Cost-effective packet switched telephony

by Elke Jahn, COO, Pandatel AG

The benefits of transmitting voice and data in the same network, such as with VoIP, for example, are widely accepted. Notwithstanding the attention paid it, VoIP is not the best solution for many applications. A Circuit Emulation Service over Packet Switched Network, or CESoP, is a low-cost and easy-to-implement technology for transporting TDM data streams via packet-based networks. CESoP is a packet-based network that emulates the behaviour of a TDM connection and permits the connection of existing terminal equipment.

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Historically, voice and data networks have been treated separately, even to the present. Since there were significant differences in the requirements for each network, there was little demand to merge the two worlds into a common system. Even providers of both services saw little reason to eliminate this strict division.

Times have changed, and so have the requirements. In the past, networks were used predominantly to transport voice. Today they are mainly used to transport data. In the past, voice telecommunications accounted for 90 per cent of the traffic. Today, voice has dropped to about 10 per cent and data communications, in compensation, has risen in proportion. This is due in good part to the Internet.

In Germany alone the number of Internet users is estimated to be 35 million, and fast, affordable, broadband connections are becoming increasingly popular. At the end of September 2004, the German telecom provider T-Com welcomed its 5 millionth T-DSL customer. According to estimates by experts, more than 73 million DSL connections were in use worldwide by the middle of 2004.

Another trend has made an enormous contribution towards the increase in data volume. In the past few years, more and more companies have set up local area networks, LANs, used for digital communication, either for email or to exchange data with branch offices. In addition, a growing number of business processes now use Internet technologies. The growing availability of low-priced components for setting up LANs, such as network interface cards, routers and switches, favours this development.

Data using voice networks

Despite the increase in data volume, network providers are still achieving somewhat higher sales with voice services than with data services, for example through the installation of a simple telephone connection to a primary multiplex connection for operating private automatic branch exchanges in companies. It is therefore no wonder that they do not want to lose these lucrative sources of revenue.

Network providers are thus faced with the wish to offer two different groups of services that have different demands upon their networks. Operating two separate network structures is not cost-effective, since this requires high investment costs and high operational costs. It is better to merge the two worlds, although originally designed for different requirements and, thus, based on different technologies.

Voice networks largely use line-oriented techniques such as PDH (Plesiochronous Digital Hierarchy) and SDH (Synchronous Digital Hierarchy). These technologies transmit voice with very high quality, since there are only very small and, generally speaking, constant delays and the required bandwidth is reserved exclusively.

The individual voice-connections are transmitted using time multiplexing TDM (Time Division Multiplexing), in which no overlapping of the voice channel occurs. The transport of the correct clock information, to precisely allot the transmission time slots, is an essential part of TDM systems, as well as an integral part of the transmission procedure in both PDH and SDH.

In contrast to this, the transmission of data packets leads to the overbooking of the available bandwidth, which can lead to fluctuating delay times and packet losses. Since the packet network is not connection-oriented, the sequence of the data may change when individual packets use different routes.

The distribution of highly precise clock pulse information is not necessary, since there are no continuous data streams; rather, the focus is on the fast and reasonably priced transport of large quantities of data. In networks transmitting packets, also called Packet Switched Networks or PSN, Ethernet technologies and the Internet Protocol (IP) predominate today. Packet data transmission handles all types of data, and has proven its worth as a universal platform.

Depending upon the infrastructure available, different methods have been developed to transport data using voice networks or voice using data networks. Initially, attention was focussed upon technologies such as ISDN, which allowed the transmission of computer data using the widely available voice networks.

Today, there are broadband services for the direct transmission of data using TDM networks, e.g. Packet over Sonet (synchronous optical networks).

Voice over IP – the pioneering technology

The most widely known technology for transporting voice using data networks is Voice over IP, or VoIP, also generally known as Internet telephony. How does VoIP work? The voice stream is divided up into packets, converted into the IP (Internet Protocol) format and tagged with data that identifies both the sender and the receiver, and then fed into the data network. When the separate packets reach the recipient, they are transformed back into a continuous stream of data.

What is difficult with VoIP, however, is the necessary signalling between the transmitting and the receiving units in order to achieve 'normal' telephone behaviour. If you

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wish to telephone from the VoIP network into the existing public telephone network, gateways and signalling converters are required. However, if the receiver has VoIP-capable telephones, the data stream can be fed directly into the receiver's LAN, to which these devices are connected.

VoIP is already being successfully used today in many applications, from the simple transmission of conversations between two subscribers to professional customer management systems.

Internet telephony is not only suitable for talking, very cheaply, to colleagues and workers all over the world, but also, for example, to set up a low-cost hotline. If, for example, a customer calls the hotline of a company in Germany, his call is fed into the Internet, forwarded to a call centre abroad and answered by the hotline staff there.

For many service providers, the transmission of voice using data networks has adequately proved its practicability and suitability for everyday use. Many providers then had the long-term wish only to offer services with packet networks, which are suitable for transporting data as well as voice information.

In the past few years, therefore, numerous efforts have been made to use the technologies available in the LAN sector in the long-distance sector as Wide Area Networks, or WANs, as well as in the urban sector for Metro Area Networks, or MANs. For example, the IETF (Internet Engineering Task Force) defined MPLS, Multi Protocol Label Switching, and the Metro Ethernet Forum defined new services.

Packet Switched Networks (PSN) – versatile, flexible, reasonable

The switch to PSN is especially interesting from a cost point of view. These are networks that transmit packets using existing bandwidth more efficiently by means of equipment that is cheaper to acquire and cheaper to run.

Moving over to packet switched networks cuts costs for network operation (Opex), while the cost-effectiveness of the existing bandwidths increases.

The service providers' customers thus benefit from favourable connection costs for Internet telephony. Since Internet phone calls are transmitted via data networks, only the costs for the Internet connection, either dial-up or always-on, are charged even for conversations around the world.

Furthermore, customers also benefit from numerous new services. For example, one receives calls from any location in the world through defined IP addresses. The call numbers are not tied to a given location and travel with the user.

If you are travelling with your notebook, you can still be reached by phone at your common number, whether you are in Munich, Paris or Rome – you only need a working Internet connection. Meanwhile, companies are increasingly aware of the many economic and practical advan-

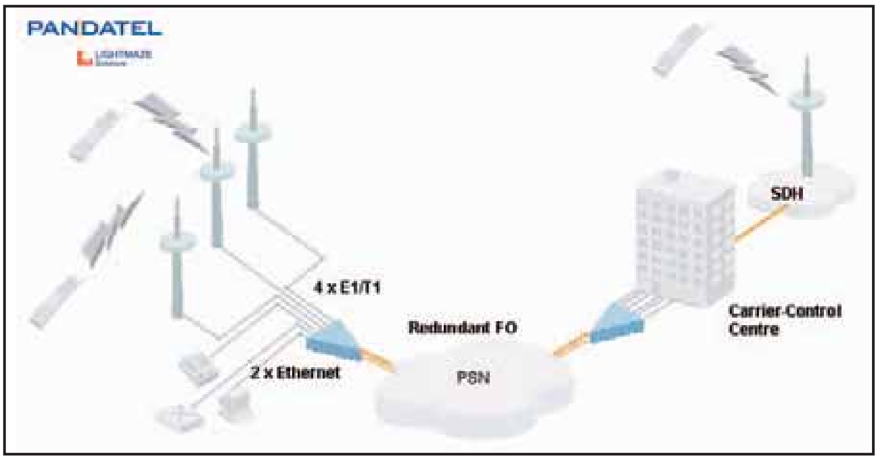

Figure 1: Ethernet ring topology with remote management access.

tages linked to Internet telephony, so they are becoming more and more interested in the new technology.

Accordingly, analysts are prophesying lucrative times for the telecom companies that provide their customers with cheap telephony, and associated services, and features that the Internet provides.

The use of VoIP, however, calls for investments. Existing telephones or telephone systems, for example, must be replaced by a complete new VoIP-capable system, or a so-called hybrid system, which combines the new and the old telephone worlds in one system. Several obstacles must be overcome in the process. For example, the products of different manufacturers are frequently incompatible with each other because of numerous competing VoIP standards (H.323, SIP, MGCP, Megaco/H.248, SCTO, etc.).

Furthermore, hybrid systems are based on two platforms, the voice network and the data network, each of which has to be operated and maintained parallel to the other. This pushes up the operational costs.

Migration to a straight VoIP architecture improves matters. In this case, however, the network providers and their customers are dependent on each other. If network providers would like to switch completely to VoIP, customers, often at considerable cost, must go with them.

Conversely, if customers wish to switch systems, network providers have to install different connections for them, which again leads to increased costs. The consequence is that many companies and network providers stick to a hybrid solution to avoid these additional costs.

Circuit Emulation Service over Packet (CESoP)

Notwithstanding the attention paid to it by the industry, VoIP is not the only option for sending voice data via a packet-based network. In fact, VoIP is not the best solution for many applications.

In addition to VoIP, there is a cost-saving and easy-to-implement technology for transporting TDM data streams via packet-based networks

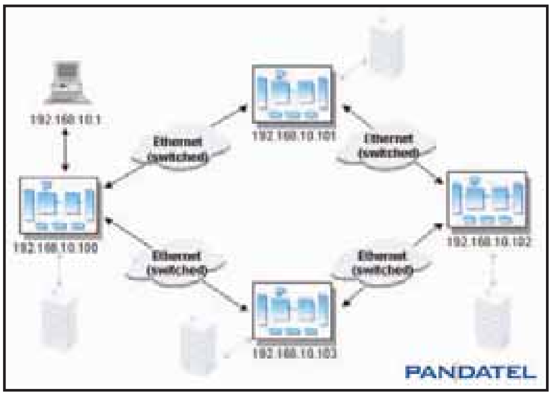

Figure 2: A suitable TDM multiplexer is ideal for setting GSM infrastructures.

called Circuit Emulation Service over Packet Switched Network (CESoP). CESoP emulates circuit-switched services in a packet switched environment. CESoP, unlike VoIP, does not use IP technology, and meets the requirements of Metro Ethernet networks.

Circuit Emulation originally comes from the ATM (Asynchronous Transfer Mode) world. It was developed for the transmission of TDM data streams using ATM networks. However, CES over ATM generates relatively high amounts of overhead (15-20 per cent). CESoP is more efficient. Since larger packets are possible in PSN than is the case with ATM (53 bytes per cell), the proportion of overhead is considerably lower.

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Because of its low cost and high efficiency, CES is an attractive voice service transmission alternative using packet-based networks.

Another point in favour of CESoP is the fact that the final customers do not have to replace existing equipment if the provider changes his network, but can, nevertheless, benefit from the more favourable charges.

The CESoP concept works with a simple, yet effective, idea: a packet-based network that 'tunnels' a TDM connection, so that the TDM devices at either end do not 'notice' that they are connected using a PSN. The packet-based network emulates the behaviour of a TDM connection.

Summary

CESoP is a cost-effective solution for transmitting TDM services using reasonably priced packet switched networks. The concept of 'circuit emulation', upon which the CESoP standard is based, has already proved its worth in the ATM world for many years, and is by now highly sophisticated.

CESoP has various advantages compared to VoIP:

√ Low-cost implementation of voice services, since with CESoP, legacy transmission networks, e.g. the existing PSTN, does not have to be modified;

√ Reduced lag, since CESoP packs the data faster than conventional VoIP systems;

√ More efficient utilisation of available bandwidths due to reduced overhead;

√ Greater flexibility, since voice and data transmission is possible using IP networks as well as Ethernet and MPLS networks (Multiprotocol Label Switching);

√ Powerful CESoPSN-equipment now available offers companies and network providers considerable savings;

√ Access costs are reduced, since dedicated lines are no longer required;

√ Investment costs (Capex) are low, since existing equipment such as switches or telephone systems can still be used;

√ Operating costs do not change, since it uses the existing data infrastructure without modification.

Network providers also benefit from additional sales by being able to offer the new services made possible by transporting data with their existing packet switched networks. CESoPSN-equipment is a low-priced, cost-effective, alternative to VoIP for the transmission of voice using data networks.