Packet-switched telephony : the "electric toy" of the 21st century?

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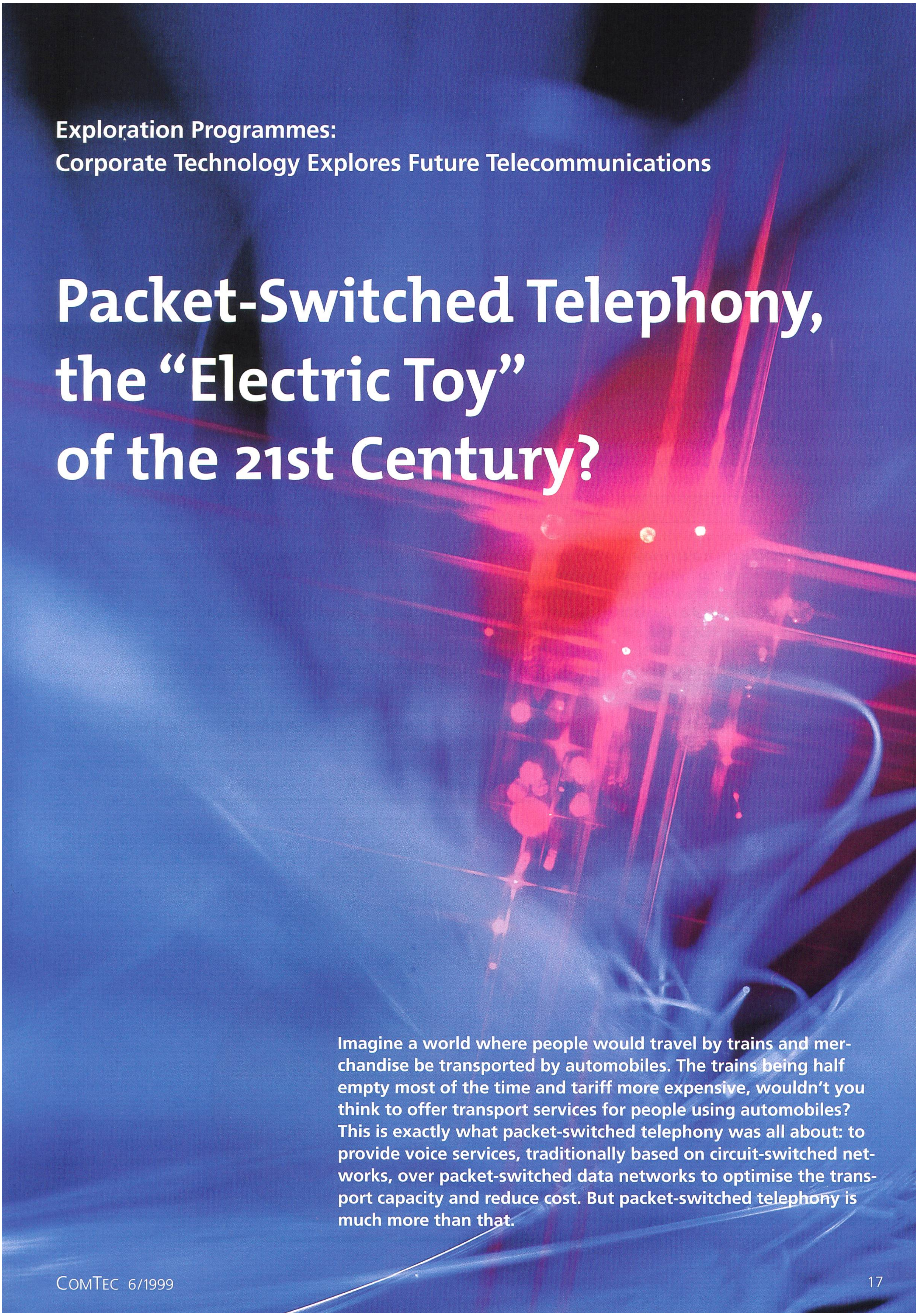

Exploration Programmes:
Corporate Technology Explores Future Telecommunications

Packet-Switched Telephony, the "Electric Toy" of the 21st Century?

Imagine a world where people would travel by trains and merchandise be transported by automobiles. The trains being half empty most of the time and tariff more expensive, wouldn't you think to offer transport services for people using automobiles? This is exactly what packet-switched telephony was all about: to provide voice services, traditionally based on circuit-switched networks, over packet-switched data networks to optimise the transport capacity and reduce cost. But packet-switched telephony is much more than that.

Exploration Programme "Voice Services Opportunities" investigates the potential use of new technologies influencing the evolution of traditional voice services. The focus of the programme is put on Internet telephony, web driven voice services, as well as automated operator and information services using advanced voice processing technologies (speech recognition, speech synthesis and speaker verification).
With Exploration Programmes Corporate Technology is exploring telecommunication technologies and new service possibilities within projects having a long-term focus of 2–5 years to build up expertise enabling active business innovation support.

With computer communications and the Internet in particular, data traffic will clearly surpass voice traffic as the primary consumer of bandwidth, so telecommunications service providers are nowadays looking to

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change their networks from a technology called "circuit-switching" which was created for voice services to "packet-switching", created and suitable for data communications services. Not linked any more to the closed world of traditional voice networks, using the flexibility of data networks and the intelligence of computers, packet-switched telephony will start a new era of multimedia real-time communication. Three worlds: voice networks, data networks and information technologies (computers) are converging, as illustrated in figure 1, to provide packet-switched telephony.

Traditional circuit-switched networks reserve an entire circuit for a call, allowing only one call to be placed there at a time. Packet-switched networks, such as the Internet, break up information such as a person's digitised voice or a fax into small pieces of data, send them over the network and reassemble them at the end. That way, several calls can go over the same circuit at once, reducing the cost of a circuit.
The primary reason to use data networks for carrying voice services was network optimisation and cost savings, but with the Internet, multimedia communications and hence the need for a unified network have emerged. Voice services (telephony, fax, modem) provided on this network are starting a new era in the world of telecommunications solutions, services and products.

But beyond the first challenge of carrying real-time voice services over data networks, a lot of work has still to be done before packet-switched telephony will become the mass market service that is the telephone today. The principal obstacles are the absence of end-to-end quality of service, scalability and standardisation. Starting from its discovery, this article shows the evolution of telephony towards what will probably come in the 21st century and covers some of the base technologies used today in traditional circuit-switched and packet-switched telephony.

Programme Scenario

The basic lead question for the Exploration Programme Voice Services Opportunities relates to the possible migration paths of today's circuit-switched network based voice services towards packet-switched network based services.

The "Electric Toy" History

To understand the present and the future, it is important to understand the origin of a technology and its evolution.

The Discovery of the Telephone

The Idea

The first idea of telephony takes its origins in 1854 when a French, Charles Bourseul (1829–1912), wrote an article in the magazine "L'Illustration de Paris" about transmitting speech electrically. In that important paper, Bourseul described a flexible disk that would make and break an electrical connection to reproduce sound. His construction was similar to the future microphone, but the construction of a receiving part converting the electrical current back into sound failed. Bourseul never built another instrument or pursued his ideas further.

The Name

The first electric telephone was constructed by a German physicist, Johann Philipp Reis (1834–1874). He was experimenting on electricity and was working on hearing aids which led to his

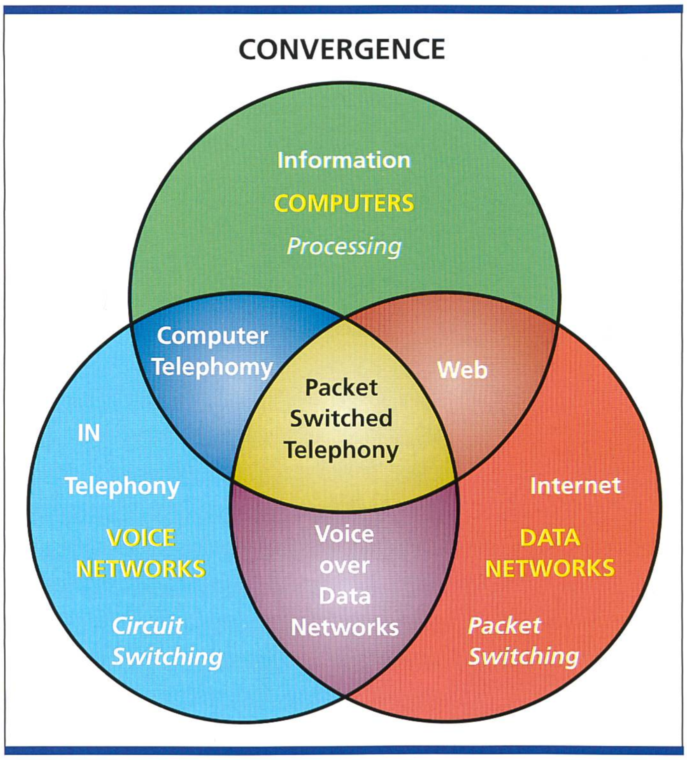

Fig. 1. Packet-Switched Telephony. The convergence of three worlds: Voice, data and computers are converging to provide packet-switched telephony.

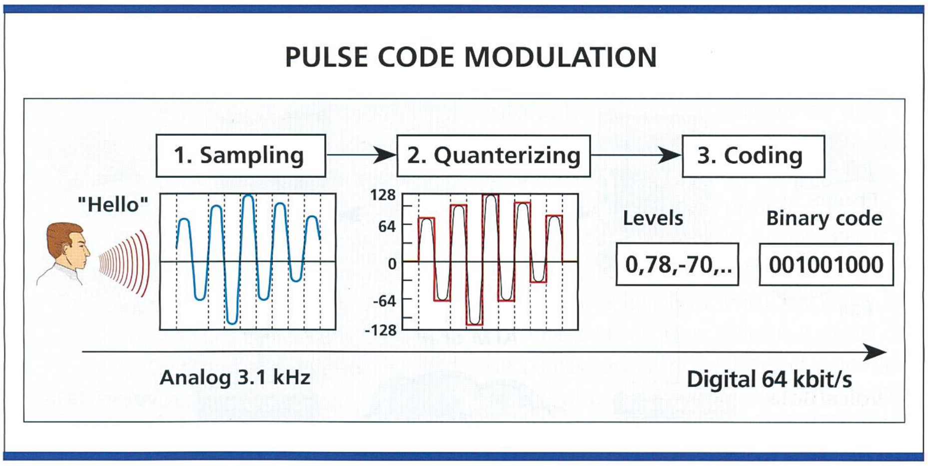

Fig. 2. Pulse Code Modulation (PCM): The digitising process. PCM is the base technology for the digitisation of voice used in traditional circuit-switched telephony networks. For packet-switched telephony several mechanisms are used to compress the voice stream during the coding phase to get as little as 5,3 kbit/s.

interest in electrical transmission of sound. In 1861, he designed a system for the transmission of musical sounds which he called "telephone". This was the first attempt to convert acoustic energy to electrical energy and back again.

The Invention
It was Graham Bell who finalised the idea of telephony and patented it in 1876. One year later, he founded Bell Telephone Association, started to lease telephones and soon reached a rate of 1000 a month. His company delivered and installed 50 000 telephones within the first three years and was soon the world's largest telephone company known as the American Telephone and Telegraph Company (AT&T).
The discovery of the telephone was at first greeted sceptically. The head of the telegraph monopoly in the United States, the Western Union, labelled it an "electrical toy". But before the turn of the century every major city in the world had a telephone service and the telegraph companies were competing for ownership.

The Telephone Exchange
The crucial element in the success of the telephone was the invention of the telephone exchange. Without exchanges, telephones could only operate if they were wired to each other.
In the beginning these exchanges were manual. All employed operators were mostly young women, who, on request, connected the caller's line to another line and noted the call for later billing.

There was several types of exchanges, each finding its own way of meeting the requirements of a public exchange. Eventually, exchanges were connected to each other by trunk lines. While it was then fairly simple for operators to connect calls to other exchanges, it was practicable only if long-distance calls remained a small proportion of the traffic. Accordingly, a surcharge was imposed, making long distance calls much more expensive than local calls.

The Automatic Exchange
The automatic telephone exchange, also called "switch", was invented in 1891 by an American funeral director (Almon B. Strowger) as a response to the unfairness of the manual operators. He created the first automatic exchange using electro-mechanical selectors.
After World War II, with electronics technologies, the electro-mechanical selectors have been replaced by relays, and since the late 70's , most of the electro-mechanical or electronic automatic exchanges have slowly been replaced by digital computerised exchanges.

The Control Part of the Exchange
Any method of automatic switching needs a control system to establish the communication between the users. Its job: to direct the connection and disconnection of lines and perform other subsidiary tasks, such as checking the status of called lines.
In Strowger exchanges, control was built into the work of selectors. As each digit is dialled, a selector connects a path to another selector, which is then ready to accept the next digit. Then, however long the call, the selectors remain in place until it is over. Since bi-motional selectors were complex, expensive and rather fragile, it was better if they could set up other calls while the first call continued. For that, the control system needed to be separate from, rather than built into, the switching mechanism.
That is the principle of common control, called "signalling": the basis of all modern exchange systems. This signalling has initially been performed using electrical impulses, later using audio multiple frequency codes and today using computer messages.

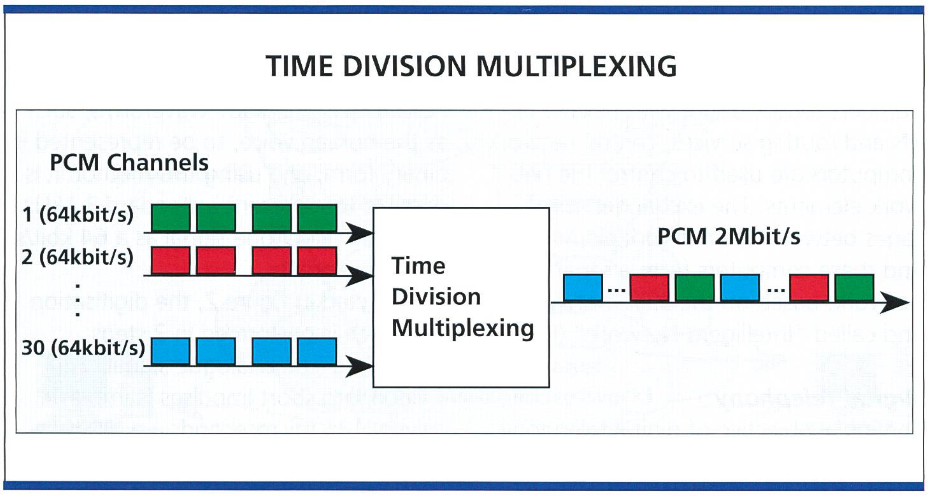

Fig. 3. Time Division Multiplexing (TDM). TDM is the technology used in circuit-switched networks for the transmission of up to 30 voice calls on the same physical link, reserving a 64 kbit/s channel for each communication. Unlike data networks, empty time slots are not used for other data.

Today's Telephony

Some 120 years after Bell's discovery, the telephone nowadays still remains the world principal system of telecommunications. The telephone network not only serves as a means for transmitting articulate speech and other sounds at a distance via the medium of electric waves but its circuits carry facsimile signals (fax) and computer data (modem) in a form that can be fed directly into processing devices. Telephone circuits enable also to connect computers to the Internet. As a consequence, telephone systems have become the most used telecommunications system in general.

The Telephone Network

The core of Swisscom's national telephony network, build up with almost 1000 exchanges (local concentrators included) and 16 500 km of optic fibre, has reached complete digitisation which started in 1981. Only the access to the network, the local access line, still uses analogue technology for the traditional fixed public switched telephone network (PSTN). With the Integrated Services Digital Network (ISDN), the local access line is also digitised. In mobile telephony (GSM), the local access line is replaced by a digital radio link.

Today's telephone exchanges communicate with each other by exchanging messages to perform the necessary tasks such as establishing a circuit for a call. This exchange of messages between the network elements forms a separate network for the control of the transmission, the signalling network, using an international framework, the Signalling System 7 (SS#7).

For the creation of value added telephony services, such as the business numbers (0800, 0900) with flexible tariffs and routing services, central network computers are used to control the network elements. The exchanged messages between the network elements and these computers form a separate network, based on the SS#7 network and called "Intelligent Network" (IN).

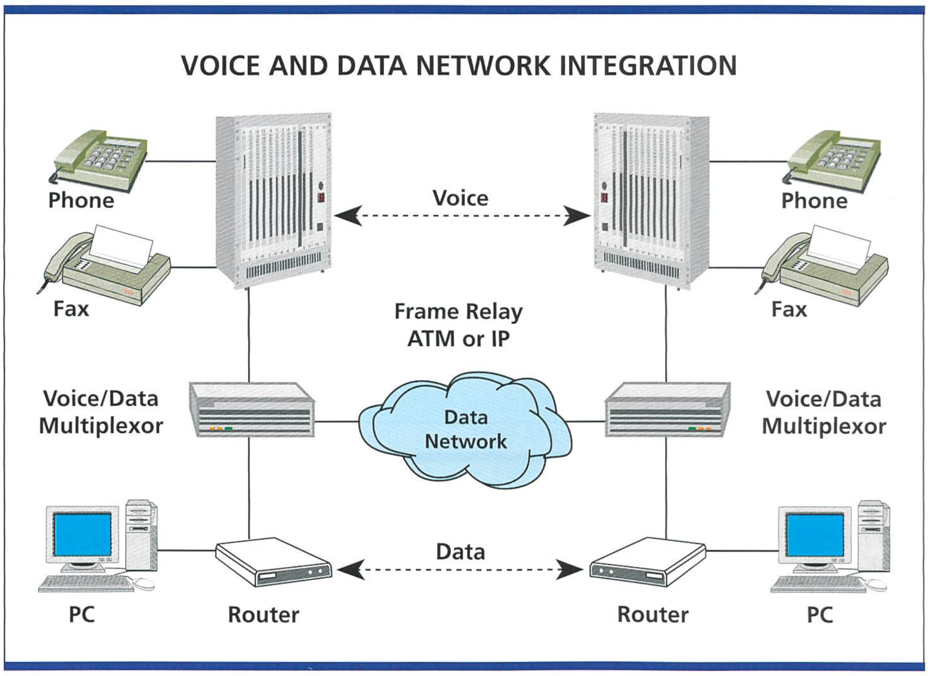

Fig. 4. Voice and Data Network Integration. Voice and data integration is usually used in corporate networks to transport voice and data communications over a unified data network, mainly for optimising and reducing the cost of their telecommunication infrastructure and services.

Digital Telephony

The initial objective of digital telephony is to improve the quality and cost of long distance communications, focusing on two directions:

- digitisation of voice calls enabling better regeneration along the communication path to guarantee the same voice quality end to end and
- multiplexing of several voice calls on a single line to more efficiently using the physical lines.

The digital telephony age takes its origin with the invention of the PCM modulation (Pulse Code Modulation) by Alec H. Reeves in 1938, who was working on Time Division Multiplexing (TDM). These technologies have been made financially affordable after World War II with the invention of the transistor by the Bell laboratories in 1948, but digital transmission systems began really to appear in the early 1970s.

Pulse Code Modulation (PCM)

PCM allows analogue waveforms, such as the human voice, to be represented in binary form, and using this method it is possible to represent a standard 3.1kHz analogue telephone signal as a 64 kbit/s digital bit-stream.

As depicted in figure 2, the digitisation of speech is performed in 3 steps:

- sampling (the analogue signal is divided into short impulses (samples) every 125 microseconds, i.e. 8000 times per second);
- quantisation (the level of each sample is measured, 128 positive values and 128 negative values are possible), and
- coding (each level value is coded into a binary code, 8 bits are necessary to code the 256 possible values of a sample). Every second 8000 × 8 bits = 64 000 bit = 64 kbit are transferred.

When a digital signal is received, the binary codes are read and the amplitude impulses are reconstructed to regenerate the original analogue signal.

Time Division Multiplexing (TDM)

Engineers saw the potential to produce more cost effective transmission systems by combining several PCM channels and transmitting them down the same copper twisted pair as had previously been

Abbreviations

IP	Internet Protocol
ISDN	Integrated Services Digital Network
MCU	Multipoint Control Unit
PCM	Pulse Code Modulation
PSTN	Public Switched Telephony Network
QoS	Quality of Service
RTP	Real-Time Protocol
RSVP	Resource reSerVation Protocol
SS#7	Signalling System 7
TDM	Time Division Multiplexing

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[2] "H.323 Technology Overview", Intel Corporation

[3] "Net Heads vs Bell Heads", Steve G. Steinberg, Wired Corporation

occupied by a single analogue signal. This technology is called TDM (Time Division Multiplexing). It enables the transmission of 30 conversations (32 channels with signalling and synchronisation) over a single transmission channel, using a type of multiplexing that combines data streams by assigning each stream a different time slot on the transmission line, as shown in figure 3.

Evolution Towards Packet-Switched Telephony

There are actually two trends which drive the evolution of current circuit-switched telephony services towards packet-switched telephony: Transporting voice over data networks and computer telephony.

Transporting Voice over Data Networks

With voice as audio signal being converted to data, there was just one step to make to transmit it over data networks. Magill and Cohen were the first to explore the possibilities of transmitting voice over the packet-switched networks in the early 1970's.
Nowadays this is done for two reasons: the first one, as shown in figure 4, is the use of data networks to substitute the expensive telephony leased lines, the second one is the integration of voice and data networks into a single network. Most of the time, that is done by multiplexing voice and data communications between two points over a data network. The data network can be either local (LAN: Local Area Network) or widely dispersed (WAN; Wide Area Network).
Transporting voice over data networks has its strongest application in the WAN area, because long distance telephony and fax toll rates are much higher than long distance data communication rates. Telephone calls (and fax calls treated equally by the network) are converted into data, placed on a wide area data network (usually a Frame Relay, an ATM or an IP network) and transported like other data traffic to a remote location. The first difference with the transport of voice in traditional telephony is that in order to use less bandwidth, voice is compressed using as little as 8 kbit/s, to be compared with the 64 kbit/s needed by the phone network. The second difference is that in the phone network the bandwidth is allocated and reserved for the duration of the call, which is not the case with data networks, where the unused bandwidth can be allocated to other users.

Computer Telephony

The basic idea is simple: link a computer to the telephone system to enhance the whole process of making and receiving calls. By providing open interfaces to control the telephony equipment at customer premises, computer telephony enables the development of several applications to enhance the telephony service.
Call Centres are actually the first solutions using computer telephony on an industrial scale. This type of solutions enables to quickly distribute a call to the first available operator, or more efficiently, using an interactive voice response system, to query what the caller wants and then route the call to the person with the most appropriate skills.
Instead of using a computer to control telephony services and a phone to use them, computers can be used for both tasks exchanging voice and data using the same network, which leads therefore to packet-switched telephony.

Internet Telephony

Beyond the basic idea of transporting voice over data networks just for economic reasons, the idea of packet-switched telephony is to recreate the equivalent of telephony services on data networks with all the appropriate mechanisms to control the establishment of real-time communications.
"IP telephony" is actually the first available packet-switched telephony service. The term "IP telephony" and its related "Voice over IP" technology have been broadly applied to a family of applications where real-time voice communication occurs over a packet-switched network based on the Internet protocol.
As shown in figure 5 several types of IP telephony are possible: "PC to PC", "PC to Phone" or "Phone to PC" using one gateway, and "Phone to Phone" using two gateways.
Choosing IP networks for the foundation of packet-switched telephony is very attractive. But while packet-switching is one of the strengths of IP networks, this

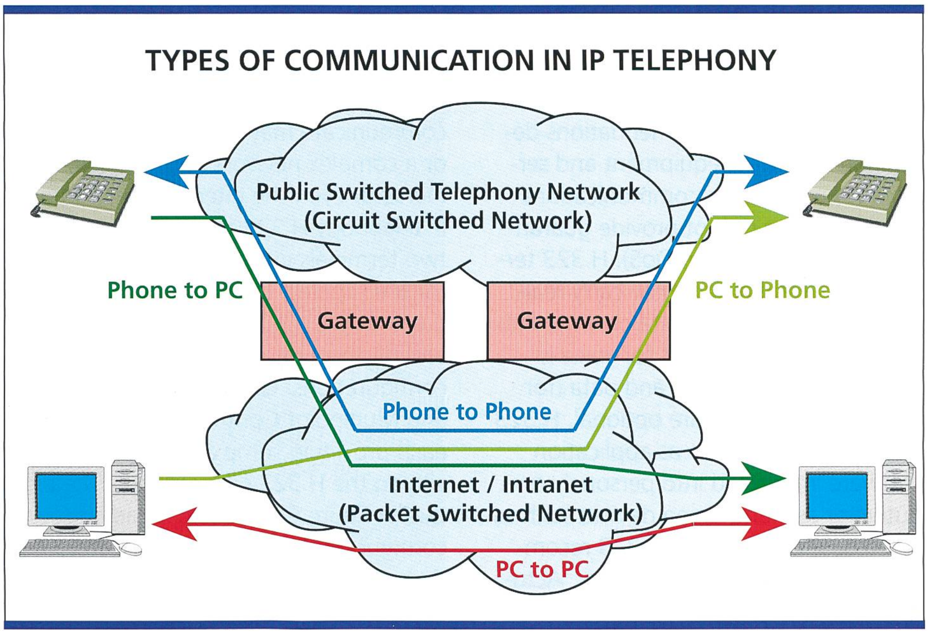

Fig. 5. Types of communication in IP Telephony. There are 4 types of communication possible in IP telephony. "PC to PC", "PC to Phone" or "Phone to PC" using one gateway, and "Phone to Phone" using two gateways.

approach also suffers from their weaknesses: Because packets are routed individually, they may follow different paths between origin and destination, and, due to the related individual delays along these paths, may arrive at the destination out of order – if they arrive at all. Such characteristics of the IP network are manifested in a phone conversation as broken speech and as jitter – the end result being a sound quality poorer than with traditional telephony. Current research into additional protocols (such as RSVP, Resource reSerVation Protocol and RTP, Real-Time Protocol) is aimed at eliminating these limitations associated with today's packet-switched telephony.

The Internet Telephony Standard H.323

IP telephony takes its origin in 1997 when some computer freaks implemented a computer software to make phone calls over the Internet. This application uses the specification of the audio part of a telecommunication standard initially created for video-conferencing, the H.323 recommendation. Other standards are emerging, but H.323 remains the most used one.
H.323 is the International Telecommunications Union – Telecommunications (ITU-T) standard for real-time multimedia communications and conferencing over packet-switched networks. It's an extension of the H.320 original standard defining video-conferencing over ISDN. Because it supports the set of audio/video coding/decoding recommendations for ISDN and PSTN conferencing, H.323 has become the de-facto standard for IP telephony.
H.323 is a set of recommendations describing terminals, equipment and services for multimedia communications over LANs that do not provide guaranteed quality of service (QoS). H.323 terminals and equipment may carry real-time voice, video and data or any combination of them. Support for voice is mandatory, while video and data (for shared applications) are optional. H.323 terminals might come as application software integrated into personal computers or as stand-alone devices such as video or IP phones. The H.323 recommendation is open and complex. Actually there are still interoperability problems between some products. But vendors are currently working together to solve open issues for the implementation of this standard.

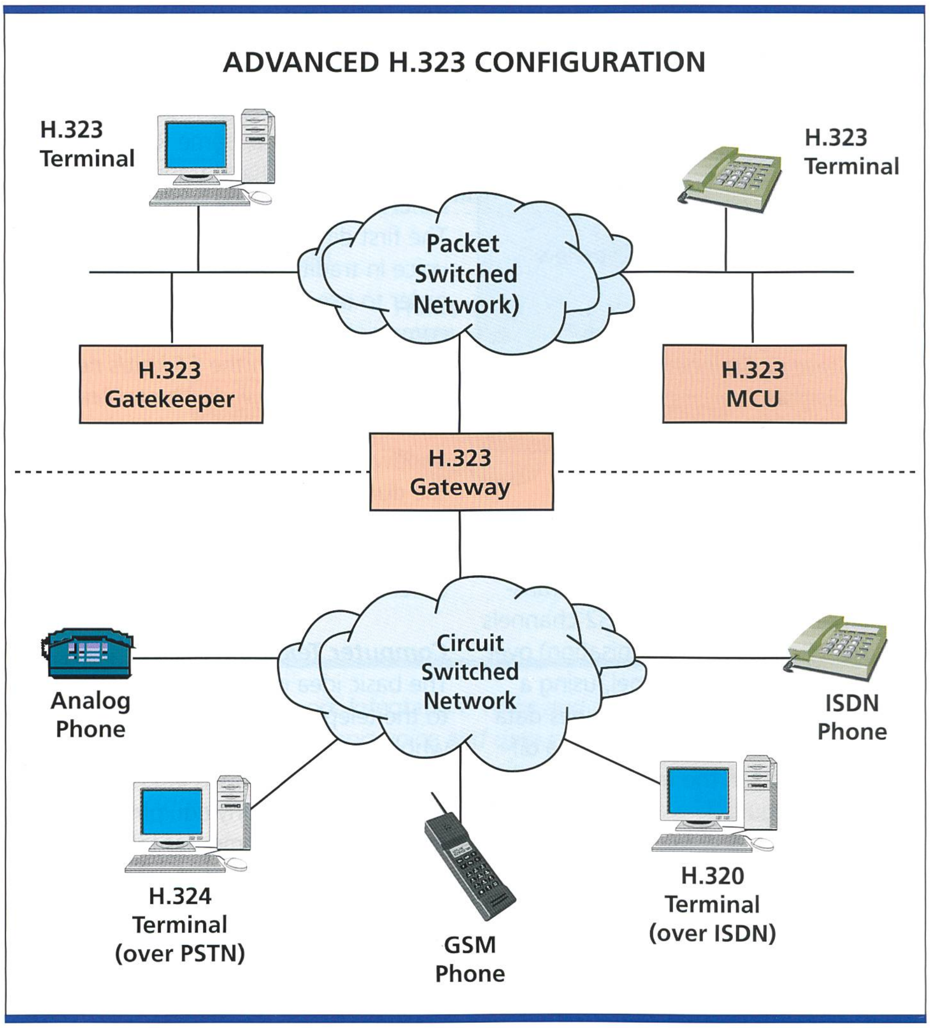

Fig. 6. Advanced H.323 configuration. Although for H.323 requires as little as two terminals interconnected with a LAN for the simplest configuration, the standards define a number of H.323 entities (Gateway, Gatekeeper, Multipoint Control Unit) and protocol interactions for advanced configurations.

H.323 Entities
The LAN over which H.323 terminals communicate may be a single segment or a complex network with multiple segments or even the Internet. Although for IP telephony H.323 requires as little as two terminals interconnected with a LAN for the simplest configuration, the standards define a number of H.323 entities and protocol interactions for advanced configurations: Gateways, Gatekeepers and Multipoint Control Units (MCU), each providing some additional functionality to the H.323 environment. The picture in figure 6 illustrates an advanced configuration including these entities.
Gateways provide the ability for H.323 devices to operate in heterogeneous network environments by enabling connectivity between H.323 terminals and terminals in other networks. The H.323 standard requires that gateways provide call signalling support, control channel messages, multiplexing, audio compression, and audio trans-coding.
Gatekeepers are software entities usually associated with a gateway. They provide interoperability services such as address translation and bandwidth management and may provide security services.
Multipoint Control Units (MCU) are hardware or software entities that are needed for multipoint conferencing. An MCU mixes all the audio coming from the terminals and switches video, based on some criteria such as the loudest person speaking.

Conclusions

Circuit-switched or packet-switched networks? The comparison of these two approaches makes it clear that circuit-switched networks, which can be compared to railways networks, offer a

proven, highly reliable solution for voice and data transmission, but at a price. On the other side, packet-switched networks, which can be compared to roads networks, are less expensive, especially for long-distance and international communications, where the difference between voice and data services pricing is significant. The most important factor is the charging structure itself, the flat rates charging scheme used for data services impacting on traditional methods of charging by distance, duration and time of the day, used for voice services. Data services however, for which packet-switched networks were made, do not suffer from packet-switched networks imperfections such as delay, delay variation and packet-loss. Real-time voice services, such as telephony, represent therefore a new and difficult challenge to such networks.
Using data networks to carry voice communications just to save money is a short term view. It is the statistical sharing of a potentially constrained resource – bandwidth – that leads one to believe at first thought that packet-switched telephony is more efficient, and, hence, less expensive than traditional telephony. The existence of an inherent cost advantage of packet-switched telephony is very debatable and not immediately clear. Although there is actually a window of opportunity for a possible cost arbitrage using packet-switched telephony, the future is elsewhere.

Outlook

Although initially one was led to think that packet-switched telephony would be an interesting technology for cheaper telephony, the future for packet-switched telephony is more in multimedia communications and solutions bringing real added value for the customers. Web driven applications (push to talk, click to fax), integrated voice and Internet call centre, multimedia conferencing, unified messaging, broadcasting of presentations and meetings, documents and applications sharing, distant learning, and network entertainment are applications which will provide this added value for packet-switched telephony to become the basis for the real-time communications of the 21st century.
The real advantage of packet-switched telephony is its ability to be integrated with other communication services, such as Internet services. The success of packet-switched telephony will really be there when the merging of three worlds, voice, data communications and information technology, will occur. Packet-switched telephony will have to provide new services with the same quality, reliability and ease of use of the current telephony service. So we will have to wait before all the old Bell head's telephones will be replaced by the Net head's ones.

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Résumé

Un peu plus de 120 années après la découverte du téléphone par Bell, le monde de la téléphonie va changer. L'idée de la téléphonie à commutation de paquets, telle que la téléphonie sur Internet, est de recréer les services dits vocaux (téléphonie, fax, modem) en utilisant les réseaux de données utilisés par les ordinateurs plutôt que les réseaux à commutation de circuits traditionnellement utilisés pour les services téléphoniques.
L'idée de base de la téléphonie à commutation de paquets est d'utiliser plus efficacement les ressources du réseau grâce à une compression de la voix et au partage de la bande passante.
Initialement intéressant, pour des raisons purement économiques, la téléphonie à commutation de paquets prendra son essor surtout grâce à la création de nouveaux services de communication multimedia combinant des services Internet, tels que le Web ou E-mail, offrant une réelle valeur ajoutée pour l'utilisateur.