InterPrise 3200D™

An IP Telephony Gateway for
Enterprise Network Applications

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Overview

Delivering integrated voice and data communications throughout your organization has never been easier or more cost-effective. Today, you can bring office locations worldwide closer together by relying on Internet Protocol (IP)-based networks to deliver both voice and data communications across the Internet or a private intranet. Inter-Tel's InterPrise 3200D IP telephony gateway can make it happen. By accepting voice transmission from any standard PBX or key system and converting it into data packets, the InterPrise 3200D transmits real-time voice alongside your organization's e-mail or fax communications. One network for voice and data means lower costs and greater efficiency throughout your enterprise.

As a result, you'll be able to eliminate costly intra-company long distance telephone toll charges. Instead, you'll pay a single flat rate for both voice and data communications between facilities. In addition to cost savings, you'll make it easier for employees and customers alike to get the information or assistance they need, regardless of geographic location.

Ease of Integration

Regardless of how you choose to take advantage of IP telephony, the InterPrise 3200D makes it easy to implement and as easy-to-use as your office phone or fax machine. You can use your existing telephone to place calls across your Wide Area Network (WAN) just like you would any other call. You can even send intra-office faxes across your data network with the InterPrise 3200D.

By delivering both voice and data network connectivity in an 8, 16, 24 or 30-port configuration, the InterPrise 3200D delivers reliable, high-quality voice and data communications to small branch offices as well as main corporate facilities.

Quality Voice Communication

The InterPrise 3200D delivers full duplex, toll-quality voice conversations by relying on superior voice encoding technology and advanced features like echo cancellation. This ensures that the delay or loss of voice packets, and the effects of that loss, are minimized so that voice conversations are free from background noise, delay and "clipping."

In addition to ensuring voice quality, the InterPrise 3200D delivers voice communications efficiently by compressing voice from 64 kilobits per second (kbps) to as low as 8 Kbps per voice channel. This dramatic decrease in bandwidth usage means you can deliver as much as eight times the amount of voice communications across your data network as you could with an uncompressed voice transmission. Less bandwidth for voice means more for data, ensuring that both will flow reliably across your network to their final destination.

Web-based Calling with ClearConnect – Talk to Agent

Capturing the sale and improving customer support over the Internet has never been easier than now. With ClearConnect – Talk to Agent, you can voice-enable your Web site and allow web-based customers to talk to your call center agents with a click of a button. Talk to Agent is an application that "plugs" into the customer's Web browser. When the "touch to talk" link on your Web site is clicked on, the Web browser becomes voice-enabled and a call is then established between the customer and your support agents. Whether you wish to increase your online sales or simply improve your level of customer support, Talk to Agent puts your customers a click away from a live representative of your company.

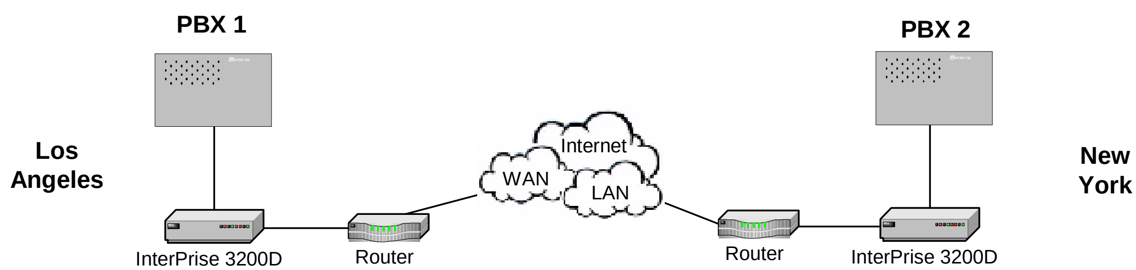
The Voice of the Internet

Whether you've already implemented a Wide Area Network, or are looking for ways to increase the return on investment for a new network project, Inter-Tel offers the right technology, in a package that's easy to implement and easy to afford. For more information, contact your Inter-Tel representative.

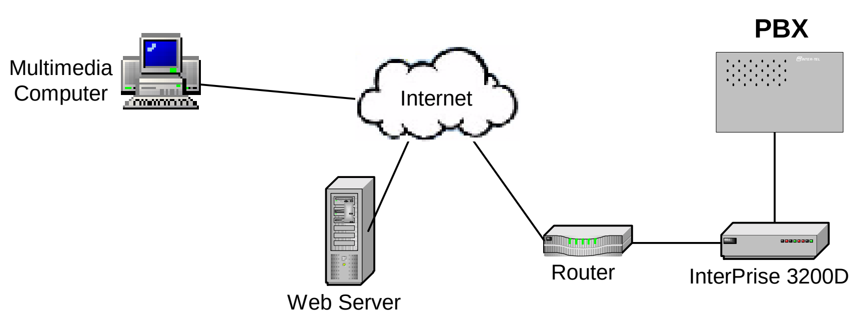
Applications

Following are a set of example applications using the InterPrise 3200D. These drawings make the assumption that the WAN is already in place. Therefore, the InterPrise gateway is located behind the existing router. In some cases, when the WAN is not in place, the InterPrise 3200D can be installed as the router.

Connecting Phone Systems Together Over the WAN



Connecting Web Surfers to a PBX Telephone with Talk to Agent



Specifications

Voice and Call Capabilities

- T1 available in 8-channel, 16-channel, and 24-channel configurations
- T1 PRI available in 7-channel, 15-channel, and 23-channel configurations
- E1 available in 8-channel, 16-channel, 24-channel, and 30-channel configurations
- E1 PRI (ISDN) available in 7-channel, 15-channel, 23-channel, and 30-channel configurations
- G.711 (ITU 64 Kbps PCM A-Law/µ-Law Standard)
- G.726 (ITU 40, 32, 24 Kbps) ADPCM
- G.728 (ITU 16 Kbps LD-CELP)
- G.729 (ITU 8 Kbps CS-ACELP)
- A-Law to µ-Law PCM code conversion
- G.165 (ITU echo cancellation Standard)
- Transparent DTMF/MF Processing
- Supports MF to DTMF conversion
- Call Progress Tone Handling
- Call Routing and Number Translation
- Local unit loop back calling
- Auto ring-down
- Automated Speech, Fax and In-Band Data Discriminator
- Digital Speech Interpolation (DSI) and Voice Activity Detection
- Voice over Frame Relay
- Voice over IP/Internet
- Voice over Data Routing Prioritization
- Data fragmentation
- Low Latency
- Adaptive Jitter Buffer Management
- Call Aggregation

Fax Capabilities

- Group III Fax Relay T.30 (demodulation and re-modulation up to 9600 baud)

Protocol Support

- Ipv4 Host and Routing (RIP v1, static routes)
- Frame Relay
- Q.922 HDLC
- RFC1490 Encapsulation
- LMI ANSI T.167 Annex D
- Frame Relay Encapsulated IP
- Simple Network Management Protocol (SNMP)

Physical Interfaces

- (2) V.35: Synchronous Data up to 2.048 Mbps
- RS-232 Console Port
- T1 or E1
- Ethernet 10Base-T IEEE 802.3

Telephony Interface for T1

- Circuit Type: T1 E&M (two-way, wink-start)
- Framing Type: D4 Superframe or Extended Superframe (ESF)

- Zero code suppression scheme: AMI (Alternate Mark Inversion), AMI ZCS (Alternate Mark Inversion Zero Code Suppression; also known as bit-7 stuffing or jam bit 7), or B8ZS (Bipolar Eight Zero Substitution)
- Line build-out (LBO) attenuation: For example, 0dB, 7.5dB, 15dB, or 22.5dB depends on the telephone system and how close it is to the 3200D; set according to the manufacturer's recommendation)
- Reference clock programming: Private Network Master (the InterPrise 3200D acts as a slave)

Telephony Interface for E1

- Circuit type: E1 PRI (ISDN), E1 MFC-R2, or E1 E&M Immediate
- Zero code suppression scheme: HDB3 (High-Density Bipolar-3 zeros) or AMI (Alternate Mark Inversion)
- Reference clock programming: Private Network Master (the InterPrise 3200D acts as a slave)

Routing

- Voice over data prioritization
- IP Precedence bit for QoS
- Data Fragmentation for QoS
- Frame Relay termination
- IP protocol routing at 720 packets per second

Management and Control

- Comprehensive on-line help, configuration, control, traffic monitoring, alarm and status recording, software download/upload
- Configurable with both IP, console port, and modem access
- In-Band Host to Host Management and Control
- SNMP Management
- Centralized call routing and CDR generation through optional Connectivity Manager software

Additional Compatibilities

- Server real-time billing support via ODBC-compliant DBMS
- CDR generation
- Connectivity Manager support
- ClearConnect SoftPhone client
- ClearConnect – Talk to Agent web-based client

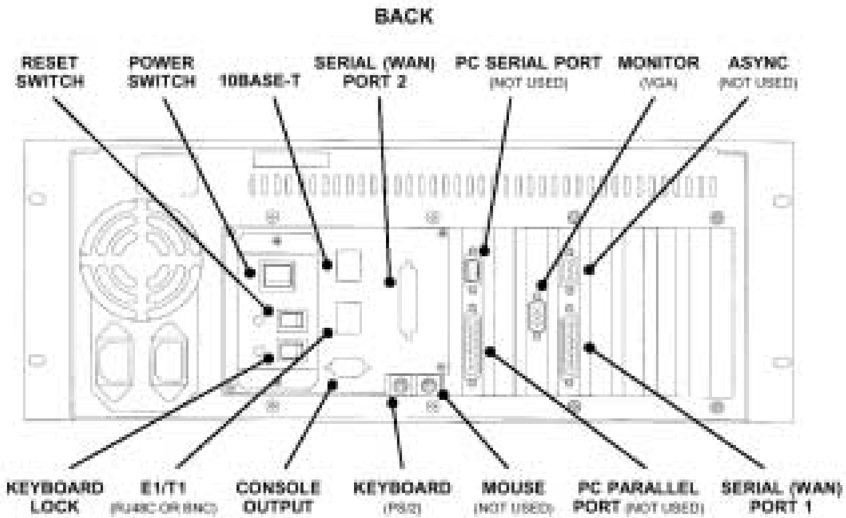
Power Supply Requirements

- Operating Voltage: 115V/230V AC with 0.6A/0.25A, 47–63 Hz
- Peak Power Consumption: 60 Watts

Physical Dimensions

- Chassis: All aluminum construction
- Handles: Extruded, formed aluminum
- Sliding Rail Options: 457 - 762 mm (18 - 30 in.)
- Weight: 14 kg. (30 lbs.)
- Width: 483 mm. (19 in.)
- Height: 177 mm. (7 in.)
- Depth: 432 mm. (17 in.)

Main Control Unit Rear Panel



Network Provisioning

Routers

When a Voice over IP (VoIP) device communicates through a router, it is highly recommended that the router be able to prioritize the packets. Since the voice packets have to reach the destination in “real time” it is important for the router to pass the voice packets through before any other data. In some configurations, a dedicated serial or Ethernet port is available on the router for the VoIP traffic. In this case, the router needs to prioritize the traffic from that physical interface over all other interfaces.

If the VoIP device is talking to the router through the same physical Ethernet interface as the LAN traffic, then you need to prioritize the VoIP traffic by TCP and UDP port numbers. When installing or configuring the router(s) used to pass the VoIP traffic, here are the port numbers you should prioritize:

UDP port 16384 (audio)
UDP ports 5000-5018 (call control)

Firewalls

If your firewall is using a fully qualified set of IP addresses, then you simply need to open up the ports listed above to allow the inbound voice traffic through. If you are using NAT, then you need to be able to “map” or “bind” the above ports to the IP address of the InterPrise 3200D.

Proxy Servers

If you intend to pass Voice over IP traffic through a Proxy server, the server needs to be capable of providing proxy services for any IP device, not just Windows PCs. Some proxy servers require a client-side application on any IP-based unit that needs access through the firewall. InterPrise 3200D is not a Windows PC, as many proxies require. Therefore, an alternate path for the Voice over IP traffic may need to be established.

Bandwidth Requirements

The bandwidth required for IP calls is determined by three factors. The first factor is the vocoder selection. The most popular vocoder is G.729 with a compression ratio of 8:1. This vocoder takes a traditional 64 kbps call and compresses it to 8 kbps. The second factor is a custom feature called Frame Multiplier. This user-definable feature adjusts the number of voice samples that get placed inside a single network packet. The third factor is the total number of InterPrise gateways communicating together on the network. This is due to another custom feature called Call Aggregation. Simply put, this allows IP calls destined to the same place to share the network protocol overhead. This allows the overhead to be distributed over multiple calls, thus reducing the required bandwidth per call.

The three factors all affect the amount of bandwidth required for the voice traffic. Although the average total rate per call is around 12-14 kbps (using G.729), certain configurations can require as much as 18 kbps and as little as 8.5 kbps when using the G.729 vocoder. The values can be predicted using the formulas provided in the installation manual. Until the actual layout of a network is known, it is impossible to know exactly what the required bandwidth will be. However, once the layout is known, precise calculations can be generated and used in designing even the most rigid of networks.

Regulatory Approval

- Approved for direct PSTN or PBX connectivity in the following countries:
  - United States
  - Canada
  - Austria
  - Belgium
  - Denmark
  - Finland
  - France
  - Germany
  - Greece
  - Holland
  - Iceland
  - Ireland
  - Italy
  - Luxembourg
  - Norway
  - Portugal
  - Spain
  - Sweden
  - Switzerland
  - UK
- The 100-ohm T1 Line Interface Card (LIC) for this system complies with Part 68 of the U.S. Federal Communications Commission (FCC) rules (FCC registration number: BE2USA--27119--CN--N).
- The 120-ohm E1 Line Interface Card (LIC) complies with the pan-European requirement CTR 4 and is approved by NMi (Notified Body # 0122).