For additional information contact:

Terry Shugart - tshugart@analogic.com
http://www.analogic.com/cti
TEL: 978-977-3000
FAX: 978-977-6813

White Paper: Voice Over IP Networks

VOICE OVER DATA NETWORKS

Long distance telephone charges continue to be a major expense item for businesses and individuals despite deregulation of the Public Switched Telephone Network (PSTN). A successful strategy for reducing long distance telephone costs is to utilize the existing capacity on private and public data networks for voice traffic. This strategy can take two forms:

- **Voice over Circuit Switched Data Networks**. Historically, much data has been transported over leased telephone lines. A market has developed for equipment to transport voice traffic over these same connections, where in most instances the voice traffic is compressed to reduce bandwidth requirements.
- **Voice over Packet Data Networks**. Recently, many companies and individuals have switched to packet data networks, including Frame Relay and the Internet, to lower costs. The use of the *Internet Protocol (IP)* over packet data networks for the transmission of voice traffic is discussed in this paper as an emerging tool for reducing long distance telephone costs.

VOICE COMPRESSION FOR CIRCUIT SWITCHED NETWORKS

Many private data networks are constructed with dedicated leased telephone lines. Other expensive switched circuits include satellite communications links and international long distance telephone lines. When voice is transported long distances across expensive circuit switched networks, it is often compressed to reduce bandwidth. This process is complicated, however, by the fact that voice traffic is usually mixed with fax and data modem traffic. To compress mixed traffic, special DSP-enabled compression systems are required.

Voice Compression and Bandwidth Reduction

Using a DSP-based compression system that has been ported to the Analogic TAP-800 Family of DSP Resource Boards, incoming voice traffic can be compressed using a voice coder (vocoder). If the call is from a fax machine or data modem, the incoming signal is demodulated and sent digitally to the receiving station. Overall, traffic can be compressed by as much as 12:1, reducing circuit switched channel costs by the same factor. One end of the system is shown in Figure 1. A similar system at the other end of the leased line decompresses the voice call and remodulates the fax and data modem traffic. DTMF signaling is also passed through, making the system transparent to users. Modern voice compression is made possible by new, high-quality parametric vocoders. *Parametric vocoders* model the human vocal tract, transmitting only the parameters needed by the model. Vocoder technology *achieves high levels of compression, low delay* and near-toll-call-quality reproduction of human speech. These same advances also make voice transmission over packet data networks possible.

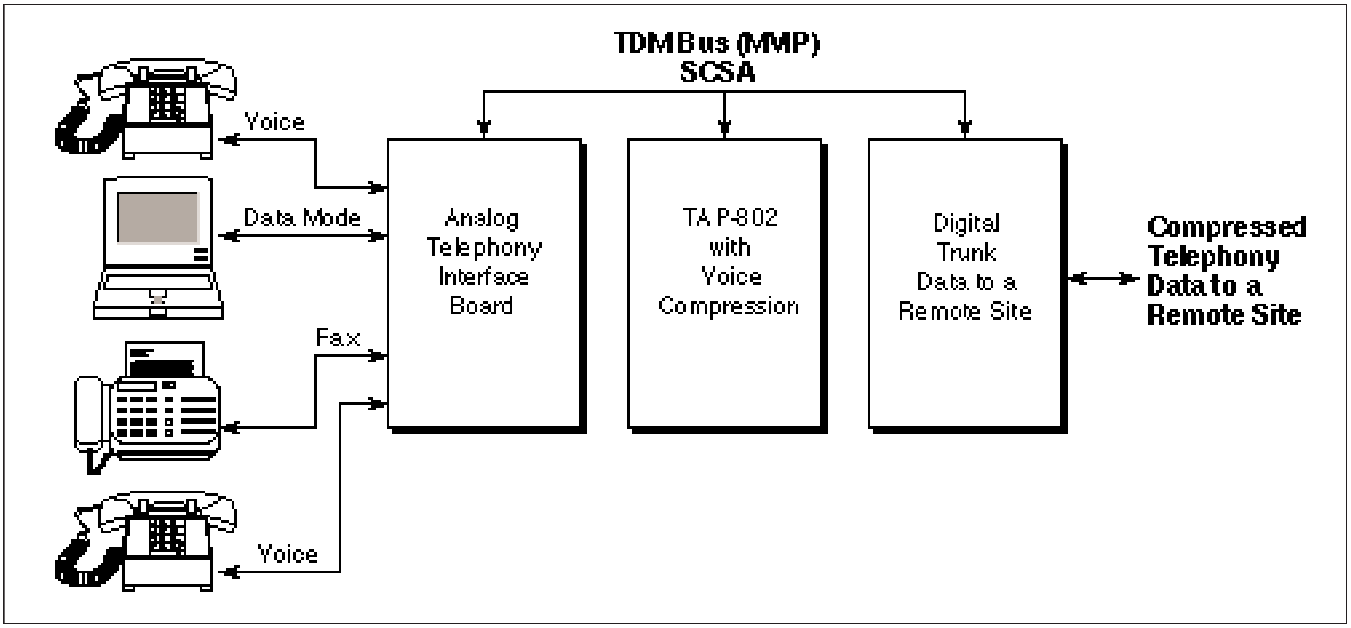

Figure 1. Compressed Voice, Data and Fax Traffic on the TAP-802.

SPEECH COMPRESSION FOR PACKET DATA NETWORKS

Voice over Packet Data Networks

Two types of public packet networks are in use today.

- **Frame Relay**. Frame Relay networks are rapidly replacing the older X.25 networks and adding new customers. Frame Relay technology not only costs less, it has much lower delays than X.25 because it does less error checking and uses newer, faster switches. Frame Relay also provides constant delay because it sets up a *Permanent Virtual Circuit (PVC)* that remains for the entire call. *The PVC is a single path* through the Frame Relay network. All the packets in the call will have the same delay and can't get out of order. Networks using X.25 switching have too much delay for voice traffic but Frame Relay, using the IP protocol, can deliver excellent voice quality over very long distances at very low cost.

- **The Internet**. The world wide Internet is a vast network with a large variety of equipment providing packet switched services. Most local connectivity is through data routers. Many backbone transmission facilities are being upgraded to Frame Relay and ATM switches. Although Voice over the Internet has longer delays than Voice over Frame Relay, the quality is acceptable for most users, and the Internet has the advantage of near universal availability.

Two types of phone-to-phone services are available over *packet networks:*

- **Service Provider applications**

- **Intranet applications**

Each of these applications requires different equipment configurations.

IP VOICE SERVICE PROVIDERS

Voice over IP network service providers fall into two categories:

- **International Telephone Companies**. These companies provide international telephone service into and out of the US at a discount over what the local PTT or US long distance carrier charges. Today most of these companies make money by aggregating traffic on leased lines. To further reduce costs, they often provide "call back" service so that calls originate in the US and carry the lower US tariffs. Many of these companies and many new international telephone companies are looking to further lower costs by deploying Voice over IP services.

- **Internet Service Providers (ISP's)**. The Internet's growth continues with usage doubling every nine months. However, competition for new customers is fierce and low cost is a selling point. To improve margins, ISPs are seeking new services to provide to users. Many ISPs are looking to add voice as a new service and a new source of revenue. Note that in the US, ISP's provide low, often fixed-cost access to the Internet because they are considered "enhanced service providers" by the US Federal Communications Commission (FCC). As such, they are exempt from the access changes that the Regional Bell Operating Companies (RBOC's) charge long distance telephone companies. This situation may change in the future but a considerable cost advantage over long distance switched circuit technology is expected to remain.

Service Providers deliver IP Voice telephone services to individuals or corporate customers over Frame Relay networks or the Internet. This IP Voice application involves placing a phone call over the local PSTN to the *Service Provider (SP)* at its Local *Point of Presence (POP)*. (See Figure 2.) At the Local POP, the voice data is compressed, packetized and sent over the Internet or a Frame Relay network to a Remote POP. The voice data is compressed, not only to save bandwidth but also to allow the voice data to fit in between other data packets in the data packet network. At the Remote POP, the call is placed to the customer over the remote PSTN. This phone-to -phone connection bypasses the long distance telephone carrier, providing substantial cost savings to the service provider, who passes these savings on to the customer. The savings are particularly dramatic for international calls.

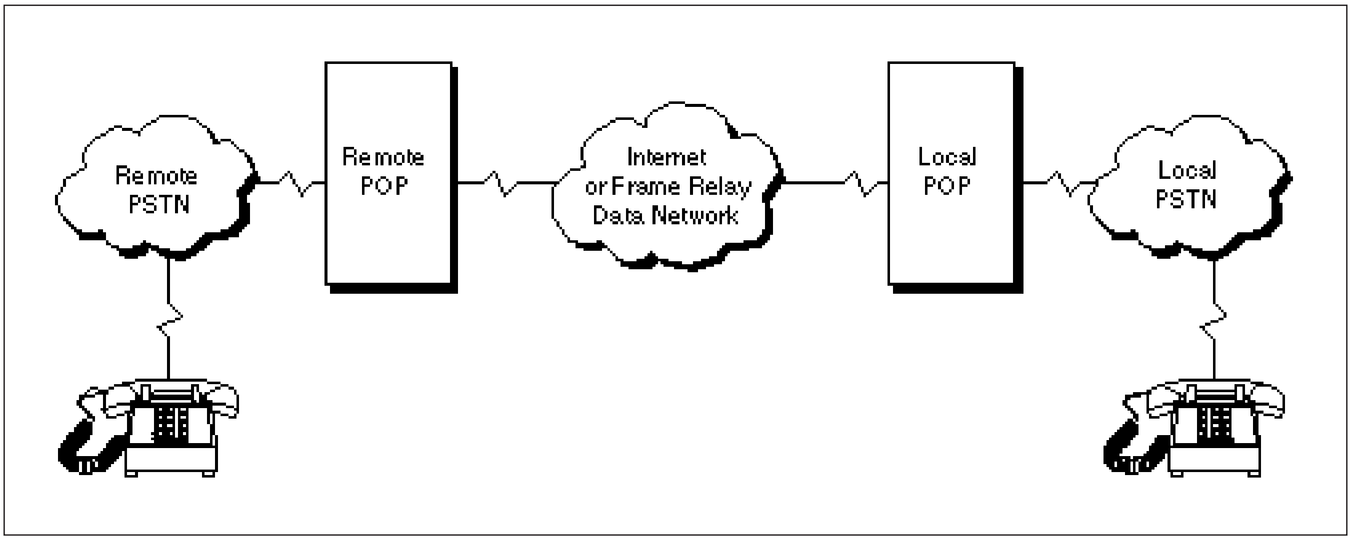

Figure 2. Phone to Phone Service Over Packet Data Networks.

Inside the POP

Telephone calls made to the service provider arrive by *T1 (North America) or E1 (Europe and much of the rest of the world)* digital telephone connection. A T1 line handles up to 24 simultaneous telephone calls using TDM multiplexing. An E1 handles 30 phone calls. The T1/E1 connection is terminated in the IP Voice Gateway on a T1/E1 telephony board (see Figure 3 below). The IP Voice Gateway is an industrial-grade Pentium PC with plug-in telephony boards connected by either the MVIP or SCSA local TDM bus.

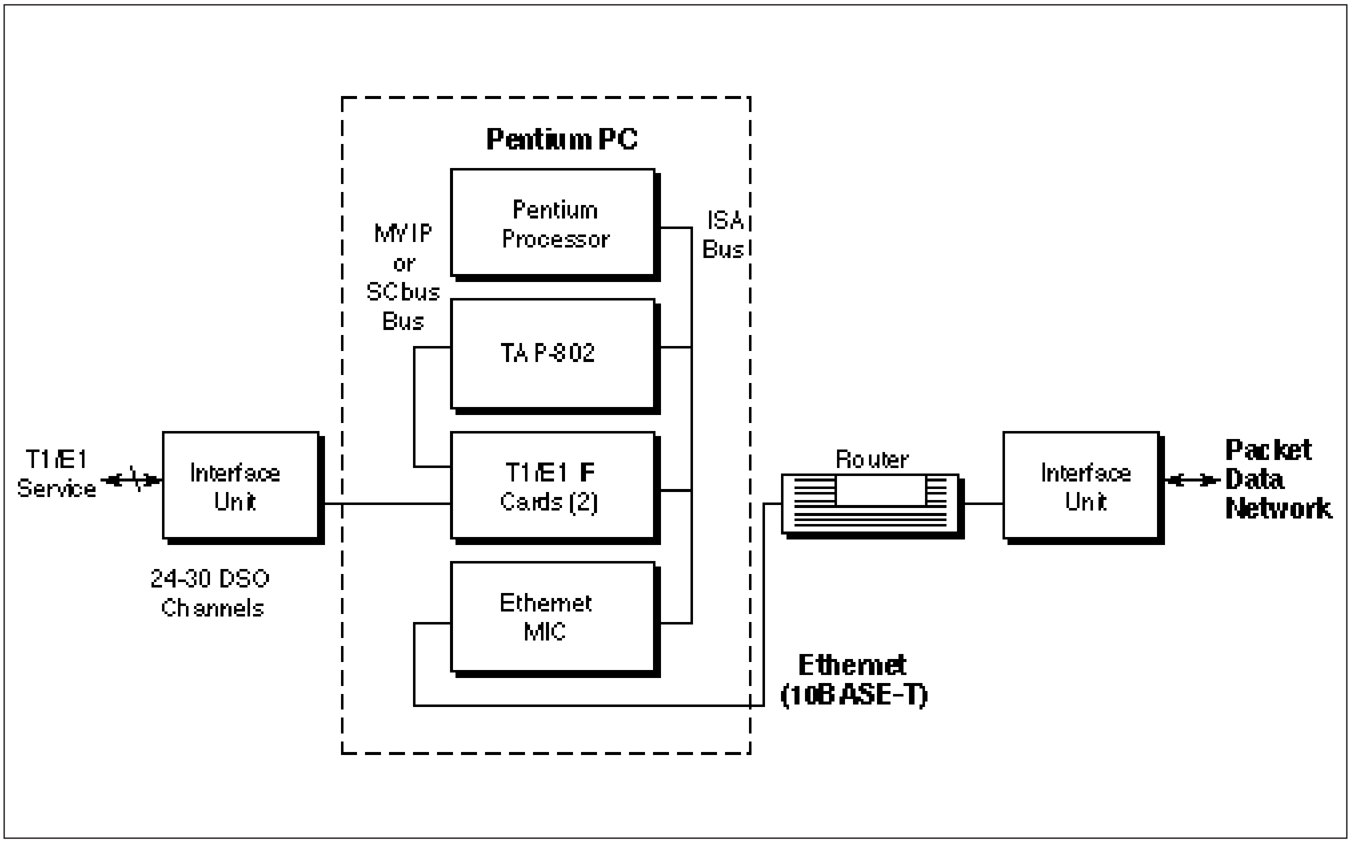

Figure 3. IP Voice Gateway for Service Providers.

The voice data is passed from the T1/E1 board through the local TDM bus (MVIP or SCbus) to the TAP-802, where it is compressed. The compressed voice data is then sent to the Pentium processor via the ISA Bus where it is packetized and sent to a router through an Ethernet connection. The router places the voice data on an IP packet data network, either the Internet or a Frame Relay network.

DSP Software

The compression software on the TAP-802 is more than just a vocoder. It also includes an echo canceller, *Voice Activity Detector (VAD)* and DTMF tone detector/generator. The DSP Speech Compression System that has been ported to the TAP-800 family of DSP resource boards is shown in Figure 4 (next page).

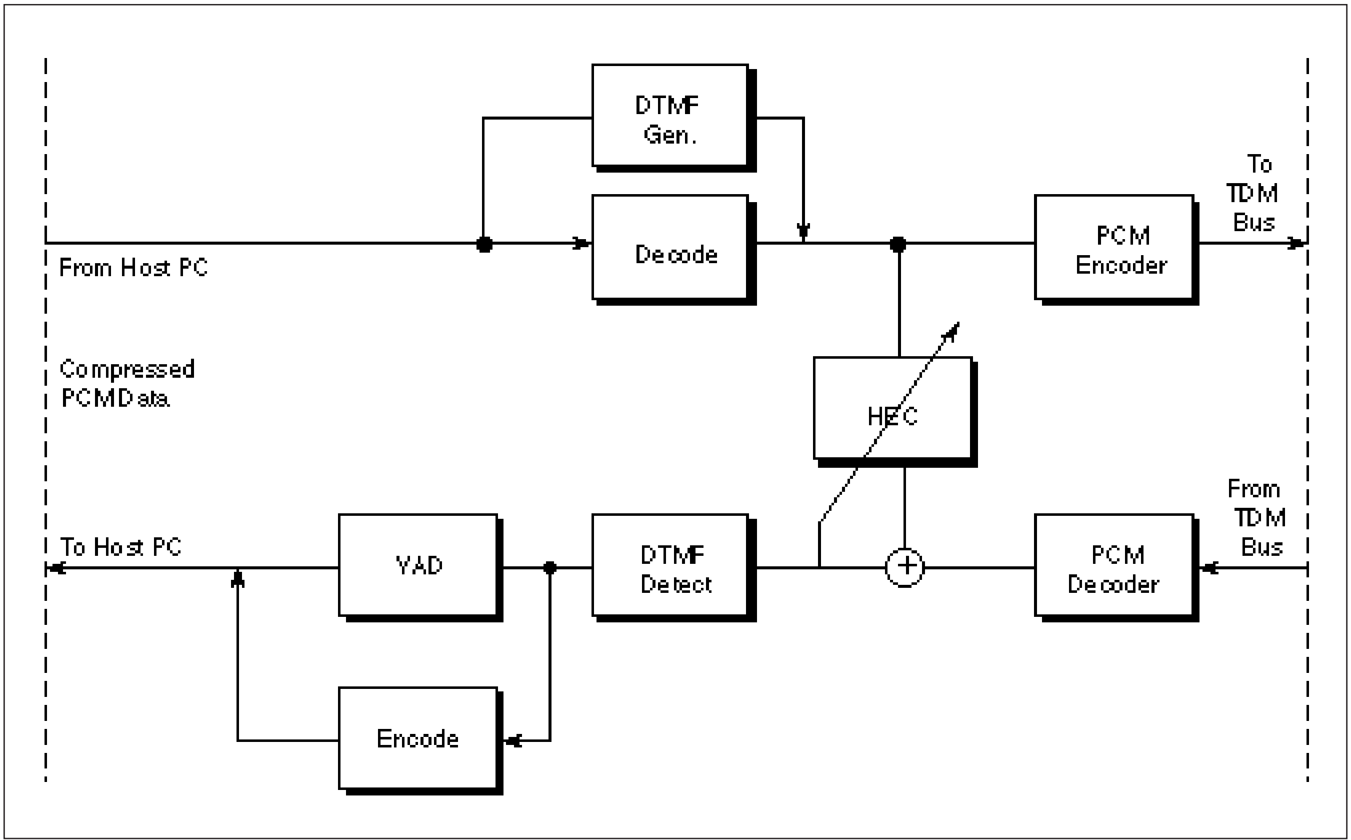

Figure 4. DSP Compression System Block Diagram .

IP VOICE OVER INTRANETS

A business with remote offices can use its own intranet facilities for IP Voice services to save on long distance telephone changes. The savings will be especially dramatic for traffic to international facilities. Any company with access to a packet data network including Internet access is a candidate for installing an Intranet Voice Gateway.

The Intranet Voice Gateway

An Intranet Voice Gateway system employing the TAP-802 DSP Resource Board is shown in Figure 5. In this system, the Intranet Voice Gateway is a PC attached to a local PBX. The connection from the PBX to the gateway is completed by a commercial voice board with either an SCSA or MVIP bus interface. The user calls an internal telephone (PBX) extension connected to the gateway and is greeted by a voice prompt that is provided by the voice board. The user is asked to enter an access code and the phone number for the call, using a telephone key pad. The gateway then compresses the voice data in the TAP-802, using a DSP Compression System, packetizes the compressed voice data on the host, and completes the call through a packet data network to another gateway at another company location. *The process is the same as in a Service Provider system except that the Gateway is accessed from a PBX internal telephone instead of a local telephone call to a Service Provider.*

HOST SOFTWARE

Packetization Software

The compressed voice data is packetized in the host PC using the Real Time Protocol (RTP). This thin protocol at the Transport Layer provides time stamping capability for audio transmissions over packet networks, including Ethernet LANs, the Internet and Frame Relay. The RTP utilizes the User Datagram Protocol *(UDP)* and the Internet Protocol (IP). UDP is a fast transmission protocol that omits error checking and flow controls in the routers used in the network. It is unreliable to the extent that some packets can be lost, but with voice a few lost packets won't be easily noticed. UDP has a much shorter delay than other protocols and is required for real time voice. However, the transmission of the telephone number and call control data must be completely reliable. For this data, the Internet Transmission Control Protocol *(TCP)*, which does complete error checking, is used.

Standards

Considerable technical work is being done to make the Internet and other packet networks more appropriate for real time data such as *voice and video*. To improve the delivery of real time data, the capability of prioritizing traffic is being added to routers and switches. Control for prioritization will be provided by the Resource Reservation Protocol (RSVP), which is being standardized by the Internet Engineering Task Force (IETF).

Other standards involved in IP Voice networks come from the International Telecommunications Union's (ITU's) H.323 suite of specifications *for video conferencing* over IP packet networks. Of particular interest are the H.245 call control specifications and the vocoders, which include G.711, G.722, G.723, G.728 and G.729. *In principle, interoperable IP Voice systems will negotiate an appropriate vocoder from the list of standard vocoders*, much as modems negotiate a common transmission rate. New vocoders are still being introduced and it isn't yet clear what the final list of standard vocoders will be.

Other Host Software

To complete the Voice Gateway system, the host has to provide an Interactive Voice Response (IVR) application, a Directory of Gateways and a Monitor/ Billing system.

- **IVR Application**. The voice prompt and access/code entry dialog that takes place at the beginning of each IP Voice session is a simple IVR application. The software that comes with the voice board in the Intranet Voice Gateway and with the T1/E1 telephony boards provides the tools for building this part of the application.

- **Directory of Gateways**. The Directory of Gateways is a database where telephone numbers are mapped into IP addresses for the various Voice over IP Gateways in the network. For today's systems, where the number of gateways in a particular Service Provider network or corporate intranet is small, this is a simple task. When standardization makes many systems compatible, this software will also have to be standardized.

- **Monitoring/Billing Software**. The monitoring/billing software can also be simple for small systems but larger systems will require SNMP compatibility and standard database interfaces for billing software packages.

IP Fax and IP Voice

Another new technology gaining acceptance is Fax over IP networks. Like Voice over IP networks, Fax over IP is a cost-reduction strategy for long distance telephone bills. It is estimated that many corporate fax machines cost $7500 per year to operate because of long distance telephone charges. Fax modems use the same DSP board technology as voice compression systems and can co-exist on the same multi-processor DSP Resource Board. Soon these capabilities will combined into the Voice/Fax Gateway where both voice and faxes will be given least-cost routing through either Service Providers or the corporate intranet (see Figure 5).

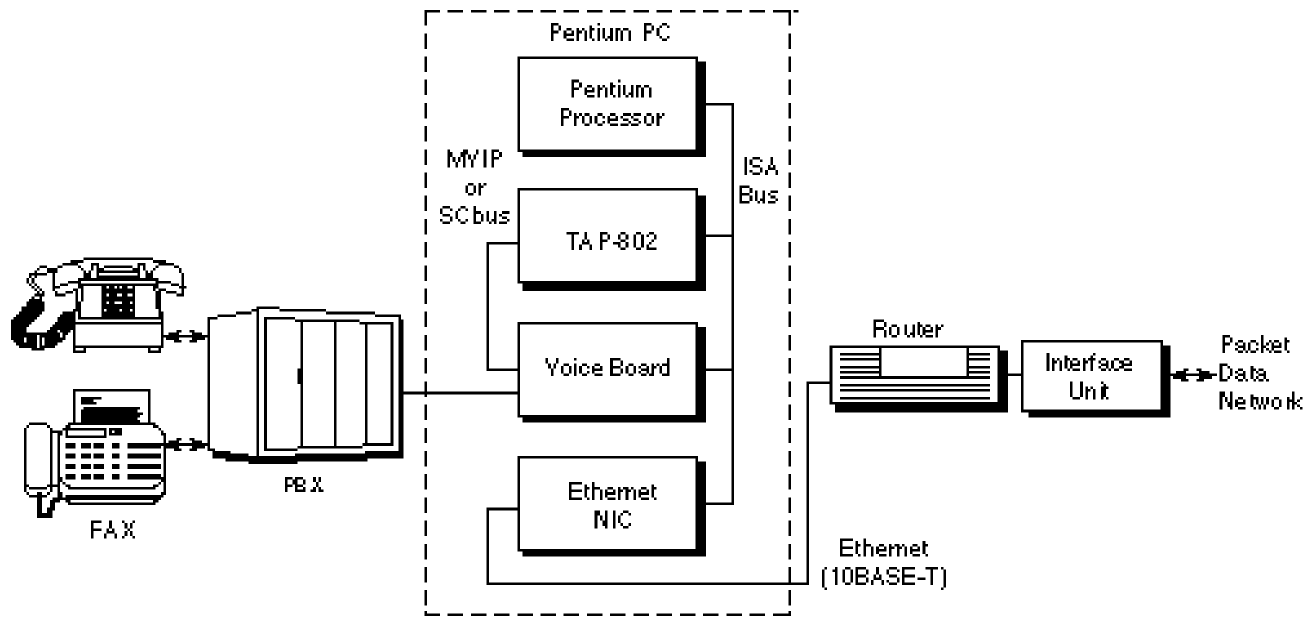

Figure 5. Intranet Voice/Fax Gateway.

Voice Over IP Networks Wrap Up

Whether over Circuit Switched Data Networks, Packet Switched Data Networks, Frame Relay, Intranets or the Internet there are a variety of situations where carrying compressed voice over IP networks can save money – or make money. There are also additional features and capabilities that a Voice over IP gateway can offer. Analogic's TAP-800 family of DSP Resource Boards can significantly reduce time to market for developers and integrators who are building gateways.