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Voice over IP in Alcatel OmniPCX 4400 and OmniOffice

> Voice over IP: a new Eldorado for PABXs?

Introduction

The transmission and switching of voice in packet form over data networks has aroused considerable interest in recent years, mainly as a result of the emergence of powerful, low cost Digital Signal Processors (DSP).

Although Voice over Frame Relay (VoFR) was not as successful as expected, except perhaps in the United States, it introduced several fundamental mechanisms for transporting voice in packets, particularly by standardizing the voice encoding and compression algorithms.

In addition, the explosion of the Internet, with the resultant adoption of the Internet Protocol (IP) as the universal network protocol, has resulted in active standardization of voice signaling and transport over the IP network. The combination of voice and data networks that is made possible by Voice over IP (VoIP) is of considerable interest to Internet service providers because it allows them to compete with conventional telephone operators without having to deploy new networks.

At enterprise level, the convergence of voice and data networks also has several advantages:

- Single wiring system all the way to the office.
- Single network infrastructure. For example, voice-data multiplexers on leased lines can be replaced by IP Virtual Private Networks (IP VPN).
- Common network management.
- Better integration of voice and data applications, for example, workgroups, telecommuting or traveling subscribers.
- New applications can be created. For example, the Web Enabled Call Center (WECC) allows an Internet browser to contact a company agent either by e-mail or by telephone, using VoIP, to request further details about the page currently being viewed.

To provide this convergence, the suppliers of network products have had to pool their skills. This has resulted in an unprecedented wave of mergers and buyouts in the telecommunication sector.

OmniPCX 4400 and OmniOffice: the IP Evolution of the Alcatel 4400 and Alcatel 4200 PABXs

The Alcatel 4400 is a medium to high capacity PABX, while the Alcatel 4200 is a small and medium capacity version aimed at Small and Medium Enterprises (SME). Because these PABXs provide a wealth of features, in terms of telephone services and applications, and have a large installed base (more than 15 million subscriber lines currently in use), it was natural to base the design of Alcatel's VoIP products on the existing systems. **OmniPCX 4400** and **OmniOffice** are the names given to the reincarnations of these two systems.

This migration process has several advantages. For example, it means that the installed base can evolve and standard solutions can be used when parts of the IP network do not have the facilities required for voice transport.

Basic Techniques of VoIP

Voice over IP techniques can be divided into four main categories:

- general "voice in packet" techniques;
- voice encoding techniques;
- signaling protocols;
- Quality of Service (QoS) functions that are required in the network to provide optimum sound quality.

Voice in packets

The speech quality of a telephone call made over a packet network depends on two factors:

- Mouth-to-ear delay (transit time), that is the time between the moment the speech is uttered by the speaker to the moment it is received by the listener. This is a physiological aspect for which the commonly accepted upper limit is 150 ms.
- Rate of packet loss; although

this depends on the code used, the generally accepted upper limit is 3%.

Like any asynchronous network (asynchronous transfer mode or frame relay), the IP network introduces jitter, which means that packets sent at regular intervals do not take the same time to travel through the network. On arrival, this jitter must be absorbed in a buffer because the restored voice signal is a continuous flow. The time spent by a packet in this buffer is added to the propagation time; it is equal to the size of the buffer multiplied by the time represented by a voice packet.

Correct dimensioning of this buffer is crucial since, if too big, the total delay may become too great and, if too small, the loss rate may become unacceptable. The OmniPCX 4400 and OmniOffice PABXs use adaptive, dynamic management of this buffer.

Speech Encoding

Speech is encoded and packetized by software running on signal processors. The coding algorithm standards for VoIP are G.723.1, G.729a and G.711. The software that applies these algorithms must also be able to cancel echo, perform voice activity detection, adjust the gain, detect whether the call is from a fax or a modem and, if this is the case, apply other algorithms.

For these PABXs, Alcatel decided to take control of the speech coding software, primarily to ensure that the software can be ported to different signal processors and even to general usage processors. In practice, the VoIP equipment for the OmniPCX 4400 and OmniOffice ranges from a simple, single-channel IP telephone set to gateway boards, each of which can handle up to 60 channels. Consequently, a broad range of signal processors must be used.

When the processor produces its coding result (24 bytes for 30 ms of speech in G.723), the information is encapsulated in a 12 byte Real-time Transport Protocol (RTP) header and then sent to the IP stack to be transmitted in User Datagram Protocol (UDP) / IP. In return, the Real-time Transport Control Protocol (RTCP) messages give a general idea of the jitter and the rate of packet loss experienced by the recipient.

RTP and RTCP are Internet Engineering Task Force (IETF) recommendations which are used universally for transporting voice over IP.

Signaling

There are several standard signaling protocols for VoIP: H.323, the Session Initiation Protocol (SIP) and the Media Gateway Control Protocol (MGCP). The existence of different protocols partly reflects the multiplicity of standards organizations that have worked on VoIP and partly the different approach between intelligent IP equipment and IP equipment controlled by a central intelligence which only handles low level functions (speech encoding and packet processing). H.323 and SIP belong to the first category, and MGCP to the second. Without going into the details, H.323, developed by the International Telecommunications Union (ITU-T) is the IP development of the Q.931 Integrated Services Digital Network (ISDN) protocol, whereas the IETF's SIP uses messages similar to HyperText Transfer Protocol (HTTP) ones.
H.323 is a recommendation which introduces the following protocols:

- H.225/Q.931 for the call setup protocol.
- H.225/RAS: The Registration, Admission and Status (RAS) protocol is used between the H.323 terminals and a dedicated server, the gatekeeper. This is mainly so that the terminals can register, obtain a translation of a telephone number into an IP address, manage the bandwidth used in the network, and obtain authorization to set up a call.
- H.245 is for opening and closing the RTP and RTCP channels and interchanging the speech encoding algorithms accepted by the terminals.

Quality of Service

Today's IP networks use a "best effort" approach to transport packets to their destinations. This means that the routers forward the packets in "first in, first out" order and, if congestion occurs, delete excess packets without any regard to their importance. This is clearly unacceptable for transporting voice.

By QoS functions, we mean the software and hardware used by the network elements to identify, classify and retransmit packets according to their traffic flow requirements. Physically, realizing QoS involves several queues, reflecting different traffic priorities, which are emptied according to an appropriate algorithm.

QoS mechanisms can be explicit or implicit. They are explicit when an Ethernet frame or the IP packet contains a mark to identify the priority the packet requires (IEEE 802.1p/Q or IP Type of Service) or when prior signaling is used to identify the voice flow (Resource Reservation Protocol; RSVP). Implicit mechanisms automatically recognize the data flows according to information in the packets, such as the source and destination addresses, the layer 4 protocol, port numbers and layer 7 information.

All the IP equipment in the OmniPCX 4400 and OmniOffice is capable of setting explicit QoS indicators and applying RSVP, which is a perfect match with Alcatel's data communication products.

Migration Components

Alcatel's objective has been to develop the Alcatel 4400 and Alcatel 4200 architectures to realize an "ideal" all-IP solution , as illustrated in *Figure 1*.
The three main components required for this migration are the gateway, the IP telephone, and the multimedia workstation.

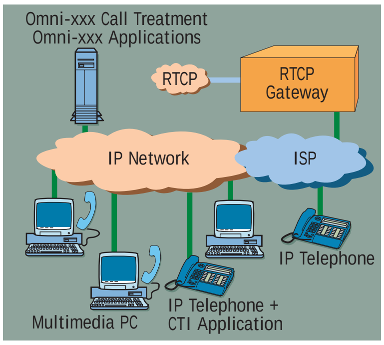

Figure 1 – All-IP PABX

Voice over IP Gateway

The gateway, the first application of VoIP, interconnects several networked PABXs over an IP network without losing the virtues of the OmniPCX 4400 networks, that is, total transparency of telephone services for all the network's users. Thanks to the inter-node ABC-F (Alcatel Business Communications –Features) network signaling protocol and the concept of hybrid data links, it is possible to separate ABC-F signaling from the physical link transporting the voice. The transport of inter-node signaling over the IP network has been part of the Alcatel 4400 for a number of years.

Because of the virtual private network concept, an inter-node data link can be a switched network. Diagrammatically, when the inter-node link is a switched network (for example, the ISDN), the ABC-F signaling transmits a virtual directory number in the call setup messages. When a call is received from the switched network with a virtual directory number, such as the called party's number, the called PABX links this call with the initial ABC-F call.

This simple and elegant technique for producing virtual private networks of PABXs has been proposed to the European Computer Manufacturers Association (ECMA).

The extension of this mechanism to an IP network is immediate, to the extent that the gateway is seen by the OmniPCX 4400 call handling function as an ISDN port (see *Figure 2*).

The gateway monitors network quality using the RTCP protocol. When it detects that the voice quality is unacceptable, it rejects further calls so that the OmniPCX 4400 and OmniOffice call handling functions can reroute them if another path is available.

In hardware terms, the gateway function is performed by the Local Area Network (LAN) VoIP board, which is seen by the OmniPCX and OmniOffice PABX central processor units as a series of ISDN primary interfaces. Thus it receives ISDN signaling messages (Q.931) which it converts into H.323 signaling messages (H.225). The board has the following characteristics:

- Up to 60 compressed channels (G.723.1 or G.729a or G.711), A law or µ law, using TI 62xx processors.
- Fax relay compliant with the T38 protocol.
- H.323 V2 protocol developed by Alcatel.
- Channel aggregation.
- Dynamic jitter buffer.
- Client RAS.
- Network interface: Ethernet 10/100 baseT or X.24-V.11.
- Linux operating system used on a Power Quicc II processor.

The gateway can also handle H.323 terminals because it uses this standard. The terminals are PCs using a microphone and sound card for the voice part (e.g. Microsoft Netmeeting). To realize this, a gatekeeper needs to be installed in the network so that the terminals and the gateway register with it and convert directory numbers into IP addresses. Services provided to these terminals are limited to those available via the H.323 protocol.

The LAN-VoIP board, with software changes, has many other roles in the OmniPCX 4400.

IP Telephone Set

Today, users can take advantage of OmniOffice 4200 and OmniPCX 4400 telephone services via a range of dedicated digital telephones, known as the Reflexes™ range. The user-friendliness of these telephones has been designed to reflect the advanced services offered by the PABX call handling software (programmable keys, keys with functions dependent on context, call by name, display screen, etc). These telephones have a socket into which an optional card can be inserted.

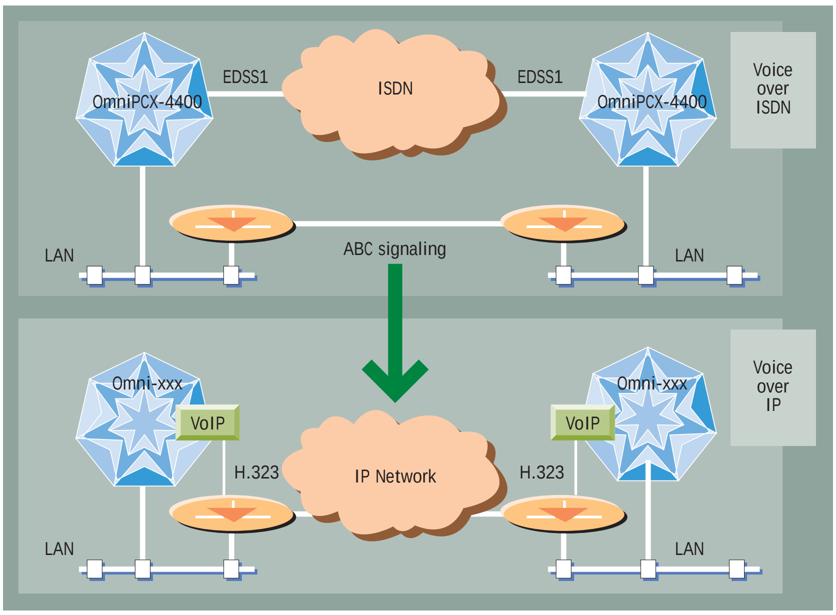

Figure 2 – OmniOffice and OmniPCX 4400 network using H.323 gateways

It was therefore natural to design the IP telephone as a standard Reflexes™ telephone with an additional card, called the Transparent System Connector IP (TSC-IP), which performs the specific VoIP functions (see *Figure 3*). In addition to being quick to implement, this approach has the advantage of being able to upgrade the installed base of telephones, so that our customers have a future-proof investment. However, in the case of a complete IP telephone, it is more costly than a fully integrated solution. As regards speech encoding, TSC-IP uses the same principles as LAN-VoIP.

TSC-IP connects to an Ethernet network via an Ethernet 10 baseT interface; in a second stage, this will evolve to a 10/100 baseT interface. Customers might consider that it is costly to monopolize a second 100 Mbit/s port on the parent LAN switch for an ordinary telephone that generates only a few tens of kbit/s when a 100 Mbit/s connection is already being used for a workstation. To overcome this, TSC-IP uses a 100 Mbit/s Ethernet hub. The PC is connected to the TSC-IP hub and the TSC-IP is connected to the network so that the entire setup operates as a 100 Mbit/s Ethernet (see *Figure 3*).

Problems where the office has only one LAN outlet are resolved in the same way.

IP equipment requires a minimum basic configuration to be able to start up the network. It needs its own IP address, the IP address of the default router and the masks of the associated subnetworks. Although manual configuration is possible via TSC-IP, it is hard to envisage asking the telephone end user to enter these parameters. To begin with, the user probably doesn't know anything about IP and the telephone keypad was not designed for entering this type of data. Consequently, the telephone configures itself automatically using Dynamic Host Configuration Protocol (DHCP). The telephone broadcasts a discovery message to the DHCP server (relayed by routers, if necessary). At

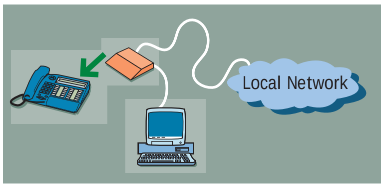

Figure 3 – Reflexes™ telephone set and TSC-IP card

the end of the protocol interchange, the DHCP server provides the telephone with the configuration parameters and the download server's address. After downloading the processing software using the Trivial File Transfer Protocol (TFTP), the telephone is ready for use.

The customer may already have a DHCP server. However, if the network is not equipped with one, the OmniPCX central processor unit supplies one. If several servers exist, the OmniPCX server is chosen first.

Using DHCP simplifies address management and telephone mobility, making life easier for the users. All they need to do is disconnect their telephone and reconnect it in another office where it is immediately operational. Security is also improved because it is impossible to copy the IP address for malicious purposes or by accident.

Signaling to the OmniPCX and OmniOffice CPU is the same as for standard Reflexes™ phones. It is simply encapsulated in the UDP/IP packets and addressed to the central processors of the OmniPCX and OmniOffice via a LAN-VoIP board which relays the signaling. A layer above UDP provides reliable signaling transport.

This architecture complies with the MGCP architecture while the OmniPCX and OmniOffice call handling function acts as the gateway controller and TSC-IP plays the unintelligent media gateway role. Migration of current non-standard signaling to extend the MGCP relating to IP telephones (Megaco IP Phone Media Gateway), which is being drafted by the IETF, will pose no implementation problems.

Finally, the telephone can be powered locally from the mains or remotely from the exchange.

Multimedia PC

The multimedia PC has a telephone connected to its sound card. Voice and data pass through the PC network interface card. The PC software is divided into two distinct modules:

- call control;
- speech channel processing.

The Alcatel 4980 or PIMphony applications control the calls. The speech processing module has been added to these applications.

Alcatel 4980 and Alcatel PIMphony are Computer Telephony Integration (CTI) applications developed to run under Windows and control and monitor a PABX telephone set. Using a multimedia PC, the telephone is simulated by the speech channel processing module. OmniPCX and OmniOffice treat this module like an IP telephone. In fact, the multimedia PC uses only the signaling messages relating to the setup of the RTP channels, since the telephone services are controlled by the CTI application.

The speech channel processing module is also responsible for encoding or decoding the speech packets going to or coming from the RTP data flow.

These last two functions are performed by the PC processor and not by a DSP. The result is sent to the sound card.

The telephone handset has a hookswitch which has a state that is accessible from the PC COM port. For ringing, when the telephone is on hook, the audio signal is amplified so that it can be heard through the earpiece.

Derived Products

This part focuses mainly on the OmniPCX. It describes products specifically for user groups working either close together or at a distance.

Hub for Reflexes™ Telephones

Logically, the hub for Reflexes™ telephones is a TSC-IP capable of connecting four Reflexes™ phones and an analog line. It is used to combine DSP resources and maximize the number of ports on the LAN exchange so that the cost of the IP connection is considerably reduced. Physically, the hub looks like a desktop box, as shown in *Figure 4*.

The hub has been designed for use in shared offices or in open plan environments. Technically, it uses TSC-IP principles. The number of speech encoding channels can be increased to six by using a more powerful DSP.

Remote Crystal on IP

An OmniPCX 4400 consists of a number of modules, known as crystals because there is full meshing between each board in a module. These modules are organized in a tree structure in which each crystal is linked to the stage below by an 8 Mbit/s synchronous time division multiplex (see *Figure 5*).

When all these crystals are installed in a single rack, there is no point in linking them by IP. On the other hand, if the crystals are remote (branches or suboffices), it becomes worthwhile replacing telephone-only leased lines with the lines used by the data network. The functions for interconnecting the crystals and packetizing voice are handled, in the OmniPCX, by the LAN-VoIP card equipped with special software. For this role it has been renamed INT-IP, as shown in *Figure 5*.

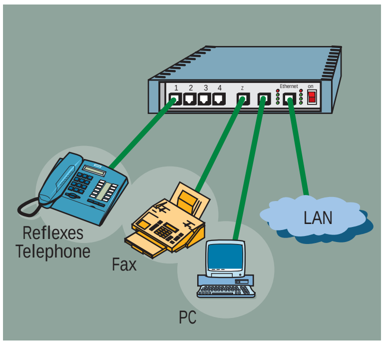

Figure 4 – Hub for Reflexes™ telephones

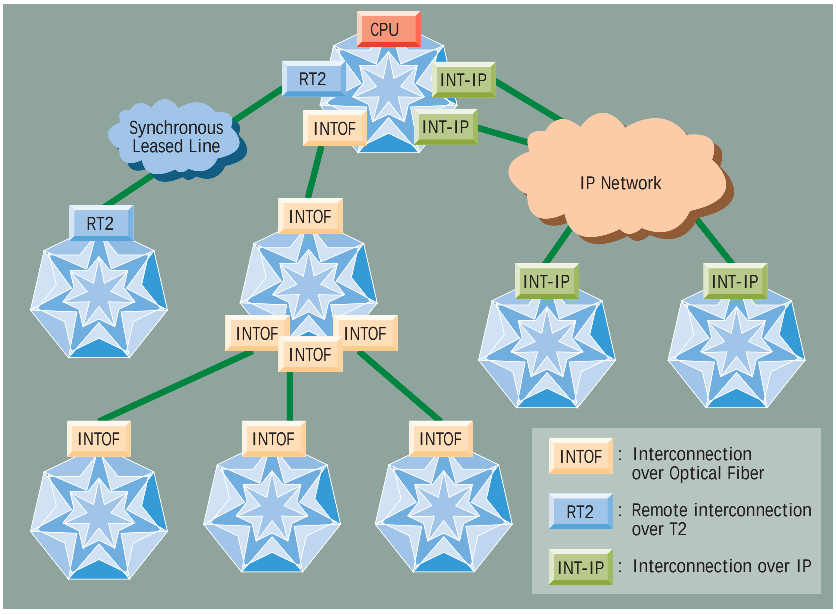

Figure 5 – OmniPCX configuration with remote crystals interconnected via IP

INTOF : Interconnection over optical fiber **INT-IP :** Interconnection over IP
RT2 : Remote interconnection on T2

Conclusion

Voice over IP has become a reality and, as always, the migration will be progressive. To make it work will require a great deal of discipline and collaboration between the voice and data teams. It involves migrating the PABX's dedicated switching equipment to data network switching equipment. Consequently, VoIP will not be Eldorado in hardware terms. However, by identifying and developing the high added value applications that VoIP allows, the pay off will be extremely beneficial. ■

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