5- OmniPCX *Office* native VoIP

Native VoIP

IP Telephony

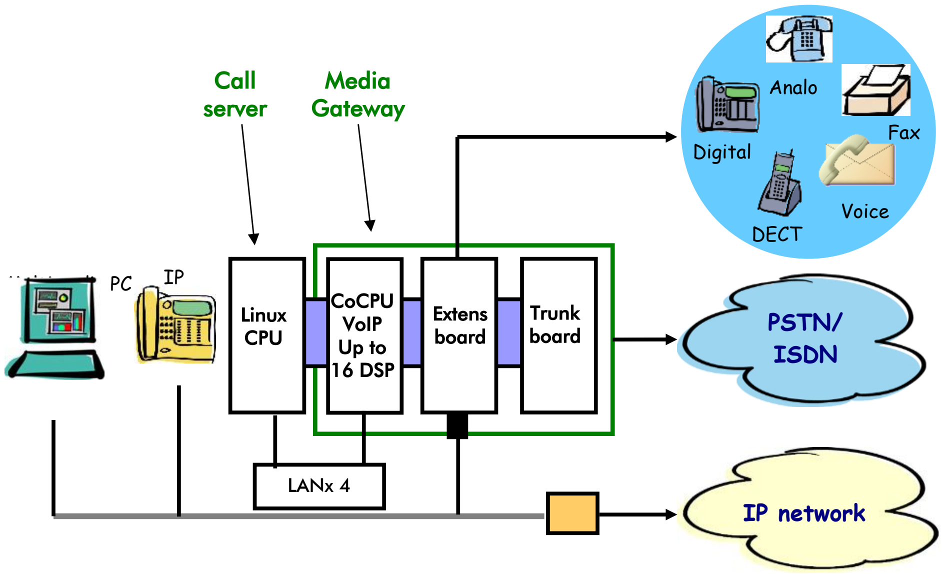
IP Trunking

Bandwidth

Native VoIP

Based on the all-in-one concept, Alcatel OmniPCX Office is natively IP. It is capable to manage not only packet but also circuit world…with unmatched voice features and applications.

OmniPCX Office is a native VoIP appliance. It integrates a C~~c~~all S~~s~~erver and a Media Gateway ~~and a H323 Gateway~~.
The C~~c~~all S~~s~~erver is the OmniPCX Office CPU which runs on Linux
The M~~m~~edia G~~g~~ateway is the combination of the different circuit trunk boards, circuit extension boards and the CoCPU VoIP ~~where the DSPs are pluged~~.
The Call server and Media gateway incorporate~~d~~ H323 protocol to take into account standard H323 devices and interoperta with Alcatel and non Alcatel systems supporting this standard.



Native VoIP

DSPs (Digital Signal ~~Protocol~~Processor) are an important part of central system resources for VoIP They allow codings and transmission between an IP phone (or Pimphony IP) and the ISDN/PSTN world (or a circuit extension of the company). Those DSPs are located on the CoCPUVoIP.

Capacities

Up to 6 CoCPU VoIP per system
- **Overall limit 96 DSP compression channels**
- **Nb of CPU+CoCPUvoip+CoCPU@ must not exceed 3 per main module or extension module.**

A CoCPU VoIP supports ~~provide connection for~~ one daughter board
- **VoIP daughter board 4 DSP or**
- **VoIP daughter board 8 DSP or**
- **VoIP daughter board 16 DSP**

Use of those DSPs is subject to channel/DSP software keys
- **key 2, key 4, key 8, key 16**

If you need 10 DSP you must buy
- **CoCPU VoIP + VoIP daughter board 16 DSP+ key 2 + key 8**

In oder to define the number of DSP for a given configuration, please refer to the automatic tool embedded in the OmniPCX Office quotation tool (Actis)

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IP Telephony

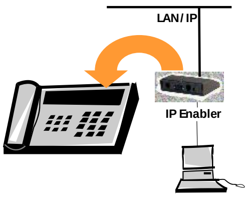
OmniPCX Office , being native VoIP appliance, provides all of the benefits of IP telephony including complete scalability, improved management, and reduced phone costs.
It supports Alcatel IP phones, ~~H323 client software, as Microsoft NetMeeting~~Alcatel PIMPhony IP edition and H323 clients (as Microsoft NetMeeting). All these IP terminals have the same voice quality as standard Alcatel Reflexes terminals.
~~**includes several security mechanisms to protect company network and data from external enemy and secure internet accesses.**~~

Capacities

- **IP subscribers : 200 (IP phones & ~~IP~~ PIMphony IP)**
- **Number of DSP channels :up to 96**

Refer to the OmniPCX Office quotation tool (actis) to define the number of DSPs needed for a given configuration

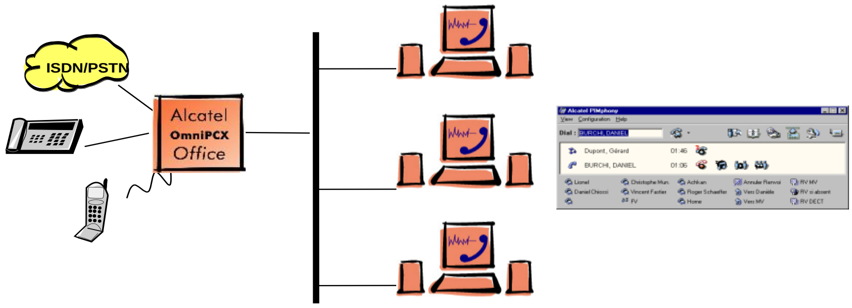
Fast IP enabler



This plugware is an Ethernet LAN switch 10/100 base T. It transforms an UA set into an IP set. It allows the company to use a single port in the corporate LAN to connect the employee PC and telephone
Towards the corporate LAN, it has an uplink which can be configured in 10 or 100baseT.
The UA set is connected internally to one port of the switch. The PC is connected externally to the other port of the switch which is auto-sensing and automatically adapts to the PC configuration (10 or 100BaseT).
The fast IP enabler is able to tag the voice packets ~~at the level 2 (Ethernet) using 802.1P and 802.1Q and~~ at the level 3 (IP) using TOS and DiffServ. ~~When connected to OmniPCX Office, only level 3 is used because in small SME configurations LAN bottlenecks can usually be avoided.~~
The fast IP enabler gives priority to the voice and voice signalling packets ~~in the direction of the corporate~~on the LAN.
The fast IP enabler will choose dynamically the best compression G711 when packets will stay in the LAN, and G723.1 when packets will be sent over the WAN, or G729a when requested.

IP Telephony

IP PIMphony



PIMphony basic, Pro and Team can be installed in IP mode on a multimedia PC. To do that the customer must buy e PIMphony software keys. The Protocol stacks (identical to the one of IP Reflexes) to put on the PC are in the PIMphony CD. They include G711 and G723.1 voice compression algorithms.

Level 3 QOS is provided : Microsoft Windows (98, 2000, Millenium, NT4) use TOS and will perform the tagging of IP Voice & Voice signalling packets in the PC. Those packets will have priority when sent by the PC

Voice can then be delivered through the PC microphone and Loudspeaker. An optional "IP handset Comfort kit" is available in the catalog to provide increased confidentiality and comfort to the user. This handset can be hung up by the side of the PC screen. An alternative is a professional headset.

☺ **Benefits :**

Moving, adding and changing :
With the embedded DHCP server, moving or adding an IPphone will only require plugging it into the new ethernet connection.
Adding an IP phone can be done either by pluging directly the IP phone to any ethernet port. If no ethernet port is available, the ethernet network is easily expandable by cascading a new LAN switch. Of course if the user already has an ethernet connection for his PC, this connection can be shared between the PC and the IP phone thanks to the IP plugware. In this case no supplementary Ethernet port, nor cabling is needed.

Branch office and Home worker solutions
Customers more and more request a transparence of features between Headquarters and Branch Offices (same directory, voice mail, features etc...). This is a standard feature of remote IP phones in the branch or at home, managed by the call server in the headquarters. The IP phones at the remote site have the exact features as they would at the company site.
This implies a managed data link between both sites (leased line, IPVPN...).

Mobile worker solution
Mobile workers usually don't benefit from a managed data connection, but connect from time to time their multimedia PC to the company through the internet to download price lists or files. If their PC integrates VoIP (e.g. IP PIMphony), they can benefit remotely from telephony features on the call server.

IP Trunking

Companies with "state-of-the-art managed WAN data network" are today willing to mix voice and data on this network to save money on inter site telephone calls. This can be done by sending voice over IP over this "managed WAN data network". These solutions are particularly popular with multi-site businesses that call long-distance and/or internationally on a frequent basis.

Capacities

- **IP trunks : 120**
- **Number of DSP channels: up to 96**

Refer to the OmniPCX Office quotation tool (actis) to define the number of DSPs needed for a given configuration

- **For IP trunking, OmniPCX Office delivers not only G723.1 but also G729a compression mechanism. G711 coding can also be used but does not optimize bandwidth.**

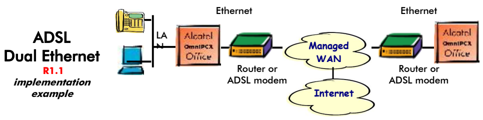
WAN Access Methods

IP sets and IP phones managed by the OmniPCX Office can access to the WAN through the OmniPCX WAN connection or directly through an external router.

<u>WAN access through OmniPCX Office</u> (refer to chapter internet)

-1- ISDN 2 B connection to an Internet Service Provider. Either the 2 B channels from a Basic Rate Interface (BRA) or any 2 B channels from a Primary rate Access (PRA)

-2a- Ethernet connection to any existing router (called dual Ethernet) in R1.1
-2b- PPPover Ethernet (PPPoE) connection to an ADSL modem in R1.1



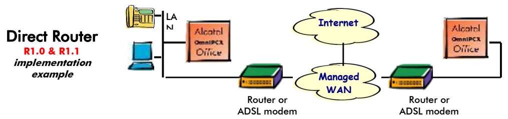
In those configurations, VoIP packets share the bandwidth with OmniPCX Office internet applications (E mail, VPN, etc...if used) and benefit from OmniPCX Office WAN QOS mechanism.

- **Policing : Based on TOS/Difserv header, OmniPCX put Voice over IP in its high priority queue**
- **Queuing/ Shaping : OmniPCX Office manage 2 queues in a buffer memory. High priority queue and Lower priority queue.**
- **Congestion control : OmniPCX Office use Random Early Detection mechanism (RED)on the Lower priority queue to inform packet sources to decrease their transmission rate.**

IP Trunking

<u>**Direct WAN access through router**</u>

Any existing router directly connected to the company LAN



In this configuration VoIP packets benefit only from the external router QOS mechanism (if any).
Nota :OmniPCX Office internet applications (firewall,E mail, VPN, etc...)cannot be used in such a configuration, therefore external devices will be required for such applications.

IP trunking overflow and back up

Whatever the connection method, OmniPCX Office offers ISDN Overflow and Backup mechanism fully integrated with ARS.

- **Overflow : when VoIP calls exceeds atreshold**
- **Back up : In case of failure of the WAN link**

Interoperability

OmniPCX Office being H323 V2 compliant, is able to interoperate with Alcatel and non Alcatel systems supporting this standard

☺ **Benefits**

International/Long distance savings :By replacing « E1 tie-lines boards and dedicated voice leased lines » between dispersed PBXs with « VoIP gateways and data leased lines », multi-site customers can circumvent expensive telephone charges, by using the data Wan infrastructure. A company can reduce long-distance charges for intra-company calls by integrating voice, in the enterprise data networks.

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Bandwidth

Voice Activity Detection (VAD)

In a phone conversation, one party speaks at a time. The packets sent by the listening party are just "silence"packets which spoil valuable bandwidth. The target is then to detect the beginning of a silence, inform the other device (which will generate background noise to avoid the impression of "cut line")and stop emission. It is assumed to decrease necessary bandwidth by 35%

Bandwidth needed for a voice call

The bandwidth needed for a voice call depends on a number of factors :

- **Voice call transported on the LAN (Ethernet) or on the WAN?**
- **Compression mechanism used : G723.1, G729a , none (coding G711) ?**
- **Voice Activity Detection (VAD) used?**
- **Compressed Real Time Transport Protocol (CRTP) used at the WAN access ?**

Vocoder	Bit rate	Packetisation time	Payload	IP frame =payload +RTP(12) + UDP(8)+IP(20)	Bandwith at IP level	Bandwith at Ethernet level (*)	Bandwith at WAN level (**)	Bandwith at WAN level (**) with CRTP (***)
G.723.1 (MP-MLQ)	6.4 Kb/s	30 ms	24 Bytes	64 Bytes	17.1 Kb/s	27.2 Kb/s	19.2 Kb/s	9.1 Kb/s
G.729a	8 Kb/s	30 ms	30 Bytes	70 Bytes	18.7 Kb/s	28.8 Kb/s	20.8 Kb/s	10.7 Kb/s
G.711	64 Kb/s	30 ms	240 Bytes	280 Bytes	74.7 Kb/s	84.7 Kb/s	76.8 Kb/s	66.7 Kb/s

(*) IP frame + MAC (14) + CRC(4) + preamble (8)+ silence inter-frame (12), (nota :without VAD)
(**) 8 bytes layer 2 overhead (=max for PPP, MLPPP, FRF.12, HDLC)
(***) Some Switch/routers are able to compress RTP/UDP/IP headers from 40 bytes to 4bytes (or even 2 bytes in certain conditions). It is called Compressed Real-Time Transport Protocol or CRTP. It brings the bandwidth at IP level down to 9 or 11Kb/s for speech transmission. OmniPCX Office can benefit from that.
(****) In the table we have not taken into account bandwidth saved with Voice Activity Detection. For information, VAD is assumed to bring down bandwidth needed at 65% of full rate.

OmniPCX Office mechanisms for bandwidth optimization

OmniPCX Office provides

- **In the LAN, G711 coding and G723.1 compression**
- **In the WAN, G723.1 compression, G729a compression and G711 coding**
- **Voice Activity Detection**

OmniPCX Office is fully compatible with routers using CRTP

~~***Easier management*** **For corporations with multiple sites, management of PBXs is not centralized.. As a result, IT managers can reduce the time and effort that they spend managing the telephone system.**~~

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