing data network.

Company Data Network



Local Area Versus Wide Area Networks

We can classify IP data network technologies as belonging to one of two basic groups: Local Area Networks (LANs) or Wide Area Networks (WANs). A LAN connects many devices that are relatively close to each other, usually in the same building. A WAN connects a smaller number of devices that can be many miles apart. Different transmission facilities can be used in a WAN to support remote operations — everything from digital connections (e.g. ISDN, cable and DSL) to dedicated T1/E1 and frame relay connections. This is one of the reasons that IP data networks offer so much flexibility and cost-effectiveness in reaching all types of remote locations and workers. See the chart below for a bandwidth comparison of various WAN connection types.

WAN Connection Types

Bandwidth	ISDN	Digital (T1/E1/Frame Relay)	Cable	DSL
Max Download Speed	128K	1.54M bps/ 2.0M bps	1.5M bps or >	1.5M bps
Max Upload Speed	128K	1.54M bps/ 2.0M bps	256K bps or >	1.5M bps

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Using the Internet to Extend the Network

Today, instead of simply dealing with local or regional concerns, many businesses now have to think about global markets and logistics. Many companies have facilities located across the country or even around the world. And, they all need a way to maintain fast, secure and reliable communications wherever their offices are.

Before the Internet, this meant using leased lines to maintain a private Wide Area Network between the offices. This private WAN has obvious advantages over a public network, like the Internet, when it comes to reliability, performance and security. But maintaining a WAN, particularly when using leased lines, can become quite expensive, rising in cost as the distance between the offices increases.

As the popularity of the Internet grew, businesses turned to it as a means of extending their own networks. First came intranets, which are password-protected sites designed for use only by company employees. Today, many companies are creating their own intranet-based VPNs (Virtual Private Networks) to accommodate the needs of remote employees and distant offices. A VPN is a private network that utilizes dedicated equipment and data encryption to securely connect remote sites or users together over the public Internet. Now, fast, secure, reliable, and cost-effective data communications are a reality for branch offices, telecommuters and road warriors.

The Opportunity - Unprecedented Connectivity Options

In the past, there have been many attempts to merge voice and data networks, but it wasn't until the Internet revolution and the widespread deployment of IP data networks that the industry at large finally had the right transport mechanism to support voice **and** data. Having a universal language that virtually all worldwide networks can understand has opened up unprecedented connectivity options now available to visionary telecommunications managers.

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Understanding the Converged Network

Voice over IP

Voice over IP uses the data network packet-switching method to provide a more efficient way of sending voice communication. Packet-switching optimizes the use of network resources (bandwidth) because the channel is only occupied during the time the packet is being transmitted. Many users can share the same channel because individual packets can be sent and received in any order and the network can balance the load across various pieces of equipment. This allows several telephone calls to occupy the amount of space occupied by only one in a circuit-switched network. By migrating telephone networks to packet-switching technology they immediately gain the ability to communicate more efficiently the way computers do.

In the IP world, voice is another data application running over the IP network. In a converged environment, the PBX becomes the equivalent of a super-server (like a mainframe) that sits on the network and is accessed by remote clients (e.g. handsets or even PCs, using converged applications) anywhere on the network over any type of transmission lines. Therefore, Voice over IP solves the PBX's networking limitations by providing a cost-effective, efficient means of communicating over the company's existing data network, or the Internet. Voice over IP gateways, operating alongside the company's PBX, make it possible to maintain all existing systems and simply extend voice and the PBX's features and functionality out to remote locations and home users. It can seamlessly tie together dissimilar proprietary PBX systems and provide networking capabilities to key telephone systems that previously weren't available.

Voice over IP Gateways

The device that bridges the voice network and the data network together is called a Voice over IP (VOIP) gateway. A VOIP gateway connects directly to an existing voice network and plugs into an IP data network. It uses the network's router to access the Internet or a private intranet. A VOIP gateway is a point-to-point, or point-to-multipoint, solution (one is required for each location). It merges voice/fax from telephones onto the IP network and then utilizes another VOIP gateway, at the remote end, to separate the voice/fax from the data network and send it back to the PBX, telephone, or fax machine.

Voice/Fax "Ride Free" on the Data Network

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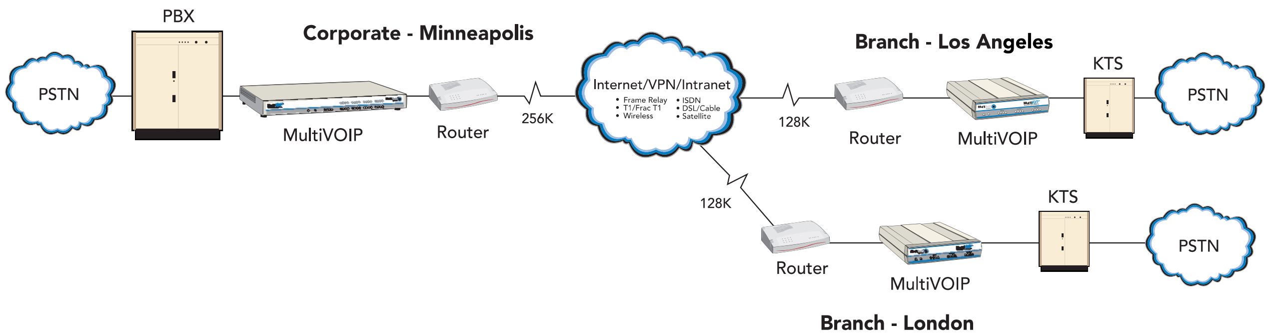
The MultiVOIP Gateway

The MultiVOIP family of Voice over IP gateways is available in analog and digital models ranging from one to 60 ports. MultiVOIP gateways connect directly to phones, fax machines, key systems, or a PBX and plug into the data network to provide real-time, toll-quality voice connections to any office on your VOIP network. With MultiVOIP, your customers will avoid the hassle and expense of replacing their existing routers, WAN connections or phone systems required by other VOIP solutions. It supports industry-standard protocols to ensure optimum interoperability and voice quality. And, users are not required to master new operating procedures. In fact, once configured, a MultiVOIP network can be used immediately and companies can start saving money.

In our company example, the following diagram maps out their converged network using MultiVOIP Voice over IP gateways.

Converged Network



How Does it Work?

Because VOIP gateways are point-to-point solutions, they first need to establish the call setup. This process requires the caller to access the local MultiVOIP gateway and then dial a telephone number/extension to reach the remote MultiVOIP gateway. A call processing technique maps this dialed number into an IP address for delivery to the remote VOIP gateway.

Next, in order to merge the voice call onto the IP data network, it must be encapsulated into an IP packet. To encapsulate voice, the call is broken up into frames. The size of the frame is defined by the voice compression used (see bandwidth requirements on pg. 8 for more details on voice compression).

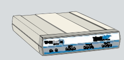
Then, several frames of voice data are collected. A protocol header is added to the beginning of the voice data to indicate the destination address and data type. Finally, the packet is queued for transmission. This entire process of voice packetization takes only 10 to 20 milliseconds (ms), making it undetectable to the human ear.

How a VOIP Network Works

Phone Equipment:

- *Uses standard dialing to make a call*
- *Routes the call to a channel on the VOIP gateway*

VOIP Gateway:



- *Establishes call setup*
- *Packetizes voice and telephone signaling*
- *Applies audio compression*

Router:

- *Routes packets over the Internet/ intranet to the remote office VOIP gateway*

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Voice over IP Frequently Asked Questions

The following section covers some of the frequently asked questions a data communications manager or telecommunications manager may have as they begin to learn how a VOIP solution can solve their telephony challenges and save them money.

Bandwidth Requirements

"How can I be assured that my data "pipe" will not be flooded by voice traffic and negatively impact the timely delivery of data services?"

One common misconception about VOIP is that it is a bandwidth hog. In reality, with the use of voice compression, voice is a very efficient type of traffic. A vocoder (voice encoder/decoder) provides multiple voice compression standards which range from G.723 (5.3K bps/6.3K bps) to G.729 (8K bps) to G.711 (full, uncompressed 64K bps) and can be selected on a system or a per port basis. This allows the administrator to minimize network bandwidth requirements or maximize voice quality on an office-by-office or user-by-user basis. With MultiVOIP, the majority of applications are optimally configured for voice quality with minimal bandwidth requirements by simply using the factory defaults for voice compression.

As a rule of thumb, 14K bps of bandwidth per call is ideal. This includes the compressed voice packet and the IP overhead. To determine total VOIP bandwidth needed per location, take the number of VOIP ports or channels being utilized and multiply by 14K (ideal bandwidth). Then double this number, to accommodate for both voice and data traffic, to get the total bandwidth required for optimum voice quality.

Company Example: Los Angeles branch office is using 2-ports
2 x 14K = 28K x 2 = 56K bps minimum bandwidth

Using the formula, the company needs a minimum bandwidth of 56K bps. Since their data network already has a 128K connection, bandwidth will not be an issue (see diagram on p. 4).

It should also be noted that bandwidth is used only when someone is speaking. With MultiVOIP, a silence suppression/Voice Activation Detection (VAD) feature is an option that frees unused call bandwidth for data traffic. This is significant, since callers are usually silent for 60 percent of the call.

Voice Quality

"I'm not yet convinced that Voice over IP can deliver business quality voice."

Independent tests of VOIP systems have shown that they are perfectly capable of delivering "toll-quality" voice. Earlier implementations were criticized for excessive noise and other quality of service issues. Today, better algorithms, quicker voice compression, and the availability of high-speed communication links have all made VOIP implementations a viable technology.

The actual voice quality is affected by a number of factors: WAN bandwidth (the higher the better), voice compression (as discussed previously) and network conditions including latency, jitter and packet loss.

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Latency is defined as the average "travel" time it takes for a packet to pass through the network, from source to destination. The average time varies according to the amount of traffic being transmitted and the bandwidth available at that given moment. If the traffic is greater than the bandwidth available, packet delivery will be delayed.

MultiVOIP deals with the latency issue in a private network as well as over the public Internet. In a private network, when network traffic is at peak levels, voice can be given priority over data to ensure consistently high voice quality using the Differentiated Services (DiffServ) Quality of Service (QoS) protocol. This is an end-to-end requirement, which means it must be supported at various points on the network in order for the voice traffic to receive the proper priority from every device it encounters. Another way to enforce Quality of Service is to use the Resource Reservation Setup Protocol (RSVP). RSVP-enabled routers set aside bandwidth along the route from source to destination based on the IP addresses associated to the MultiVOIP gateways.

Optimum Latency Thresholds and Voice Quality:

Up to 150 ms = excellent
150 - 250 ms = good
250 - 350 ms = usually acceptable
> 350 ms = depends on application

When running Voice over IP on the public Internet, the issue of latency cannot be controlled due to the ever-changing path and router hops that your voice packet may take before it reaches its destination. However, the MultiVOIP gateway does a good job of not adding any additional latency through the box itself. Therefore, if you have a good Internet service provider, and they are able to provide you with a quality of service guarantee, you should be able to manage any latency you may encounter.

If you have concerns about latency on your network, or the public Internet, use the above threshold chart to determine its possible affect on your voice quality.

Jitter is defined as the variability in packet arrival at the destination. Voice packets must compete with non real-time data traffic, therefore, if there are bursts of traffic on the network, they can result in varied arrival times. When consecutive voice packets arrive at irregular intervals, the result is a distortion in the sound, which if severe, can make the speaker unintelligible.

The MultiVOIP gateway utilizes a Dynamic Jitter Buffer to collect voice packets from the IP network, store them, and shift them to the voice processor in evenly spaced intervals. During high latency periods, the jitter buffer size is dynamically increased to receive delayed voice packets. During low latency periods, the jitter buffer is dynamically decreased to minimize the end-to-end voice delay.

Packet loss is the percentage of undelivered packets in the data network. When data packets are lost, a receiving computer can simply request a retransmission. When voice packets are lost, or arrive too late, they are discarded instead of retransmitted. The result is disconcerning gaps in the conversation (like a poor cell phone conversation).

The MultiVOIP gateway utilizes Forward Error Correction to increase voice quality by recovering lost or corrupted packets. The current Forward Error Correction implementation can recover one of two consecutive lost/corrupted packets or every other lost/corrupted packet, thereby eliminating any noticeable voice degradation.

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MultiVOIP also utilizes Bad Frame Interpolation to increase voice quality by making the voice transmission more robust in bursty error environments. It interpolates lost/corrupted packets by using the previously received voice frames. Interpolation of one or two voice packets will not cause a noticeable degradation in voice quality. Typically, Bad Frame Interpolation is invoked if Forward Error Correction cannot recover the lost/corrupted packets. By utilizing both Forward Error Correction and Bad Frame Interpolation, MultiVOIP is continually optimizing voice quality regardless of the conditions.

Security

"Will adding Voice over IP affect the security of my existing data network?"

On a private network, security is not an issue because the network is private to outside intruders. If the VOIP connection is over the Internet, or through a VPN connection, the network security will not change. MultiVOIP does not interfere or change the way the current data security is set up.

Standards

"Will the MultiVOIP gateway talk to other VOIP solutions?"

At the application level, standards for Voice over IP interoperability are still evolving. The H.323 standard is the one most widely deployed and is the only approved protocol adopted by the International Telecommunications Union (ITU). It is an umbrella standard that specifies the components, protocols and procedures providing multimedia communication over packet-based networks.

Another emerging standard, developed by the Internet Engineering Task Force (IETF), is the Session Initiation Protocol (SIP). This protocol, designed specifically for VOIP applications, is gaining popularity in the area of IP phones and soft phones (MS Messenger) because of its simplicity.

MultiVOIP utilizes both the H.323 and SIP protocols to provide complete interoperability with other Internet telephony solutions. The inbound IP call protocol is automatically detected and the voice channel is dynamically configured to match. The outbound IP call protocol is configured with the phone number allowing you the flexibility to call H.323 or SIP devices from the same port.

Reliability

"Can you assure me MultiVOIP is going to work all of the time?"

Telecommunications managers have been accustomed to delivering a 99.999% reliable service. Because the MultiVOIP gateway works with the existing phone system, there is little risk in deployment. If the data network should go down, or if all the VOIP channels were busy, the user can always revert automatically or manually to the standard PSTN to make the call.

"What happens if my LAN/WAN goes down?"

MultiVOIP utilizes a feature called PSTN fail-over that allows it to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heightens reliability and augments QoS when conditions threaten to undermine voice quality. Utilizing user definable controls, MultiVOIP continually checks if the LAN/WAN is threatened by packet loss, jitter or latency, or to see if the network is completely down. If it detects a problem, MultiVOIP switches to "survivability mode" transparently routing all calls over PSTN lines connected to the MultiVOIP gateway. MultiVOIP continues to monitor the connection and automatically switches back to the LAN/WAN once the conditions improve.

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Ease of Use

"Will my users require extensive training to use the MultiVOIP system?"

No, placing calls with MultiVOIP is like using your existing phone system. It uses single-stage dialing by utilizing a Uniform Dialing Plan that is consistent with the E.164 (PSTN) standard numbering plan. This includes automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make VOIP calls.

Networking Dissimilar Proprietary PBX Systems

"Will MultiVOIP work when networking dissimilar proprietary PBX phone systems?"

Yes, as long as the PBX has analog extension ports, CO ports, E&M ports, or T1/E1/PRI cards available, or the ability to add cards with the appropriate interface. There is nothing proprietary about an analog or digital interface. This is the benefit of utilizing the MultiVOIP gateway. It simply bridges the two systems together.

Supplementary Services

"Does the MultiVOIP support PBX-like features such as call transfer, call forwarding and call hold?"

Yes, MultiVOIP supports H.450 supplementary services to provide for call transfer, call forwarding, call hold, call waiting, and name identification. It also supports Q.SIG, an inter-PBX signaling protocol, for networking PBX supplementary services in a multi- or uni-vendor environment. In addition, MultiVOIP supports SIP extensions providing call forward and call transfer capabilities.

Management

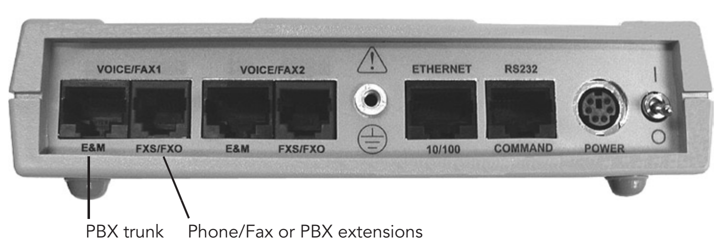
"Can I manage my MultiVOIP gateways from a central location?"

Yes, the MultiVOIP gateway is easily managed locally using a windows-based software application or remotely by the central office with a web browser or SNMP. Multi-Tech also includes its own SNMP management software called MultiVOIPManager, which provides central site configuration, management and call monitoring for all MultiVOIP gateways on the network. It utilizes a Windows interface that makes it easy to view events like usage tracking, live use reporting, call history, and voice quality statistics. In addition, MultiVOIPManager eases administration by automatically e-mailing call logs based on volume or time.

Plugging into the Voice and Data Network

"How does MultiVOIP plug into my existing voice and data network?"

For maximum investment protection, the MultiVOIP 2-, 4- and 8-port models accommodate changing communication needs by providing a programmable FXS/FXO and E&M interface for each port. This means you don't have to worry about ordering the right interface to connect directly to the customer's phones, fax machines, key phone systems or PBX system. On the digital MultiVOIP model, an industry standard RJ-45 jack is provided to connect directly to either a digital port on the PBX or directly to a T1/E1 or PRI line. On the data network side, the MultiVOIP gateway simply plugs into the Ethernet network.



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Port Configuration

"How do I determine the number of ports I need and which MultiVOIP to order?"

You do not need a port for every telephone on the PBX system. You simply need to determine the calling ratio to determine how many ports you need at each location. The following guidelines can help you:

1) If you are replacing tie/trunk lines, for every line that you support you need one port on the MultiVOIP.

   Ex. 4 Tie Lines = 4-port MultiVOIP (MVP410)

2) If you are not using tie/trunk lines, you can determine the ratio based on the location's long distance communication bills. First, determine what percentage of the bill is used for intra-office communication (typically between 25% to 40%), then multiply the percentage by the number of available PSTN lines at the location. The result will determine the minimum number of ports needed.

   Ex. Minneapolis, Corporate:
   25% is intra-office calling. They have 16 lines.
   25% x 16 = 4
   Recommendation: 4-port MultiVOIP (MVP410)

   Los Angeles Branch office:
   30% is intra-office calling. They have 5 lines.
   30% x 5 = 1.5
   Recommendation: 2-port MultiVOIP (MVP210)

   London Branch office:
   40% is intra-office calling. They have 4 lines.
   40% x 4 = 1.6
   Recommendation: 2-port MultiVOIP (MVP210)

Model	Description
MVP130	1-Port VOIP Gateway
MVP210	2-Port VOIP Gateway
MVP210-G	2-Port VOIP Gateway/Gatekeeper
MVP410	4-Port VOIP Gateway
MVP410-G	4-Port VOIP Gateway/Gatekeeper
MVP810	8-Port VOIP Gateway
MVP810-G	8-Port VOIP Gateway/Gatekeeper
MVP2410	24/48-Port T1/PRI VOIP Gateway
MVP3010	30/60-Port E1/PRI VOIP Gateway

If you do not know what percentage of the phone bill is being used for intra-office communication, the rule of thumb is 30%.

If you need more than 16 ports, we recommend the digital MultiVOIP (MVP2410 or MVP3010).

For a worksheet designed to help you calculate your customer's bandwidth and port configuration needs, reference our Configuration Guide located in the back of this primer.

Gatekeeper Models

"When do I use a MultiVOIP Gateway/Gatekeeper (-G Model)?"

The MultiVOIP Gateway/Gatekeeper (-G Model) includes an integrated gatekeeper to facilitate call management in a Voice over IP network. These cost-effective MultiVOIP gateways provide centralized phone book management as well as deliver the power to define and control how H.323 voice traffic is managed over IP networks. With the integrated gatekeeper, network managers can configure, monitor and manage the activity of registered end points. In addition, they can set policies and control network resources, such as bandwidth usage, to ensure optimal implementation.

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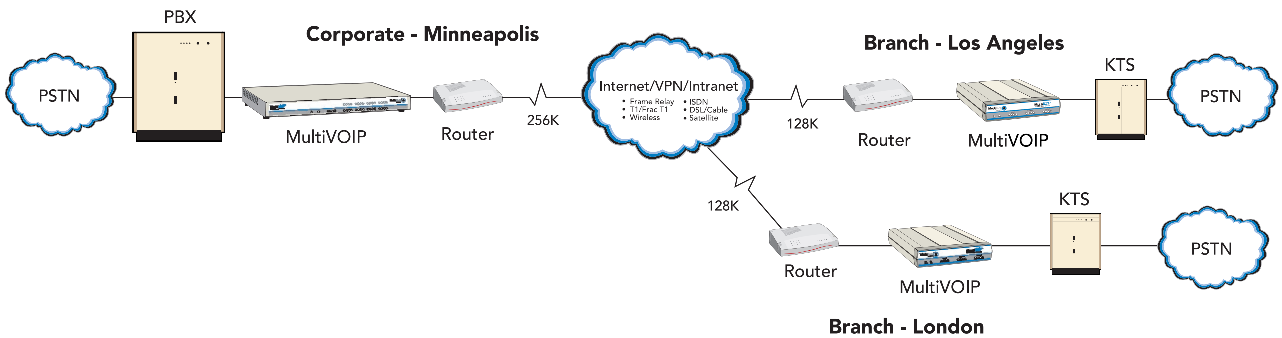
Voice over IP Applications

MultiVOIP is ideal for multi-location businesses looking to reduce toll charges associated with intra-office calling. It is designed to help a company maximize investments they've already made in their data network infrastructure and voice equipment. Using our company example, the following are some of the many applications for a VOIP network:

Office-to-Office Communication

A VOIP network can be as small as two offices or as large as hundreds of offices. Each office installs and configures a VOIP solution on their network to begin placing calls or sending faxes to other offices on the VOIP network. This allows a company to extend its telecommunications network to remote offices without the expense of replacing the phone system at each location. Our company example shows a typical three office VOIP network.

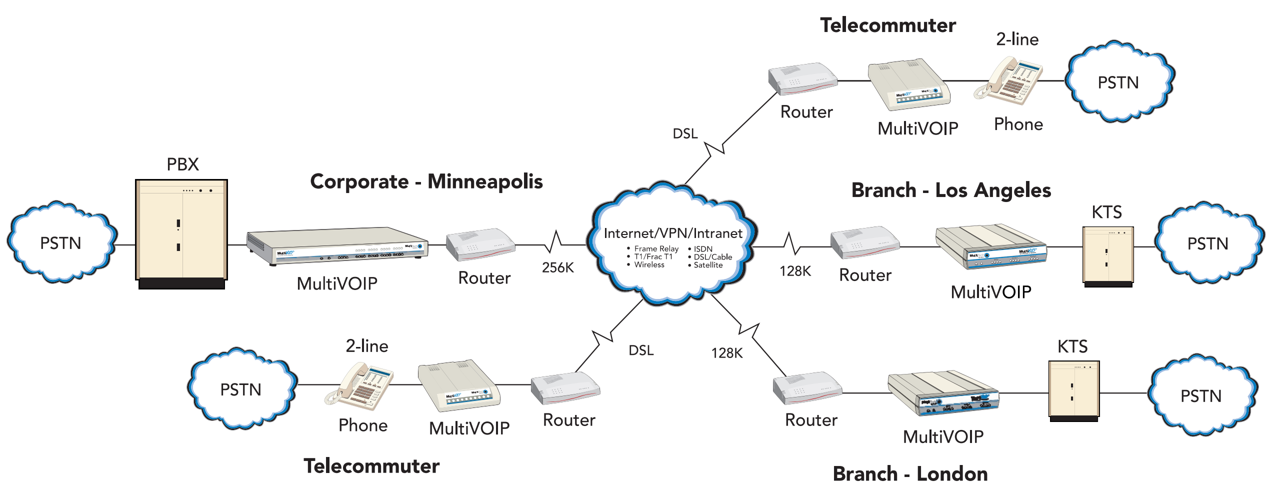
Office-to-Office



Create Off-Premise Extensions for Telecommuters

Extend the reach of a customer's PBX into home office locations. Simply connect a VOIP solution to the PBX at the corporate office, and another VOIP solution at the home office. Now, anyone can place calls to the home office by dialing an extension number. And, the home office can dial others on the VOIP network without incurring long distance charges. In our company example, we have now added two telecommuters to the VOIP network.

Off-Premise Extensions



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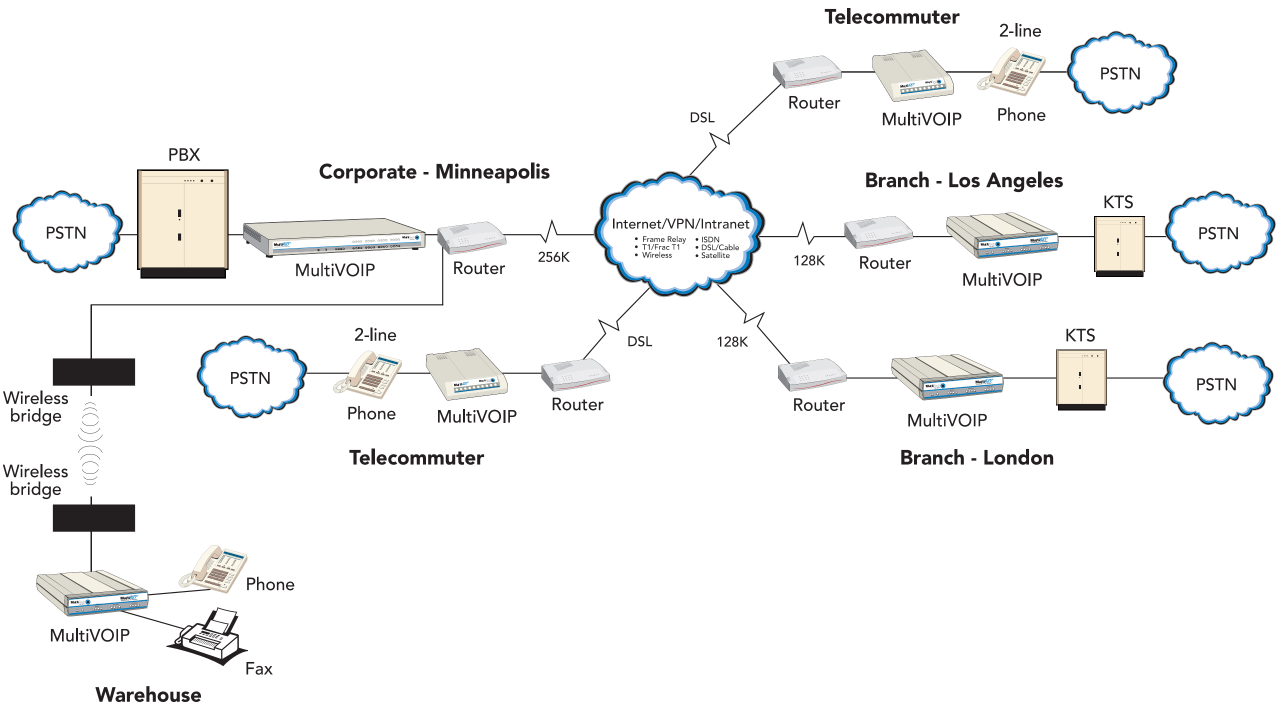
Wireless Connections

To extend a customer's PBX to a building across the street, utilize a wireless bridge to connect the two networks. Now, your customer has voice and data connectivity without laying cables or paying monthly charges for dedicated lines. Building off our example VOIP network, we have now added a wireless connection to the company's warehouse.

Wireless Building-to-Building



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Configuring a MultiVOIP Network

Now that you have a basic understanding of Voice over IP, the MultiVOIP gateway, and its applications, the next step is to learn how easy it is to configure the solution around existing telephone and data networks.

Configuring the Telephony Interface

Let's first discuss how to configure the MultiVOIP gateway to the various telephony options. We will look at both the analog and digital MultiVOIP solution.

Configuring an Analog MultiVOIP:

The MultiVOIP gateway is equipped to support one of three voice-port signaling types*:

1. FXS (foreign exchange station) interface: connects directly to phones, faxes, and CO ports on PBXs or key telephone systems (KTS)
2. FXO (foreign exchange office) interface: connects directly to an analog PBX extension, PSTN, or KTS extensions
3. E&M interface (Ear and Mouth): connects directly to analog PBX trunk ports

* MVP130 supports FXS and FXO only.

The type of phone equipment that you use to connect to the MultiVOIP gateway will determine which interface port you will use:

Device Type	FXS	FXO	E&M	T1/E1/PRI
Analog phone/fax machine	X			
Multiline (2- or 4-line) phone	X			
Key Telephone System (KTS)	X	X		
PBX	X	X	X	X
PSTN		X		X

You will note that with a key telephone system and a PBX, you have a couple of interface options. The interface that you use will create a different path and dialing procedures for the Voice over IP call. In general, MultiVOIP does not modify the behavior of the telephone equipment. It simply provides a connection to the IP data network instead of the PSTN and passes along the equipment's features and functionality.

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Key Telephone System and PBX Interface Options:

1. **FXS interface -** Use this interface when connecting the line side (CO port) of your key telephone system or PBX to the MultiVOIP gateway. The system will act similarly to connecting directly to a CO line. Here's how it works:

   Outgoing calls: The caller needs to access the MultiVOIP gateway in order to receive a dial tone. An easy approach would be to associate a button on the phone system that will access the MultiVOIP gateway. At this point, MultiVOIP will generate a dial tone and the caller can call any other location on the network that is also equipped with a MultiVOIP gateway.

   Incoming calls: MultiVOIP will generate the ring signal, just like an outside CO line. At this point, you can use the features of the telephone system to determine how to handle the call. For example, on a key telephone system all of the phones may be programmed to light up. On a more advanced phone system, you can route the incoming call to a receptionist or to an Interactive Voice Response (IVR) system.

2. **FXO interface -** Use this interface when connecting the station side (extension port) of your telephone system to the MultiVOIP gateway.

   Outgoing calls: The caller will need to access the extension in order to receive a dial tone from the MultiVOIP gateway. This can be done by simply dialing the extension number that the MultiVOIP is connected to, or by preprogramming a button to access the extension. At this point, the MultiVOIP gateway will generate a second dial tone. Now, the caller can call any other location on the network that is also equipped with a MultiVOIP.

   Incoming calls: The MultiVOIP gateway will utilize an extension on the phone system. At this point, the caller will hear an extension dial tone generated by the remote phone system. The caller can then complete the call by dialing any other extension on the phone system. An easy approach to eliminate the need for this "two stage" dialing would be to use the phone system's ability to route the incoming call immediately to a reception desk or IVR system.

3. **E&M interface -** E&M is the preferred interface, on a PBX system, because it provides the most reliable and quickest disconnect among calls allowing the MultiVOIP to re-establish the port for availability. It is also the interface used when you are using the MultiVOIP solution to replace tie lines. For outgoing and incoming calls, the dialing behavior is the same as the FXO interface (as described above).

All channels on the MultiVOIP gateway do not have to be configured the same way. For example, one channel could be connected to an extension line off the PBX (using the FXO interface), and the other channel connected directly to a fax machine (using the FXS interface).

Digital MultiVOIP Configuration:

The digital MultiVOIP uses a T1, E1 or PRI interface to connect to a PBX system. For outgoing and incoming calls, the dialing behavior is the same as the FXO interface (as described above). Multi-Tech recommends utilizing a digital MultiVOIP solution when you are connecting 16 or more lines.

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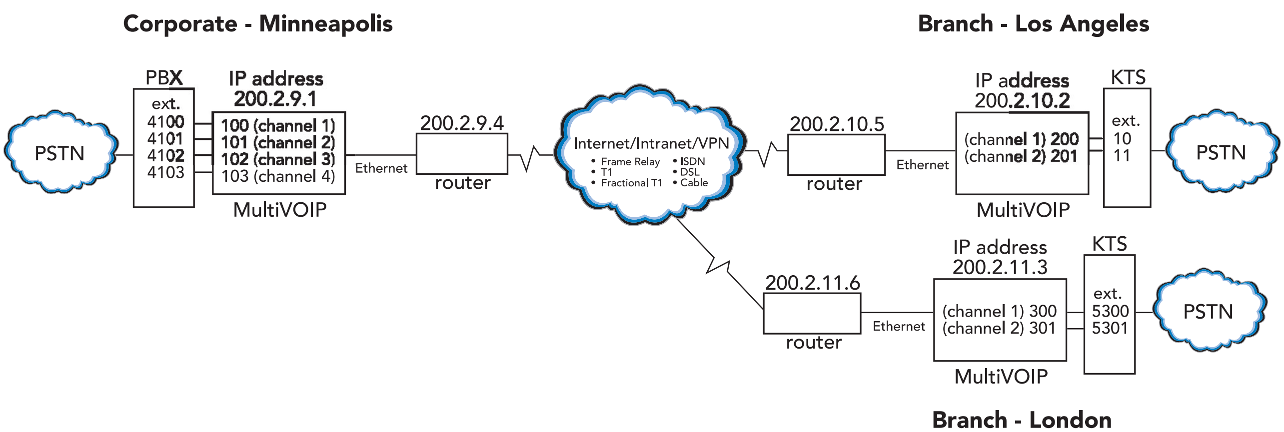
Building the VOIP Dialing Plan

MultiVOIP provides single stage dialing by utilizing a Uniform Dialing Plan that is consistent with both the PBX and the E.164 (PSTN) standard numbering plan. This means placing calls is like using your existing phone system, no user training is needed.

In order to accomplish this, MultiVOIP utilizes an Inbound and Outbound phonebook. In our company example, the system allows employees in any office to dial employees in any other office using only three digits. The following is a worksheet used to layout this company's dialing plan.

Outbound Phone Book - Minneapolis		
Destination Pattern	IP Address	Description
200	200.2.10.2	Outbound Call to Ch. 1 Los Angeles
201	200.2.10.2	Outbound Call to Ch. 2 Los Angeles
300	200.2.11.3	Outbound Call to Ch. 1 London
301	200.2.11.3	Outbound Call to Ch. 2 London
Outbound Phone Book - Los Angeles		
100	200.2.9.1	Outbound Call to Ch. 1 Minneapolis
101	200.2.9.1	Outbound Call to Ch. 2 Minneapolis
102	200.2.9.1	Outbound Call to Ch. 3 Minneapolis
103	200.2.9.1	Outbound Call to Ch. 4 Minneapolis
300	200.2.11.3	Outbound Call to Ch. 1 London
301	200.2.11.3	Outbound Call to Ch. 2 London
Outbound Phone Book - London		
100	200.2.9.1	Outbound Call to Ch. 1 Minneapolis
101	200.2.9.1	Outbound Call to Ch. 2 Minneapolis
102	200.2.9.1	Outbound Call to Ch. 3 Minneapolis
103	200.2.9.1	Outbound Call to Ch. 4 Minneapolis
200	200.2.10.2	Outbound Call to Ch. 1 Los Angeles
201	200.2.10.2	Outbound Call to Ch. 2 Los Angeles

Inbound Phone Book - Minneapolis	
Prefix to Remove	Description
100	Inbound Call to Ch. 1
101	Inbound Call to Ch. 2
102	Inbound Call to Ch. 3
103	Inbound Call to Ch. 4
Inbound Phone Book - Los Angeles	
200	Inbound Call to Ch. 1
201	Inbound Call to Ch. 2
-	-
-	-
-	-
-	-
Inbound Phone Book - London	
300	Inbound Call to Ch. 1
301	Inbound Call to Ch. 2
-	-
-	-
-	-
-	-



A more sophisticated phonebook setup can be created that requires an "intelligent" PBX and the MultiVOIP gateway to automatically append and strip digits. So, for example, when an employee dials another remote company location using a standard 12-digit number, the PBX would know to hand that call to the MultiVOIP gateway. MultiVOIP, in turn, would strip the unnecessary 5-digit destination pattern (9+1+area code) and direct the call to the IP address of the remote MultiVOIP. For details on this type of dialing plan, refer to the user's guide.

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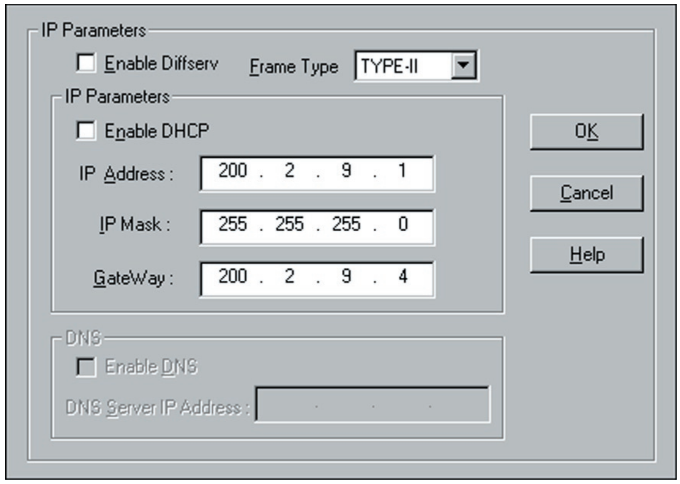
When configuring the MultiVOIP gateway for the network, you must identify, within the software, its IP address, subnet mask, and gateway address. The IP address is your unique LAN IP address. The subnet mask is the number that identifies the sub network to which your MultiVOIP is connected. The gateway address is the IP address of the device connecting your MultiVOIP to the Internet/intranet. Each MultiVOIP on the network will need to be configured this way so that the phone directory can be mapped to the IP address of the individual MultiVOIPs on the network.



Deploying the VOIP Network

The VOIP administrator can take each individual MultiVOIP and pre-configure it before sending it to the remote sites. The remote site administrators need only connect power to the pre-configured MultiVOIPs, and connect them to the Ethernet LAN and predefined telephone equipment. At this time, the VOIP network will be fully operational.

Advanced Feature Configuration

The MultiVOIP software provides a number of advanced features that can be configured to enhance voice quality. Our recommendation is to use the factory defaults. Most users find these are more than adequate for the application.

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Multi-Tech Pre-Sales and Post-Sales Support

At Multi-Tech, we believe our resellers are truly an extension of our sales force. Therefore, we've put together a Voice over IP support program to provide you with both pre-sales and post-sales support.

Optimum Reseller Program

Our pre-sales support starts with our Optimum Reseller program. Once you sign on, you will receive a complete sales kit on Multi-Tech solutions. This kit includes contact information, product brochures, and other sales tools. In addition, as a registered Multi-Tech Optimum Reseller, you will receive the following:

- **Free Pre-Sales Support -** unlimited access to our Inside Sales Specialists, your first line of support to help you win business. Your second line of support includes access to our Product Support Specialists who can assist you with your product configuration needs.
- **Free Technical Training -** an invitation to our traveling roadshow seminar series in which you'll receive detailed sales and technical training on Voice over IP, VPN and other Multi-Tech solutions.
- **Exclusive Promotions -** offered monthly, these special promotions are designed to help you profit by selling more Multi-Tech solutions.
- **Demo Equipment -** You'll be eligible to purchase "not-for-resale" (NFR) demo equipment.
- **On-going Communication -** Our business is about communication so we believe that keeping you abreast of the latest advances in technology is key to a successful reseller partnership.
- **Free Marketing Materials -** customize promotional direct mail, faxblasts, and e-mail blasts with your logo and contact information. These marketing materials, available free of charge, are available for promoting the MultiVOIP and RouteFinder VPN solutions to your customer base.

To join the Optimum Reseller Program, simply fill out an application on our web site at: www.multitech.com/PROGRAMS/Opt_App.asp

MultiVOIP Marketing Literature

Multi-Tech has designed a detailed product brochure and an educational VOIP technology guide for your customers. Both are available on our web site, or can be ordered, free of charge, for customer mailings, trade shows, or other promotional activities. Just call 1-888-288-5470 (U.S. & Canada) or 763-785-3500 and ask for our reseller marketing specialist.

- MultiVOIP brochure (#86002030) - an overview of the product features and benefits.
- Voice over IP Technology Guide (#86000461) - an introduction to Voice over IP.

Technical Support

Multi-Tech provides FREE, toll-free, post-sale, technical support for the product. Our technical support team can be reached by calling: 1-800-972-2439 (U.S. & Canada) or 763-785-3500.

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Warranty and Overnight Replacement Service

The MultiVOIP product warranty is two years. In addition to our warranty, we offer an Overnight Replacement Service to eliminate concerns of downtime on the VOIP network*. The Overnight Replacement Service provides the following benefits:

- Maximizes equipment reliability
- Streamlines problem resolution
- Includes all overnight shipping charges
- One-time fee
- 2-year coverage

* For U.S. customers only.

For more information, visit our web site at www.multitech.com/programs/orc.

In summary

We hope you found that this primer addressed your basic questions regarding Voice over IP and Multi-Tech's MultiVOIP gateway solution. We feel strongly that Voice over IP can help you differentiate your business and give you a competitive advantage. But don't take our word for it, you be the judge. Call our toll-free demo line at **1-877-TRYVOIP** and bring convergence to your customers today.

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VOIP Glossary of Terms

Bad Frame Interpolation - Interpolates lost/corrupted packets by using the previously received voice frames. It increases voice quality by making the voice transmission more robust in bursty error environments.

Bandwidth - The transmission capacity of a communications line. It is a factor in determining the amount of information and the speed at which a medium can transmit data or voice.

Bps (bits per second) - A unit to measure the speed at which data bits can be transmitted or received.

Cable Connections - Cable modems allow a PC or networked computer to transmit and receive data over a cable TV network (CATV). Because existing CATV networks already employ high-bandwidth coaxial cable into the home or office, these modems are much faster than dial-up analog modems offering speeds from 3 to 10M bps.

Central Office (CO) - The lowest, or most basic level of switching in the PSTN network. A business PBX or any residential phone connects to the PSTN at a central office.

Circuit-switched Network - A technology used by the PSTN that allocates a pair of conductors for the exclusive use of one communication path. Circuit switching provides a temporary connection of two or more communications channels using a fixed, non-shareable path through the network. Users have full use of the circuit until the connection is terminated.

CODEC - Coder-decoder compression scheme or technique. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

Compression - Used at anywhere from 1:1 to 12:1 ratios in VOIP applications to consume less bandwidth and leave more for data or other voice/fax communications. The voice quality may decrease with increased compression ratios.

DiffServ (Differentiated Services) - Is a quality of service protocol that prioritizes IP voice traffic to help preserve voice quality even when network traffic is heavy.

DSL (Digital Subscriber Line) - A technology that allows a provider to use the excess bandwidth found in a copper line for the provision of data services. Its maximum download speed is 1.5M bps.

E&M (Ear and Mouth) - The interface on a VOIP device that allows it to be connected to analog PBX trunk ports (tie lines).

Echo Cancellation - The elimination of an echo in a two-way transmisison.

Ethernet - A 10-megabit/100-megabit baseband local area network that allows multiple stations to access the transmission medium at will without prior coordination.

Forward Error Correction - Increases voice quality by recovering lost or corrupted packets.

Frame - A group of data bits in a specific format to help network equipment recognize what the bits mean and how to process them. The bits are sent serially, with a flag at each end, signifying the start and end of the frame.

Frame Relay - A fast-packet data communications standard that allows a network to carry data frames in packets of varying length; usually used to connect LANs or for LAN-to-WAN connections. They are protocol independent making it a less expensive, high-speed network.

FXO (foreign exchange office) - The interface on a VOIP device for connecting to an analog PBX extension.

FXS (foreign exchange station) - The interface on a VOIP device for connecting directly to phones, fax machines, and CO ports on PBXs or key telephone systems.

H.323 - An industry-standard call setup protocol designed to standardize VOIP communications between other H.323 telephony solutions.

ITU (International Telecommunications Union) - A civil international organization established to promote standardized telecommunications on a worldwide basis.

Internet - Refers to the computer network of many millions of university, government and private users around the world. Each user has a unique Internet address (IP address).

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Internet Protocol (IP) - A protocol used to route data from its source to its destination in an Internet environment. It is a highly distributed protocol (each machine only worries about sending data to the next step in the route).

IP address (or Internet Address) - A 32-bit address used by IP data networks to uniquely identify the location of a device on a network. Normally printed in dotted decimal format (e.g. 129.128.44.227).

IP Gateway - A network device that converts voice and fax calls, in real time, between the PSTN and an IP network.

IP Gatekeeper - An H.323 entity that defines the policies that govern the multimedia system (e.g. dialing plans, user privileges, bandwidth consumption, etc.). It also provides the means to extract information from such a system for billing or other purposes.

ISDN (Integrated Services Digital Network) - Provides a digital telephone service which allows both data and voice communication over the same telephone line and at significantly faster speeds than the traditional Plain Old Telephone Service or analog service. There are two types of lines which provide access to ISDN, Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI provides two bearer or B channels and one signaling or D channel. PRI provides 23 B channels and one D channel in the U.S. and 30 B channels and one D channel in Europe.

Jitter - The variability in packet arrival at the destination. When consecutive voice packets arrive at irregular intervals, the result is a distortion in sound, which if severe, can make the speaker unintelligible.

Key Telephone System (KTS) - Phone devices with multiple buttons that let you select incoming or outgoing CO phone lines directly. Similar to a PBX, except with a KTS you don't have to dial a "9" for a call outside the building.

LAN (Local Area Network) - Two or more computers linked together in a contained location; such as an office building, allowing users to share files and access to printers.

Latency - Average "travel" time it takes for a packet to pass through a network. The lower the latency, the better the voice quality.

Leased Lines - Dedicated common-carrier facilities and channel equipment used by a network to furnish exclusive private line service. Also called a leased circuit.

Packet - A sequence of binary digits, including data and control signals, that is transmitted and switched as a composite whole.

Packet-switched Network - A method of transferring information in which data is broken into small pieces, called packets, and transported over shared communications channels.

PBX (Private Branch Exchange) - A phone exchange located on the customer's premises. The PBX provides a circuit switching facility for phone extension lines within the building, and access to the PSTN.

POTS (Plain Old Telephone Service) - The basic analog phone service consisting of standard telephones, telephone lines, and access to the public switched network.

PSTN - The public switched telephone network that traditionally routes voice calls from one location to another.

QoS (Quality of Service) - Refers to the measure of service quality provided to the user.

Router - A device that connects two networks using the same networking protocol.

Silence Suppression/Voice Activation Detection - In Voice over IP, silence suppression/voice activation detection (VAD) is a software application that allows a data network carrying voice traffic over an Internet/intranet connection to detect the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.

SIP (Session Initiation Protocol) - A signaling protocol for setting up conferencing, telephony, multimedia and other types of communication sessions over the Internet.

SNMP (Simple Network Management Protocol) - The TCP/IP standard protocol that is used to manage and control IP gateways and the networks to which they are attached.

Static IP Address - An IP address that is permanently assigned to a network device by an ISP.

Subnet Mask - A mask used to determine what subnet an IP address belongs to.

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Telnet - The TCP/IP standard network virtual terminal protocol that is used for remote terminal connection service and that also allows a user at one site to interact with systems at other sites as if that user terminal were directly connected to computers at those sites.

T1 - A high-speed (1.544M bps) digital telephone line with the equivalent of 24 individual 64K bps channels time division multiplexed together. A T1 can be used to transmit voice or data, and many are used to provide connections to the Internet.

Tie Line - A dedicated circuit linking two points without having to dial a phone number (i.e. the line may be accessed by lifting the phone handset or by pushing a button).

Trunk - Service that allows quasi-transparent connections between two PBXs, a PBX and a local extension, or some other combination of telephony interfaces to be permanently conferenced together by the session application and signaling passed transparently through the IP network.

Vocoder (voice encoder/decoder) - Provides multiple voice compression standards which range from G.723 (5.3K bps) to G.711 (full, uncompressed 64K bps). These standards are used to minimize the bandwidth required for voice.

VOIP (Voice over Internet Protocol) - The technology that turns voice conversations into data packets and sends them out over a packet-switched Internet protocol (IP) network.

VPN (Virtual Private Network) - A private network that utilizes dedicated equipment and large-scale encryption to connect remote sites or users together over the public Internet.

WAN (Wide Area Network) - The result of the connection of two or more LANs.

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Notes

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MultiVOIP Voice Over IP
Configuration Guide

Headquarters Location:

City: ____________ State/Prov.: ______ Country: ____________ No. of remote locations______

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: __________
B. % intra-office communication: __________
(if unknown, the rule of thumb is 30%)
C. Number of outside phone lines: __________
D. Total VOIP ports needed: (line B x line C) __________
E. Total monthly savings: (line A x line B) __________

MultiVOIP Model:

(Results from Line D = # of ports)
❐ MVP130 (1 port)
❐ MVP210 (≤ 2 ports)
❐ MVP410 (≤ 4 ports)
❐ MVP810 (≤ 8 ports)
❐ MVP2410/MVP3010 (≥ 16 ports)

2. Bandwidth Needed:

F. Voice Bandwidth: (line D x 14K) __________
G. Total Bandwidth: (line F x 2) __________

3. Return on Investment:

H. MultiVOIP cost: __________
I. No. of months for payback
(line H / line E) __________

Branch Office Location:

City: ____________ State/Prov.: ______ Country: ____________

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: __________
B. % intra-office communication: __________
(if unknown, the rule of thumb is 30%)
C. Number of outside phone lines: __________
D. Total VOIP ports needed: (line B x line C) __________
E. Total monthly savings: (line A x line B) __________

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G. Total Bandwidth: (line F x 2) __________

3. Return on Investment:

H. MultiVOIP cost: __________
I. No. of months for payback
(line H / line E) __________

Branch Office Location:

City: ____________ State/Prov.: ______ Country: ____________

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3. Return on Investment:

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I. No. of months for payback
(line H / line E) __________

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Branch Office Location:

City: ______________________ State/Prov.: _______ Country: ________________

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: ________

B. % intra-office communication: ________
(if unknown, the rule of thumb is 30%)

C. Number of outside phone lines: ________

D. Total VOIP ports needed: (line B x line C) ________

E. Total monthly savings: (line A x line B) ________

MultiVOIP Model:

(Results from Line D = # of ports)

❒ MVP130 (1 port)

❒ MVP210 (≤ 2 ports)

❒ MVP410 (≤ 4 ports)

❒ MVP810 (≤ 8 ports)

❒ MVP2410/MVP3010 (≥ 16 ports)

2. Bandwidth Needed:

F. Voice Bandwidth: (line D x 14K) ________

G. Total Bandwidth: (line F x 2) ________

3. Return on Investment:

H. MultiVOIP cost: ________

I. No. of months for payback
(line H / line E) ________

Branch Office Location:

City: ______________________ State/Prov.: _______ Country: ________________

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: ________

B. % intra-office communication: ________
(if unknown, the rule of thumb is 30%)

C. Number of outside phone lines: ________

D. Total VOIP ports needed: (line B x line C) ________

E. Total monthly savings: (line A x line B) ________

MultiVOIP Model:

(Results from Line D = # of ports)

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❒ MVP210 (≤ 2 ports)

❒ MVP410 (≤ 4 ports)

❒ MVP810 (≤ 8 ports)

❒ MVP2410/MVP3010 (≥ 16 ports)

2. Bandwidth Needed:

F. Voice Bandwidth: (line D x 14K) ________

G. Total Bandwidth: (line F x 2) ________

3. Return on Investment:

H. MultiVOIP cost: ________

I. No. of months for payback
(line H / line E) ________

Branch Office Location:

City: ______________________ State/Prov.: _______ Country: ________________

1. Configure Number of Ports Needed:

A. Total monthly long distance bill: ________

B. % intra-office communication: ________
(if unknown, the rule of thumb is 30%)

C. Number of outside phone lines: ________

D. Total VOIP ports needed: (line B x line C) ________

E. Total monthly savings: (line A x line B) ________

MultiVOIP Model:

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F. Voice Bandwidth: (line D x 14K) ________

G. Total Bandwidth: (line F x 2) ________

3. Return on Investment:

H. MultiVOIP cost: ________

I. No. of months for payback
(line H / line E) ________

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