The Multi-Tech MultiVOIP provides toll-free voice and fax communications over the Internet or Intranet. By integrating voice and fax into your existing data network, you can realize substantial savings on inter-office long distance toll charges. The MultiVOIP family is available in analog and digital models ranging from one to 60 ports. MultiVOIP products connect directly to phones, fax machines, modems, key systems, or a PBX to provide real-time, toll-quality voice connections to any office on your VOIP network.
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what's it do?

MultiVOIP is ideal for multi-location businesses looking to reduce toll charges between frequently called sites. MultiVOIP is a voice over IP gateway that integrates seamlessly into your data network and operates alongside existing PBXs, or other phone equipment to simply extend voice capabilities to remote locations. It is designed to help you maximize investments you’ve already made in your data and voice network infrastructure.

Office-to-office communication

A MultiVOIP network can be as small as two offices or as large as hundreds of offices. Each office installs and configures a MultiVOIP on their network and connects it to their existing phone equipment to begin placing calls, sending faxes, or making modem connections to the other offices on the VOIP network.

Off-net calling

Telecommuters or customers off the IP network can make long distance calls by dialing into a local MultiVOIP and placing toll-free calls to any location on the VOIP network. You can even have a MultiVOIP at a remote site dial a local phone number for a free person-to-person long distance call.

Create off-premise extensions

Extend the reach of your PBX to home office locations. Simply connect a MultiVOIP gateway to the PBX at the corporate office, and another MultiVOIP gateway at the remote office. Now anyone can place calls to the remote office by simply dialing an extension number. To extend your PBX to a building across the street, utilize a wireless bridge to connect the two networks. Now, you have voice and data connectivity without having to lay cables or paying monthly charges for dedicated lines.

Replace expensive tie lines

A corporation that utilizes tie lines to connect branch office PBXs to the corporate PBX can now use the company’s IP-based Wide Area Network to complete the call.

MultiVOIP Gateway Gateway Gateway Gateway Gateway Lifecycle Extension
why MultiVOIP?

TOLL-FREE VOICE OVER IP SOLUTIONS
CONNECTS DIRECTLY TO PHONES, FAX OR PBX
turnkey solution

Save thousands of dollars each month

MultiVOIP can save your company substantial amounts in long-distance charges. Even if your company uses one of the most inexpensive calling plans, a MultiVOIP network can quickly return your investment and begin paying you back:

| Location         | IP Addresses (4 lines) | PSTN Voice/Fax 100 Min/day | Payback
|------------------|------------------------|----------------------------|---------|
| Central Site     | 3291                   | 105 days                   | 1.5 months
| Remote Site      | 1904                   | 60 days                    | 2 months
| Chicago          | 2095                   | 60 days                    | 2 months
| Partner Site     | 1905                   | 60 days                    | 2 months
| Partner Site     | 1908                   | 60 days                    | 2 months

Easy integration

With MultiVOIP you have the hassle and expense of replacing your existing router, WAN connections or phone systems required by other VOIP solutions. MultiVOIP simply plugs into your Ethernet network. Neither your phone service or network is placed at risk. Minimum requirements:

- Ethernet network
- IP addresses
- IP telephone

Award-winning voice quality

With MultiVOIP you’ll experience consistent toll-quality voice connections. Using the Perceptual Speech Quality Measurement (PSQM), Internet World Magazine found that MultiVOIP delivered exceptional voice quality in fact. MultiVOIP outperformed the competition.

Complete support for multiple telephony interfaces

For maximum investment protection, the MultiVOIP four, two and four port models accommodate changing communications needs by providing a programmable T.38/F.77 and an E&M interface for each port. This allows MultiVOIP to connect directly to a phone, fax machine, key phone system or PBX. It automatically detects whether the incoming call is a voice or fax call. The single port MultiVOIP supports F.77 and E&M interfaces, while the digital MultiVOIP connects directly to a PBX or PSTN (see Table 1). T1 or PRI.

Bandwidth management

Bandwidth is used only when someone is speaking. The silence suppression/ Voice Activity Detection (VAD) feature in an option that freezes unused call bandwidth for data traffic. This is significant, since callers are usually silent for 40 percent of a call. When using silence suppression, MultiVOIP also offers Comfort Noise Generation (CNG) at the receiving end so the user knows the line has not dropped. In addition, MultiVOIP supports voice compression standards like G.729 (8:1) and G.723 (5:1). These standards help maximize the bandwidth required for voice: G.729, for instance, in the maximum compression rate and requires only 5.36 kbps (plus an added 7.84 kbps for IP overhead). Even at maximum compression, your voice solution will still provide toll-quality voice.

No user training required

MultiVOIP provides single stage dialing by utilizing a Uniform Dailing Plan (UAP) that is consistent with the 4-digit PSTN standard numbering plan. This includes the automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make VOIP calls. In fact, placing calls with MultiVOIP is like using your existing phone system.

Advanced speech technologies

MultiVOIP supports the Integrated Services Digital (Diffferent) Quality of Service (QoS) protocol which sets priorities for voice and fax traffic and allows transparent delivery. Diffferent helps more sensitive voice traffic across even low-bandwidth WAN connections, like ISDN, and with the priority and quality required by voice. Other features such as adaptive echo cancellation, forward error correction, bad frame interpolation, tunable latency and dynamic jitter buffer, further enhance voice quality.

Interoperability

MultiVOIP utilizes the 1413 and SIF protocols to provide complete interoperability with other Internet telephony solutions. The enhanced 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 10 or 1413 or SIIF devices from the same port. In addition, MultiVOIP also supports T38 real-time fax relay for interoperability among other VOIP equipment.

PSTN fail-over

PSTN fail-over allows MultiVOIP 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 degrade voice quality. Utilizing user-definable control, MultiVOIP continually checks if the LANE/4ip is threatened by packet loss, jitter or latency so no use of the network is completely down. If it detects a problem, MultiVOIP switches to “survivability mode” in real-time routing all calls over PSTN lines connected to the MultiVOIP gateway. MultiVOIP continues to monitor the connection and automatically switches back to the LANE/4ip once the conditions improve.

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Supplementary services

MultiVOIP supports nearly 100 supplementary services to provide for call transfer, call forwarding, call hold, call waiting, and more. In addition, MultiVOIP supports Q.931, with its call processing capabilities, for networks requiring supplementary services in a multi- or uni-vendor environment. In addition, MultiVOIP supports SIP extensions providing call-forward and call-transfer capabilities.

Integrated gatekeeper

The MultiVOIP family now includes models with an integrated gatekeeper to facilitate call management in a Voice over IP network. These cost-effective MultiVOIP gateways provide central phone book management as well as deliver the power to define and control how voice traffic is managed over IP networks. With the integrated gatekeeper network management 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.

Management

MultiVOIP is easily managed using a Windows-based software application, Web-based or SNMP. MultiVOIP 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 sn

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Avaya small office media gateway solution

Avaya and Multi- Tech have partnered together to provide an affordable small office media gateway solution that delivers the features of Avaya’s MultiVoice™ software to the branch offices of large corporations. The Multi-Tech/Multi- Voice gateway with integrated gatekeeper, cost-effectively extends the call features and networking benefits of a centralized Avaya Mediant Series of small branch offices and discov

Kudos! You’re the judge. Call our toll-free line and hear how easy it is to use our products. Schedule your free Product Test Center Recommended Government Computer User’s Choice

Avaya and VDI Solutions

Avaya and VDI Solutions are dedicated to providing a seamless telephony experience for their customers. With Avaya and VDI Solutions, customers can experience the best in voice and data communications, allowing them to stay connected and productive even when they’re on the go. Whether it’s remote or mobile, Avaya and VDI Solutions offer a comprehensive portfolio of telephony solutions that deliver the power to define and control how voice traffic is managed over IP networks. With the integrated gatekeeper network management 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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MultiVOIP is a cost-effective and reliable solution for deploying a voice network. It supports a variety of voice compression standards and offers features such as PSTN fail-over, which allows calls to be transferred to a PSTN line when the IP network is down. This ensures that calls are not interrupted, even in the event of a network failure. MultiVOIP also includes features such as Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) to improve voice quality. Additionally, MultiVOIP supports Bandwidth Management, which helps optimize bandwidth usage and ensures that voice traffic is prioritized over other data traffic.

When choosing a voice network solution, it is important to consider factors such as cost, reliability, and ease of integration. MultiVOIP is a cost-effective solution that is easy to integrate and offers a high level of reliability. It is also compatible with a variety of other voice network solutions, including Avaya and Nortel products. As a result, MultiVOIP is a popular choice for businesses looking to deploy an affordable voice network solution.

Success is about making it easier. Multi-Tech is about making it easier. We create better ways of sharing information—remotely and over the Internet. Multi-Tech solutions set the standard for efficient and effective communication in an information-hungry age. Our products are known for their reliability, performance, and flexibility. With Internet access, remote access and telephony products, Multi-Tech is creating a world where technically, everyone's possible.
why MultiVOIP?

TOLL-FREE VOICE OVER DATA NETWORKS PROVIDE:

CONNECTS DIRECTLY TO PHONES, FAX OR PBX TRUNK/SUITE SOLUTION

Save thousands of dollars each month

MultiVOIP can save your company substantial amounts in long-distance charges. Even if your company uses one of the most inexpensive calling plans, a MultiVOIP network can quickly return your investment and begin paying back your

Award-winning voice quality

PSTN fail-over allows MultiVOIP to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heightens availability and augments QoS when conditions threaten to<br>

Easy integration

advanced speech technologies

Quality Measurement (PSQM), fax relay for interoperability among other VOIP equipment. In addition, MultiVOIP also supports T.38 real-time phone number allowing you the flexibility to call H.323 or SIP devices configured to match. The outbound IP call protocol is configured with the protocol is automatically detected and the voice channel is dynamically operability with other Internet telephony solutions. The inbound IP call<br>

Advanced speech technologies

MultiVOIP supports the Differentiated Services (DiffServ) Quality of Service (QoS) protocol which sets priorities for voice and fax traffic and allows transparent delivery. DiffServ helps more-sensitive voice traffic across even low-bandwidth WAN connections, like 56K and ISDN, with the priority and quality required by voice. Other features such as adaptive echo cancellation, forward error correction, tunable latency and dynamic jitter buffers, further enhance voice quality.

Complete support for multiple telephony interfaces

For maximum investment protection, the MultiVOIP four, four and eight-port models accommodate changing communications needs by providing a programmable FAX/FAX and an IP interface for each port. This allows MultiVOIP to connect directly to a phone, fax, machine, phone system or PBX. It automatically detects whether the incoming call is a voice or fax call. The single port MultiVOIP supports FAX and FAX interfaces, while the digital MultiVOIP connects directly to a PBX or IP/FAX line. (Q711 or Q931)

Bandwidth management

Bandwidth is used only when someone is speaking. The silence suppression/ Voice Activity Detection (VAD) feature is an option that frees unused call bandwidth for data traffic. This is significant, since callers are usually silent for 80 percent of a call. When using silence suppression, MultiVOIP also offers Enhanced Voice Quality (EVQ), an option for preserving voice quality while providing silence suppression.

Interoperability

MultiVOIP utilizes the 1143 and ELF protocols to provide complete interop-

PSQM scores:

PSTN fail-over allows MultiVOIP to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heightens availability and augments QoS when conditions threaten to degrade voice quality. Utilizing user-definable controls, MultiVOIP continually checks if the LAN/WAN is threatened by packet loss, jitter or latency, or noise of the network is completely down. If it detects a problem, MultiVOIP switches to a "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.

Advanced speech technologies

MultiVOIP supports the Differentiated Services (DiffServ) Quality of Service (QoS) protocol which sets priorities for voice and fax traffic and allows transparent delivery. DiffServ helps more-sensitive voice traffic across even low-bandwidth WAN connections, like 56K and ISDN, with the priority and quality required by voice. Other features such as adaptive echo cancellation, forward error correction, tunable latency and dynamic jitter buffers, further enhance voice quality.

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Interoperability

MultiVOIP utilizes the SS 1143 and ELF protocols to provide complete interoperability with other Internet telephony solutions. The enhanced IP call protocols 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 either SS 1143 or ELF devices from the same port. In addition, MultiVOIP also supports 776 real-time fax relay for interoperability among other VOIP equipment.

PSTN fail-over

So, why MultiVOIP? Why not use MultiVOIP?

MultiVOIP is easily managed using a Windows-based software application, with features of SNMP. MultiVOIP also includes its own SNMP management software called MultiVSM, which provides complete site configuration, management and call monitoring for all MultiVOIP gateways on the network. It can buffer a telephone system that is easy to view events like nose tweaking, live use reporting, call history and voice quality statistics. In addi-

Supplementary services

MultiVOIP supports IP 420 supplemental services to provide for call transfer, call-forwarding, call hold, call waiting, and call identification. It also supports Q.120, an inter-PBX signaling protocol, for networking PBX supplementary services in a multi- or co-located environment. It also supports SIP extensions providing call forward and call transfer capabilities.

Integrated gatekeeper

The MultiVOIP family now includes models with an integrated gatekeeper to facilitate call management in a Voice over IP environment. These cost-effective MultiVOIP gateways provide complete call management and as well as deliver the power to define and control how voice traffic is managed over IP networks. Within the integrated gatekeeper network management can be configured, monitored and managed to ensure optimal performance.

Avaya small office media gateway solution

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why MultiVOIP?

FULL-DISCOURSE VOICE TECHNOLIGIES

CONNECTS DIRECTLY TO PHONES, FAX OR PBX

TURNKEY SOLUTION

Save thousands of dollars each month

MultiVOIP can save your company substantial amounts in long-distance charges, even if your company uses one of the most expensive carrier plans. A MultiVOIP network can quickly return your investment and begin paying you back.

Advanced voice quality

PSRN fail-over

PSRN fail-over allows MultiVOIP 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. Using user-definable controls, MultiVOIP continually checks if the Line/Modem is threatened by packet loss, jitter, or latency, and automatically switches to the PSTN network when the IP network is completely down. It will detect a problem, MultiVOIP switches to "survivability mode" automatically routing all calls over PSTN links forever if the network is completely down.

Interoperability

MultiVOIP utilizes the In 913 and EIF protocols to provide complete interoperability with other Internet telephony solutions. The embedded IP call protocol is automatically detected and the voice channel is dynamically configured to match the bandwidth for data traffic. This is significant, since callers are usually silent for 60 percent of a call. When using silence suppression, MultiVOIP also offers a Comfort Noise Generation (CNG) at the receiving end so the user knows the line is consistent with the E.164 (PSTN) standard numbering plan. This includes automatic appending and stripping of digits to dialed numbers to ensure that PSTN call setup is consistent and requires only 5.3K bps (plus an added 7-8K bps for IP overhead). Even at maximum compression, your PSTN solution will still provide toll-quality voice.

Bandwidth management

Bandwidth is used only when someone is speaking. The silence suppression/ Voice Activity Detection (VAD) feature in an option that few unused call bandwidth for data traffic. This is significant, since callers are usually silent for 60 percent of a call. When using silence suppression, MultiVOIP also offers a Comfort Noise Generation (CNG) at the receiving end so the user knows the line has not dropped. In addition, MultiVOIP supports voice compression standards like G.729, G.721 and G.723 (1/2). These standards help maximize the bandwidth required for voice: G.723, for instance, in the maximum compression rate and requires only 5.3K bps (plus an added 7K bps for IP overhead). Even at maximum compression, your PSTN solution will still provide toll-quality voice.

No user training required

MultiVOIP provides single stage dialing by utilizing a Uniform Dialing Plan (that is consistent with the 4-digit PSTN standard numbering plan). This includes the automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make PSTN calls. In fact, placing calls with MultiVOIP is like using your existing phone system.

Supplementary services

MultiVOIP supports the full suite of supplementary services to provide call transfer, call-forwarding, call hold, call waiting, and more. It also supports QSIG, an inter-PBX signaling protocol, for networking PBX supplementary services in a multi- or uni-vendor environment. In addition, it supports SIP extensions providing call-forward and call-transfer capabilities.

Integrated gatekeeper

The MultiVOIP family now includes models with 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 voice traffic is managed over IP networks. With the integrated gatekeepers, network managers can configure, monitor and manage the activity of registered end-users. In addition, they can set policies and control network resources, such as bandwidth usage, to ensure optimal implementation.

Complete support for multiple telephony interfaces

For maximum investment protection, the MultiVOIP four, four and eight-port models accommodate changing communications needs by providing a programmable FXS/FXO and an E&M interface for each port. This allows MultiVOIP to connect directly to a phone, fax machine, key phone system or PBX. It automatically detects whether the incoming call is a voice or fax call. The single port MultiVOIP supports FXS and FXO interfaces, while the digital MultiVOIP connects directly to a PBX or PBX trunk.

Media Server to small branch offices, utilizing traditional analog devices, over the Internet. Avaya and Multi-Tech have partnered together to provide an affordable small office media gateway solution that delivers the features of Avaya's Multiplatform™ software to the branch office of large corporations. The Multi-Tech/MultiVOIP gateway with integrated gatekeeper, cost-effectively extends the call features and networking benefits of a centralized Avaya Media Server to small branch offices, utilizing traditional analog devices, over an IP infrastructure. MultiVOIP also renders local office survivability in the case of a LAN or WAN failure, by providing local, reliable PSTN trunking.

Kudos!

Avaya small office media gateway solution

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No user training required

MultiVOIP provides single stage dialing by utilizing a Uniform Dialing Plan (that is consistent with the 4-digit PSTN standard numbering plan). This includes the automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make PSTN calls. In fact, placing calls with MultiVOIP is like using your existing phone system.

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what’s it do?

MultiVOIP is ideal for multi-location businesses looking to reduce toll charges between frequently called sites. MultiVOIP is a voice over IP gateway that integrates seamlessly into your data network and operates alongside existing PBXs, or other phone equipment to simply extend voice capabilities to remote locations. It is designed to help you maximize investments you’ve already made in your data and voice network infrastructure.

Office-to-office communication
A MultiVOIP network can be as small as two offices or as large as hundreds of offices. Each office installs and configures a MultiVOIP on their network and connects it to their existing phone equipment to begin placing calls, sending faxes, or making modem connections to the other offices on the VOIP network.

Off-net calling
Telecommuters or customers off the IP network can make low distance calls by dialing into a local MultiVOIP and placing toll-free calls to any location on the VOIP network. You can even have a MultiVOIP at a remote site dial a local phone number for a free person-to-person long distance call.

Create off-premise extensions
Extend the reach of your PBX into home office locations. Simply connect a MultiVOIP gateway to the PBX at the corporate office, and another MultiVOIP gateway at the remote office. Now, anyone can place calls to the remote office by simply dialing an extension number. To extend your PBX to a building across the street, utilize a wireless bridge to connect the two networks. Here, you have voice and data connectivity without having to lay cables or paying monthly charges for dedicated lines.

Replace expensive tie lines
A corporation that utilizes tie lines to connect branch office PBXs to the corporate PBX can now use the company’s IP-based Wide Area Network to complete the call.

The Multi-Tech MultiVOIP provides toll-free voice and fax communications over the Internet or Intranet. By integrating voice and fax into your existing data network, you can realize substantial savings on inter-office long distance toll charges. The MultiVOIP family is available in analog and digital models ranging from one to 60 ports. MultiVOIP products connect directly to phones, fax machines, modems, key systems, or a PBX to provide real-time, toll-quality voice connections to any office on your VOIP network.
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**Audio Models**
- analog models: MVP120, MVP130, MVP210, MVP2410, MVP410, MVP428
- digital models: MVP3010, MVP30-60, MVP428, MVP810, MVP24-48, MVP2410, MVP410, MVP410-g

**Digital Models**
- MVP120: 2 FXP voice ports
- MVP130: 1 FXP voice port, 1 FXP digital port
- MVP2410: 10 FXP voice ports
- MVP410: 4 FXP voice ports
- MVP810: 8 FXP voice ports

**Protocols**
- 312.5 Kbps, 56 Kbps, 28.8 Kbps, or 14.4 Kbps synchronous and asynchronous, 2.048 Mbps
- HDLC, PPP, V.35, X.25
- Voice over IP (VoIP): RTP, RTCP, SIP
- Voice over Frame Relay: H.323, H.450
- Voice over ISDN: H.450.6, H.450.8
- Fax over IP: T.37, T.30
- Modem over IP: V.90, V.34
- Network protocols: ARP, IP, TCP, UDP, ICMP

**Dimensions**
- MVP120: 6.2" x 1.4" x 9.0"; 2 lbs
- MVP130: 4.2" x 4.9" x 11.3"; 7 lbs
- MVP2410: 17.4" x 1.75" x 8.75"; 7.5 lbs
- MVP410: 10.8" x 2.5" x 14.2"; 23 kg
- MVP810: 44.2" x 4.5" x 22.2"; 3.4 kg

**Power Consumption**
- MVP120: 10W
- MVP130: 15W
- MVP2410: 27W
- MVP410: 46W; Digital models - 27W

**Safety**
- CE, UL 60950, EN60950, cUL, TS001
- EN61000-3-2, EN61000-3-3
- FCC Part 15 Class A, EN55022 EN55024, ETSI EN50082-1

**Ordering Information**
- MVP120: 1 or 2 available
- MVP130: 1 available
- MVP2410: 1 available
- MVP410: 1 available
- MVP810: 1 available

**What’s it do?**
- Create off-premise extensions
- Off-net calling
- Telecommuters or customers off the IP network can make low distance calls by dialing into a local MultiVOIP and placing toll-free calls to any location on the VOIP network. You can even have a MultiVOIP at a remote site dial a local phone number for a free person-to-person long distance call.

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**Create off-premise extensions**
- ADI, a Multitec value
- A corporation that utilizes tie lines to connect branch office PBXs to the corporate office, and another MultiVOIP gateway at the remote office. Now, anyone can place calls to the remote office by simply dialing an extension number. To extend your PBX to a building across the street, utilize a wireless bridge to connect the two networks. Now, you have voice and data connectivity without having to lay cables or paying monthly charges for dedicated lines.

**Replace expensive tie lines**
- A corporation that utilizes tie lines to connect branch office PBXs to the corporate PBX can now use the company’s IP-based Wide Area Network to complete the call.

**Office-to-office communication**
- A MultiVOIP network can be as small as two offices or as large as hundreds of offices. Each office installs and configures a MultiVOIP on their network and connects it to their existing phone equipment to begin placing calls, sending faxes, or making modem connections to the other offices on the VOIP network.