Your Connection to the Future

IP Networking Module

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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 closer together by relying on Internet Protocol (IP)-based networks to deliver both voice and data communications across the Internet or a private intranet. The Inter-Tel IP Networking Module can make it happen. By accepting voice transmission from your Inter-Tel Axxess or Eclipse phone system, and converting it into data (Voice over IP), the IP Networking Module transmits real-time voice alongside your organization's computer traffic, such as e-mail. 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 inter-office long distance telephone 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.

By utilizing the networking capabilities of your Inter-Tel PBX, you can seamlessly bring all your offices together to act as one. By adding the IP Networking Module to each PBX, you can connect your offices together right over your data infrastructure.

Ease of Integration

When you add the IP Networking Module to your Inter-Tel PBX, you can send and receive calls to any office in your organization, right over the data network. It doesn’t matter if that office is down the road or in another country. The phone systems will still act as one, and your employees may never even know that the call is being carried over your data network instead of the phone company.

Quality Voice Communication

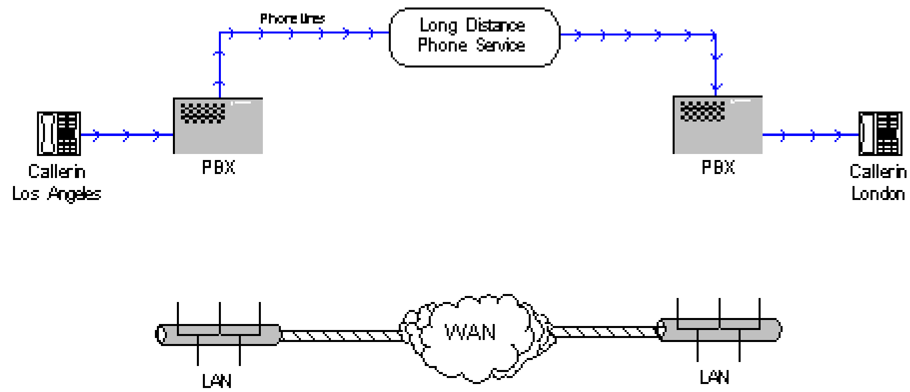
The Inter-Tel IP Networking Module delivers full duplex, toll-quality voice conversations by relying on superior voice encoding technology and advanced features like echo cancellation. This ensures that the delay or loss of voice packets, and the effects of that loss, are minimized so that voice conversations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

In addition to ensuring voice quality, the IP Networking Module 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, helping both to flow more reliably across your network to their final destination.

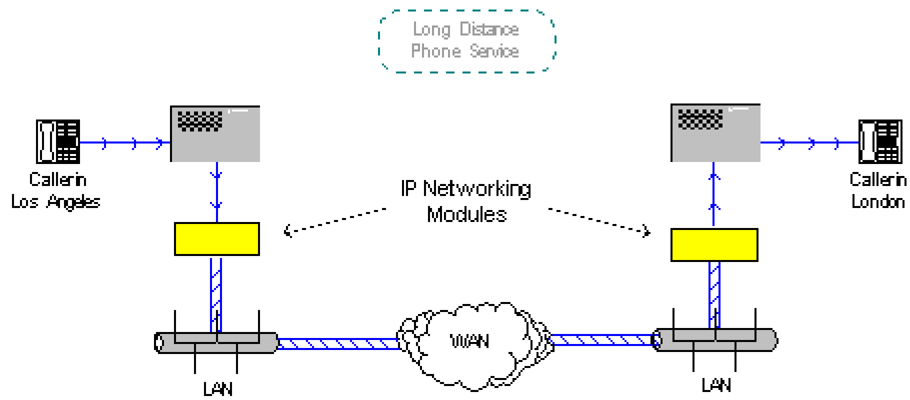
How does it work?

Think of the IP Networking Module as a converter that sits between your phone system and the data network. When someone in one office calls to another office, the PBX will send the call to this converter. The voice transmission turns into data and is sent over the network to the converter at the other office. From there, the data changes back into voice and is sent to the number dialed. The caller doesn’t do anything different, so there is no learning curve to overcome. The pictures below illustrate this concept.

Normal Long Distance Call



Long Distance Call Using the IP Networking Module



Your Inter-Tel PBX will need to be equipped with a T1-PRI card. This is where the communication link is established between the PBX and the IP Networking Module. A simple cable connects them together without the need for any extra equipment. The IP Networking Module will then connect to your data network in one of the following ways:

- Directly to your router through a serial connection
- Directly to your router through a 10Base-T Ethernet port
- To a 10Base-T Ethernet port on your LAN

If you have small offices that are not currently connected together by a data network, you can save on the cost of a new router by using your IP Network Module as your router. If your data routing needs are modest, you can save more money by letting the IP Networking Module take care of this for you. When you choose this option, the connection to the PBX is the same. Your LAN will then connect to the IP Networking Module, and it will connect your whole office to the wide area network (WAN) over Frame Relay, dedicated lines, or even the Internet.

Specifications

Voice and Call Capabilities

- T1 available in 7-channel, 15-channel, and 23-channel configurations
- E1 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 IP Networking Module 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 IP Networking Module 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

- Configurable with both IP, console port, and modem access
- In-Band Host to Host Management and Control
- SNMP Management
- Call Detail Record generation

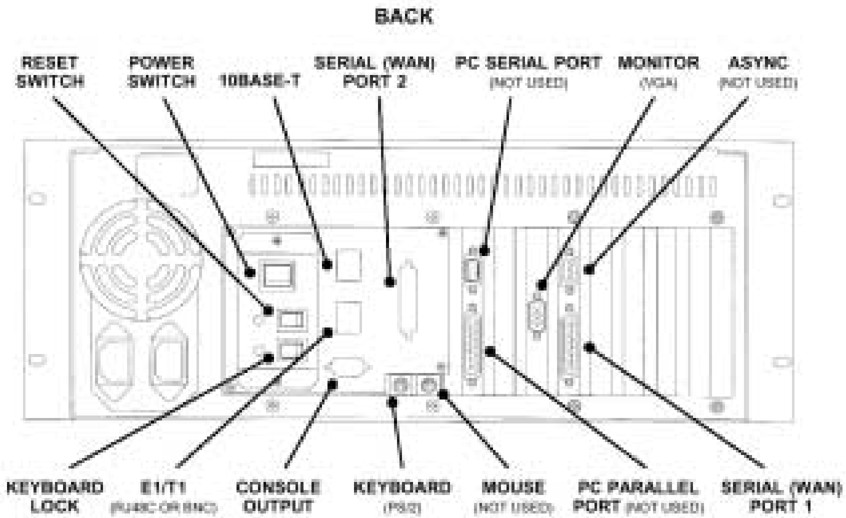
Power Supply Requirements

- 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 IP Networking Module.

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. The IP Networking Module 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 IP Networking Modules 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 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).