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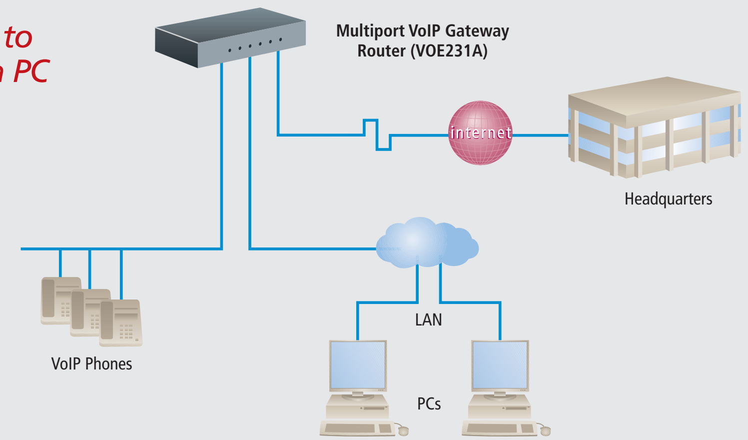
Multiport Analog VoIP Gateway Routers

Connect up to eight voice/fax analog calls to VoIP.

FEATURES

- Transparently connect your analog phones and fax machines to VoIP.
- Combines voice and data traffic for routing over your IP network or the Internet.
- Sized for use in enterprise environments.

Transparently connect up to eight analog phones plus a PC to the Internet.



OVERVIEW

Multiport VoIP Gateway Routers combine voice and data traffic for for routing over any IP network.

They enable you to transparently connect up to eight analog phones and fax machines to VoIP. Because these routers work with ordinary analog devices, there's no need to buy special IP telephones—just use the phones and fax machines you already have.

Models with FXO ports also connect to the local PSTN for backup in case your network connection goes out, so you can call special numbers such as emergency 911.

All models feature dual autosensing 10-/100-Mbps Ethernet ports.

Total VoIP connectivity.

Multiport VoIP Gateway Routers are full-featured VoIP gateways featuring SIP as well as H.323v4 signaling.

They offer up to eight FXS ports and two or four FXO ports. FXS analog ports connect any analog phone or PBX and provide services such as dial-tone, ringing, and Caller ID. FXO ports access the local PSTN for backup.

The routers are smart and adaptable, too! Telephony over IP (ToIP) enables them to handle calls based on variables such as hunt groups, Caller ID, Called ID, and time of day. And of course, they provide all basic phone services such as dial tone and ringing.

The routers provide advanced Quality of Service (QoS) and traffic management. QoS provides clear voice throughput every time by prioritizing VoIP packets on your network. Voice priority and traffic shaping ensure optimal up- and downstream voice quality.

It's a secure router, too!

VoIP Gateway Routers are capable enterprise routers that connect both voice and data to the WAN through a single secure device. They support full IP routing with IPSEC, PPPoE, DHCP, NAT, and VLAN tagging.

DNS maps domain and host names and IP addresses, enabling you to "hide" your real IP address from the Internet. An access control list enables you to control who can get into your network.

IPSEC VPN sets up secure virtual "tunnels" that protect your data and keep private calls private.

PPPoE encapsulates PPP frames inside Ethernet frames, enabling the router to establish a link from one machine to another over an IP network and then transport data packets over the connection. PPP enables services such as user authentication and usage metering that are missing from the Ethernet protocol.

DES, 3DES, and AES encryption keeps out unwanted users when you send data over the Internet.

Easy setup and management.

VoIP Gateway Routers are quite straightforward to set up. Just connect one of the dual 10/100 Ethernet ports to your LAN. The ports automatically sense and adjust to network speed and duplex as well as cable pinning.

Configure and manage the router through any standard Web browser, SNMP, or through Telnet using a command line interface.

Stay connected to the PSTN.

VoIP Gateway Routers with both FXS and FXO ports (VOE235A–VOE236A, VOE238A) enable you to access your local analog PSTN telephone service as well as your Ethernet nework. The router automatically falls back to your analog voice line if your network goes out.

You can also program the router to choose the PSTN or the network, depending on which number is dialed. This enables you to retain emergency 911 service and to route only certain calls—to your branch offices, for instance—through the IP network. Telephony over IP (ToIP) provides real-time PSTN and VoIP call switching for any PBX, switch, or phone.

VoIP.

Voice over Internet Protocol (VoIP) is a cost-saving alternative to traditional telephone service that enables voice data to be transported over IP networks, like the Internet, instead of the public switched telephone network (PSTN) or a cellular network.

VoIP, which operates strictly over IP networks, can connect to other VoIP nodes or traditional phone lines. The IP network used may be the Internet or a private network.

In either instance, the actual data-transport portion of this network can still be made up of the full gamut of network services: high-speed leased lines, Frame Relay, ATM, DSL, copper, fiber, wireless, satellite, and microwave signals. VoIP simply digitizes voice data and adds it to other information traveling along the same network.

With this technology, a phone call can be placed between two PCs, between a PC and a standard telephone, between a PC and an IP phone, between an IP phone and a standard telephone, or between two IP phones. It will take a long time for the PSTN to support this technology seamlessly, but this seems to be the direction in which phone systems are headed.

Benefits of VoIP

Because VoIP is inexpensive, has a worldwide reach, and operates on a few simple principles, it's exploded in popularity recently—especially among both small and large businesses that incur significant long-distance telephone expenses.

Savings

Without question, the primary benefit of a VoIP system is decreasing or eliminating long-distance telephone charges. Organizations with a high volume of long-distance voice traffic stand to save quite a lot of money by implementing a VoIP system. However, this factor alone may not warrant a full commitment to VoIP for some companies.

Setup fees for VoIP are usually quite low so your organization can generally start saving money after only a month or two of service. And with the wide variety of VoIP products and services on the market, it's easier than ever to set up a VoIP phone system over your network.

Convenience

VoIP can be set up in a way that enables you to use phone numbers in exactly the same way as you did before VoIP. Most of the services you get with traditional phone service—Voice Mail, Call Waiting, and Call Routing, for instance—are also available with VoIP.

VoIP doesn't interfere with other network services either, so you can surf the Web while making a VoIP call.

Portability

VoIP doesn't tie you to one phone or to a single location. Anywhere you find high-speed, reliable Internet access, you can use VoIP. Your phone number stays the same wherever you are—office, home, hotel, or even traveling overseas.

Standards

Although the ITU standards for VoIP have evolved significantly in the last few years, VoIP is still suffering from a lack of generally accepted interoperability standards.

H.323, a standard for real-time audio, video, and data communications across IP-based networks (including the Internet), is almost universally accepted as the primary standard for VoIP call setup and signaling. It's actually a collection of standards that works together for sending multimedia and data over networks that don't provide guaranteed Quality of Service (QoS).

The H.323 standard includes:

Real-Time Transport Protocol (RTP) specifies end-to-end network transport functions for applications transmitting real-time data such as video. RTP provides services like payload type identification, sequence numbering, time stamping, and delivery monitoring to real-time applications. Plus, it works with RTCP.

Real-time Transport Control Protocol (RTCP) works with RTP to provide a feedback mechanism, providing QoS status and control information to the streaming server.

Registration, Admission, Status (RAS) is a gateway protocol that manages functions such as signaling, registration, admissions, bandwidth changes, status, and disengage procedures.

Q.931 manages call setup and termination.

H.245 negotiates channel usage and capabilities.

H.235 provides security and authentication.

As VoIP product manufacturers began conducting interoperability tests for more complex operations, they recognized that they needed a simpler and more adaptable standard for call handling and signaling protocol.

To this end, the IETF developed the **Session Initiation Protocol (SIP)**. SIP is built with less computer code than H.323 is, so it's less cumbersome. Because SIP is similar in nature to HTML—it uses ASCII text for configuration—users can adapt it more easily for specific VoIP systems. In contrast, modifying H.323 for VoIP applications requires a knowledgeable computer programmer.

Both H.323 and SIP are considered "thick clients," where intelligence is maintained in the end devices such as IP telephones. In this respect, H.323 has a head start, although most VoIP systems today support both H.323 and SIP.

Providers

Despite the fact that VoIP standards are still developing, providers are already flooding the market with products and services while forming partnerships and matching expertise to strengthen their position in this new market. The biggest of these players and alliances—the ones who have the size and experience to grasp technical issues and quickly build infrastructures over which to offer VoIP services—are able to keep up with (and often influence) the continual changes in this market and keep rolling out new services.

Components

A VoIP system depends on devices that connect your traditional phone or phone system to an IP network. Components that you'll see in a VoIP system include:

- End-user devices
- Gateways or gatekeepers
- IPBXs
- IP Networks

End-user devices are usually VoIP telephones or PCs running VoIP software. End-user devices have their own IP address and make a direct connection to the IP network.

A gateway is a device that converts circuit-switched analog voice calls from a traditional PBX into VoIP packets and transmits them over an IP network either to another gateway or directly to an end-user device.

A gateway can have additional features such as voice compression, echo cancellation, and packet prioritization.

Because VoIP-enabled end-user devices can communicate directly with each other over an IP network, a gateway is not a required component of a VoIP system as long as the VoIP devices are connected directly to the IP network.

An IPBX is a PBX with a built-in gateway. IPBX systems are equipped for hundreds of telephone ports, with WAN support for trunk connections to the PSTN, and with high-speed IP WAN links. In addition to VoIP features, these systems usually include other features typical of traditional PBX systems such as music on hold, auto-attendant, and call management. Often, they include Ethernet ports to support VoIP telephones.

VoIP can be set up with or without a connection to standard PSTN phone service. You can, of course, place calls over the Internet directly from your PC or IP phone to another VoIP-enabled device. But what makes VoIP so versatile is that, through the use of a gateway service, it can also be used to call the numbers of phones connected to standard land-line or cellular phone services. They can also receive calls from standard telephones.

Not all fun and free calls

There are still things to consider when you're deciding whether or not to invest in VoIP.

Regulation vagaries: Much of the government regulation of VoIP is still being worked out. The U.S. government hasn't decided whether VoIP is going to be regulated as phone service or whether to tax it. VoIP isn't available in all countries.

Compatibility: Although older VoIP equipment may still have some compatibility issues, current VoIP products from different vendors generally work together.

Cost: For all the popular talk about VoIP being free, it isn't truly free. Any VoIP system has costs associated with its implementation—equipment, high-speed Internet access, and gateway service.

QoS: VoIP depends on having a fast, reliable network to operate. A fast network connection with guaranteed bandwidth is not a problem in a corporate intranet where you have complete control over the network. However, if you're using the Internet for VoIP, you're using a public network that may be subject to slowdowns that cause drop-outs and distortion. You may find that your high-speed Internet connection is faster than the actual Internet and that the quality of your connection is generally unacceptable or is unacceptable at times when Internet usage is high.

Emergency services

If you subscribe to a VoIP gateway service that enables you to use your VoIP phone like a regular phone, be aware that you may not be able to call 911 for emergencies. If 911 service is important to you because you don't have an alternative way to call 911, shop for a VoIP provider who provides this service.

Consider, too, that VoIP needs both working Internet access and power to work. If you lose your Internet service, your phone goes, too. And, unlike regular phone service that can keep basic telephones working when the power goes out, VoIP needs power—if you lose power, you lose your phone.

Moving forward

Before VoIP technology becomes truly universal, the current worldwide PSTN will have to migrate to a packet-based IP equivalent. Industry inertia alone dictates this will not occur instantly. The current worldwide PSTN system has grown to what it is over a period of 125 years. Given the sheer complexity of the existing PSTN, the migration to an IP packet network will probably occur during several decades.

As migration from the PSTN to IP-based networks proceeds, businesses and home users will gradually discover reasons of their own to implement VoIP. It won't happen right away, but we predict that VoIP will become a big part of telecommunications in the not-so-distant future.

Although it's not quite as convenient as conventional phone service, VoIP can offer serious savings—particularly if you now regularly pay for multiple overseas phone calls. Keep in mind though, VoIP isn't a one-size-fits-all solution. But with a little planning, VoIP could spell savings for you!


VOE231A: rear view

TECH SPECS

Caller ID Type — FSX Ports: ½ FSK, ITU V.23/Bell 202 generation;
FXO Ports: Caller ID FSK CLI reception and relay (Bellcore/ANSI and ETSI/ITU), Call routing based on Caller ID

Call Routing —
Virtual interfaces;
Routing criteria: Called party number (Destination), Calling party number (source), time of day, day of week, date, longest prefix match, wildcard match, regular expression match;
Number manipulation functions: Replace numbers, add/remove digits, regular Expressions;
Fallback routing: Soft fallback to alternative interface or call router table

Environmental — Operating temperature: 32–104° F (0–40° C);
Operating humidity: 5–80% (noncondensing)

Fax and Modem Support — G.711 fax and modem bypass;
T.38 Fax relay (9600 bps, 14.4 kbps)

IP Services — IPv4 router;
Static routes, ICMP redirect (RFC 792), RIPv1, v2 (RFC 1058 and 2453);
Static and dynamic NAT and NAPT;
DHCP server and client;
Access control lists;
IPSEC AH & ESP modes, preshared keys;
AES/DES/3DES encryption

Management — Web GUI;
Industry standard CLI with local console (RJ-45, RS-232) and remote Telnet access;
TFTP configuration and firmware loading;
SNMP v1 agent (MIB II and private MIB);
Built-in diagnostic tools (trace, debug)

QoS — Traffic classification by ACL;
TOS and DiffServ labeling, configurable TOS/Precedence bits or DiffServ codepoints;
IEEE 802.1p/Q;
Traffic scheduling: Priority, weighted fair queuing (WFQ), hierarchical traffic classes;
Policing of traffic classes;
DownStreamQoS™ dynamic restriction of inbound (downstream) TCP traffic to free bandwidth for voice packets;
Improves voice quality in the receiving direction

Voice Processing — Voice codes: G.711 A-Law/µ-Law (64 kbps), G.726 (ADPCM 40, 32, 24, 16 kbps), G.723.1 (5.3 or 6.3 kbps), G.729ab (8 kbps);
Up to 8 parallel voice connections;
G.168 echo cancellation;
Carrier tone detection and generation;
Silence suppression and comfort noise;
Configurable dejitter buffer;
Configurable tones (dial, ringing, busy);
RTP/RTCP (RFC 1889)

Voice Services — Anonymous Caller ID block;
Call blocking;
Call forward on busy;
Call forward, selective;
Call forward, unconditional;
Call hold/retrieve;
Call return;
Call transfer, blind;
Call transfer, with consultation;
Call waiting/retrieval;
Caller ID;
Conference drop;
Conferencing (3-way calling);
Distinctive ring;
Do not disturb;
Hotline calling;
Incoming Caller ID on/off;
IP URL dialing;
Message waiting indication;
Self Caller ID block;
Speed dial;
Voicemail message retrieval;
Warmline calling

CE Approval — Yes

Connectors — All: (2) RJ-45 10-/100-Mbps Ethernet;
VOE231A: (2) RJ-11 FXS;
VOE232A: (4) RJ-11 FXS;
VOE233A: (6) RJ-11 FXS;
VOE234A: (8) RJ-11 FXS;
VOE235A: (2) RJ-11 FXS, (2) RJ-11 FXO;
VOE236A: (4) RJ-11 FXS, (2) RJ-11 FXO;
VOE238A: (4) RJ-11 FXS, (4) RJ-11 FXO

Indicators — LEDs: (1) Power, (1) Run (1) VoIP Link, (2) Link, (2) 100M, (2) Activity, (2–8) Voice Port

Power — 100–240 VAC, 50–60 Hz, external

Size — 2.5"H x 8.5"W x 7.1"D (6.4 x 21.6 x 18 cm)

Weight — 1 lb. (0.5 kg)

Item	Code
Multiport VoIP Gateway Routers	
2-Port FXS	**VOE231A**
4-Port FXS	**VOE232A**
6-Port FXS	**VOE233A**
8-Port FXS	**VOE234A**
2-Port FXS, 2-Port FXO	**VOE235A**
4-Port FXS, 2-Port FXO	**VOE236A**
4-Port FXS, 4-Port FXO	**VOE238A**