Enterprise Solutions Division
Enterprise Business Unit/Product Marketing

A4400 IN RELEASE 3.0 & 3.1

PRODUCT DESCRIPTION

Product description A4400 R3.0 & 3.1	
Reference number 127/99/GM	
Edition	Date
1	30/09/99

PRELIMINARY

This document describes the additional services provided by Alcatel 4400 releases 3.0 & 3.1 in comparison with Alcatel 4400 release 2.It is a complement to the Alcatel 4440 release 3 & 3.1 marketing presentation.

The reader is supposed to know the detailed content of release 2 which has been described in marketing document referenced 264/97/PW Ed. 1 dated 19/08/97.

In the same way, this document does not include country specificity, a similar document has to be produced for each country.

New management (47xx) and call center services brought by Alcatel 4400 are described in separate documents.

Cross-compatibility (Alcatel 4400, applications, releases …) is also explained in an other document.

SUMMARY

1. INTRODUCTION

Releases 3.0 & 3.1 are not merely an upgrading or updating of release 2. Nor do they simply have the aim of increasing the number of PBX functions and features on offer to the user.

Alcatel 4400 releases 3.0 & 3.1 surpass the typical evolution between past releases by going beyond the provision of mere PBX functions. Releases 3.0 & 3.1 are the first stepping-stone in a smooth transition to the communication system of the future.

With release 2, Alcatel 4400 already enhanced communication so that it gained its present state-of-the-art positioning in terms of

- Networking
- Call handling
- Mobility

Another main achievement of release 2 was the move towards a single global software release for all countries.

Releases 3.0 & 3.1 maintain all these high-performance features whilst introducing a whole new concept.

Now Alcatel 4400 not only offers new PBX functions as you might well expect but also integrates and combines them with a whole host of other concerns focusing on:

- Evolution of the user's working environment
    - Desktop PC
    - Mobility
    - …
- evolution of the communication environment
    - Internet
    - Messaging : E-Mail, voice mail,…
    - Wireless
    - Multi-carriers
    - …
- Cost reduction
    - At acquisition time
    - At installation time
    - For running cost
- *Voice over IP*

The Alcatel 4400 PBX acts as a host platform for a whole new set of services available in Alcatel 4400 Release 3.0 & 3.1 such as :

R3.0

- **Large networks**
- **Large systems**
- **Large DECT coverage**
- **Inexpensive DECT infrastructure**
- **Wirefree system**
- **A4980 Click & Phone**
- **Visual messenger**
- **Secured DISA**
- **Second call for free**
- **Leased line optimization**
- **Use of Frame Relay instead of leased line**
- **ABC on demand**
- **A new multimedia PC based attendant**
- **ARS server**
- **A new range of sets**
- **New call center services**
- **…**

R3.1

- **VoIP for networking**
- **remote crystal with voice compression**
- **DECT encryption/authentication**
- **A4615 voice mail for small systems**
- **Automatic DISA for non ISDN signaling**
- **AHL protocol on Ethernet**
- **A4980 Homeworker**
- **Integration of voice services and voice mail services with the user desktop**
- **.....**

All these new sellable services brought by Alcatel 4400 release 3.0 & 3.1 involve not only the new PBX release 3.0 & 3.1 but also new applications.

With the Alcatel 4400 release 3.0 or 3.1, we are now moving towards a new communication system where the PBX will take the place of a server and pave the way for a complete evolution and transformation of the communication world.

2. LIST OF NEW SEVICES INTRODUCED BY RELEASE 3.0 & 3.1

SYSTEM CAPACITY ENHANCEMENT

- **Standalone system**

	R3.0	R3.1
Data users per node (circuit mode only)	Up to 1000	Up to 1000
Users per node	Up to 5000	Up to 5000
Phone-book entries	Up to 60000	Up to 60000
Digits for external numbers	Up to 30	Up to 30

- **Wireless system**

	R3.0	R3.1
DECT sets	Up to 5000	Up to 5000
Base stations RBS2G Advanced	Up to 1000	Up to 1000
Base stations IBS Optimised	Up to 256	Up to 256

- **ABC-F homogeneous network**

	R3.0	R3.1
Nodes	up to 100*	Up to 100*
Users in the network	Up to 50000	Up to 50000

* Only 32 nodes in backbone

NETWORKING

R3.0

- **Integrated voice compression platform**
  - up to 6 compressed voice channels + ABC-F or ISDN signalling on an unique 64Kb/s channel
    - On leased line.
      - Serial link X24/V11 or V36
      - S0FV leased line
      - G703/G704 E1
      - Overflow
      - Multi-directions leased lines
    - .On ISDN network
      - ABC networking on demand (with voice compression)
      - ABC networking on demand (without voice compression)
      - ABCVPN with voice compression
      - Second call for free
    - On Frame Relay

- **100 nodes network**
  - Adaptive routing enhancement
  - ABCVPN

- **ABC Hybrid links and ABCVPN enhancements**
  - Backup ABC signalling

- ABCVPN over non-ISDN switched network

- **Open Networking**
  - New QSIG-GF supplementary services
  - Integrated DPNSS
  - ABCVPN and QSIG-GF interworking
  - ABCVPN over a QSIG-GF networking (ABC in QSIG-GF D channel)

- **Multi-carrier enhancements**
  - Capacity enhancement of ARS
  - ARS is time dependent
  - ARS is dependent on calling party
  - Centralised ARS

- **New networking software licence**

- **SO Data pool to connect Remote Access Server**

- **VN6 for French ISDN network**

R3.1

- **VoIP**
  - **new LIOE board**
  - **connection to Ethernet for ABC networking, teleworker Wecc**
  - **H323 compliant**
- **remote crystal with voice compression (LIO-B board)**

MOBILITY

R3.0

- **Extension of Capabilities**
  - Extension of Capacity

	R3.0	R3.1
DECT sets	Up to 5000	Up to 5000
Base stations	Up to 1000	Up to 1000
PARI per system	8	8
PARI per ACT	1	1

- Enhancement of DECT features in ABC homogeneous Network
  - voice mail notification & voice message consultation
  - supervision
  - boss/secretary features
- "Twinset" function
  - unique voice mail box
  - single dialling number
  - simultaneous ringing on both sets

- **Support of a new type of base station A4070 IO (IBS)**
  - radio base station connected to an UA board via UA link (2)
  - up to 256

- **Wireless desktop "Reflexes" terminals**
    - UA 3G sets connected through DECT radio link
    - **A4097 CBL** (TSC) wireless desktop sub-device

	R3.0	R3.1
per system with "Advanced" RBSs	up to 100	Up to 100
per system with "Optimised" RBSs	Up to 500	Up to 500

- Complete range of DECT handsets
    - * A4072 "Speech"
    - * A4074 GB (Business), GC (Comfort), GI (Industry)
    - * A4074 GB Ex (Intrinsically safe) and A4070 EA Ex (Intrinsically safe "Advanced" RBS)

- Alcatel telephony on PC : **A4980 "PC phone"**
- Ubiquity : association & integration of GSM handsets in the A4400 environment

R3.1

- **DECT encryption/ authentication**
- **Call by name on A4072 "Comfort"**

UBIQUITY

R3.0

- **association & integration of GSM handsets in the A4400 environment**

R3.1

- **Ubiquity enhancement**

TELEPHONY ON PC

R3.0

- **A4980 "PC PHONE"**

R3.1

- **Alcatel 4980 enhancements**
    - **analog set**
    - **GroupWare (supervision, phone book for groups)**
    - **integration with voice mail and unified messaging**
    - **openness :**
        - **MAPI & TAPI with Outlook,**
        - **Notes,**
        - **LDAP directories**

VOICE MAIL : 4635 release 2.0

- **R2.0 new services**
- Visual Messenger
- Digital networking

- *R2.x new services*

    - **Move mail box**
    - **hôtel mail box**
    - **8 languages**
    - **ubiquity fully integrated (management)**
    - **increased number of IVR port up to 250**

VOICE MAIL : 4615 release xx

- **A4615**
    - **for SME « an answering machine for each PBX »**

Visual messenger

- **integration with mail applications : Outlook, Notes & with A4980**

NEW UA TERMINAL RANGE

- **New UA 3G line "*Reflexes*" sets**
- A4004 "*First*"
- A4010 "*Easy*" : display
- A4020 "*Premium*" : internal keyboard & display
- A4035 "*Advanced*" : internal keyboard & display

- **New subdevice range (TSC): indoors (4020/4035) or in stand alone position (all sets)**
    - A4091 CTI: CTI interface
    - A4093 ASY/CTI: V24 & CTI interfaces
    - A4094 ISW: S0 interface
    - A4094 ISW/CTI: S0 & CTI interfaces
    - A4095 AP: analog terminal interface
    - A4097 CBL: wireless desktop subdevice

ATTENDANTS SERVICES

R3.0

- **New range of Attendants**
    - Alcatel 4059 MAC offer

- **New services for Attendants**

R3.1

- **50 attendants per node and per attendant group**

CALL HANDLING

R3.0

- **Hunting groups in ABC-F homogeneous network**

- **ISDN Calls logging on all sets with display**

- **Internal Calls logging**

- **New services for security**

- **Enhancements for external Directory connection**

R3.1

- **Automatic DISA for non ISDN signaling (e.g. PCM)**
- **DISA allowed with GPA board**

SECURITY

- **Security solutions:**
    - identity & password for all possible accesses to configuration applications
    - DISA access control: **Security Dynamics**
    - Blocking DISA after **n** attempts
    - Set (guest) locked after **n** password entry errors (hotel)
    - Specific barring categories for external forwarding
    - Specific category for trunk to trunk transfer
    - Detailed history file for management operations
    - IO2/OBCA CPU access: ISDN number identification
    - TCP wrapper: filtering per facility of TCP or UDP access
    - Firewall offer

CALL CENTER

- **Remark: detailed product description of call center is found in separate documents.**

HOTEL/HOSPITAL

R3.1

- **multiple wake-up**
- **AHL protocol on Ethernet**
- **8 digits for guests and rooms numbers on AHL**
- **centralized outgoing trunks : accounting ticket on guest AHL node**
- **non answered call repertory**

MISCELLANEOUS APPLICATIONS

- **Voice guidance**
    - higher number of voice guides
    - voice guides chaining
    - detection of presence behind Z interface

- **CSTA**
    - multi-line monitoring (without supervision keys)
    - DECT monitoring

MANAGEMENT TOOLS

Remark: detailed product description of 4715/30/40 & 4755 is found in separate documents.

Alcatel 4715/30/40, release 4.4

- **General**
  - management of A4400 release 3.0 & 3.1
  - management of A4200 C,D,E release 3.1, 3.2
  - Windows 98 & NT4 Workstation compatibility (W95 from R4.2)
  - security access control

- **Connectivity**
  - remote access to A4400 through RMA for: configuration, data collection (accounting, performance)
  - reception of A4400 alarms through RMA

- **Configuration**
  - management of UA 3G sets
    - new functions implemented in A4400 release 3
    - new label set tool for UA 3G
  - configuration per domain
  - tool for configuration of ARS tables

- **Accounting**
  - reporting on ABC-F and local calls
  - Reporting on large network (outgoing calls)
  - accounting for dual currency including Euro
  - time based metering: configuration of time in milliseconds
  - import of a complete code book in tabular mode: Excel format
  - accounting package on a non dedicated PC: only for capacity less than 250
  - Accounting tickets on subscriber node

Alcatel 4755 release 4.2

- **General**
  - disk mirroring
  - evolution of the SUN stations

- **Connectivity: reception of A4400 alarms through RMA**

- **Alarms and topology: export of SNMP traps to hypervisor**

- **Configuration:**
  - broadcasting for multiple subnetworks -> A4400 & A4400/4300L
  - broadcasting for nodes without permanent connection: -> stand-alone or ISVPN

- **Accounting: reporting on ABC-F and local calls for A4400**

- **Past time performance for: trunk groups, attendants, DECT, subscribers, called numbers**

- **Directory:**
  - infocentre
  - directory improvements
    - automatic dialling on A4300L
    - handling of PC-client installation improvement

- **Alcatel 4200 management: A4200 E release 3.0; A4200 C, D, E release 3.1**

Alcatel 4755 release 4.4

Remark: A4400 release 1.5.2 is no longer supported by A4755 release 4.4

- **General**
  - management of A4400 release 3 & 3.1
  - security access control

- **Connectivity:**
  - remote access to A4400 through RMA for:
    - configuration,
    - data collection (accounting, performance)

- **Alarms and topology: display of A4400 racks**

- **Configuration:**
  - configuration per domains

- **Accounting:**
  - accounting for dual currency including EURO
  - import of complete carrier tariffs in tabular mode (Excel format)

- **Past time performance: import of new predefined reports**

- **Directory:**
  - STAP protocol (for automatic dialling) on TCP/IP
  - link between accounting (organisation map) and directory (for automatic update)
  - pull-down menu on each field on the PC client window

3. SYSTEM CAPACITY ENHANCEMENT

Releases R3.0 & 3.1 of the A4400 system provides a lot of interesting capacity enhancements, at node level, as well as network-wide.

3.1 STANDALONE SYSTEM ENHANCEMENTS

	R3.0	R3.1
total physical voice terminals (Z, UA)	5000	5000
total digital sets (all types)	5000	5000
total Data subdevices & TA	1000	1000
total DECT & GAP sets	5000	5000
max. Trunk groups	200	200
digits for external numbers	up to 30	up to 30
phone book entries	60000	60000

3.2 WIRELESS SYSTEM ENHANCEMENTS

	R3.0	R3.1
TotalDECT sets & GAP sets	5000	5000
Total 4070IA (RBS2G) base stations	1000	1000
Total 4070IO (IBS) base stations	256	256

3.3 NETWORK CAPACITY ENHANCEMENTS

	R3.0	R3.1
Max. nodes in the network	100	100
nodes in the back-bone (included)	32	32
Max. Trunk groups	2000	2000
total users (physical & fictive)	50000	50000

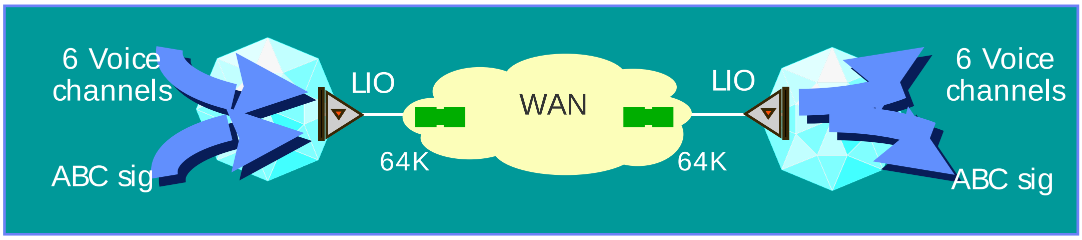
4. NETWORKING

4.1 OVERVIEW

4.1.1 Release 3 of A4400 (before C1.530)

The major evolution of networking in R3 concerns the **Integrated Voice Compression** platform and its applications.
Integrated voice compression will not only help cut private traffic costs, but also bring a set of new solutions to interconnect nodes.
These solutions complete the wide portfolio of solutions to interconnect A4400s without threatening existing network infrastructure.

÷ **Voice compression can be used on leased lines**, as voice and data multiplexers can. Two A4400 nodes can be connected through a 64kb/s leased line, with a capacity of 6 voice channels and a signalling ABC channel (see diagram).



÷ In R2, ABCVPN solutions require a permanent connection between nodes to carry ABC signalling, for instance on IP or on an X25 network.
In R3, it will be possible to have a complete VPN solution, that is without permanent ABC signalling channel: ABC signalling can be carried on a B-channel on ISDN, this B-channel being set up on demand, when there is traffic to exchange between nodes. This solution is called **ABC networking on demand** and provides all the ABC services.

With ABC networking on demand, nodes are interconnected only when there is voice or signalling traffic to exchange. When there is no traffic, the ABC node is isolated from the rest of the network, thus generating no traffic costs.

ABC networking on demand can also benefit from the services of the voice compression platform. Instead of using a dedicated B-channel on ISDN to carry ABC signalling, with voice compression this B-channel is able to carry the complete ABC link with 6 voices and the signalling, and this B-channel is set up on demand. The result is a powerful and inexpensive solution to interconnect small sites to an ABC network.

÷ Another application of voice compression is to cut the traffic costs of 2 companies with high phone traffic between them. This is called **2nd call for free**, and is something that looks like ABC on demand.

÷ A4400 will also provide **Voice over Frame Relay** natively thanks to the same integrated voice compression platform.

÷ Note: the integrated voice compression platform cannot be used to interconnect a main ACT to a remote ACT.

In R3, there is an increase in the homogeneous network capacity to **100 nodes and 50,000 users**, which brings a better service level than the supra-network configuration

In the multi-carrier environment, there are also considerable enhancements of the Automatic Route Selection (ARS) package. For instance, in R3, **ARS is time and day dependent**, and the **ARS server,** where ARS can be centralised in an ABC network, simplifies management and update. A4400 can now handle up to **"30-digit" public numbers**.

4.1.2 Release 3 of A4400 (from C1.530)

This version supports a new generation of LIOB, LIOP, LIOX boards called LIO2. The "voice compression" services are the same as in previous versions of R3, except that LIO2 boards support 8 voice compression channels instead of 6 before.

4.1.3 Release 3.1

In R3.1, Voice over IP (VoIP) is integrated.
It allows to provides toll bypass, that is telecom costs reduction, but also ABC services, throughout a private IP network.

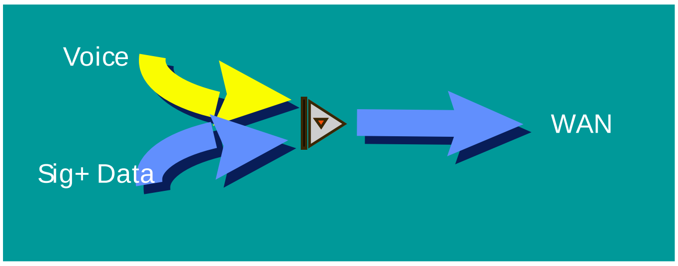
Remote ACT can be connected through a cheap 64kb/s leased line, thanks to the voice compression. This provides a low equipment cost, low cost of ownership solution for branch offices.

A)Release 3 (beforeC1.530)

4.2 INTEGRATED VOICE COMPRESSION PLATFORM: TECHNOLOGY

The Integrated Voice Compression platform is realised on one of a range of boards called LIOx. The type of LIOx to choose in the range depends on how the A4400s are interconnected. LIO means Link Optimiser.
Interconnection can be on a leased line, ISDN switched network and Frame Relay

This platform is in fact a voice and data multiplexer. On a multiplexer, there are 3 aspects:
- Voice
- ABC signalling and data
- Wide Area Network, also called composite channel



4.2.1 Voice part

The compression algorithm used on the Integrated Voice Compression Platform is G723.1.
The compression rate is about 6.4 kb/s
It is the default standard for low rate VoIP (voice over IP), and implemented on the Microsoft communication application Netmeeting.

- The quality of G723.1 is rated by what is called M.O.S. (Mean Opinion Score). 5 is the maximum quality, G711 (64k) voice is rated 4.2, G723.1 is around 3.9, like G729 at 8kb/s, and GSM is 3.5.
- Echo cancellation is integrated because of the delays due to compression.
- Fax G3 is supported at up to 9600b/s. Fax is demodulated and re-modulated to avoid using 64kb/s bandwidth on the Wide Area Network.
- Modem calls do not go through compression equipment.
- MFQ23 codes are interpreted, coded and regenerated on the other side. This avoids any problems with distortion, on which Q23 receivers are very sensitive.
- Lost Frame Interpolation helps to overcome loss of voice frames. This is particularly helpful on Frame Relay network where frames can be discarded in case of saturation.
- Voice activity detection, silence suppression and silence regeneration sends voice frames only when the parties actually talk and send nothing when the parties do not talk.
  This means that, in association with dynamic bandwidth allocation, 50% of time (therefore bandwidth) is saved and reallocated for other traffic e.g. data.

To avoid this, the parties think that the line is dead during silence, silence is regenerated on the other side.
- Loss less voice transit is provided to have only one compression decompression for the call even if it transits through several A4400s.
- The compression rate is static (6.4kb/s). When other vendors multiplexers provide dynamic compression rate, this mode is never used on the field because changing quality during one conversation is not accepted by customers.
  Note: dynamic voice compression means that the compression rate is reduced (e.g. from 8 to 4kb/s) in case the link is saturated and increased when saturation disappears.

4.2.2 Data part

Data support is necessary on Integrated Voice Compression Platform because ABC signalling needs to be exchanged between nodes. But low rate HDLC data can also be supported: see the voice and data integration chapter below.

ABC signalling is in fact HDLC data, and contains not only call handling signalling but also management, routing, supervision etc. of traffic.

Because of the dynamic bandwidth management on the A4400 voice compression platform, bandwidth not used by voice is used by ABC signalling and data (see Wide Area Network part).

Note for experts: 2 ABC signalling channels and 2 data channels can be supported per board. 2 ABC signalling is for multi-directions (see cost optimised solution for small traffic)

4.2.3 Wide Area Network part (composite channel)

Before being sent on the WAN, voice and data traffic are multiplexed. There are 2 methods of multiplexing and managing the WAN bandwidth:

- Time Division Multiplex
- Frame, cells or packets.

TDM books a fixed bandwidth for each channel. The advantage is that delay and delay variability are small. But the drawback is a waste of bandwidth which is not acceptable when using low speed access (64Kb/s, 128Kb/s): when a channel is not used or when voice is not active, bandwidth can not be used by other traffic e.g. data.

In the A4400 integrated voice compression platform, frame mode is used. This means bandwidth is allocated dynamically. Bandwidth can be reused by ABC signalling or data when there is one or more voice channels not set up or during silence part of conversations (see example in the voice and low speed data integration chapter). This means that at night for instance when there is no voice, charging tickets using almost all the bandwidth available can be transferred (eg.64Kb/s).

The problems brought by frame mode multiplexing are delay and managing the priority of traffic. To overcome these problems, several solutions have been implemented on A4400:

- Data frame fragmentation. User data frames can be very long: to avoid facing long delays when one long frame is sent, long frames are fragmented into smaller ones, according to data frame fragmentation.
- Frames (with voice) are sent regularly to minimise delay and delay variability.
- Priority is for voice. Frames to be sent on the WAN are first filled with voice information and space left is filled with ABC signalling and user data.

The WAN link can be a leased line, an ISDN channel or a Frame Relay Permanent Virtual Circuit.

Delay in the network (except Frame Relay)

- The delay for a voice call between 2 nodes end to end is between 40 to 90ms
- Each transit adds 10 to 40ms (loss less voice transit is available)

4.2.4 LIOP, LIOB and LIOX boards

The voice compression platform is realised with one or several units of a range of boards called LIOx: LIOP, LIOB, LIOX
Each board has a common part of 6 voice compression modules, each of the compression modules handling dynamically one voice or fax channel. The difference is in the external interface:

- LIOP has also one PRA access

- LIOB has one serial link and 4 BRA network access
- LIOX has no external interface

4.2.4.1 LIOP

LIOP is 2 boards in one: one LIOX plus one PRA board. LIOP is adapted for new systems

- The PRA access can be connected to the public ISDN network (T2): the same access will be used to carry private traffic (ABC networking on demand) and public traffic (the normal incoming/outgoing traffic with the public network).
  E.g. it is possible to build an autonomous network node with just a VH basic package (with CPU) and LIOB board, without any dedicated link to the other node.
- The PRA can also be connected to a G703/G704 Fractional E1 leased line. This is when high voice traffic is requested between the 2 sites.

4.2.4.2 LIOB

LIOB is a LIOX plus 4 BRA network access plus 1 serial link interface. LIOB is for new systems or for installed base when serial link interface is requested.

- The serial link on LIOB is either X24/V11 or V36 and is connected to a small capacity leased line (64, 128kb/s) or to a Frame Relay network.
  The serial link interface of LIOB can not be used to connect a terminal. It is to connect to the network. Furthermore, LIOB is mandatory in case of compression on serial link: the combination LIOX plus synchronous terminal adapter is not valid.
- The BRA access can be connected to a S0FV leased line, available in Germany or in Austria. The other BRA access on the same board can be used as normal ISDN access for public calls.
- The BRA access can also be connected to the public ISDN network (T0). the same access will be used to carry private traffic (ABC networking on demand) and public traffic (the normal incoming/outgoing traffic with the public network).
- The BRA access cannot connect to a S0 set or S0 PC board: the BRA access is only T0, not S0.

4.2.4.3 LIOX

LIOX is an extension board without external interface.

- LIOX can be used as an extension board to LIOP or LIOB when no external interface is requested anymore, but additional voice compression resources are needed.
- LIOX can also be used on installed base when the A4400 has already a PRA or a BRA board connected to the public network.

4.2.4.4 Pool of voice compression resources

Voice compression resources are in a pool on a node basis, and are shared between all the directions in a node. The benefits of this are:
- security, in case some compression resources are out of service
- in a central site, fewer compression resources are requested than the sum of compression resources on remote sites (Erlang law). But the loss rate should be very low.

4.3 INTEGRATED VOICE COMPRESSION ON LEASED LINES

Integrated voice compression can be used on leased lines. The benefit for the customer is to reduce the bandwidth of the leased line, thus saving money.

The A4400s can be interconnected through a lot of different interfaces: serial link X24/V11or serial link V36 or S0FV in Germany or G703/G704 (E1).

4.3.1 Integrated voice compression vs external voice and data multiplexers

Traditionally voice compression on leased lines is done by multiplexers, but the main tasks of multiplexers is to integrate the different data traffic of the company.

The applications of integrated voice compression platform are focused on voice, and therefore can be used in separate voice and data environment, which is not the case with multiplexers. Separate voice and data networks avoid redesigning the data network to carry voice, and can still bring big savings.
But the platform can also be used in integrated voice and data environment, with existing voice & data multiplexers or voice & data can be integrated by the A4400, without using external multiplexers (see further in this chapter).

Of course in some cases, external voice compression is better adapted, for instance in case of heterogeneous networks (PABXs from different manufacturers), or if data networking is the main driver and voice is only marginal.

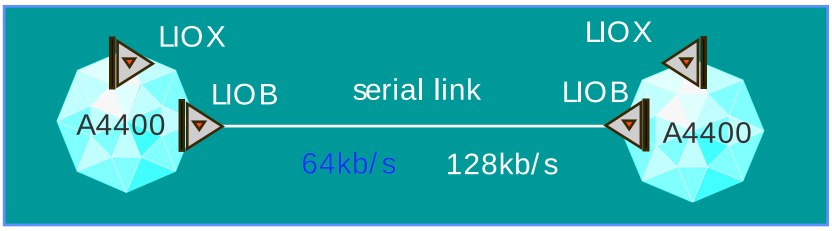
Because voice compression is integrated in the A4400, this has several advantages over external solutions with voice and data multiplexers.

- For the distribution unit:
  - Engineering is complex because every external multiplexer has different features and limits (analog or digital interface, transparent or switched mode, ...) requiring a case by case study and validation. With integrated voice compression, engineering is simplified . For instance, the A4400 OPS integrates voice compression platform
  - The logistic process also is simplified: installation, support and maintenance is done through the same channel as the PABX.
  - No need for a PRA or BRA to connect to the multiplexer, which corresponds to a price advantage.
- For the customer:
  - One single management for PABX and integrated multiplexer.
  - Secured power feeding: the same as the PABX

4.3.2 Integrated voice compression platform: connectivity on leased lines

4.3.2.1 Serial link leased line

- The A4400 can be connected to a serial link interface for networking.
- The physical interface is either X24/V11 or V36. LIOB use is mandatory.
- The access speed is either 64kb/s or 128kb/s
- A 64kb/s access is able to support up to 6 voice channels and ABC signalling
- A 128kb/s access is able to support up to 12 voice channels and ABC signalling
- If more voice channels are necessary, a G703/G704 leased line is necessary (E1 or Fractional E1).



4.3.2.2 <u>S0FV leased line</u>

* In Germany, leased lines are available on a 2B+D access called S0FV. This type of access is also available in Austria.
* The physical interface is a basic rate access. LIOB can be used, but also a BRA access on BRA8, or on BPRA (along with LIOX).

* A S0FV access with one B-channel (64kb/s) can support 6 voice channels
* A S0FV access with one 2B-channel (2*64kb/s) can support 12 voice channels
* If more voice channels are necessary, a second S0FV access can be added or a G703/G704 E1 access can be used

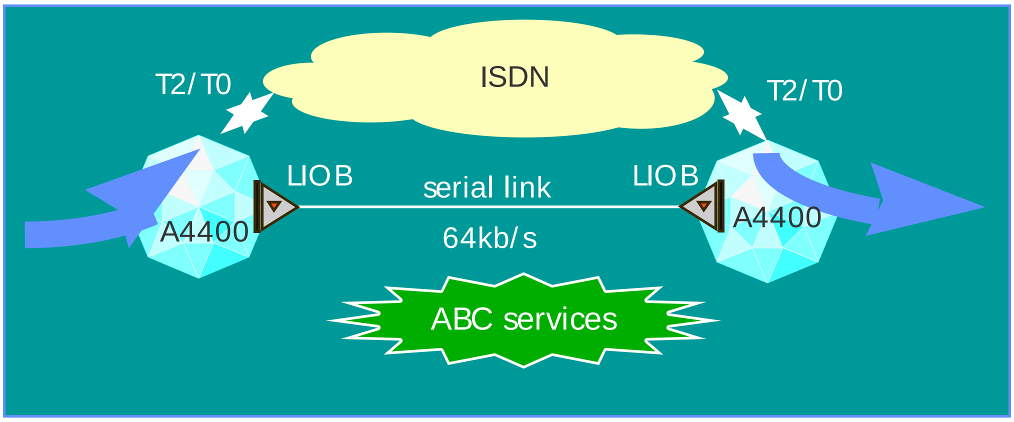
4.3.2.3 <u>G703/G704 E1 leased line</u>

* When more than 12 voice channels are necessary, a Fractional E1 leased line can be used.
* The physical interface is a Primary Rate access. LIOP can be used but also PRA.

* With a 256kb/s E1 leased line (the minimum in France), up to (6*4)=24 voice channels and ABC signalling are supported.

4.3.3 Overflow on leased line

When all the voice channels between 2 nodes are busy (e.g. 6 channels on a 64kb/s leased line), the 7th call can overflow on the public network, with ABC level of services, thanks to ABCVPN.

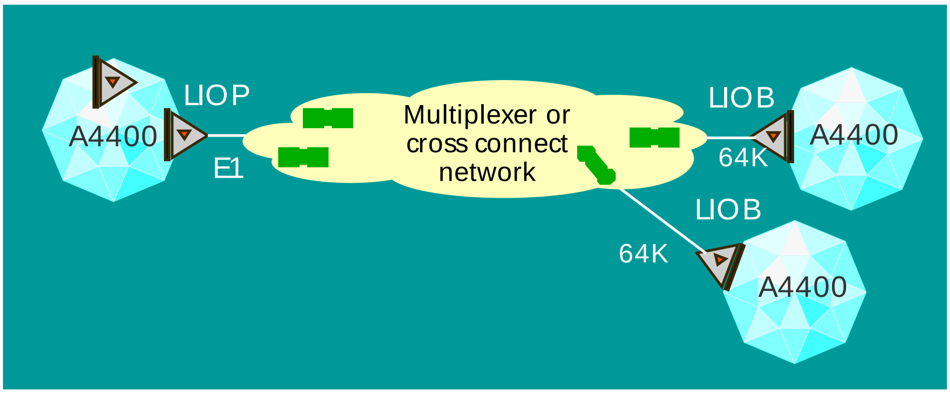


4.3.4 Multi-directions leased line

When building a star network with leased line, because leased lines are point to point, it is normally necessary to have as many accesses in the central site as distant sites: e.g. with 10 distant sites over 64K serial link it is necessary to have 10 serial link on the main site which requires a lot of hardware.

Some operators, in Portugal and now also in France (France Telecom) are offering multi-direction leased lines: this means that in the central site the leased lines from the distant sites are multiplexed on a single G703/G704 (E1) access. Some B channels are dedicated to direction A, other to direction B etc...

Distant sites can be connected with E1 or serial link 64K (to check with operator offer)



4.3.5 Voice and data integration

4.3.5.1 Downstream A4400 with installed TDM muxes

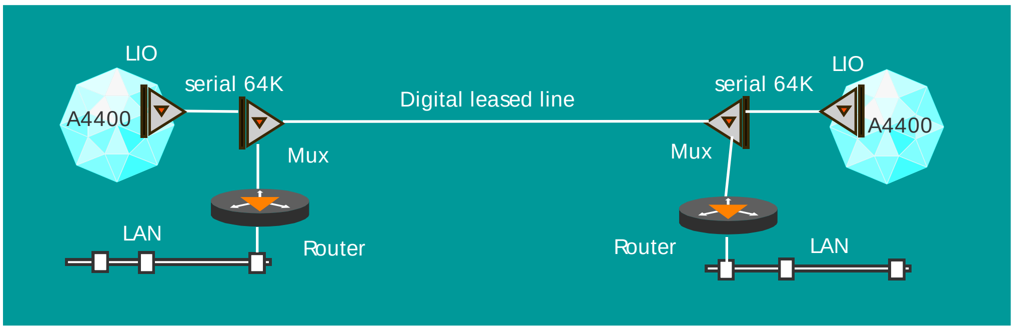
Instead of using voice compression on muxes, it is possible to combine integrated voice compression on A4400 and voice and data integration with TDM muxes (e.g. Newbridge).

If a customer is not satisfied with the voice compression offered by its muxes (compression algorithm not up to date) , he can replace it by Integrated Voice Compression on A4400, while keeping the TDM muxes for multiplexing (see diagram).

The serial link output of LIOB can be connected to a synchronous serial link input of the mux and carried as bit transparent data by the mux. Data coming from the LAN is multiplexed in that point, that is downstream the A4400.

In fact, the mux network is seen as a leased line network by the A4400 ABC network.

Instead of LIOB, if there is an existing PRA board on the A4400 to the TDM muxes, it can also be reused: channels are not voice anymore but data on the TDM muxe side.

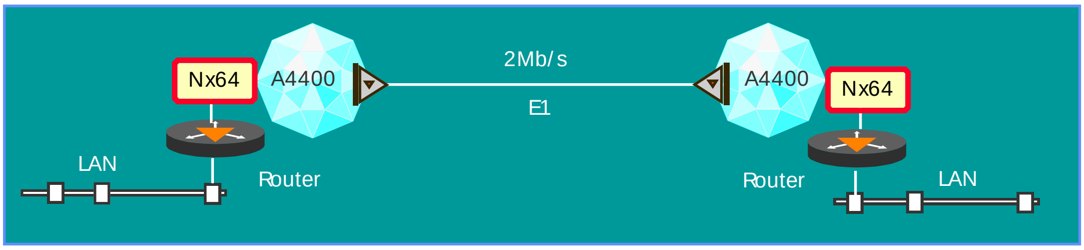


Restriction: This configuration can not be used with statistical muxes, because of delays. TDM muxes are in circuit mode, with guaranty of low delay mandatory for voice.

4.3.5.2 Upstream A4400 for high speed data up to 2Mb/s

In R2, by using the N*64 board, it is possible to interconnect routers up to 2Mb/s trough A4400s connected with a G703/G704 E1 or fractional E1 leased line. The leased line bandwidth is shared in TDM mode between voice plus ABC signalling and data: some B channels for voice, some B channels for data.

In R3, it is possible in addition to mention the benefits of voice compression. Integrating voice compression on A4400, will reduce the bandwidth required by voice by a factor of 6. The bandwidth saved can then be allocated to data, meaning higher service level for data users.

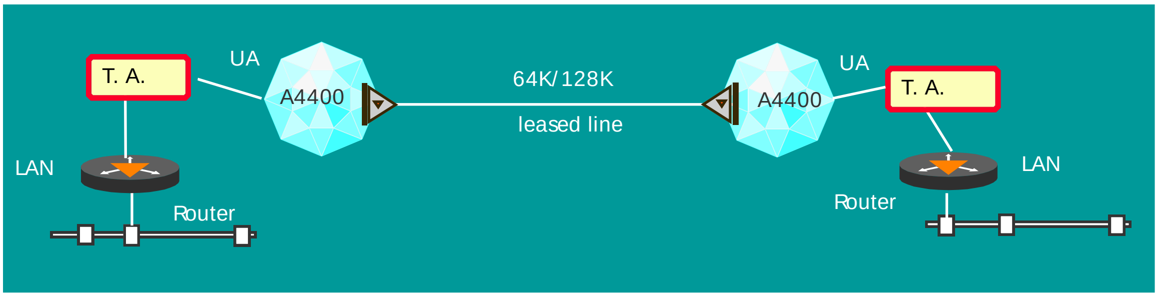


4.3.5.3 <u>Upstream A4400 for low speed data up to 64kb/s</u>

The integrated voice compression platform is a voice and data multiplexer, because it handles a special kind of data called ABC signalling.
It can also multiplex low speed HDLC data stream, for instance to interconnect a small branch router to a larger router in main site

At worst on a 64K leased line, with 6 voice channels, (64-6*7)=22kb/s is available for ABC signalling and data. But thanks to dynamic bandwidth allocation and silence suppression, (64-6*3,5)=36kb/s is available when 6 channels are busy and at night (when no active voice channel) the complete 64kb/s bandwidth can be used by data.

The data equipment must be connected to a synchronous 64kb/s Terminal Adapter (4083 SP + 4088) of the A4400. The data traffic must be HDLC, which is the case if the traffic from router is PPP or Frame Relay.
The A4400 is HDLC transparent in that case.
Voice has the priority over ABC signalling and HDLC data. ABC signalling has priority over HDLC data.



Note : when leased line is 128K and connection through serial link, TDM multiplexing is used : 64K for data, 64K for compressed voice.
This mode of voice and data integration only works on a leased line: it is not possible to integrate voice and data when the WAN link is either ISDN (ABC networking on demand) or Frame Relay.

* Concerning ISDN, data link between routers is permanent which is not compatible with ABC networking on demand where connection is up only from time to time.
* Concerning Frame Relay, A4400 is not a FRAD and is not able to multiplex different FR traffic: data is integrated downstream the A4400 in that case. See Frame Relay chapter.

4.4 INTEGRATED VOICE COMPRESSION ON ISDN NETWORK

Several applications are possible on ISDN

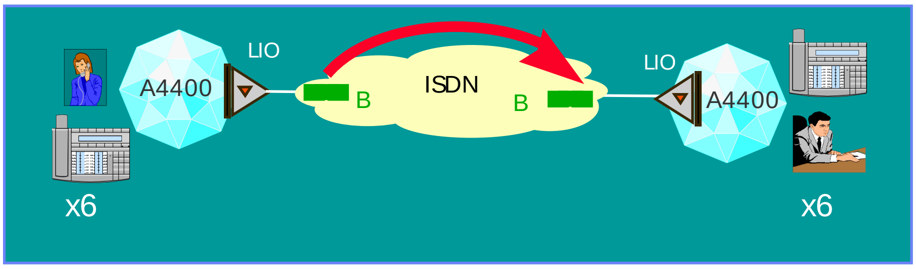
- ABC networking on demand (with voice compression platform)
- ABC networking on demand (without voice compression platform)
- ABCVPN with voice compression
- 2nd call for free

4.4.1 ABC networking on demand (with voice compression platform)

4.4.1.1 What it is

ABC networking on demand (with voice compression) is a new VPN solution to build an ABC network. There are in fact 2 aspects to this application:

* Voice compression on ISDN, which brings further savings compared to voice compression on leased lines
* ABC signalling on demand, without having a permanent link for signalling between nodes



This is a unique A4400 offer: no multiplexer can do this!

a) Voice compression on ISDN:
Compared to voice compression on leased line, with ABC networking on demand, a "virtual leased line" (supporting up to 6 channels and the ABC signalling) is set up on the public ISDN only when there is traffic and is cancelled when there is no more private traffic. With ABC networking on demand, associated with voice compression. The saving can be larger than with voice compression on leased line, especially when ISDN tariffs are "inexpensive" compared to leased lines, for instance in the local zone: at 10 km in France, a call of 9 hours on ISDN is the same as the daily price of a 64kb/s leased line.

Saving begins where there are overlapping calls, that is several calls in conversation at the same time. This is called the "2nd call for free" effect. The higher the traffic, the higher the probability of having simultaneous calls and higher the savings, as it can be seen on the diagram below. Savings increase up to a certain point, when the virtual ABC link is too often saturated, and overflow occurs.

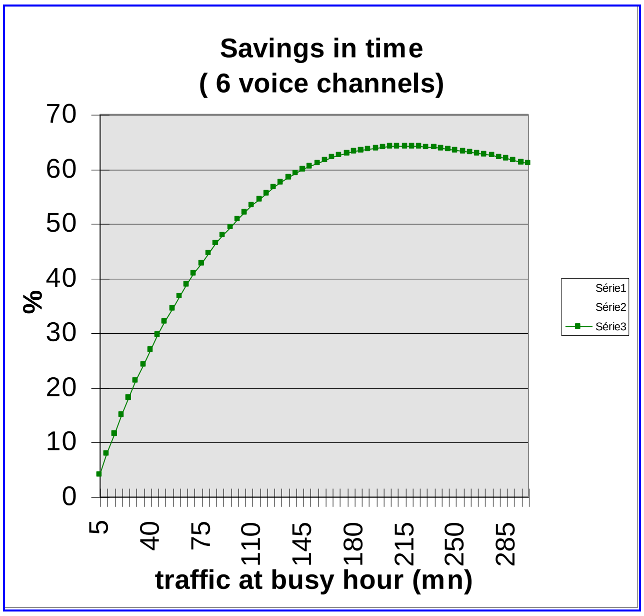


Diagram showing the savings realised with voice compression on ISDN, depending on the customer traffic between 2 nodes at the busy hour.

b) ABC signalling on demand

In R2, ABCVPN solutions require a permanent connection between nodes to carry ABC signalling, for instance on IP or on an X25 network. The IP network is not always present, or access to the PABX is not always granted. Using X25 network generates additional costs and cannot be used in every country.
Hence the idea of having a VPN solution independent of the data environment of the customer: the ABC signalling is carried in an ISDN B-channel. But having an ISDN B-channel set up 24H on 24H is too expensive: in R3, the ABC signalling channels can be established (respectively, released) on demand, i.e. when there is (respectively, no longer) traffic from users.

This does not require the integrated voice compression platform, but using this platform will enable the ABC signalling on demand to be carried in the same support channel as the 6 voices, without additional cost: this is ABC signalling for free!

Compared to ISVPN, this solution provides a higher level of service: ABC services. There is no need to pay for UUS, where the price is highly variable from country to country.

4.4.1.2 Benefit of ABC networking on demand with voice compression for a customer

The customer who can benefit from this service is a multi-site customer where A4400s are not networked.
With ABC networking on demand associated to voice compression, it is possible:

- to provide ABC services, that is services that will enable better services to be provided to users at lower equipment cost.
  e.g. a centralised voice mail will provide voice mail to users without installing voice mail on small site
- cut the private running costs of the customer (the telephone operator bill), thanks to voice compression on ISDN. e.g., if the customer has 120 minutes of communication savings will be as high as 57% at busy hour.
- some ABC services will increase the traffic between the nodes. This is the case of centralisation of attendants, or voice mail. But impact on cost is limited, let us take an example:

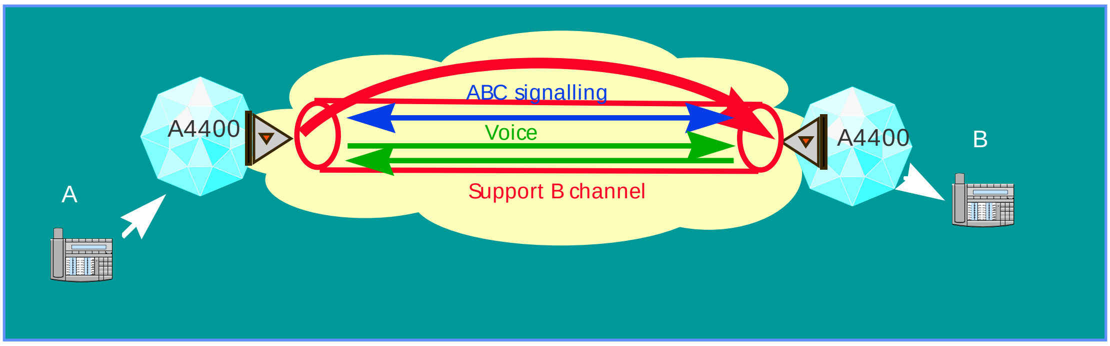
If traffic doubles from 60min to 120min at the busy hour, increase of cost is only 40% according to the diagram above:

  - 60 min of user traffic means 37 min of LIO traffic
  - 120 min of user traffic means 52 min of LIO traffic
  - and 52/37= 1.4, i.e. 40% running cost increase only with the user traffic doubling!.

  This is a strong argument for resource centralisation!

- without changing its WAN environment: the customer is on ISDN and stays with ISDN. No need to go to the operator to ask (and wait and pay) for a new service: leased line (installation cost), X25 packet network.
  There is no need either to use the existing data network, in general the IP network, which is sometimes closed to voice applications.

4.4.1.3 How it works



Originally ABC nodes are not connected to each other. When there is an internal call from a node to another, a data B-channel is set up on ISDN network. This B-channel is in fact a tunnel where ABC signalling is transported. When ABC signalling is exchanged between the 2 nodes, the first voice call will go through the same tunnel. A second voice call will go through the same tunnel ... up to 6 calls.
When there is no more voice traffic, the ISDN channel supporting the tunnel is released.

4.4.1.4 Capacity/modularity

* In the case of the saturation of the 6 channels, the 7th call will "overflow" and be a normal 64Kb/s ISDN voice call.
* If more than 6 compressed channels are necessary between the 2 A4400s, more channels can be supported between 2 nodes, for instance 12, 18 etc...When the first support B-channel is saturated (6 simultaneous calls), the 7th call will set up a second support B-channel. In order to avoid having several B-channels established and not fully loaded, new user calls are distributed in a sequential mode: the call will first try the first support B-channel, if saturated, the 2nd B-channel will be used etc...
  Of course, additional LIO boards are necessary in this case.

4.4.1.5 ABC services

A voice network is to transport voice but also to provide a lot of other services...

To provide these services, A4400s need to exchange information through ABC signalling ... and this will set up the link if it is down.

Some applications have evolved to avoid setting up the link to exchange information without any drawback on the service offered. This is the case of:
* Configuration data broadcast: these applications will wait for the link to be set up to exchange information
* Adaptive routing (see chapter below)

But other services cannot face any delay, they need to be real time and therefore will set up the link. For instance, in case of centralised voice mail in a node, if there is a Message Waiting Indication to notify to a set in another node, this information needs to be sent immediately.
* More generally, all call handling services will set up the link, independently of voice, for instance, automatic call back on busy subscriber.
* Alarms also must be real time: alarms also can set up the link, but to avoid excessive traffic due to alarms centralisation, it is better that alarms are managed through RMA.

Of course, if the link is set up, these services will not generate additional costs.

<u>Note</u>: to realise ABC networking on demand, it is necessary to understand in depth the ABC protocol, to know when to set up and when to release the link. This service is therefore almost impossible to be offered by external equipment like multiplexers.

In case the signalling traffic is almost permanent, for instance in case of a lot of set supervisions (boss secretary) or parallel hunting group in network, it is still possible to benefit from Voice Compression on ISDN. This is described below in the "ABCVPN with voice compression" chapter.

4.4.1.6 <u>Topology/ routing</u>

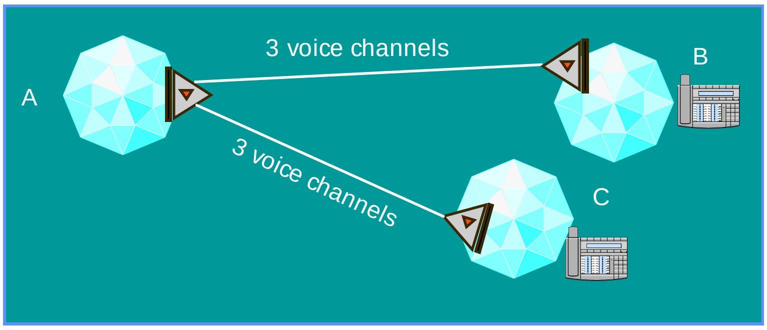
* When using ABC networking on demand (with voice compression), the best topology is a star. This is because star topology concentrates the traffic and this is good to maximise savings with voice compression on ISDN.
* Adaptive routing has evolved due to the non permanent nature of ABC on demand. Adaptive routing requires regular updates of the status of the network (nodes and links). This is not economically compatible with ABC networking on demand where the link must be up to exchange status information.
  On the other hand, routing is very simple because of the star topology mentioned above. To reach a set in another node, a branch node always sends the call to the same adjacent node: the main node.
  In the branch A4400, a routing table indicates to which node the call must be routed depending on the node of the called party. This adjacent node will then use adaptive routing to calculate the route. Because of this, update of status of network is no longer necessary in the branch node, which translates in avoiding unnecessary calls.

4.4.1.7 <u>Cost optimised solution for small traffic</u>

When connecting small sites to a network using ABC networking on demand, 6 voice channels are often largely oversized. 3 channels are in general enough (equivalent to 50 sets at 0.02 E). Furthermore thanks to ABCVPN, if the ABC link on demand is saturated, calls can "overflow" on clear 64Kb/s ISDN channels.

To offer an attractive solution to connect small sites special software licences for compression are per voice channel. Furthermore, fewer LIO boards will be necessary in the central site.

Let us take the following example with a central site A and 2 branches B and C:
* instead of ordering **4** LIO boards and 3 times licenses for 6 channels (**18** total)
* only **3** LIO boards and licenses for 6 channels in A and licenses for 3 channels in B and C, are necessary (**12** total)



This is of course to configure in the A4400 the right number of channels necessary per direction (eg.3 from A to B and 3 from A to C).

<u>Note: One LIO board can handle 2 directions, which means 2 ABC signalling channels.</u>

4.4.1.8 Technical requirements

* The only requirement is that the ISDN network should provide SUB-addressing service. Specific DID numbers are not required for the ABC networking on demand. It is therefore possible to use this solution on a node connected to ISDN that has no DID numbers.
* The ISDN support channel has the "data" Bearer Capability. In general, voice calls and data calls cost the same on ISDN.
* ABC networking on demand can be set up only over ISDN, but also QSIG-BC or QSIG-GF network.

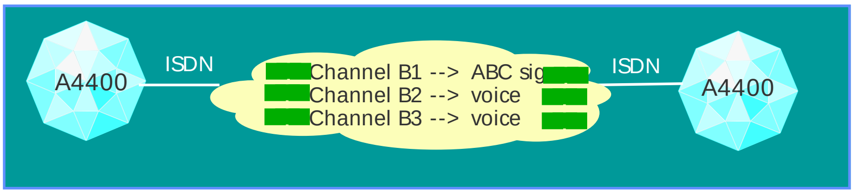
Licences: This solution requires voice compression plus ABC softaware licences.

4.4.2 ABC networking on demand without voice compression platform

Some customers are interested in ABC networking on demand, but are not willing to use voice compression platform:
* either because they want top quality G711 voice (64kb/s) without compression
* or the traffic is so low that investment in voice compression cannot be cost justified

Therefore, ABC network on demand is also available without the voice compression platform.

How it works: initially, ABC nodes are not connected. When there is an internal call from a node to another, a data B-channel is set up on ISDN network to carry ABC signalling. When ABC signalling is exchanged between the 2 nodes, a second ISDN call (ABCVPN) is established between the 2 nodes for the first voice communication and will benefit from ABC services. A second internal call will set up a third ISDN channel.
When there is no more voice traffic, the ISDN channel supporting ABC signalling is released.

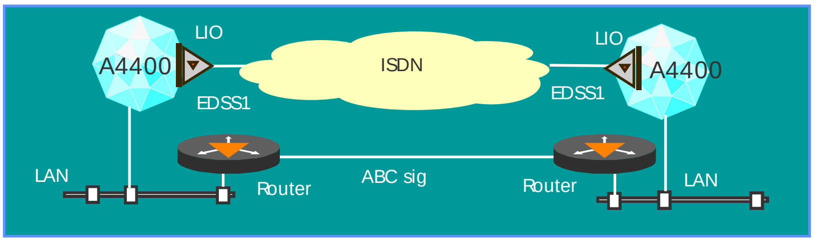


The drawback of this solution is that it is necessary to pay for the B-channel supporting ABC plus the normal voice calls at the difference of voice compression where the B-channel supporting ABC also includes up to 6 simultaneous voice channels.

Licences: This solution requires IO2 board and ABCVPN licences.

4.4.3 ABCVPN with voice compression

If the customer already has an ABCVPN network with signalling on IP and voice on ISDN, he can nevertheless benefit from integrated compression on ISDN.
The benefit will be reducing its running costs, without changing its WAN environment, just by adding voice compression boards and updating the system to R3.



In this solution, all the ABC signalling (call handling) will go through the IP network, and an ISDN call supporting the tunnel is set up (released) only when there is (no longer is) voice traffic.
Compared to ABC network on demand, there is no additional traffic generated for exchange of ABC signalling: this is an advantage when there is an almost permanent signalling traffic, even if the voice traffic is low, for instance in case of supervision (for Busy Lamp Field or boss secretary).

Licences: voice compression, ABCVPN software licences are required.

4.4.4 Second call for free

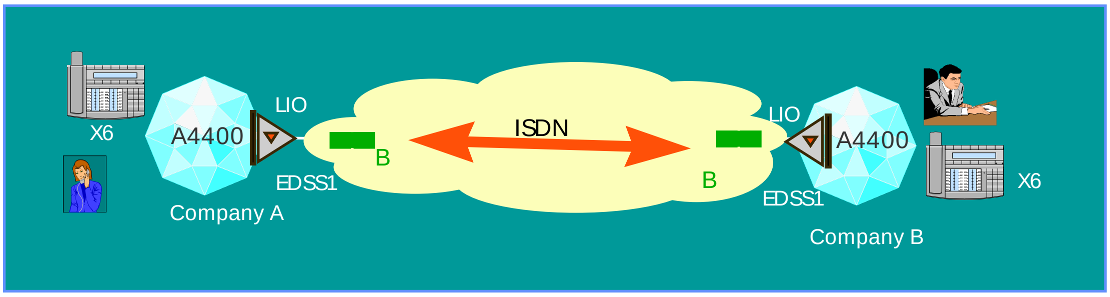
4.4.4.1 What it is

This is another application of voice compression on ISDN.

"Second call for free" is a solution to cut the traffic costs between 2 different companies or 2 sites of a multi-site company, without implementing an ABC network, which means changes to stand alone systems, like having an homogeneous numbering plan, common management etc...

This solution is for instance for 2 companies exchanging considerable voice traffic, for instance a car manufacturer with its suppliers.

Just by implementing "2nd call for free" (voice compression platform), 2 companies can cut their traffic costs:
* without changing their ISDN subscription with the operator
* without change of the dialling habits of subscribers: they dial the public number as before



4.4.4.2 Why "second call for free"

When there is a call between the 2 companies, this call will be identified (thanks to ARS) by the A4400. The A4400 will then set up a ISDN data call to the other A4400. Connection is made. The ISDN B-channel is in fact a tunnel where several compressed calls can be transported. The first call is compressed and transported in the tunnel. If a second call is set up by a user, it will be identified by the system and injected in the same tunnel. This avoids setting up a new ISDN call : this is why it is called second call for free. Of course, up to 6 simultaneous calls can be supported in one single ISDN B-channel.
When there are no more user calls on the tunnel, the support ISDN data B-channel is released. Calls can be in both directions

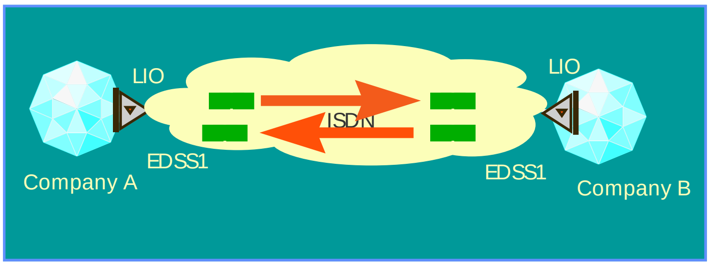
4.4.4.3 Capacity/modularity

See same paragraph in ABC networking on demand chapter

4.4.4.4 Who pays for what?

The cost of support ISDN B-channel is incurred by the PABX that set up the call until the release of the support channel and not shared between the 2 companies according to their respective outgoing traffic.

* A first approach to this is to say that this is not a problem because the next call could be in the other direction and that statistically over a long period of time, costs are allocated fairly to the 2 companies according to their outgoing traffic.
* A second approach is for customers dissatisfied with the first approach. The solution is then to specialise a support channel only for outgoing calls. This means that 2 support channels are necessary: 1 from A to B and one from B to A and each company only pays for its outgoing traffic.
  One LIO board per site is enough, with 3 voice channels in each direction. The drawback of this is that the pay-back is not as high as in the first approach due to the 2 support channels.



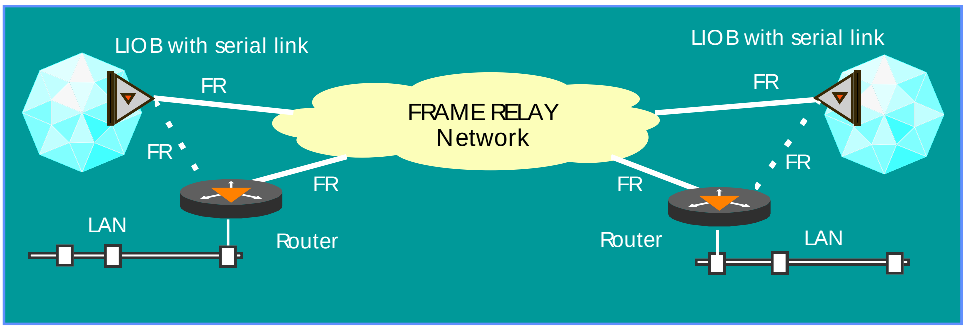
Concerning charging information, there will be a ticket for "real" cost, which is the cost of the ISDN B-channel on ISDN, allocated to the A4400 origin of the call.
There will be also "virtual" tickets, for communication supported in the tunnel channel, with cost 0, and allocated to subscribers.

4.4.4.5 Technical requirements

* only DID sets can be reached, as for standard ISDN calls
* ARS licence is necessary for "2nd call for free"
* SUB-addressing service is necessary on the ISDN network
* Note: Following ISDN services: Advice of Charge, Call Completion on Busy Subscriber, Malicious Call Identification, are not available through 2nd call for free tunnel.

4.5 INTEGRATED VOICE COMPRESSION ON FRAME RELAY

- Frame Relay is becoming a cost effective alternative to leased line for carrying data traffic. Frame Relay networks built by operators are also increasingly able to support voice, because operators announce typical delays around 70ms and this is compatible with voice support.
- As seen before, on the WAN side of the integrated voice compression platform, frames are sent and received. To adapt this frame mode multiplexing to a Frame Relay mode is technically very simple: The difference is that in the first case, the platform will try to fill the bandwidth available at access speed), in the second case the platform will follow the terms of the contract with the operator (Committed Information Rate, Excess IR, etc...) "independently" of the access speed.
- The application of this service is to build small A4400 networks over Frame Relay (3/4 nodes)
- There are 2 ways of connecting the A4400 to Frame Relay
  * directly to a Frame Relay operator, for instance when FR is cheaper than a leased line (continuous line in the diagram).
  * There can also be an unique access from the operator to the company. In this case, a piece of multiplexing equipment, such as a FRAD or a router is inserted between the A4400 and the Frame Relay network and multiplexes voice and data (dotted line on the diagram). In this case also, acceptable delay and jitter should be provided.
    The connection of the A4400 to the FRAD could also be in HDLC mode: the A4400 has the 2 modes to adapt to the existing equipment.



- The A4400 is necessarily connected to the FR equipment with a LIOB board with serial link interface.
  The access speed is up to 2Mb/s (tbc), but maximum 12 voice channels are supported by access.
  A4400 is not a FRAD, so data should not be sent through LIO when connected to Frame Relay.
  The number of PVC's per access is 4, meaning 4 directions (only 2 directions per board).
- Because each operator provides a different service and engineering is not trivial, a case by case analysis must be carried out for each business.

4.6 100 NODES NETWORK

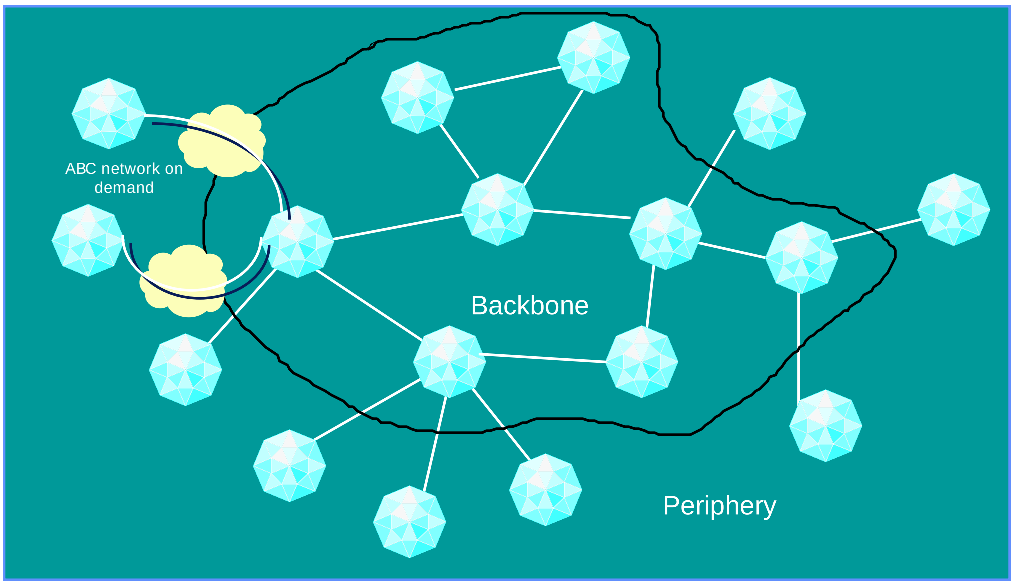
With R2, building more than 32 nodes network requires dividing the network into sub-networks (supra-network), but this has consequences on the level of ABC services: inside the sub-network the full level of ABC services is available, but between sub-networks there are some restrictions. Following services are not available:

- adaptive routing
- portability of number (moving)
- non QSIG-GF compatible ABC services like supervision, BLF in network, boss secretary
- centralized voice-mail
- management services like configuration data broadcast, audit
- ...

With R3, it is possible to build homogeneous networks of up to 100 nodes, with up to 50000 subscribers, with the whole set of ABC features.
The provisioning level is enhanced accordingly: for instance the phone-book has 60.000 names.

There has been an evolution in some applications to overcome problems generated by such big networks, for instance adaptive routing.

The typical topology of a 100 nodes network is represented in the following diagram.



The network is composed of 2 parts:

- the backbone nodes which have several connections to other A4400s
- the peripheral nodes, in general small branches, which are connected to one A4400 in the backbone.

4.6.1 <u>Adaptive routing enhancements</u>

- Adaptive routing requires regular exchange of status of links and nodes between A4400s to calculate the route in the A4400 of the calling party. In a 100 nodes network, adaptive routing would generate an important traffic of routing information and also a large load on processing resources to calculate the route.
- In the diagram above, routing from the peripheral node is very simple: if the call is to another node, the A4400 sends the call to the adjacent backbone node. There is no need here to have powerful route calculation or receive status of all nodes and links in the network.
- In R3, it is possible to move the route calculation from the node of the calling party to the adjacent node in the backbone. Call from peripheral node to another node is sent to the adjacent node, in the backbone which calculates the route. This avoids having routing traffic on the link between peripheral and backbone node and no route calculation on the peripheral node.

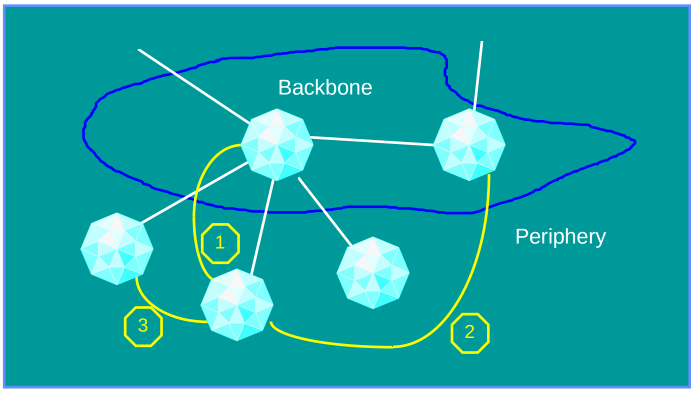
This is the same routing mode as for ABC networking on demand.

- <u>To connect peripheral to backbone node, all solutions are possible, but ABC networking on demand is particularly adapted because it is a solution optimised for low traffic.</u>

- In the backbone zone, the maximum number of A4400s is 32. Nodes in the backbone exchange status of links and nodes in the backbone zone
- To interconnected nodes in the backbone, all solutions except ABC networking on demand can be used: ABC signalling between nodes should be permanent.

4.6.2 ABCVPN

* ABCVPN is available in the instance of saturation of the link between peripheral and backbone node.
* Overflow can be carried out directly to adjacent backbone node, or peripheral node connected to the same adjacent or other backbone or peripheral node.



4.6.3 Homogeneous numbering plan

In a 100 nodes network, there are things which do not change: the numbering plan must be homogeneous and the number of digits of numbers is maximum 8 digits, like in R2.

4.7 ABC HYBRID LINKS AND ABCVPN ENHANCEMENTS

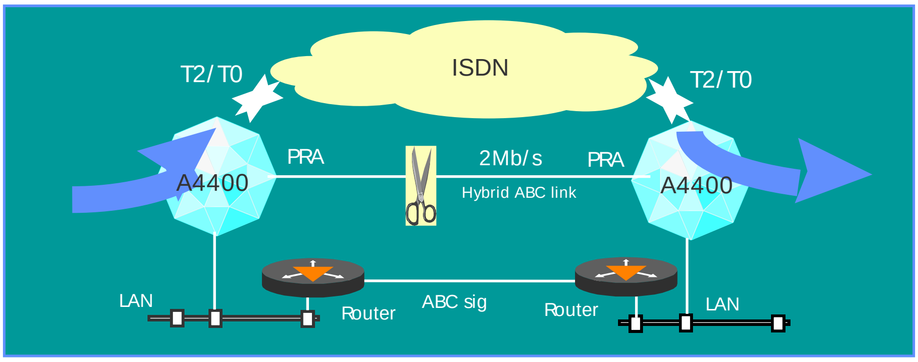
4.7.1 Multi-access hybrid ABC link

The technical means, among other applications, of handling more than 6 compressed voice channels between 2 nodes (refer to pre-sales presentation).

4.7.2 Backup ABC signalling

In R2, when a leased line (supporting voice and ABC signalling) is saturated between 2 A4400 nodes, additional calls can overflow on the ISDN network with provision of ABC services When the leased line is cut, calls can be set up but without ABC services.

In R3, a backup signalling path can be established between the 2 nodes, for instance on an IP network. When the main signalling, for instance in the D-channel of E1 leased line, is cut, ABC signalling is sent on the backup signalling path, for instance the IP network.



The backup signalling path can be permanent (IP) or on demand (on ISDN)
When the main signalling path recovers, ABC signalling is switched back to that main path.

Note or experts: Only ABC hybrid links can be backed up, not classical ABC links. But this is only a configuration issue, because it is also possible to build a hybrid link with ABC signalling in channel 16 of the E1 link.

ABC network on demand cannot be backed up (no backup signalling)

4.7.3 ABCVPN over non-ISDN switched network

* In R2, support of the voice part of ABCVPN calls is possible on ISDN type access: EDSS1, QSIG-BC, DPNSS(with gateway).
* In R3, ABCVPN can also use Channel Associated Signalling to set up voice calls on the switched network. This includes analog or PCM access, provided they support DDI: NDDI access are not supported . For instance available protocols are DC5A, R2, etc...
  This is useful in countries where ISDN is not common (Asia, Latin America) or when the customer has an installed mux network able to switch calls.
  It will be possible for instance to build an ABC network on analog lines:
  - ABC signalling on analog leased lines with modems
  - Voice on the analog public switched network
* PCM or analog interfaces can be ACT or US boards

4.8 OPEN NETWORKING

4.8.1 New QSIG-GF supplementary services

- Intrusion
- Do Not Disturb
- Call offer

4.8.2 DPNSS is integrated

- In R2, DPNSS was available on the A4400 thanks to an external ABC/DPNSS gateway (A4400 R2.1 with PRA trunks only). DPNSS is interworking with ABC.
- In R3, DPNSS is integrated in the A4400 and interworking with ABC.
- The DPNSS service level is the same as in R2.1 with the gateway.

4.8.3 ABCVPN and QSIG-GF interworking

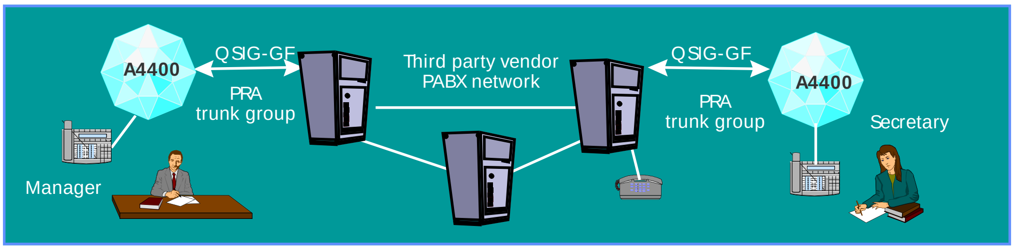
- In R2, it was not possible to have QSIG-GF and ABCVPN in the same network
- In R3, a call can firstly go on an ABCVPN path (between 2 A4400s) followed by a QSIG-GF path

(to a Siemens PABX)

- In R3, ABCVPN can be realised with the voice part on a switched QSIG-GF interface

4.8.4 ABCVPN over a QSIG-GF networking (ABC signalling in QSIG-GF D channel)

- With R2, the full ABC level of features is not available when A4400 are interconnected through QSIG-GF
  * No network update, no supervision, no centralised voice mail, no alarms...
- In R3, it is possible to use ABCVPN with:
  * Voice on B-channels of QSIG-GF access.
  * ABC signalling carried in QSIG-GF D-channel; several permanent "communications without B-channels" are set-up to carry ABC signalling channels.



- The number of directions per T2 QSIG-GF access is limited to 2, because of the load on the D-channel.
- This means that the size of the homogeneous A4400 network is limited. The application is when 2 or 3 A4400s are introduced in a big heterogeneous network based on QSIG-GF. This solution ensures that the full level of services between A4400's is provided.
- If more A4400s are added to the network, other solutions to carrying ABC signalling must be foreseen. In data B channels, on an IP network etc....

- Important notice: homogeneous numbering plan between A4400's!

Note: this has been tested, among other, with Datus multiplexers by the German R&D

4.9 MULTI-CARRIER ENHANCEMENTS

4.9.1 Enhancement of capacity of ARS

The A4400 is now able to handle up to **30** digits public numbers. The previous limit was 20.

The number of route lists is increased to **1000**. This is useful for forced on net on big public VPN networks with hundreds of sites, where there is at least one Route List per destination site.

4.9.2 ARS is time dependent

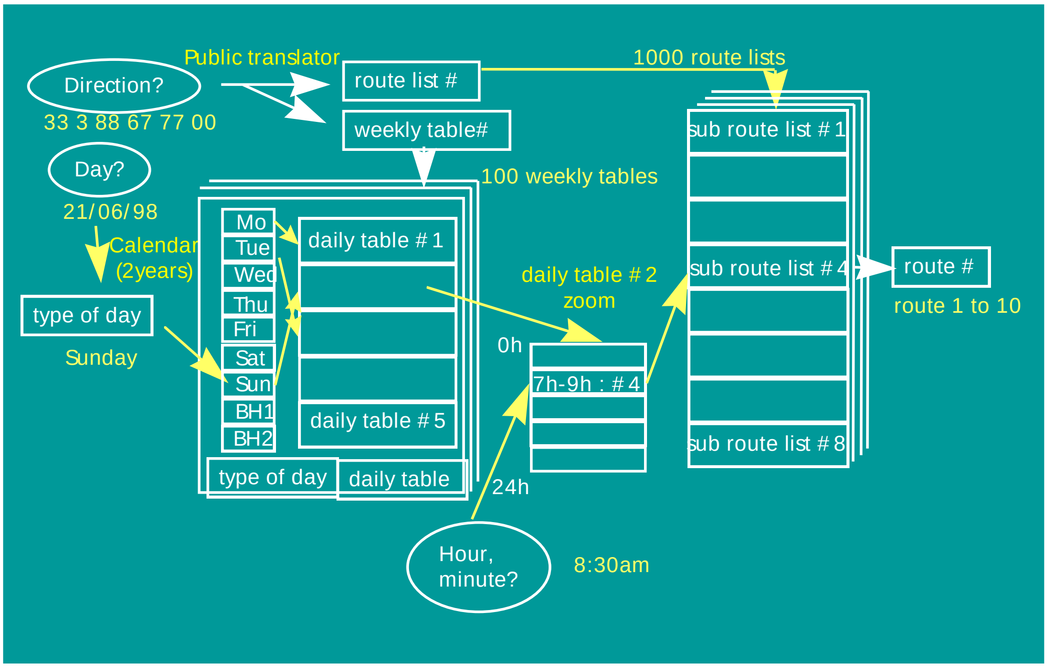
ARS is now dependent not only on direction but also on
- day of the week
- hour, minute of the day

4.9.2.1 How does it work?

The basic mechanism of R2 is kept, this means that a customer who does not want to use time of day ARS, can do so and has nothing concerning time to program.

When user is dialling a number, a direction is identified to which a route list is associated. The route list concept is enhanced in R3, because a route list is a set of sub route-lists which contains the routes. The sub route list in a route list will be selected according to time considerations.
What are the criteria?
- firstly, time zones depend on the direction: for instance off peak hours to US are not the same as for Hong-Kong. This is why a weekly table is associated to each direction.
- this weekly table (100 tables) contains for each type of day, the daily table to use.
- this daily table (5 tables/weekly table) contains the sub-route list to use depending of the hour in the day. 24 time zones are available per day.

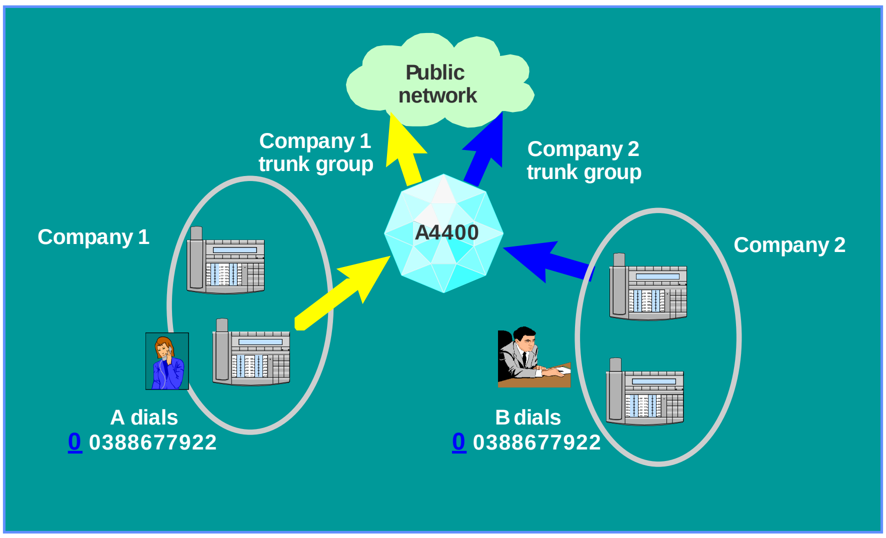
Let us take an example (see diagram below):
* a call is made on 21 June 98, 8:30am.
* A4400 checks if this day is a Monday, a Tuesday ... or a bank holiday. To check if this day is a bank holiday, the A4400 has a 2 year calendar. 2 types of bank holiday are available. 21st of June is a Sunday
* in the weekly table selected by the dialled direction, Sunday is associated to daily table #2.
* at 8:30 am sub-route list #4, is selected in the route list
* in the sub-route list, the route is selected.



4.9.3 ARS is dependent on calling party (multi-tenant)

In R3, ARS is also dependent on the calling party. This means that the routing by ARS of a public number will be different if this number is dialled by 2 different subscribers.

The application of this is mainly for multi-tenant installations. Each company has its own outgoing public trunk group to the public network (because of billing). All subscribers of all the companies dial "0" for public outgoing call, but call will be routed to the trunk-group of the company.
It is as if there were several public numbering plans in the A4400: one per company.



In fact, routing depends on the entity of the calling party. Up to 255 entities are available in the A4400.

Another application is when a company (not necessarily multi-tenant) has subscribed to 2 different carriers, one of which offers very inexpensive calls but not top quality voice (voice compression). Calls can be routed according to the requirements of the departments of the company. Commercial department requires top quality voice, R&D department looks for inexpensive calls. Calling party dependent ARS meets these requirements.

4.9.4 ARS handles data traffic

ARS tables for data and voice calls are the same. Voice and data calls use the same route lists but data calls will skip routes that are not compatible with data.

New carriers will perhaps not support all types of calls as incumbent carriers do. E.g., the first route is an ISDN access supporting only voice, but no fax, no modem, no data: a data call will not take this first route and overflow to the second.
Quality parameter is associated to a route.

* If quality of the call is not compatible with quality of the route, the route is skipped and the next route is chosen.
* Different traffic (quality) types are supported

4.9.5 ARS server (centralised ARS)

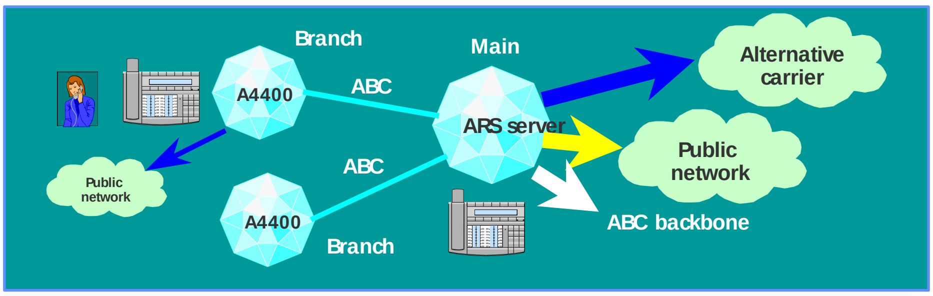
4.9.5.1 The problem

Programming and updating ARS information is a heavy task in an ABC network. Nodes must be managed one by one, there is no configuration broadcast. This is because routing is not the same in different nodes: it is easy to understand that ARS is not the same for a system in Paris and a system in London even if they are networked.

For some types of network, for instance, a star network with a big central node and several small branches located in the same area, ARS configuration is almost the same in all the nodes, except for the local area calls which go through the local trunk group: for instance, all the international calls are routed through trunk groups located in the main node.

4.9.5.2 The solution: ARS server

The idea is to have one central point (the ARS server) where all the complexity of ARS is handled and frequently updated and a branch where a very simple and stable ARS is programmed. The ARS server is located in the main node of the network.



Let us take an example:

* A subscriber in a branch dials a public number, the local ARS manages the zone barring and checks if the call is a local public call or an internal call (forced on net). If this is the case, the local ARS handles the call. If not, for instance with an international call, the local ARS hands out the call to the ARS server in the headquarters, which makes a more precise

analysis. The ARS server can analyse if the call is for the US or for Australia, the local ARS just knows that the call is an international call.

* ARS server can also be used in a campus network, where all the public calls are sent to the ARS server node, even public local calls. If the ARS server node is out of service, calls can overflow on the local trunk group.

* Zone barring is done once, in the originating node.

* The ARS privilege of the subscriber is handled by the ARS server. This means that a second level of barring can be done (in addition to zone barring). This is the case only if nodes are interconnected through ABC. If a branch is connected through QSIG, ARS privilege is not handled by ARS server.

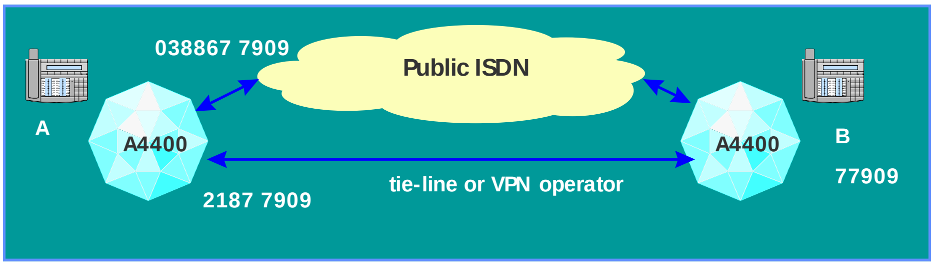
4.9.6 ARS handles private external numbering plan (several public translators)

Up to R2, private external numbering plans were managed using abbreviated numbers and trunk group prefixes. Private external numbering plans are used when the A4400 is connected with tie-lines (analog or digital) to other PABXs . This is not easy to manage due to the limitations of abbreviated numbers, concerning barring and overflow.

In R3, private external numbering plan can be managed with ARS. In fact several "network translators" are implemented in the A4400. Each translator is selected according to the prefix dialled by the user. When dialling "0", the standard public PTT translator is used. When dialling "2", the "private" network translator is used. Then specific barring can be done (up to 30 digits analysis), overflow from one trunk-group to another can be programmed in a very simple way.
There are up to 255 network translators.

When the connection is made through ABC, QSIG or DPNSS, ARS cannot be used for the private numbering plan. Existing mechanisms (routing prefix, open numbering plan...) must be used. Only tie-lines or ISDN (EDSS1) access can be used with ARS.

This is applied when 2 or more external numbering plans co-exist on a A4400. This is the case when a company has a subscription with a VPN operator providing a particular numbering plan to the company, for instance Alcanet with the prefix 2.



* For on-net directions, barring can be done and overflow programmed. For instance for a French site, access could only be allowed to French Alcatel sites. When the Alcanet trunk-group is saturated, overflow occurs: when dialling 2 187 7909, ARS transforms the number to 0388677909 and send this number on the France Telecom trunk group.
* For off-net directions, the user has 2 means of calling a destination: either with 0 or with 2. Whatever the prefix dialled by the user, ARS will choose the cheapest carrier according to what is programmed.

Note: this can also be used in case the VPN access and the public access are the same: e.g. Colisée in France and VipNet in Norway. In case of saturation, calls to Alcanet can overflow to France Telecom or France Telecom to Alcanet.

In R3, it is possible to have different zone barring according to the prefix dialled by user.

4.9.7 ARS handles low quality of service carriers: congested route tagging

Problem (see diagram):

Subscriber A cannot reach the called party because alternative carrier network is often congested.

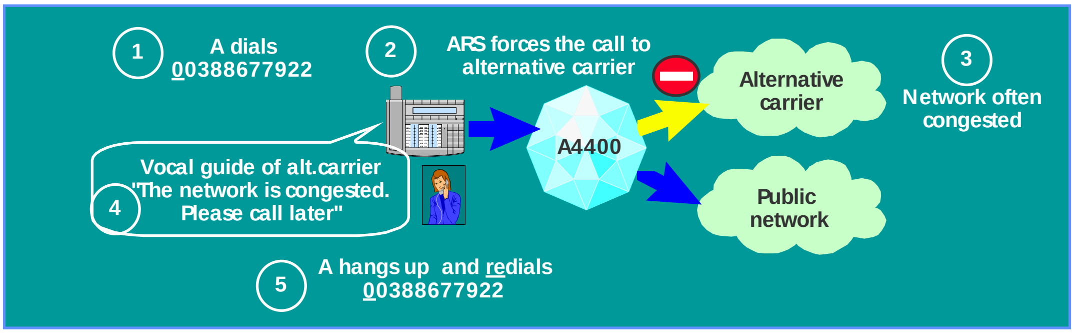
There is a route available through the public network (incumbent operator), but ARS forces the call to the alternative carrier.

* If trunk group to alternative carrier is ISDN

If the alternative carrier network is congested, automatic ARS overflow to public network (incumbent operator) when backward info from the ISDN network triggering the ARS overflow

* If trunk group to alternative carrier is analog/PCM
  ARS cannot detect that the network is congested: only voice guide from the alternative carrier heard by the caller

The solution: ARS "congested route tagging"



A dials its number, the call is routed to the alternative carrier. The network is congested. A gets a voice guide from the alternative carrier. A hangs up. The route is marked (tagged) unavailable for the set that has made the call.
If A re-dials the same number, ARS will skip the marked route and go to the public network.
The tagged route is erased after a time out.

* This mechanism is set up only when the trunk group has been qualified "suspect" by management
* Also works with indirect access (linked to the route not to the trunk group)

4.9.8 Several DID translators for multi-carrier incoming traffic

Due to deregulation, alternative carriers are not only interested in long distance traffic but are also building local loops to directly connect customers without going through the incumbent operator.
When installing the local loop, the operator also gets blocks of numbers that are distributed to customers directly connected.

This means that a customer will have 2 trunk-groups with incoming calls, 1 to the incumbent operator and another to the alternative operator with 2 different installation numbers and DID blocks.
Note: the customer will keep its numbers on the incumbent operator until number portability is implemented in the public network, which is not for today. Companies are in fact reluctant to change numbers.

In R3, the A4400 will be able to translate several blocks of DID numbers to the local numbering plan (directory number).

4.9.9 Calling party identification per carrier

In R3, A4400 has new possibilities for handling Calling Line Identification number to take into account multi carrier ISDN environments

When a A4400 is connected to 2 carriers (see 4.9.8), each set can have a DID number per carrier. The Calling Line Identification sent in case of outgoing call is dependent of the carrier that is used (the trunk group)

4.10 NEW NETWORKING SOFTWARE LICENCE

New licences for integrated voice compression platform

This is not finalised completely. The principle is the following:

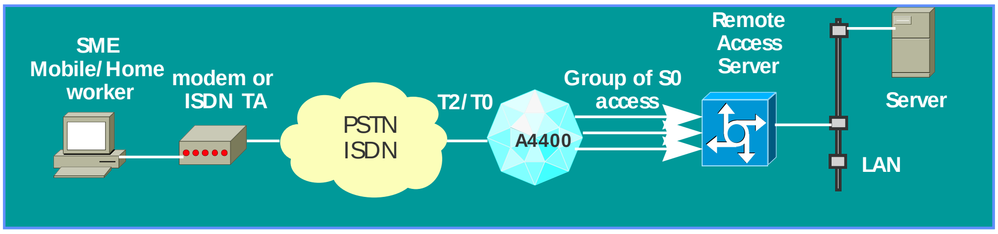
- A customer wanting to implement this service will have to pay for:
- LIO boards (LIOP, LIOB, LIOX) according to the external interfaces that are requested on the A4400 node (serial link, T2, T0 etc...)
- the number of compression resources necessary on that node. The granularity is not defined at the moment.

Refer to WPL

4.11 DATA SERVICES

4.11.1 S0 data pool to connect Remote Access Server

- R.A.S. is Access of remote terminals or small branches, through the public switched network, to the
data applications of the company. The traffic is to small to justify a leased line between the remote site and the central router. The access of the remote site is done through the switched network, either ISDN (ISDN connection on the PC) or analog (modem connection on PC- from C1.530 version).



- A4400 and R.A.S. can now be connected through several S0 access (several channels are necessary to support the traffic).
The benefit is having a common public access instead of having 2 public access: one for the PABX, one for the R.A.S. Internal access from another node is also possible. This solution is for Small to Medium Enterprises.

Example of utilisation:

- In the Cisco range of Remote Access Servers, the 3620 can be used in this configuration with the following restriction: it can only support ISDN remote terminals; not remote analog modems (to be checked in Cisco catalog).
The 3640 with PRI supports both remote ISDN and analog. For remote analog modems it has integrated digital modems...but it can not be connected to the A4400. The A4400 has no S2 interface and the Cisco no T2...

- Other R.A.S. with multi BRI: Ascend Max 2000 (with integrated digital modems) or Shiva Lan Rover E plus.

CAUTION: these are possible examples. It has not been validated.

- Other application: In Germany, E-mail server

Restrictions

- Not possible to connect a R.A.S. to A4400 with T2.RAS is connected directly to public ISDN.
- "ISDN S0 data pool" is not available for S0 voice terminals.

4.12 VN6 FOR FRENCH ISDN NETWORK

France Telecom will provide ETSI interface on T2 and T0 public ISDN interfaces on the VN6 level of its ISDN network.

There is no new ISDN supplementary service provided by ETSI interface compared to existing VN4 interface (Numéris).

FT claims that it will continue to support VN4 interface for a few years but...

To anticipate phase out of VN4 by France Telecom, first for new connected customers, afterwards for installed VN4 base, A4400 R3 is now compatible to France's ETSI (already in R2.1).

B)Release 3.1

4.13 INTEGRATED VOICE COMPRESSION: EVOLUTIONS IN R3.1

New hardware of LIOP/LIOB/LIOX boards are available in R3.1. These boards are called LIO…2 and will replace previous design.

- LIOP2 replaces LIOP
- LIOB2 replaces LIOB
- LIOX2 replaces LIOX

The main difference between LIO and LIO2 boards is the number of voice compression channels per board.
6 Voice compression channels were available on LIO range.
8 voice compression channels are available on LIO2 range.
The benefit of this increased capacity per board is that less boards are necessary to provide the same number of channels in a node.

Furthermore these new boards also provide following enhancements in R3.1

- G729A voice compression algorithm
- 8 voice channels per 64kb/s leased line

4.13.1 G729A voice compression algorithm

In R3, the compression algorithm is G723.1. In R3.1, the customer will have choice between G723.1 and G729A.

G729A is the standard voice compression algorithm for Voice over Frame Relay.
Compared to G723.1:

- G729A provides the same voice quality than G723.1. The Mean.Opinion.Scores are the same (around 4/5)
- G729A uses a little more bandwidth than G723.1: 8kb/s instead of 6.4kb/s
- G729A offers better interoperability with other voice compression algorithms: voice quality is slightly better when communication with DECT, GSM or voice mail.

But only one of these algorithms can be used in a customer network. G729A should be used in all the network and for all voice compression applications (legacy voice compression, VoIP, remote ACT with compression) for better voice quality (transit without decompression)

Conclusion: as G723.1 and G729A have comparable performances, G723.1 is the default algorithm. G729A is also provided if the customer asks for it. But G729A is an option: see "what to order).

4.13.2 8 voice channels max per compressed access (per 64kb/s channel)

8 voice channels are supported on a 64kb/s channel (leased line serial or E1, ISDN), instead of 6 in R3.0.
The benefit is faster payback, as more channels are supported in the same bandwidth

4.13.3 What to order for "legacy" voice compression in R3.1

Same as in R3 plus.
Option G729A can be ordered

Note: the number of “voice compression channels” software licences is dedicated for “legacy” voice compression (voice compression on ISDN, leased line , Frame Relay as opposed to VoIP), voice compression for VoIP and compressed remote ACT are different software licences.

4.13.4 LIO2 on R3.0 (release C1.530)

LIO2 can also be used on R3.0 on the support version of R3.0. 8 voice compression channels are available per board.
But G729A and “8 channels per 64kb/s” are not available.
Use of LIO2 boards require upgrade to C1.530 release.

4.14 INTEGRATED VOICE OVER IP IN R3.1

Up to now, VoIP was available on A4400 using an external PC based VoIP gateway, called VIP PC.
In R3.1, VoIP is integrated:

- This allow the A4400 to stay more than ever an up to date system ahead from the competition.
- From the beginning A4400 is open and is becoming more and more open to IP (47xx management through IP, ABC through IP, CSTA on IP). VoIP is therefore a natural evolution
- A4400 integrated VoIP completes the already comprehensive list of integrated “packet voice” solutions to interconnect A4400s, which includes ATM, voice compression over leased line, ISDN (ABC on demand) and Frame Relay. This allows to easily integrate A4400 ABC networks in existing or new network infrastructure.
- It is an investment for the future. At the difference of other packet voice technologies, like FR and ATM, IP goes to the terminal. So VoIP will also enable communications between standard telephone and VoIP enabled terminals.
  The VoIP tele-worker and Web Enabled Call Centre are the first application in R3.1. In the following A4400 releases local VoIP terminals are supported: it is A4400 IP PCX.

VoIP first application is cost reduction on the Wide Area Network for a multi-site company, also called “toll bypass”.

VoIP in A4400 will is used to build an ABC network over an IP infrastructure, i.e. interconnecting A4400s. The benefit is to reduce running costs and provide ABC feature transparency as well as web call centre and teleworker.

How can VoIP cut running costs?

- If the customer has an IP infrastructure linking its sites, he can route voice and fax traffic over the IP network, at no additional cost, instead of using the public “toll” network.
- With the same available bandwidth, voice compression will enable to carry 8 calls where only one can be carried on the public network (because of overhead, the actual figure is lower)
- Silence suppression will cut by a further 2 factor the average WAN bandwidth necessary for one voice channel.
- Data is transported on the same link than voice. More data can travel on the link, than if voice and data were on separate networks (statistical multiplexing)

Where will VoIP be implemented first?

VoIP will be used when the customer wants to speed up the investment he has done in a “high-performance” private IP network.
But voice quality is an issue in IP networks. The place to implement VoIP first, is in simple IP networks where performances of the IP network can be managed.

- Where VoIP runs toward end point nodes, that is without transit
- Where end point node is connected trough a point to point connection
- Where voice traffic is low

The place is networks with branch offices, with star topology. R3.1 targets branch offices with more than 30 subscribers.
Such an IP network is easier to manage than a meshed IP network with complex routing as can be found in campus networks.

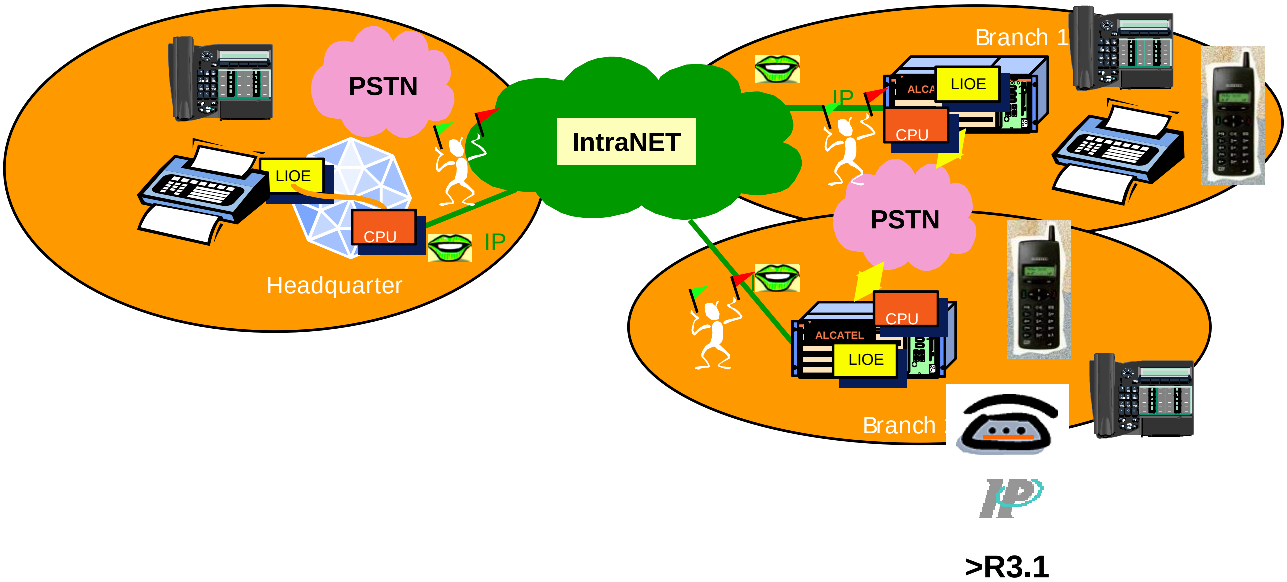
However, cost cutting is not the only benefit brought by VoIP in A4400. This solution also brings a high level of feature transparency through the network: the ABC services.

Networking on the A4400 is not just infrastructure providing basic call service; a lot of other services, enabled by the ABC protocol, bring the following benefits:

- Optimizing equipment costs, e.g. centralized voice mail, directory
- Sharing centralized resources, e.g. centralized call center, attendants, management.
- Enhancing communications between users: automatic call back, DECT roaming, boss/secretary, set supervision.

ABC also allows further running costs cutting thanks to the least cost routing services (ARS/LCR) of the A4400:
Break out service (hop off) will enable to route public calls on the IP network and step out of the private network on the closest A4400 to the public called party (take care that the public operator does not compress voice also).

To prevent quality problems on the IP network or to bypass these problems when they arrive, A4400 is able to tag VoIP packets (802.1p/q, Diffserv).and implements "guaranteed call delivery" services to ensure that:
- calls can be set up, even if number of VoIP channels is very small
- conversations are of good enough quality



4.14.1 Technology

4.14.1.1 LIOE Hardware

VoIP is realised in the A4400 by an ACT board called **LIOE**.
Note: LIOE is dedicated for VoIP and can not be used as extension board for "legacy compression"(voice compression on ISDN, leased line , Frame Relay)

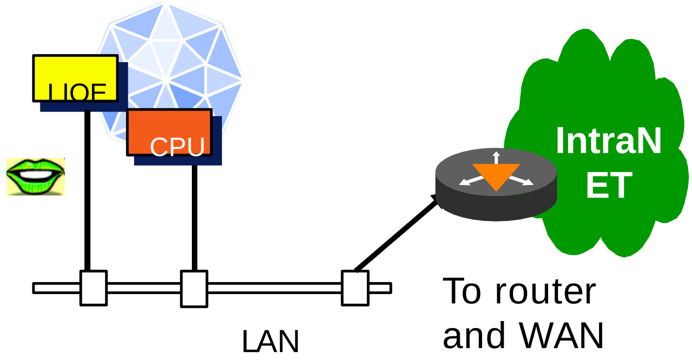
This board concentrates the voice traffic coming from various A4400 boards, compresses this voice channels, packetises the compressed voice and send VoIP packets on the Ethernet to a LIOE in another node.

To perform that, LIOE is equipped with 12 voice compression channels (on the hardware point of view).
The capacity per LIOE can be increased to 30 channels by putting a daughter board called **COMP18** (former CHC12) which brings 18 additional voice channels.

Each LIOE board has also an Ethernet interface to be connected to the LAN (the "IP trunk group").

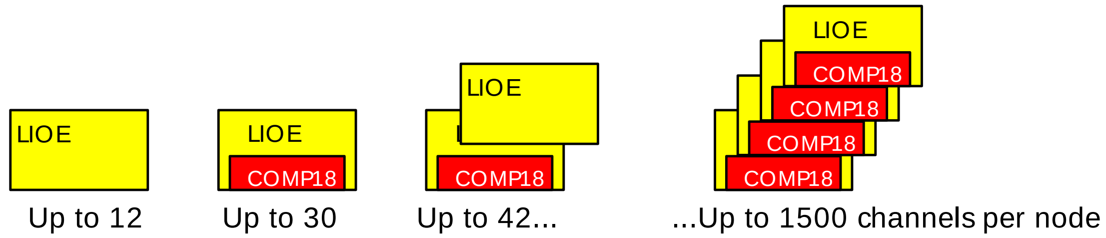
A connecting board called **CBC1**, brings RJ45 connection to the LIOE, so that the interface is 10baseT (10megabits/s)

Note: it is recommended that the VoIP access is connected to a switched port of a LAN switch for better manageability of bandwidth, therefore better voice quality



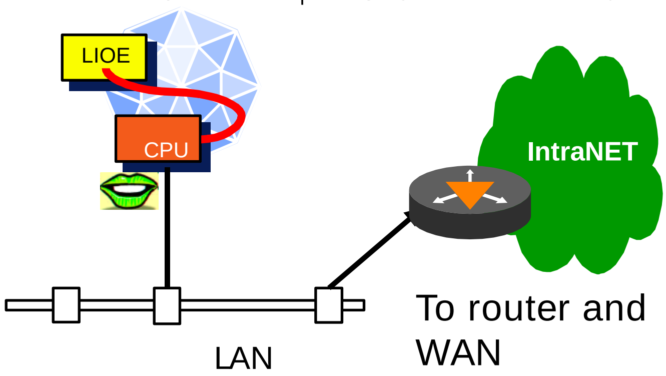
4.14.1.1.1 *Scalability*

Many LIOE boards can be plugged in a A4400. Up to 50 boards in fact.
This means than the same A4400 equipment can be used both in small branch offices (with a few VoIP channels) and very large headquarters (with hundreds of VoIP channels), which is unique on the market .



4.14.1.1.2 *Mini-hub*

As shown on the diagram below, A4400 has already a connection to the Ethernet on the CPU (for management, A4980, ABC signalling etc…)
To save ports on the hub or LAN switch, all the IP ports of the A4400 can be grouped onto one single IP port. The limit is 1 or 2 LIOE boards plus CPU on the same 10baseT access



4.14.1.2 Voice processing

A4400 Integrated VoIP is supporting voice and fax traffic. Voice can be compressed and fax demodulated to save bandwidth on the IP network. Then voice is encapsulated in IP packets and sent on the Ethernet to a LIOE in another A4400.

4.14.1.2.1 *Voice support*

There are 3 voice compression algorithms available for VoIP in the A4400:

- G711. The well known PCM coding at 64kb/s
- G723.1: the low rate compression algorithm at 6.3kb/s

- G729A: the low rate compression algorithm at 8kb/s

- The quality of voice compression algorithms is rated by what is called M.O.S. (Mean Opinion Score). 5 is the maximum quality, G711 (64k) voice is rated 4.2, G723.1 is around 3.9, like G729.A, and GSM is 3.5.

- Which compression algorithm to use?
Due to the high bandwidth required for G711, PCM coding should be used in LAN environment but is not recommended in the WAN. This is due to the 64kb/s coding but also to the overhead generated by the encapsulation in IP packets, which means around 100kb/s bandwidth at the Ethernet level (for each way A to B and B to A!)
The choice is between G723.1 or G729A.
These 2 algorithms provide almost the same performance: see G729 chapter above. In the VoIP implementation of G729A, there not a significant advantage of G729A concerning delays (packets are sent at the same rate as for G723.1, every 30ms)
So G723.1 is recommended, but G729A can be used
– if customer requires it
– if G729A is used in another part of the network, for instance voice compression on leased line, for better interoperability

- Echo cancellation is integrated because of the delays due to compression and transport on IP network.

- MFQ23 codes are interpreted, coded and regenerated on the other side. This avoids any problems with distortion, on which Q23 receivers are very sensitive.

- Lost Frame Interpolation helps to overcome loss of voice frames. This is the case in IP networks where frames can be discarded in case of congestion. But of course, packet loss rate should not be too high.

- Dynamic jitter buffer (per direction).
Successive voice packets of the same voice channel are sent regularly on the IP network. But these packets will not face the same delay in the IP network to reach the receiving A4400. This is called jitter or delay variation. To be reassembled and sent at regular intervals to the listener, voice samples are buffered in the receiving LIOE. This will introduce additional delay. If jitter is small, which is the case where the IP network is running well, the buffer is small and the additional delay is low. If jitter is high, the buffer will be larger and the buffer delay too. Dynamic jitter buffer size will allow to avoid introducing additional delay when the IP network runs well.
There is one jitter buffer per direction. This means that if packets received from site A with a high jitter and packets received from site B are with a low jitter, the delay due to buffer on calls from site B will not suffer from high jitter of packets from site A.

- Voice activity detection (VAD), silence suppression and silence regeneration sends voice frames only when the parties actually talk and sends nothing when the parties do not talk. This means that, around 50% of time (therefore bandwidth) is saved and reallocated for other traffic e.g. data.
To avoid that the parties think that the line is dead during silence, silence is regenerated on the other side: this is called comfort noise generation.
VAD can be deactivated.

  Note: silence suppression allows to limit the average bandwidth required on the Ethernet for a voice channel. But of course if a call is not set up there will be no bandwidth required in the Ethernet. This means that the average bandwidth is still lower (typically 0,7 E at the busy hour)

  Note: for designing the IP network, the full bandwidth figures (without VAD) should be used instead of figures ~~(~~after VAD~~)~~. This is specially the case when the number of VoIP channels is low on a link (there is a certain probability than all voice channels are active at the same time, meaning congestion).

- Loss less voice transit is provided. Loss less voice transit is to have only one compression decompression for the call even if it transits through several A4400s. Normally this does not happen in an IP world because, at the difference of connection oriented data (Frame Relay, ATM), IP is any to any and there should not be transit in A4400 PBXs, except in very large networks.

4.14.1.2.2 Fax support

- Fax G3 is supported at up to 9600b/s. Modulation allows to transport this signal in a 64kb/s channel . To be transported on the IP network, fax is demodulated at 9.6kb/s at the originating LIOE and re-modulated at the receiving LIOE.
  The implementation is based on "fax relay" standard (T38), without spoofing. Faxes are transported in UDP/IP packets.
  The implementation is real time fax, but tolerates less delay than fax spoofing: maximum delay is 1 second in the IP network.

- Automatic fax detection: voice compression resources are allocated dynamically for voice or fax. This avoids specialising resources for either fax or voice, therefore avoiding possible congestion. Fax is identified by detecting fax signals.

- Note: In general, modem calls do not go through compression equipment. This is also the case with VoIP in A4400.

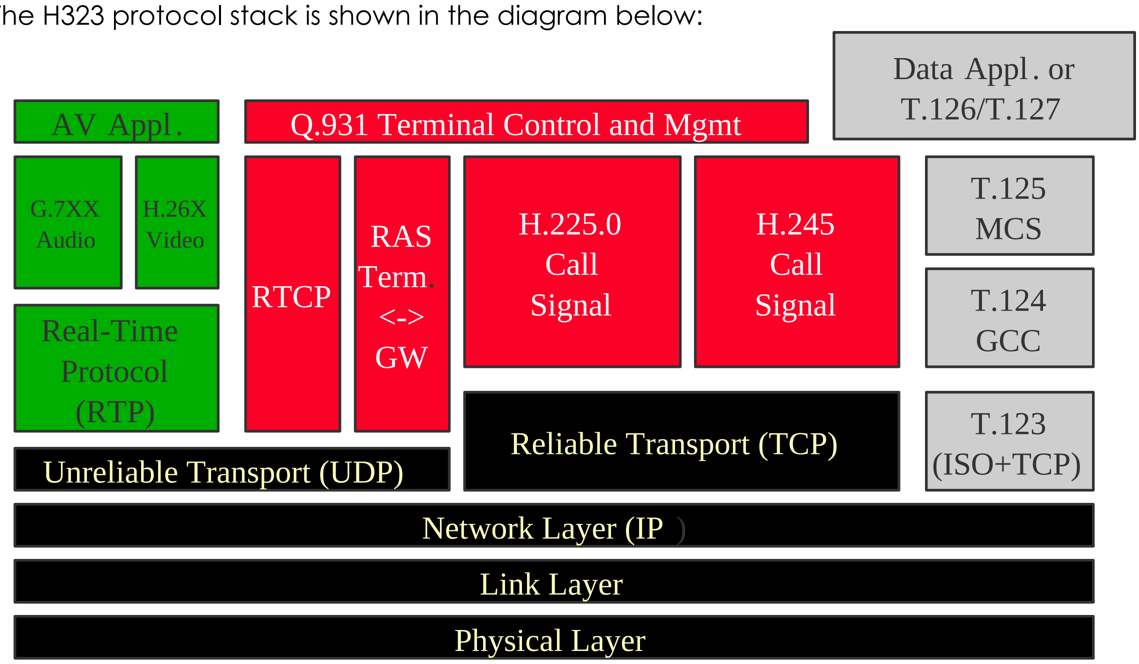
4.14.1.3 H323 support

4.14.1.3.1 A few general information about H323

VoIP equipment communicate with a protocol called H323.
H323 is not only a standard for telephony, but also for conferencing (video, data).

H323 can be compared to QSIG or EDSS1 in the "circuit switching" world

Most VoIP implementations use H323: SIP is another standard for VoIP but is far from being mature.
Same for MGCP, which aims at separating the media (voice support) from the signalling part of the VoIP gateway.
The H323 protocol stack is shown in the diagram below:

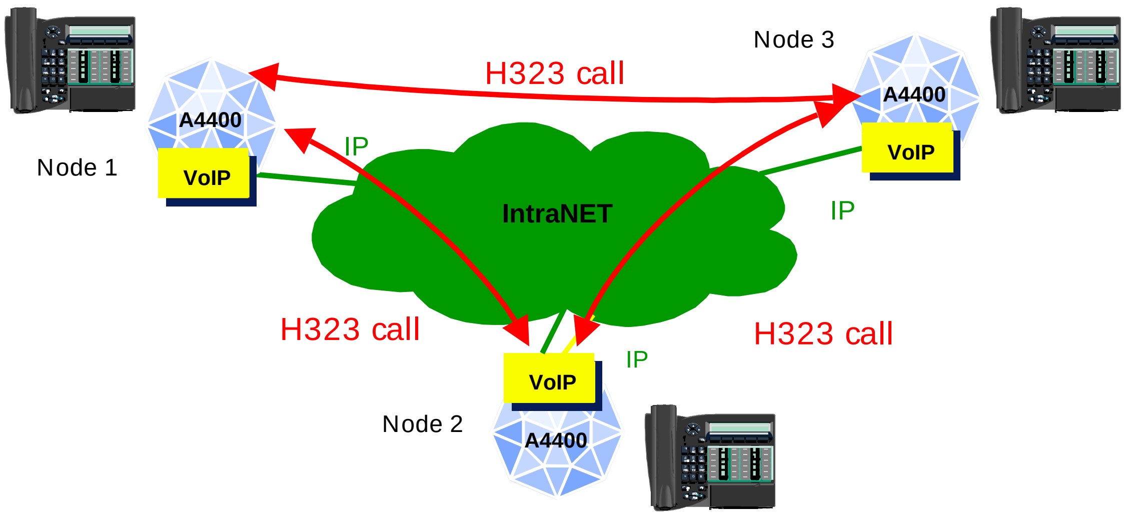


- Voice is compressed by the G7XX algorithms. The result is time stamped and encapsulated in the RTP (real time protocol) layer, encapsulated again in the UDP (unreliable data protocol) layer, put in IP packets, sent on the Ethernet network.
- The signalling (setting up, releasing the VoIP call) are realised by H225 and H245 standards (Q931 (ISDN) like). TCP sessions are set up between VoIP equipment to carry this signalling.
- RTCP (Real Time Control Protocol) allows to get feedback info about performances on the IP network (packet loss,…)
- RAS (registration, admission, status) is used for communication between a VoIP equipment and a H323 gatekeeper.

More general info can be found in the plethoric literature on this subject

4.14.1.3.2 Position of A4400 on H323

A4400 is compliant to H323 regarding telephony.
A4400 VoIP is a "VoIP gateway" in the H323 terminology
For instance (see diagram), when a set in node1 calls a set in node 2, an H323 call is done between the VoIP boards (the IP trunk group) in A4400.

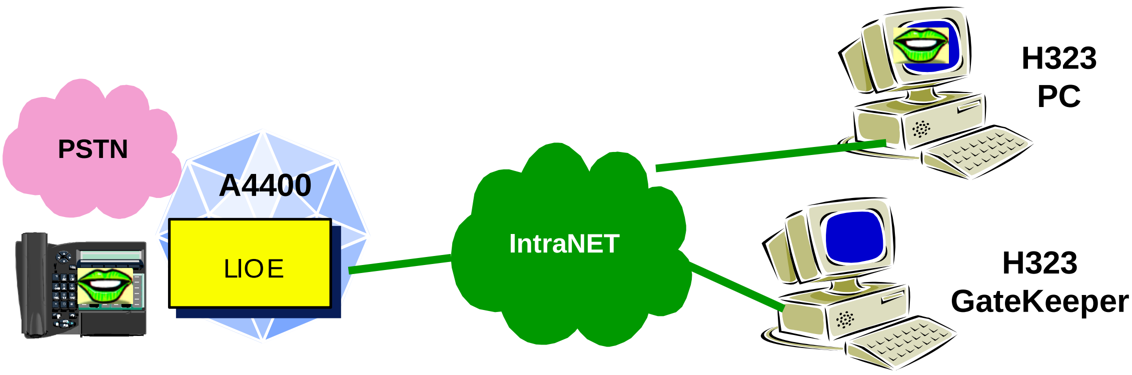


- Interoperability with other VoIP equipment (VoIP gateways)

As H323 is a standard, it should bring interoperability between gateways of different manufacturers. But only testing can give the answer.
Voice should work (provided that voice compression algorithm are the same), but fax over IP will certainly not, as many implementations of fax over IP exists. MFQ23 interpretation is not standard either: overdialling should not work.
Using gateways from different vendors is therefore not recommended

- Communication with a VoIP enabled PC

As A4400 VoIP is a VoIP gateway, VoIP enables terminals like PCs are able to set/receive VoIP calls to/from A4400 terminals through A4400 VoIP.
VoIP enabled PC means that the PC has audio capabilities and an H323 client, like Microsoft Netmeeting or Netscape Communicator (conference)



For calls from A4400 to VoIP enabled PC, A4400 has a table that translates the telephone number (E164) of the PC into the IP address of the PC.
If a gatekeeper is present in the network, A4400 VoIP can also be gatekeeper client. This means that the table E164 to IP is not used: for each call, A4400 requests the IP address of the PC to the gatekeeper. The gatekeeper maintains the table “E164 to IP”.

For calls from PC to set or trunk on A4400, PC sends the IP address of LIOE, followed by the telephone number to be reached.

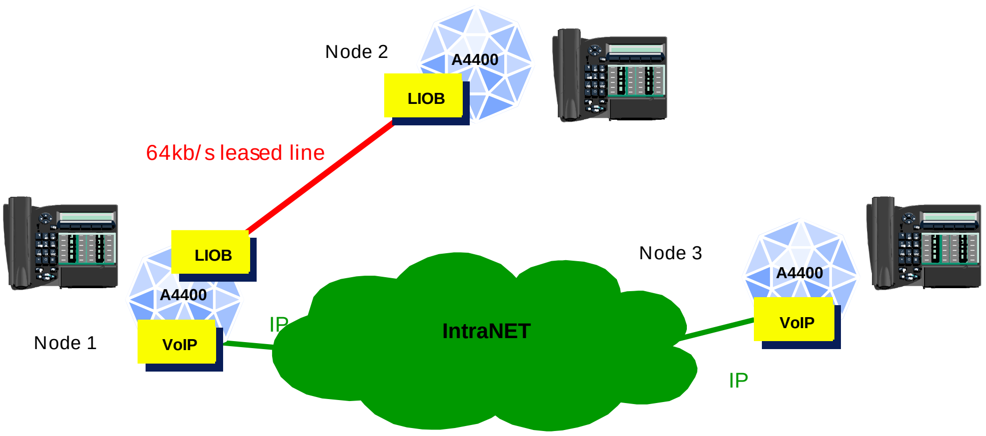
4.14.1.3.3 H323 is only basic call!

H323 addresses basic call only. This means that there is no supplementary services supported like name, call back etc…neither between 2 VoIP gateways, neither between gateways and terminals.
Supplementary services (QSIG like) begin to be defined in standardisation bodies (H450), but are only relevant for gateways not for terminals.

This means that with H323 only, it is not today possible to have real private telephony, with ABC services between nodes and Reflex (call handling) services on PC terminals.

This is why, what is presented about H323 implementation in A4400 are only technical building blocks to provide real applications to customers
. ABC networking with VoIP
. Tele-worker with VoIP
. Web Enabled Call Centre
Focus must be done on applications rather than on technology.

4.14.1.4 Interoperability with “legacy” compression



The benefit of VoIP in A4400 is that it is interoperable with “legacy” voice compression (on ISDN, leased line, Frame Relay).

For instance, see diagram above, a call from node 2 to node 3, will be compressed only once and decompressed once in the network: there is no decompression/compression in node 1.

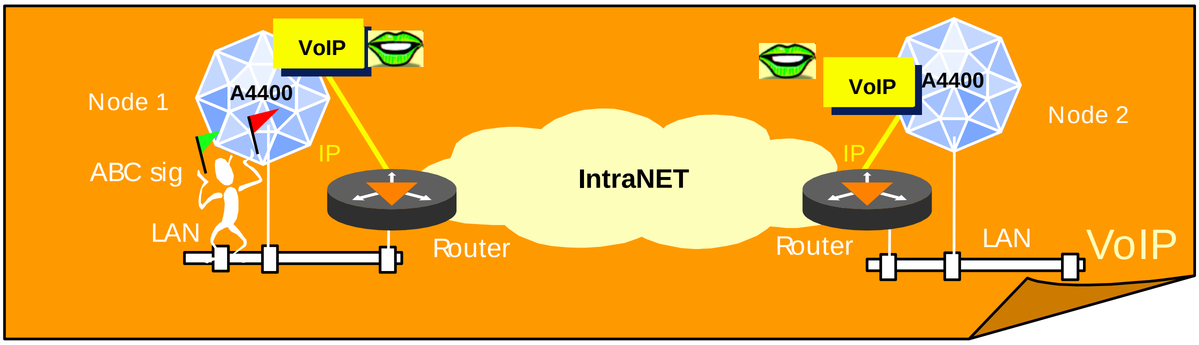
LIOB, LIOP, LIOX should be from the new generation : LIOB2, LIOP2, LIOX2

4.14.2 ABCVPN to provide ABC features tranparency

From an A4400 point of view, the IP network is seen as a switched voice network.
This means that the A4400 analyses the number dialled by the user and sends the H323 call on the IP network, through its IP trunk group, to the IP trunk group of the node of called party (see diagram in H323 chapter)

Furthermore, as discussed in the H323 chapter, H323 does not support feature transparency as QSIG does.

Therefore ABC signalling should take a different path than voice to have ABC services through the IP network: ABCVPN is used for that.



- ABC signalling goes through the CPU
- Voice and H323 signalling goes through the LIOE access.

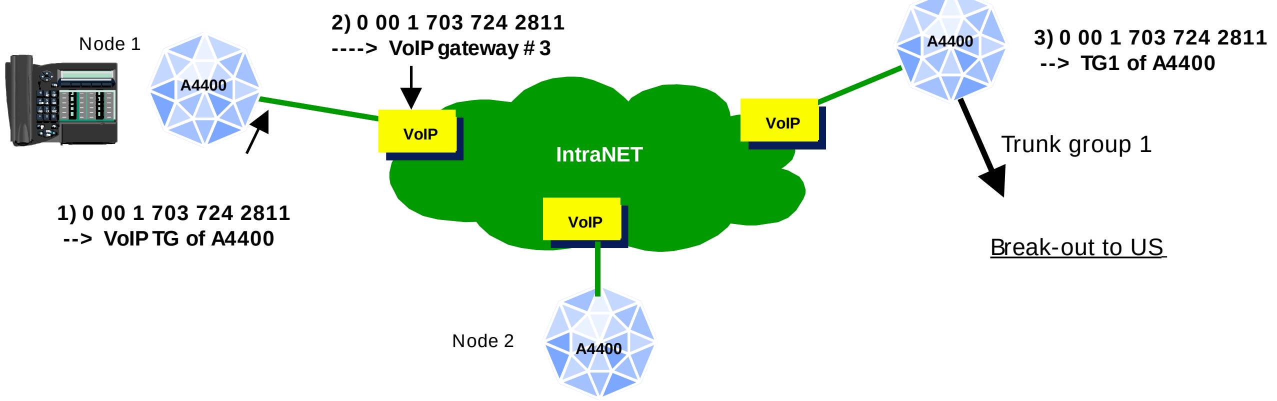
This type of architecture as following advantages:

First are the general advantages of Virtual Private Networks compared to point to point solutions (like leased lines, Frame Relay or ATM PVCs).

- The A4400 **"IP trunk group"** is dynamically **shared between all the directions** (a direction is a distant node): which means supporting a larger traffic for the same number of compressed voice channels, than if a fixed number of voice channels is allocated to each direction
- No transit in an intermediate PABX, which avoids adding delay and using additional voice compression resources: the network is fully meshed

♦ There are also benefits on the **management of the numbering plan**.
The benefit of ABCVPN is that managing the numbering plan is easier: for instance users can move from one A4400 to another while keeping the same directory number.
Furthermore, this will not impact the numbering plan in the IP network.
Range of new numbers also can be created on one A4400 without impact on the numbering plan.
This is the benefit of integration of VoIP in the A4400. When VoIP is realised by external equipment, like routers, voice numbering plan must be managed both in the A4400 and in the router : change in numbering plan in PABX will imply change in router voice numbering plan: 2 updates are necessary instead of 1 if VoIP is integrated in A4400. Implementing break out (hop off) is even more complex with external VoIP (3 updates instead of 1: see diagram below).

Managing the numbering plan when VoIP in external equipment



- Another advantage of ABCVPN is that managing the IP address of the LIOE is very simple: only the IP address of the local LIOE has to be programmed in a node.
  The IP address of the distant LIOE is sent by the receiving node at the set-up of the call.

  This way of working is also useful when an IP trunk group can be composed of several LIOE, for instance to have 60 voice channels.
  Each LIOE has an IP address. For 60 channels, 2 LIOE are necessary: IP trunk group has 2 IP addresses.
  At the set up of the call, ABCVPN will return the IP address of the LIOE of called party that has free channels avoiding failure of the call or to manage overflow from one LIOE to other.

4.14.3 Quality of service

Voice quality is a major issue in IP networks. This is because the IP network is not deterministic: at the difference of circuit switching, delay is not predictable. Voice quality is sensitive to delay, delay variability (jitter) and packet loss.
Quality in the IP network depends mostly of the IP network of the customer and not on the VoIP equipment itself.
This chapter tells what is done in the A4400 to “help” the IP network providing “voice grade” quality and what is done when, from time to time, the performances of the IP network are not good enough.

- A well designed IP network is necessary to support voice.
  Performances required from the IP network by VoIP in A4400 are described below (bandwidth, delay, packet loss).

- Voice will require priority over other data traffic. The A4400 is able to tag VoIP packets according to 802.1p and Diffserv. This will enable LAN switches and routers to identify voice packets and give them priority over other packets.

- On the A4400, it is possible to limit the traffic to the IP network. This is useful for customers wanting to start smoothly with VoIP , that is with a few voice channels in order not to disturb other applications in their IP network.

- Nevertheless, in some cases, network can be saturated, degrading quality of voice.
  In that case, to ensure delivery of calls with acceptable quality, the A4400 sends the call on another network, for instance the ISDN network

4.14.3.1 Performances required from the IP network to provide a good voice quality

4.14.3.1.1 Bandwidth requirements

The template below gives the bandwidth required at Ethernet level for the 3 types of voice compression algorithm.

Voice compression algorithm	Compression rate	Framing size(a)	Ethernet frame length (b)	Bandwidth on Ethernet (c)	Bandwidth on the WAN: with PPP and header compression (d)
G711	64kb/s	30ms	78+240 octets	84,8kb/s	72,8kb/s
G723.1	6.4kb/s	30ms	78+24 octets	27,2kb/s	15,2kb/s
G729A	8kb/s	30ms	78+30	28,8kb/s	16,8kb/s

(a) Framing size means that a packet containing voice is sent every e.g. 30 ms on Ethernet. The longer this delay, the lower the overhead and therefore used bandwidth on Ethernet. The shorter this delay, the better the quality of voice (delay)

(b) Ethernet frame length: see composition of this frame below (preamble (8) and GAP(12) must be added to obtain 78octets of overhead)

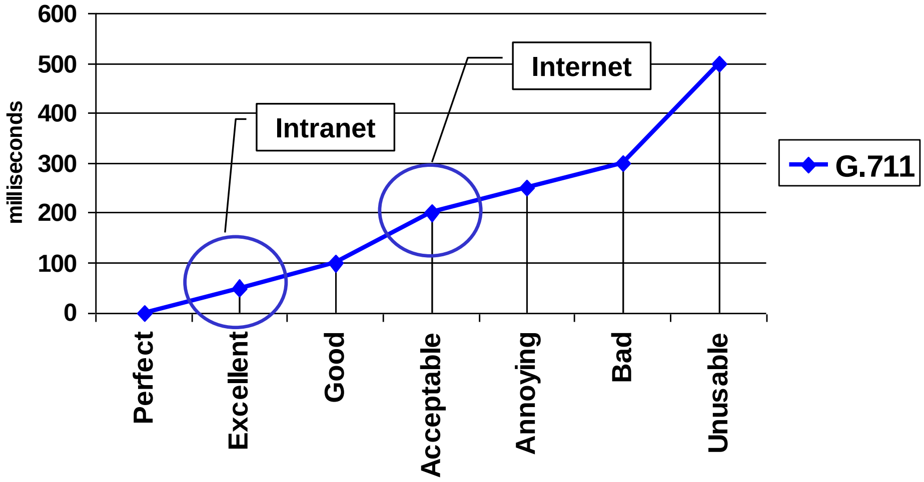
Eth header	IP header	UDP header	RTP header	Payload (1 voice)	Eth CRC
14 bytes	20 bytes	8 bytes	12 bytes	24 bytes (G723.1)	4 bytes

(c) Bandwidth on Ethernet: this is the bandwidth needed on the Ethernet interface of the A4400. This information is necessary for dimensioning of the IP network.
The bandwidth required is without VAD (Voice Activity Detection) and for one way only (half duplex)
This means that putting VoIP generates a large overhead, multiplying the bandwidth needed on the Ethernet. E.g. 27kb/s for G723.1.

(d) This is not a A4400 feature:
On the WAN, bandwidth can be reduced by using PPP and header compression techniques on routers (provided they implement it): e.g. CRTP (Compressed RTP) on PPP
Note: To interconnect routers on the WAN, leased lines are used (or Frame relay). The layer 2 protocol is either PPP (Point to point protocol: which is IP on serial link) or Frame Relay.

4.14.3.1.2 Delay in the network

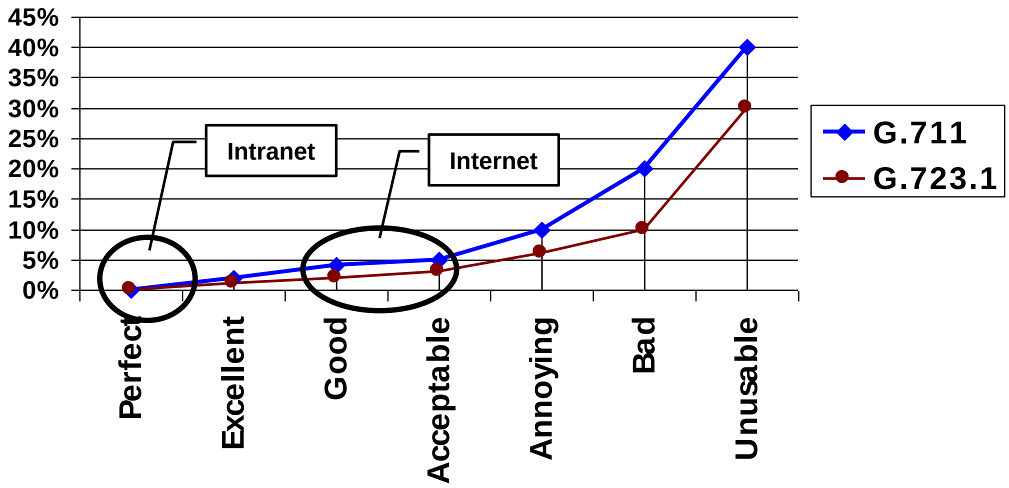
The diagram below shows the perceived degradation of voice quality depending of the delay in the IP network. The delay in the IP network should not exceed 200ms to have an acceptable voice quality in the private network



- The delay for a voice call between 2 A4400 end to end is 100 ms (If delay of the IP network is 0)
- If WAN bandwidth is limited, it is recommended to limit the maximum size of data packets to avoid delay and jitter (like frame fragmentation for VoFR)

4.14.3.1.3 Packet loss

Loss of VoIP packets can also have a bad impact on the quality.
As voice packets run on UDP (for speed) lost, packets are not sent again, as for TCP.
Lost frame Interpolation can recreate lost packets by using received packets, so that the users don't notice it to.



If packet loss is below 5%, voice quality should be acceptable for private networks.

4.14.3.2 Priority in the IP network

To provide the performances required for voice support, as several types of traffic contend for the limited bandwidth, the IP network should implement some kind of prioritisation. This means than voice packets are given priority to less delay sensitive traffic (E-mail, FTP,etc…)

To identify voice packets, routers or LAN switches have to look in header of packets or frames.
Some very sophisticated data equipment can look deeply into packets to identify application (voice, SAP, ..) but in general LAN switches only look at the Ethernet header and routers only on the IP header: There are therefore unable to identify the application.
To address this topic, several standards have been published:

- at level 2 of OSI (Ethernet): 802.1p standard (associated with 802.1Q/VLAN)
- at level 3 of OSI (IP): Type Of Service or DiffServ

Thanks to this standards, IP header or Ethernet header is supporting an additional information giving the priority.
The priority field is completed by the data network manager (associating a priority to a port, IP address, …on the data equipment connected to the application) or tagged by the application itself.
This information is used by data equipment for queue management (Weighted Fair Queuing) and packet discard when congestion (Random Early Discard)
The picture about all these standards is not clear at the moment, there is no real emerging standard. Some routers and some switches implement some (but not all) of these standards.

To provide better quality in the IP network, the A4400 will be able to tag voice packets according to:

-**802.1p/q**

A4400 adds 4 bytes to Ethernet frames. On A4400, it is possible to tag or not according to configuration. User priority (3bits-value 0 to 7) and Virtual LAN identity(12 bits) can be configured.

802.1p/q is only relevant in the same LAN. If a router is crossed, tagging is lost.

-Diffserv (Differentiated Services)

Diffserv marks the existing TOS field in the IP(V4) packet. 64values are available

Note: only voice (RTP/RTCP) and fax packets are tagged, not H323 signalling (H245/H225 traffic)

Nevertheless, with this sort of quality of service mechanisms, quality can not be guaranteed, it is only best effort.

RSVP delivers guaranteed bandwidth, but is not implemented in A4400 in R3.1, because of lack of visibility

So to always guaranty call delivery in acceptable quality conditions, a backup should be foreseen on a circuit network, for instance the public switched telephone network, which is in general available (for public calls). This is what is described below.

4.14.3.3 Automatic route selection IP network/Public switched Network

PABXs have been designed to run on circuit networks, e.g. PSTN and ISDN. Now the A4400 is also able to use a IP network to send voice and fax.

Having a foot in both worlds brings new services, using these 2 networks like in a multi-carrier configuration with automatic route selection adapted for the VoIP case. The benefits are the following:

- ♦ Smooth evolution path from voice over circuits to voice over packet (with an evaluation phase)
- ♦ Ensuring guaranteed delivery of voice calls
- ♦ Removing load from the IP network, when it offers not good enough performances

2 services are presented below: one is preventive and the other is curative

Note: It is also useful to have a public network connection on a A4400, for traffic not supported through IP (modem calls)

4.14.3.3.1 Limitation of number of simultaneous voice communications

For customers that are cautious about VoIP, that want to try before a wide scale implementation, it is possible to limit the number of simultaneous calls on the IP network, in order to limit possible bad impact on existing applications running on the IP network.

♦ Limitation of number of channels per node (per trunk group)

Through configuration, it is possible to define the number of voice channels in the IP trunk group.

For instance, if 2 is configured on a branch office, 1 or 2 calls can be routed on IP.

A third call will overflow on the public network (thanks to ABCVPN multi-carrier capability), providing a transparent service to users (guaranteed call delivery)

If quality of VoIP is not good enough, all the traffic can be switched back to public network by the manager, waiting for an upgrade of the IP network.

If quality is good, a wider scale implementation can be planned.

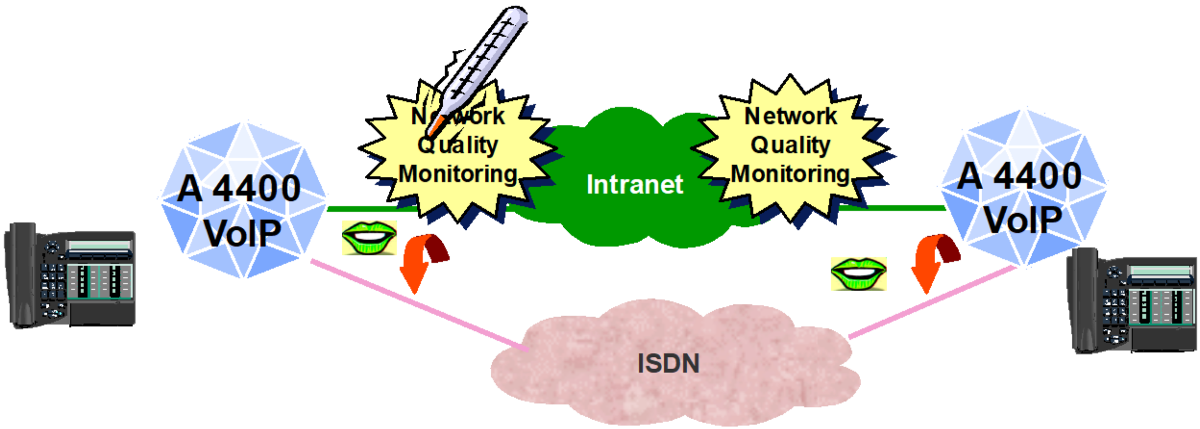
- <u>Limitation of number of simultaneous calls per direction</u>

Limitation can be more precise, it can be done by direction rather than for the whole node. For instance, if there is a low bandwidth WAN link between 2 nodes, the max number of calls can be defined on that direction, even if other VoIP channels are available on the 2 nodes for other directions.

Technically, the VoIP channels of the LIOE can be separated in several trunk groups. Then it is possible to dedicate a VoIP trunk group to a direction.

4.14.3.3.2 Backup on ISDN in case of bad quality on IP network

Even if the IP network is well designed, sometimes performances required from the IP network will not be matched.
In this case, it should be nice if calls could be routed on the public network, to give users a better voice quality
This is possible on A4400, thanks to "IP quality monitoring" service in the A4400, that allow to route calls on the PSTN when quality goes below a given threshold.



How does it work?

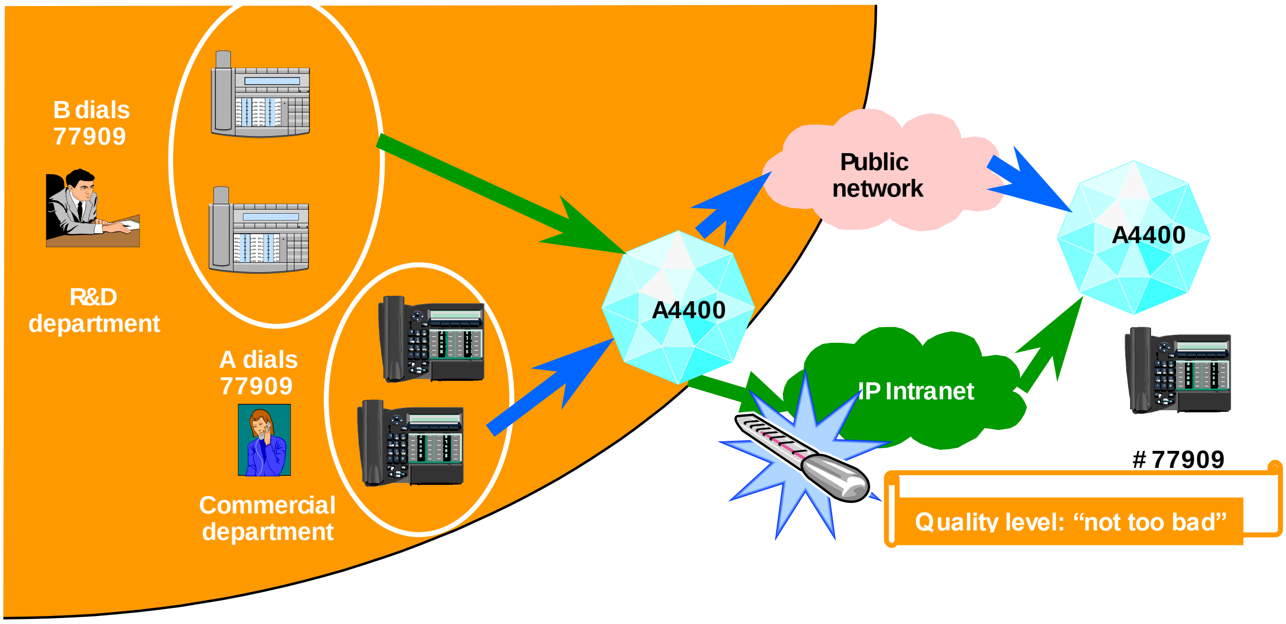
- On IP, it is possible to monitor performances parameters such as packet loss rate, jitter and delay thanks to an H323 protocol called RTCP
- When the quality is below a given threshold (composed of a mix of delay, jitter and packet loss values), new calls on IP would even more downgrade the voice quality. To avoid this, new calls are automatically re-routed on the public network, through the PSTN or ISDN interfaces of the A4400. When quality increases again, new calls are again routed on the IP network.
- It is as if the size of the IP trunk group was dynamically adjusted to send voice traffic when the IP network is able to support it, in order not to threaten mission critical data applications
- This RTCP monitoring is done direction by direction. Because, quality can be very good to one direction and bad to another depending where the bottleneck is in the IP network.
- There is no fall back to ISDN for already established calls, only for new calls.

Note: monitoring takes place during conversation phase. If the first call has quality problems, this service will not work perfectly. But in this case there is also a design problem of the IP network, which is not able to support even 1 single call.

The basic service above is in fact a little more sophisticated than described because overflow on PSTN generates additional costs and therefore should be tightly controlled, that is offered only to people that require it.

This is why user class of service for VoIP have been defined corresponding to the right to overflow on PSTN when quality is not good enough.

Several quality levels of the IP network have also been defined to have several class of calling users: users that will always use VoIP, users that will always use PSTN and several classes in between.



For instance in the diagram above, employees from commercial department have higher VoIP privilege than employees from R&D department. When quality level is measured and qualified "not too bad", R&D employees are forced on VoIP but commercial employees go on PSTN. If quality is "very good", everybody uses VoIP

Note: This is an example of the services that can be provided because VoIP is integrated on A4400.

4.14.4 Management

4.14.4.1 Configuration

Because VoIP is integrated, management is easier than with external VoIP equipment
See "numbering plan" above

4.14.4.2 Charging

Charging tickets are available for calls from node to node. These tickets are of type "ABCVPN" and give the calling party identity.
Also gives the VoIP trunk group origin of the call. This allow to identify that the call is a VoIP call.

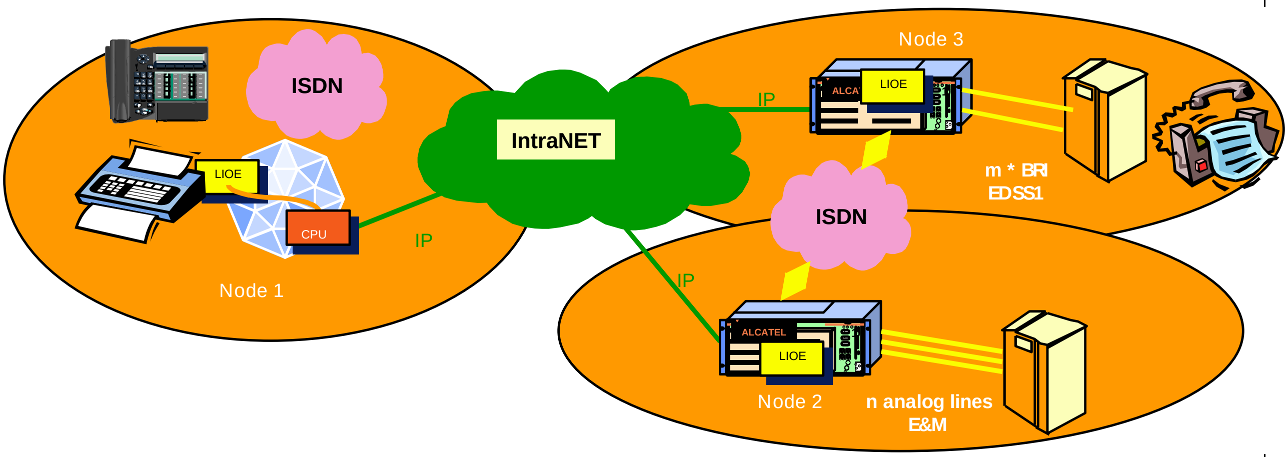
4.14.4.3 Traffic observation

The VoIP trunk group is a trunk group, so counters are available.

Storage of quality parameters is also done and are accessible using a maintenance tool (not A47XX)

4.14.5 A4400 VoIP in heterogeneous PABXs networks

VoIP in A4400 application is not only in homogeneous but also heterogeneous networks.
For instance if a network has 3 nodes, 1 A4400 and 2 3rd party PABXs, and the customer wants to implement networking with VoIP, it is not necessary to replace these 3 rd party PABXs.
A small A4400, just equipped with a LIOE board and a trunk board can be placed between the IP network and the 3rd party PABX, as shown in the diagram below



In this configuration, the A4400 is a VoIP gateway. The strong points of A4400 in this configuration are:

- The large number of interfaces (analog, digital) and signalling protocols (DC5A, EDSS1…), to be able to connect any PABX. Interoperability between protocols is also provided.
- Features transparency if connection is done with QSIG-GF or DPNSS
- Backup on ISDN

4.14.6 A4400 VoIP in applications for users

VoIP is not only a "networking" technology , which means that VoIP goes also up to the terminal.
The first applications of this with A4400 are tele-worker and Web Enabled Call Center.
In these 2 applications, LIOE ensure voice communications between A4400 classic telephone terminals and VoIP enabled PCs

4.14.6.1 Tele-worker with VoIP

See the "tele-worker" chapter in this document

4.14.6.2 Web Enabled Call Center

See CCCCC product documentation

4.14.7 What to order

Underlined items are mandatory for VoIP

- <u>LIOE board</u> 3BA23167AA
  12 voice channels (without software licence)
- COMP18 3BA23168AA
  Optional daughter board of LIOE. 18 additional voice channels (without software licence)
- <u>CBC1</u> 3BA23182AA
  Connecting board of LIOE: DIN/RJ45. Not necessary when patch panel on VH.
- <u>"Number of VoIP channels per node" software licence</u> 3BA09275AA
  This the right to use voice channels of LIOE. Includes G723.1 voice compression algorithm.
- "Number of G729A channels per node" software licence 3BA09296AA
  When G729A is requested, this number is equal to the number of VoIP channels licences. G723.1 and G729A use is exclusive.
  If VoIP and compressed ACT are also used, this licence includes also VoIP and compressed ACT voice compression channels.
  Note: If G729A is requested, the licence "number of VoIP channels… " (3BA09296AA) must also be ordered

4.15 REMOTE ACT ON 64KB/S LEASED LINE (WITH VOICE COMPRESSION)

Remote ACT can sometimes be used in place of a node, to link a small site (branch office) to a bigger one.

Advantage of remote ACT on network node is a cheaper price for the customer, both for equipment and installation, which is very sensible for sites with 20 to 50 lines.

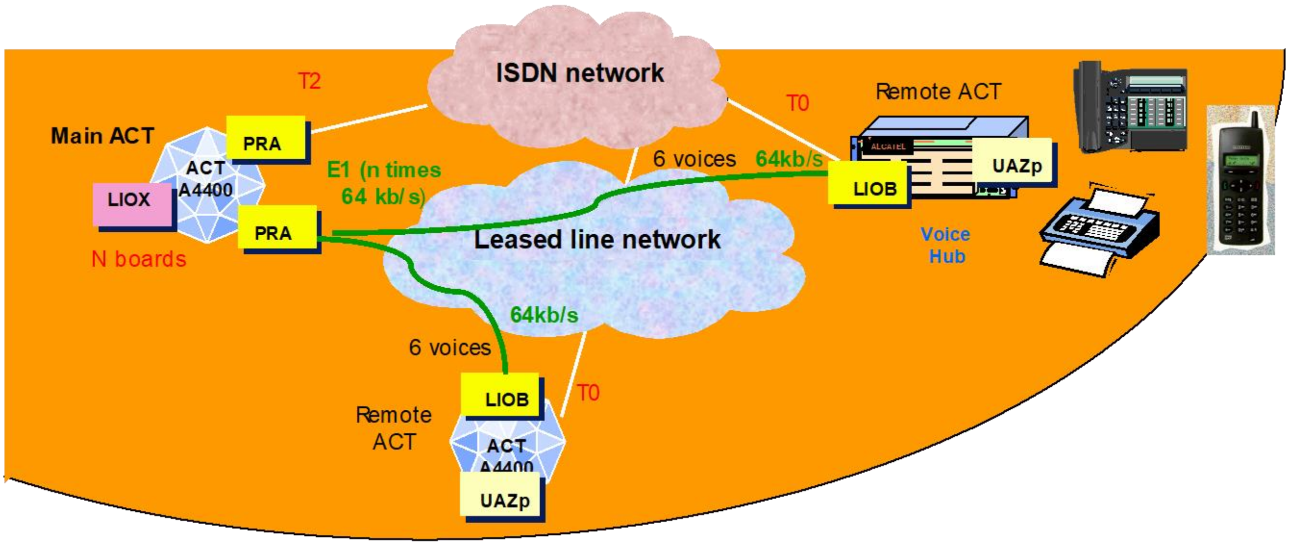
The major drawback of remote ACT solutions in previous releases is the connectivity between remote ACT and ACT in central sites:

. with INTOF, fiber optic is required, which means being in a campus network (owning the fiber) or in a city (renting dark fibre)

. with RT2, a 2Mb/s link is necessary, which is expensive for a few subscribers (even if only a part of the bandwidth is rented-fractionnal E1).

Note: Nevertheless, as they are more and more operators lying dark fiber throughout cities, customers can seize opportunities to get cheap bandwidth on an E1 leased line. Remote ACT with RT2 is therefore more than ever a solution to consider for this type of networks.

This means that these previous "remote ACT" solutions are seldom used in the WAN where there are bandwidth constraints, due to the cost of bandwidth to link the remote site.



With R3.1, this restriction is overcome, because it is possible to connect **a remote ACT through a 64kb/s leased line**.
64kb/s leased line is in general "cheap" both for installation and monthly rental.
This solution is called **remote ACT with voice compression**.
This 64kb/s link supports 8 (compressed) voice channels and the inter-ACT signalling.

Another drawback of remote ACT with RT2 is that the remote ACT is lost if the link with the "central" ACT is out of service. This is not the case with "remote ACT with compression": **backup on ISDN** is provided.
When the 64kb/s leased line is out of service, a B-channel is established between "central" and remote ACT through the ISDN network. The remote ACT is again connected to the "central" ACT. It can run as before leased line was dead.

These 2 main features allows to use remote ACT even for long distance networking (long distance between main and remote).

The drawback of this solution is that it needs the central ACT to run (not fully autonomous as a node solution) and the limited capacity of the ACT (see further)

4.15.1 Technical description

4.15.1.1 Connectivity through 64kb/s leased line

- Leased line is 64kb/s. Connectivity is done through a serial link (X24/V11 or V36) or through T0 (S0FV in Germany)
- Maximum capacity is one 64kb/s leased line per remote ACT, that is 8 voice channels. If more than 8 voice channels are necessary, another solution must be used: RT2 (without compression) or node.
- Some operators offer multi-direction leased line: instead of having many 64kb/s interfaces on the central site (1 interface per remote ACT), a G703/G704 (E1, 2Mb/s) interface is provided. Each 64kb/s channel in the E1 link goes to a different direction (see diagram)
- LIO board dedicated for "remote ACT with compression" application **and** for one direction
  For each compressed remote ACT, one LIO. is necessary in the remote ACT and one LIO. in the central site.
- In the remote site a LIOB is mandatory (when using serial link interface). In the central site, a LIOB can be used, or LIOP (for the first direction) and LIOX (from second direction), or PRA plus LIOX (for all the directions).

4.15.1.2 Backup of leased line

- If the leased line goes down, backup of the link is possible.
  This allows to provide a secure solution, and keep the remote ACT running even if there is a problem on the WAN (link failure) or a CPU problem (when CPU duplicated in main)

- When leased line goes down, the remote ACT has no connection to the CPU anymore.
  An ISDN data call is then set up from the main ACT to a BRA interface of the LIOB of the remote ACT.
  The connection is done, and there is again 64kb/s bandwidth between main ACT and remote ACT. Signalling and up to 8 voice channels go through that path.
  Level of service is the same as before link failure.
  When leased line comes back, traffic is switched back transparently from the ISDN channel to the leased line and ISDN B-channel is released.

- This requires that the LIOB board in the remote ACT has a T0 interface connected to the ISDN network and DID and data (data bearer capability).services on the ISDN network

4.15.1.3 An A4400 "2 boards" solution for a small site

- With remote ACT with compression, only 2 boards are necessary to provide a full set of A4400 services
  - LIOB board provides the "CPU" function, private and public interfaces
  - UA 16, UA32 or UAZp (VH only) allows the connection of Reflex, Z (with or without Z adapter) and DECT (with IBS) terminals

  Note: these sets also benefit from centralized services like voice mail etc…and users services like DECT roaming. Voice guides should be local for instance with new Z20VG board. A local connection for music on hold is available.

- The capacity of remote ACT with compression is limited to 30/40 sets maximum
  - Because of the limitation of 8 channels to the main site
  - Because of the 4 BRA access on the LIOB (8 public calls). Of, course it is possible to add a PRA in the ACT, but price will be impacted.

  Therefore, the VH package is adapted to this solution (see how to order).

4.15.1.4 Limits/restrictions

- 1 LIOB max per remote ACT. No backup link as with RT2.
- Requires LIO2 range of boards. Does not work with LIO1 range of boards
- Overflow on ISDN when 8 voice channels are busy is not available.

4.15.2 Features/benefit

Following diagram sums up the feature/benefit of the solution

***Feature**s*	***Benefits***
Use of a 64kb/s leased line/voice compression	*Keeps running costs low, while providing up to 8 voice channels* *Price 64kb/s line: 335EUR for 10km for one month in France*
Backup on ISDN	- *Permanence of service in case of failure of leased line* - *enables long distance networking*
No configuration to be performed in remote site	*Easy and cheap to install and maintain*
A small site solution with 2 boards: - *LIOB: CPU mode, private and public interfaces* - *UAZp or UA board*	- *Connection for Reflex, analog (direct or using A4095AP), and DECT (using A4070IO) terminals* - *2 slots free in VH for extension or Voice Guide*
Centralization of resource's in central site (voice mail, attendants, directory, management)	- *to optimize personnel and equipment costs* - *to provide services to users in remote sites without the costs:* *no SWL for centralized voice mail, accounting, DECT (compared to a node solution)*
System evolution in R3 (several DID translators, ARS multi-company, CLIP handling etc…)	*Long distance networking (main and remote ACT in different areas)*
PRA for connection to multi-direction leased line at central site (depending of carrier offer)	*Optimized configuration in central site when several remote sites (avoids to have too many LIOB)*

4.15.3 What to order

- A new package is created:
  - VH with PWS, without patchpanel
  - LIOB

- There is a SoftWare Licence for remote ACT with compression. 1 licence is necessary per remote ACT with compression.
- The "voice compression licence " is not relevant for this application (replaced by the SWL remote ACT with voice compression)
- G723.1 voice compression is included. G729A is an option.

5. MOBILITY

The Release 3 of the A4400 brings a threefold evolution :

1. the evolution of the A4400 mobility application in terms of capacity and features
   * number max. of “Advanced” RBSs multiplied by 4 to reach 1,000
   * number max. of DECT sets multiplied by 2.5 to reach 5,000
   * new features to improve the use of DECT sets in the company or in network
     "Twinset" function
     ABC-F enhancements

2. the introduction of an “Optimised” DECT radio base station (low-traffic - low-cost) to be more competitive on the small and medium size configurations segment
   * DECT Radio Base Stations with UA line interface

3. the availability of new DECT terminal equipment and applications allowing to make a major step towards full wirefree configurations
   * new range of GAP sets
   * wireless desktop Reflexe sets
   * A4980 "PC Phone"
   * Ubiquity

5.1 EXTENSION OF CAPABILITIES

The topology of a site and radio engineering considerations may oblige the use of more than 256 RBSs (size ; number of floors in buildings ; presence of natural/artificial obstacles ; etc.).

On some sites, e.g. exhibition halls, DECT users have on average far a higher traffic than the usual 0.2 Erlang assumed in standard configurations.

With the R3, the provisioning level of the A4400 becomes as follows ;
Max. number of DECT sets per A4400 node has been increased to **5000**
Max. number of DECT4 HB boards per A4400 node is **250** (24 max. per ACT)
Max. number of “Advanced” Radio Base Stations (RBS 2G) per A4400 node is **1000**

Alcatel 4400 capacity	Release 1.4 Per node	Release 1.4 Per ACT	Release 2 Per node	Release 2 Per ACT	**Release 3 Per node**	**Release 3 Per ACT**
Max. no. of DECT Terminals (1)	800		2000		**5000**	
Max. no. of Wireless Reflex sets	n/a		n/a		**1000**	
Max. no. of DECT Boards (2)	64	24	62	24	**250**	**24**
Max. no. of “Advanced” RBSs	256	96	256	96	**1000**	**96**

1 : native + hosted terminals
2 : DECT4HB board for Release 3

5.1.1 <u>256+ RBSs configuration (« Multi-PARI »)</u>

The PARI is the DECT identity of a DECT system. DECT standard limits the number of RBS in a PARI to 256. To exceed this limit of 256 RBSs, the Release 3 implements in the A4400 the « Multi-PARI » capability enabling thus a 4400 node to manage up to 8 PARIs and up to 1,000 radio base stations ("Advanced" RBS)

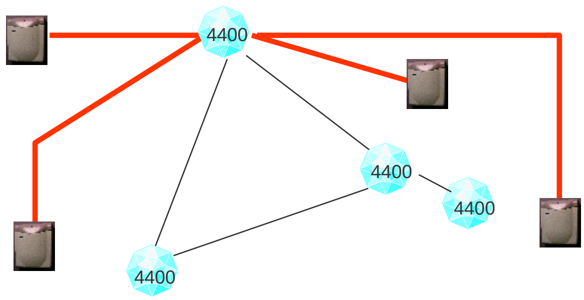
* one ACT can handle 1 PARI
* 1 PARI may spread over several ACTs
* ACTs must be connected together via INTOFs

To allow seamless handovers between any of these 1,000 RBSs, the so-called « **External Handover** » mechanism is implemented in the A4400 R3. A call initiated within one PARI of the A4400 node is not disturbed when the users moves into zones of the DECT location area belonging to different PARIs of the A4400 node.

* Attention : the handsets must support this feature too ; the A4074 GAP Step II (V3.95)handsets do
* as far as other handsets not implementing the external handover, handover is not seamless handover ; communication is cut and retrieved after a call recovery

5.1.2 Campus covered by a network

To get a homogeneous DECT coverage area allowing seamless handover everywhere in the area, one A4400 node must be dedicated to the provision of DECT to the campus subscribers. The Release 3 allows up to 1,000 RBSs and up to 5,000 DECT subscribers.



The reason for this is that the lines connecting the different nodes of a network ("Campus") can't provide synchronisation mechanisms precise enough (<4µs) to allow DECT synchronisation.

* It would not be possible to make an External Handover between two A4400 nodes
* The handset wouldn't be able to detect the presence of another DECT system as long as it would remain synchronised with the system it first locked to.
* Only after having lost the synchronisation with the system it was locked to ('out of coverage') would the set try again to locate itself on another system.

5.1.3 Enhancement of DECT features in ABC homogeneous network

For users with a DECT set as unique telephone set, the Release 3 introduces enhancements at the set level when roaming in network

* Voice Mail and Supervision : same level of service on a visited node as on the home node voice mail notification & consultation
* Call-back requests
  call-back on free requests are kept when roaming
  call-back on busy requests are turned into call-back on free when the set locates onto another node
* Secretary / Manager filtering
  this service requires "Twinset" (see "Twinset" section).
* Rappel : minimessaging is not available in network

5.1.4 "Twinset" function

5.1.4.1 Positioning

Many DECT users have 2 telephone sets ; a DECT handset to enjoy mobility and a Reflexes desktop set for the comfort and the access to the large number of telephony services it brings. « Twinset » services will make them see these two sets as only one (one dialling number ; one voice mail ; etc.).

5.1.4.2 Principles

The Twinset service is a close association between two multi-line sets - a main wired set and a secondary set (ex. : DECT handset) - with one calling number. By extension, we also call « Twinset » the set consisting of the mains set and its secondary set

The "Twinset" calling number is the main set dialling number

Incoming call on "Twinset" dialling will ring both sets

The sets must be multi-line and declared on the same node

It is still possible to call the secondary set with its dialling number e.g. if the "Twinset" function is out of service (i.e. if the main set is out of service) or to bypass the activation of the do-not-disturb or the forwarding of the "Twinset".

5.1.4.3 Services

* Easy transfer between wired set and DECT
  no more Forwarding of wired set to DECT handset
  no more Supervision of wired set by DECT handset and vice-versa
  the set with the communication will park the communication by
  - pressing the specific key with park prefix on wired set
  - pressing "OK" key on DECT handset
  then, each set can take the communication by Off-hook

* "Two sets in one"
  One mailbox (voice, mini-messaging, call-back, unanswered calls)
  - notification/consultation on both sets
  - Restriction : "Mini-messaging" and "Unanswered Calls" do not work private network wide

  One charging
  - on wired set for calls on DECT handset

  One Forwarding activation/deactivation for both sets
  - but no forwarding if secondary set is called directly with its dialling number
  - note also that forwarding of secondary set is not possible

  One « Secret Call » (CLIR) activation/cancellation for both sets
  One « Do not disturb » activation/cancellation for both sets
  One Camp-on control activation/cancellation for both sets
  One Secret code (PIN) setting for both sets (not in network)
  One Associate setting for both sets (not in network)
  One Language setting for both sets
  One Call-back Request for both sets
  One Absence message for both sets

* but ...
  lock/unlock setting per set, eg to allow the locking of wired set alone while moving with DECT handset
  malicious call activation per set

* Manager/Secretary
  the full Manager/Secretary service is provided on a multi-line wired set which will be the main set of "Twinset" (e.g. 4035)
  the DECT handset which is the secondary set of "Twinset" will offer the best compromise in terms of services and ergonomics when the Manager/Secretary is moving (see recommended profiles hereafter) :

Manager

Secretary

Restrictions for Manager/Secretary filtering

- no Direct Set Selection/Line Supervision
- no secret listening
- no Manager/Secretary text mini-messaging
- no multiple filtering tables management

5.1.4.4 Twinset services in network

When roaming (DECT), almost every "Twinset" service is available private network wide in visiting nodes
Some restrictions however...

∗ text Mini-messaging notification and consultation
∗ unanswered ISDN calls notification and consultation
∗ alarms consultation
∗ charging consultation : only charges of calls made in visiting node are available
∗ "Twinset" call transfer
∗ appointment Reminder
∗ hunting group
∗ redial : only redial of calls made in visiting node is available **(*)**
∗ call park **(*)**
∗ night set **(*)**

(*) : not possible in network but this restriction is not specific of the "Twinset" function

5.2 SUPPORT OF A NEW TYPE OF RADIO BASE STATION A4070 IO & EO (« OPTIMISED » BASE STATION)

5.2.1 Positioning

Customers look for solutions lowering up-front cost for small and medium size configurations, which means an inexpensive and fast to install DECT solution allowing to re-use the existing A4400 hardware (cabinet ; boards ; CPU ; etc.) as much as possible.

- With the R3, Alcatel introduces a new **DECT radio base station** (« Optimised » Base Station) that connects directly to a standard UA board via **2 UA links** ;
  Alcatel 4070 IO : indoor version
  Alcatel 4070 EO : outdoor version
- This radio base station supports up to 6 simultaneous communications
  1 UA B-channel = 1 DECT communication link
- It is allowed to mix « optimised » base stations and Reflexes sets on the same UA board
- It is possible to connect up to 256 « optimised » base stations on a 4400 R3

5.2.2 Characteristics

The main characteristics of this « optimised » base station are ;

Dimensions (mm)	215 x 180 x 55
Connection to PABX	2 twisted pair cable
Transmission rate	64 kbit/s per channel
Power feeding	through UA link
Traffic max.	1.7 Erlang
Mounting	wall mounting
Built-in antennae	Yes
Antenna diversity	Yes
Connections for external antennae	2

The A4070 IO (indoor) withstands the following climatic conditions ;

* temperature range : -20°C / + 60°C
* relative humidity : 15% / 90% without condensation
* altitude : 0 / 3,000 m

The following table compares the "Optimised" and "Advanced" RBSs:

Characteristics	4070 IO ("Optimised")	4070 IA ("Advanced")
Interface board	**UA16, UA32, UAZp**	**DECT2, 4 & 4HB**
Line interface	**UA (2)**	**HDB3**
Allowed per A4400 R3	**256**	**1000**
Simultaneous Comm.	6 max	12 max
Frequency Plan	10 Carriers	10 Carriers
Traffic/base	1.7 Erl	5.0 Erl
Range (no obstacle)	300 m	300 m
Distance to the PBX	up to 1200 m[1]	up to 2000 m
Handover	Yes	Yes
GAP compatibility	Yes	Yes
Duplication	Yes	Yes
Back-up	Yes	Yes
Remote feeding	Yes	Yes
Encryption ready[2]	No	Yes

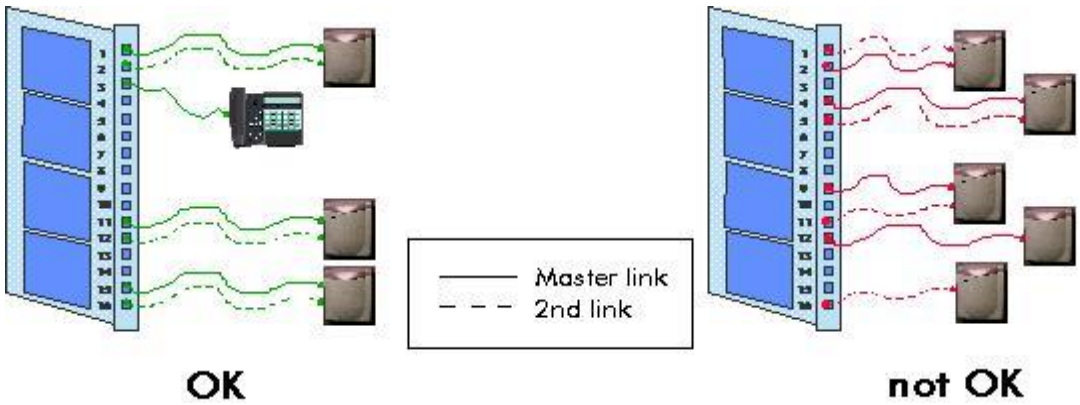
1 : 1200 m with LY278 0.6 mm cable ; 800 m with SYT 0.5 mm cable

2 : when available on A4400

5.2.3 Installation

The "Optimised" RBS connects to the A4400 via 2 UA Links. Therefore, 2 successive UA positions shall be reserved per base station on a UA board ;

* If UA positions are numbered from 1 to 16, 24 or 32 (UA16, ZUA24, UA32), the master link of the radio base must be connected on «odd» positions, the 2nd link if any being connected to the successive even position.
* both links of a radio base must be connected to the same UA-line interface chip
* mixing low-cost base stations and UA sets on the same UA-line interface chip is allowed

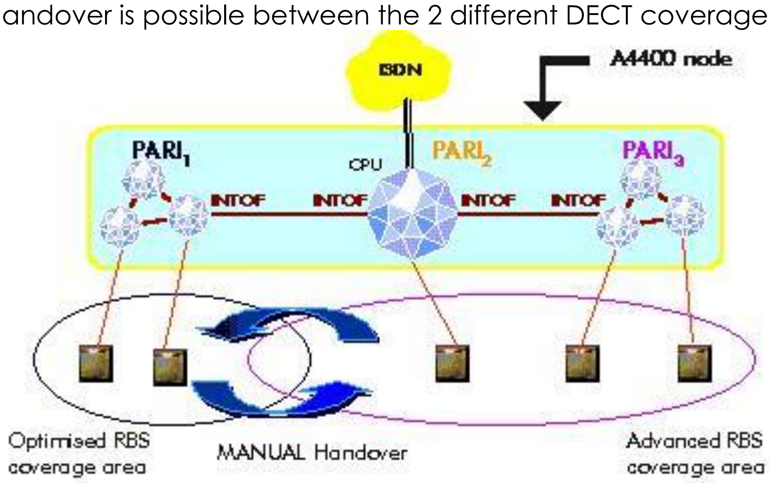


5.2.4 Compatibility

* The "optimised" base station requires the A4400 Release 3 software
* It requires handsets at least GAP compatible (ie UA/GAP or GAP) ;

compatible	Not compatible
Alcatel 4072 **Alcatel 4074 GB** **Alcatel 4074 GH** **Alcatel 4074 GI** **Alcatel 4074 GBEx**	Alcatel 4075 Alcatel 4074 B Alcatel 4074 H Alcatel 4074 Bex

* Mixing "optimised" and "advanced" base stations in the same ACT **is not** allowed.
* Mixing "optimised" base stations and "advanced" base stations in the same 4400 is allowed, but in this case
* "advanced" and "optimised" radio base stations belong to different PARIs
* only manual handover is possible between the 2 different DECT coverage areas:



5.2.5 Configurations with duplicated CPU

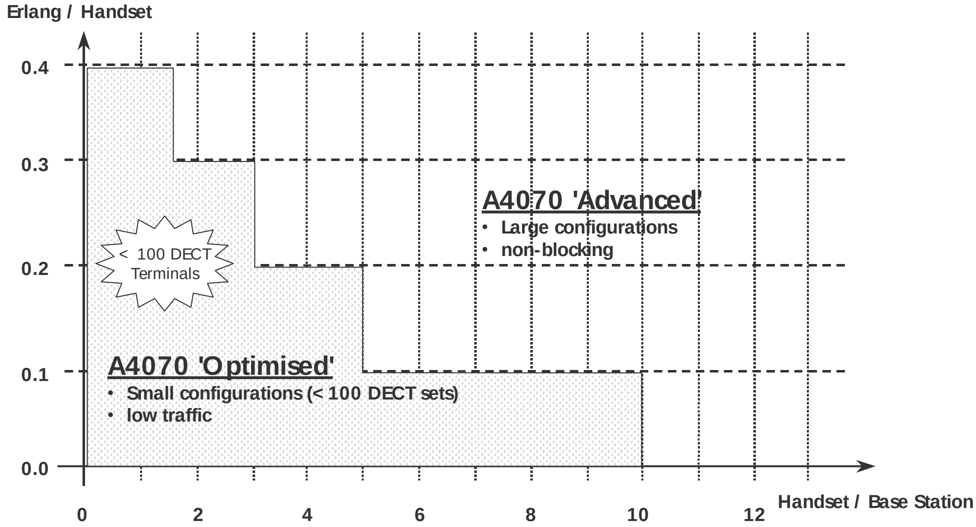
In the case of an installation with CPU duplication, it must also be ensured that DECT communications will not be cut when switching from one CPU to another.
For installations equipped with "Optimised" base stations, a specific package consisting of 2 DECT4HB boards only equipped with each a DTM daughter board will provide the necessary DECT synchronisation mechanisms (this features requires 2 free slots of the back-panel).

5.2.6 Limitations and constraints

The 4400 supports a maximum of 256 "Optimised" radio base stations
* all these base stations belong to a unique DECT domain (PARI)
* each base station gets a unique identifier at installation (PARI + RPN)

5.2.7 Use cases

* Primarily for small and medium size configurations (up to 100 users to limit the impact of traffic density variations e.g. when groups of persons exit a meeting room) where low-traffic on-site mobility shall be offered in addition to standard wired telephony.
  This abacus gives the recommended handset / "Optimised" RBS ratio to ensure a Grade of Service of 0.1% as a function of the handset traffic (Erlang)



* To extend the DECT coverage of large configurations equipped with "Advanced" RBSs to places where low traffic capabilities are requested (e.g. car park, store, basement, etc.)
  A manual handover will be necessary when passing from one coverage area to the other one

5.3 ALCATEL 4400 WIREFREE

5.3.1 Positioning

Companies are more and more concerned by costs generated by frequent removals and re-organisations, i.e. re-cabling, re-programming of the PBX, etc.

Furthermore, users with different profiles and communication needs cohabit in a company, and everybody needs to some extend to benefit from on-site / off-site mobility services : being able to reach someone or to be reached at once increases the overall efficiency and profitability of a company.

The A4400 Wirefree (R3) is a powerful and flexible solution enabling our customers;
* to offer mobility services to all their employees whatever their number and their profiles
* to move people and re-distribute offices without re-programming the PBX or re-cabling the site
* to keep however an access for the users to the full range of A4400 features
* to take their GSM fleets into account efficiently

The A4400 Wirefree solution mainly rests on;

- the Release 3 features
- a powerful DECT infrastructure
- DECT4HB boards
- “Advanced” RBSs
- a complete range of professional end-user equipment and applications
- Wireless Desktop Reflex terminals
- DECT handsets
- PC Telephony with Alcatel 4980
- Ubiquity

5.3.2 Wireless desktop "*Reflexes*" terminals with 4097 CBL (TSC)

5.3.2.1 Positioning

Taking advantage of a full wirefree infrastructure shall not be to the detriment of the level of service. A user is guaranteed

- same range of desktop terminals
- same grade of service
- same level of features (line supervision, Manager / Secretary filtering, etc.)

The A4097 CBL is a DECT device (Plugware) that allows the traditional wired link between the Reflexes set and the PBX by a DECT radio link to be replaced;

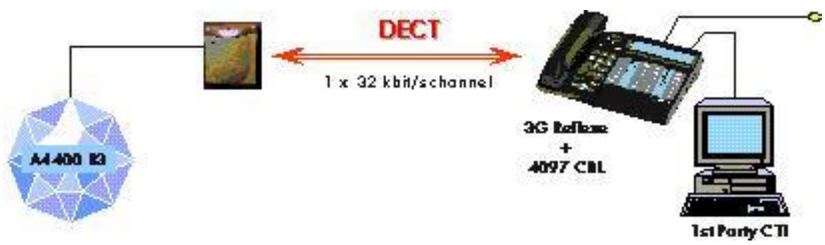
- It is primarily intended for those users who want to enjoy the flexibility of wirefree solutions with the comfort and richness of service of the "Reflexes telephony". Moving people and re-organising offices no longer requires re-programming the PBX or to re-cabling the installation…
- It is also a convenient solution for connecting a remote Reflexes terminal (no cabling is required for the set)

5.3.2.2 Product description

The Plugware is a device that is physically inserted in the casing of the 4020 and the 4035 or placed outside of the casing of the 4004 and the 4010.

Alcatel 4097 CBL

- the "wireless 3G Reflexes set" provides an access to the same services as if it was wired
- in addition, it provides a CTI plug for 1st party CTI
- the A4097 CBL is GAP compliant and gets an IPUI_O at registration



- A4097 CBL behaves like a DECT portable part
- transports the UA signalling over the DECT air interface
- no external handover nor network roaming
- connection between the UA set and the TSC via a short UA cable

- The 3G Reflexes terminal with its A4097 CBL is seen by the A4400 as a DECT portable part
  registers like a standard GAP terminal (PARK+PLI ; IPUI_O ; etc.)
  locates at power up or when roaming in a new location area
  supports seamless intra-cell bearer handover or inter-cell bearer handover within a cluster
  there are 4 new types of subscribers, one per type of 3G wireless Reflexes set
- "Wireless 4004"
- "Wireless 4010"
- "Wireless 4020"
- "Wireless 4035"
  add-on modules can still be connected to the 3G Reflexes terminal : a secretary can use a wireless 3G Reflexes terminal (e.g. for boss / secretary filtering) !

5.3.2.3 <u>Constraints and limits</u>

- A4035 operator shall not be connected behind a A4097 CBL
- authentication and encryption not supported
- No other sub-device can be connected to the A4097 CBL (i.e. no chaining)
- Provisioning level :
  - "Advanced" RBS-based configurations can support up to 1000 wireless Reflexes sets
  - "Optimised" RBS-based configurations can support up to 500 wireless Reflexes sets

5.3.3 DECT Handsets

5.3.3.1 <u>Positioning</u>

4072 "Speech"	**4074 GB "Business"**	**4074 GI "Industry"**	**4074 GBEx "Safety"**	**4074 GC "Comfort"**
BASIC LEVEL OF SERVICE **Mono-line, CLIP…**	**REFLEXE SERVICES** **Multi-line, Twinset, centralised dial by name, external handover, text messaging...**			

Alcatel has a complete range of DECT handsets fitting the different profiles of users within a company.
All these DECT handsets support **roaming in network** and are **compatible with any type of A4400 radio base station (1G, "Advanced" and "Optimised")**;

- ***4072*** : mono-line terminal giving an access to basic services and bringing comfort to the user thanks to its built-in loudspeaker allowing hands-free communication.

- ***4074 GB & GC*** aimed at white collar users
  GB : Business
  GC : Comfort : vibrator + hands-free charger

- ***4074 GI*** aimed at blue collar users, with special features for intensive use in hard conditions
  rugged
  backlight (keyboard & LCD screen)
  plug for headset

vibrator

* ***4074 GB Ex*** : intrinsically safe, intended for a use in hazardous areas (explosive atmospheres)
  certification : EEx ia IIC T4

5.3.3.2 Altiset S-GAP (« Basic »)

WARNING : This product has been taken off the A4400 portfolio and is no longer supported by A4400.

5.3.3.3 Alcatel 4072 Speech

* A4072 Speech brings more comfort to the mobile user when in their office thanks to its built-in loudspeaker allowing hands-free operations (half-duplex)
  10-digit alpha-numeric display with icons
  ITU keyboard
  antenna hidden in the casing of the set
  NiCd batteries
    Talk time / stand-by : 14 h / 195 h**
    Charging : 16 h
  Compact (60 mm x 165 mm x 30 mm) and light (149 g)
* A4072 can be used in configurations with more than 256 base stations, but handovers between different PARIs cannot be seamless
  (communications will be cut and retrieved after a call recovery)

** dependent of conditions of use

Alcatel 4072 Speech

5.3.3.4 Alcatel 4072 Comfort (A4400 Release 3.1)

* A4072 Comfort will eventually replace A4072 Speech. With the release 3.1 of A4400, it will bring additional professional services to the users and even more comfort when on the move or at the desk together with Alcatel 4980, mainly :
  * **Dial-by-name :** more comfort when on the move ; the users can access the corporate directory
  * **Voice Mail notification :** the users are immediately informed of messages left in their voice mail
  * **Automatic off-hook :** when used with A4980 PC Phone, one single click is enough to make a call in handsfree mode

Alcatel 4072 Comfort

5.3.3.5 Alcatel 4074 GB (« Business »)

* The A4074 GB is an evolution of the A4074 B DECT handset (same design)
* The UA and the GAP protocols are integrated in this new handset
  When used with a A4400, this set will work in UA mode
  When functioning in GAP mode, only the CLIP will be displayed on the screen
* The A4074 GB implements the "**external handover**" feature allowing seamless handovers in configurations with more than 256 base stations (A4400 Release 3)
* The A4074 GB is available with ITU or CCITT keypad

Alcatel 4074 GB

5.3.3.6 Alcatel 4074 GI (« Industry »)

* The A4074 GI is an evolution of the A4074 H DECT handset (same design) that comes with one major additional feature ;
  A built-in **vibrator** to allow discreet notification of incoming calls
    (Ringer and vibrator may be programmed independently). The vibrator feature may be activated via ;
    1/ long press on the «5» key and «OK» for confirmation ; ringer and vibrator are exclusive
      if ringer is on, vibrator is off
      if ringer is off, vibrator is on
    2/ long press on the «#» key (programming menu)

Alcatel 4074 GI

* The UA and the GAP protocols are integrated in this new handset
  When used with a A4400, this set will work in UA mode
  When functioning in GAP mode, only the CLIP will be displayed on the screen
* The A4074 GI implements the "**external handover**" feature allowing seamless handovers in configurations with more than 256 base stations (A4400 Release 3)
* The A4074 GI is available with ITU or CCITT keypad

5.3.3.7 <u>Alcatel 4074 GC (« Comfort »)</u>

* The A4074 GC is an evolution of the A4074 B DECT handset (same design) that comes with two major additional features ;
  a built-in **vibrator** to allow discreet notification of incoming calls
    (Ringer and vibrator may be programmed independently). The vibrator feature may be activated via ;
    1/ long press on the «5» key and «OK» for confirmation ; ringer and vibrator are exclusive
      if ringer is on, vibrator is off
      if ringer is off, vibrator is on
    2/ long press on the «#» key (programming menu)

Alcatel 4074 GC

* a **hands-free** feature provided via a specially re-designed basic parking unit offering both charging and hands-free functions (Alcatel 4071 BC)
  the feature is automatically switched on/off by swapping in/out the A4074 GC of the parking unit
  the loudspeaker volume is adjustable by a volume knob
* The UA and the GAP protocols are integrated in this new handset
  When used with a A4400, this set will work in UA mode
  When functioning in GAP mode, only the CLIP will be displayed on the screen
* The A4074 GC implements the "**external handover**" feature allowing seamless handovers in configurations with more than 256 base stations (A4400 Release 3)
* The A4074 GC is available with ITU or CCITT keypad

5.3.3.8 Alcatel 4074 GBEx (« Safety »)

* The A4074 GBEx is an evolution of the A4074 BEx DECT handset (same design).
* It is intended for use in hazardous environments with explosive atmospheres (surface industry; no mining) and has been certified « EEx ia IIC T4 »
* The UA and the GAP protocols are integrated in this new handset when used with a A4400, this set will work in UA mode when functioning in GAP mode, only the CLIP will be displayed on the screen
* The A4074 GBEx implements the "**external handover**" feature allowing seamless handovers in configurations with more than 256 base stations (A4400 Release 3)
* The A4074 GBEx is available with ITU or CCITT keypad

Alcatel 4074 GB Ex

5.3.3.9 Intrinsically safe "Advanced" RBS (A4070 EA Ex)

The intrinsically safe version of the outdoor "Advanced" RBS is intended for use in hazardous areas where the atmosphere is filled up with explosive vapours or gazes.

Ex-data :

Type of protection ....EEx d IIC T6
Certification ....INIEX 82.103.174

Technical characteristics :

Protection IP65
Dynamic pressure resistance ....min. 13.5 bar
Max. power consumption ....25 W
Diameter of connecting cable ....3.5 to 8.7 mm

Alcatel 4070 EA Ex

5.3.3.10 Alcatel 4074 Gx Step II

The main evolution of step II is, besides the support of the External Handover allowing uncut communications all over the DECT coverage area, the improvement of the way the handsets locks to a DECT system (SYS_TO_LOCK) when used in different DECT systems (heterogeneous DECT networks) :

* The handset stores the identity of the systems to which it has access rights in up to 5 **system entries**. This system entries are permanently stored in the handset non-volatile EEPROM memory.
  * The system entry to be used to lock to a system is controlled by the **SYS_TO_LOCK** parameter set in the **USER local menu** of the handset.
  * The selection can be **"automatic"** (any of the 5 systems stored in the EEPROM) or **"manual"** (one specific system entry chosen manually via the USER local menu amongst the 5 entries stored in the EEPROM)

Remarks:

* A system entry can be **inactive** meaning it contains no data defining access rights to a system. In this case the scan associated to this entry is skipped.
* No minimum radio signal (RSSI) level is considered in this algorithm because to lock to the system, the HANDSET has to receive its PARI (NT channel) and broadcast capabilities (QT channel). It is supposed that if it is able to receive this information, the RSSI level is sufficient.

5.3.3.11 On-Air registration

This procedure drastically reduces the installation time of DECT terminals. Indeed, before, all the information stored in the DECT terminal (PARK, PLI, IPUI_O, etc.) was entered manually, which was fastidious and often a source of mistakes.

Now, this information is automatically exchanged between the system and the handset;
1. The installer "defines" the sets profiles with the management application (A47xx)
   Reduced amount of information to enter (dialling number, user name, etc.)
   Use of default values
2. The installer "locates" the sets one by one with the *Dectinston* tool
   The *Dectinston* tool "opens" the A4400 to new DECT sets for 1 minute

* Any handset switched on within during this 1-minute period of time can be registered to the PBX
* the PBX and the handset exchange DECT parameters "on-air" (PARK, PLI, IPUI-N, LAL , Dialling number, IPUI-O ...)
* once the set is registered, the PBX calls it and its dialling number is displayed on the LCD of the set (which allows to detect and reject potential "hackers").

5.3.4 Authentication / Encryption (A4400 Release 3.1)

With wireless communication, voice and signalling information are potentially available in the air for anybody. Because of that, people feel extremely concerned by security (i.e. PBX intrusion and confidentiality). In some domains of activity, this may be a crucial concern (bank, insurance, police, army, etc.)

- Can someone listen to their conversations ?
- Can unauthorised persons connect to 4400 and make/receive calls ?

The release 3.1 of A4400 introduces 2 new levels of security to answer this concern in addition to the mechanism of identification already supported by A4400.

5.3.4.1 3 levels of security

5.3.4.1.1 Identification

This is the most basic (still powerful) level of security : A4400 identifies any DECT apparatus by its IPUI-N (unique world-wide) as it has been provided at registration time. This is being done before authorising any inbound/outbound call.

5.3.4.1.2 Authentication (release 3.1)

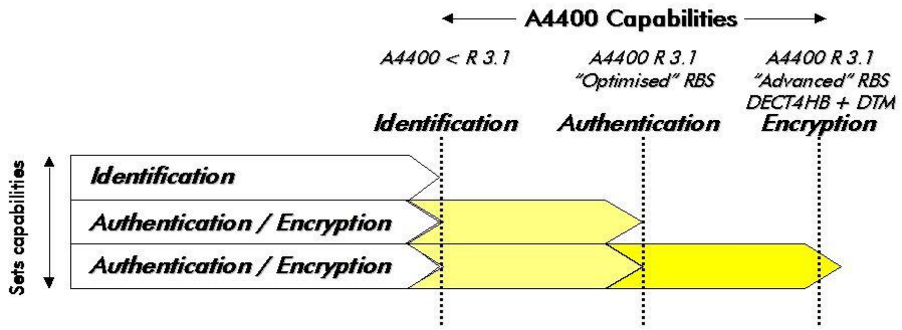
This level of security comes in addition to "identification" : A4400 not only identifies the DECT apparatus via its IPUI-N, but it also authenticates it (and so does also the DECT set) through an unerring computation using a 128-bit Authentication Key.

5.3.4.1.3 Encryption(release 3.1

This is the utmost level of security, when the DECT apparatus has been identified and authenticated as described above, the conversation is then encrypted making eavesdropping impossible. A4400 and the DECT set encrypt (64-bit random key) clear data prior to transmission and recover clear data from received encrypted data.

5.3.4.2 Conditions of use

The highest level of security available at one DECT set will depend (i) on the capability for the whole A4400 ; (ii) on the characteristics of the DECT set itself. Service level is in line with the capabilities of the "weakest" part (handset ; Radio Base Station ; A4400) as follows :



Capabilities of the different available Alcatel DECT terminals are as follows :

DECT Terminal	Identification	Authentication	Encryption
4074 GB/GH/GI/GC/GBEx step 2	X	X	X
4072 Speech / 4072 Comfort	X	- / X	- / X
Wireless 3G Reflexe set	X	-	-
4074 B/H	X	-	-
4075	X	-	-

Attention : to benefit from encryption when migrating to Release 3.1, one must re-install the legacy handsets.

Availability of Authentication and Encryption also depends on the A4400 release

- Available only on release 3.1
- Encryption licence (On/Off)
- A4400 R3.1 with licence "On" also supports non-encryption-ready DECT sets

Authentication **and** encryption are only available with "Advanced" RBS
Authentication alone is also available with "Optimised" RBS

5.3.5 A4980 "PC Phone"

DECT handsets bring mobility services with a certain level of service from basic (GAP sets) to advanced (4074). This is well accepted when on the move, but, when back in his office, the user looks for more comfort and capabilities (e.g. group telephony, multi-line, supervision, manager/secretary). Ideally, one would like to get all the services and features available on a desktop terminal… and even more !
A4980 "PC Phone" combines DECT telephony and PC to bring all this at the desktop;

* easy call by name on PC
* screen pop up
* call log
* easy programming…

The A4980 "PC Phone" application is extensively described in another section of this document.

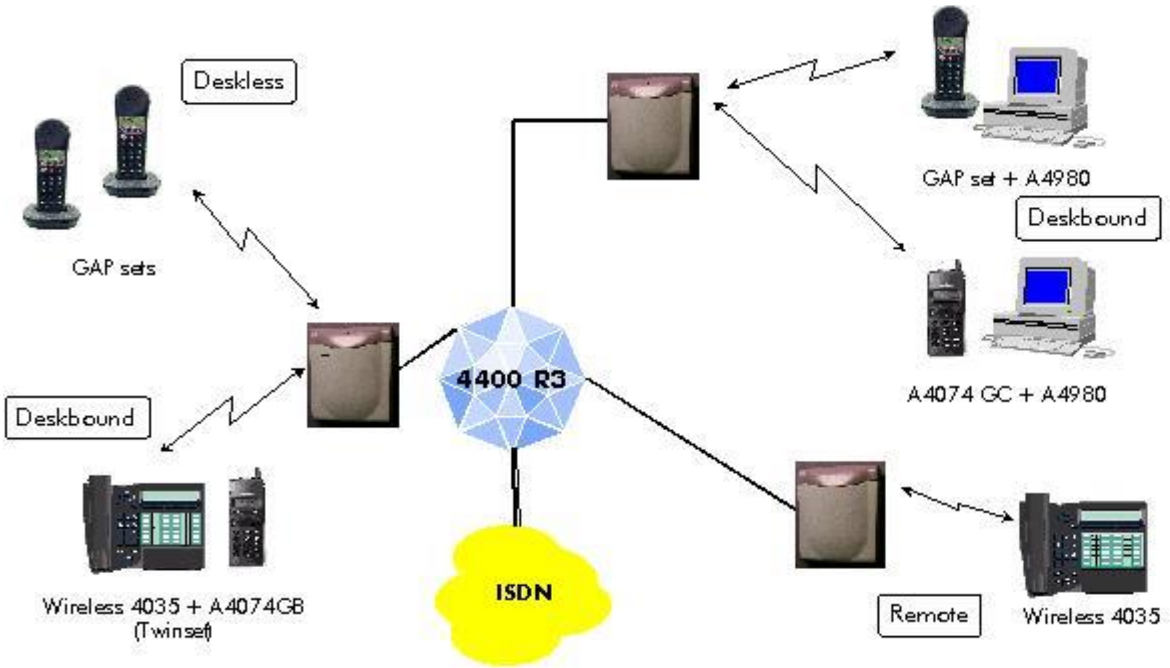
5.3.6 Ubiquity

Ubiquity is a complementary service offered on the A4400 aiming at integrating GSM phones in the A4400 environment. It is primarily intended for providing;

* ONE NUMBER service for all incoming calls,
* ONE IDENTITY service for all outgoing calls,
* ONE DIRECTORY to reach any extension of the company,
* ONE MAILBOX for all the sets of a user.

The Ubiquity service is extensively described in another section of this document.

5.3.7 Example of use



5.4 DECT SALES REFERENCES

Category	Reference	Product
Handset	3BQ28009AB	4072 Speech + Charger
Handset	TBD	4072 Comfort + Charger
Handset	4604180	4074GB UA+GAP ITU KEYPAD
Handset	4604210	4074GB UA+GAP CCITT KEYPAD
Handset	4539956	4074GI DECT Indust Term CCITT
Handset	4539949	4074GI DECT Indust Term ITU
Handset	4539932	4074GC DECT Comfort Term CCITT including charger
Handset	4539925	4074GC DECT Comfort Term ITU including charger
Handset	3BQ28008AB	4074GB Ex UA+GAP CCITT KEYPAD
Handset	3BQ28008UB	4074GB Ex UA+GAP ITU KEYPAD
Charger	35763**	4071 BA
Charger	35763**	4071 DA
Base Station	4501731	4070 IA – “Advanced” indoor RBS
Base Station	4508570	4070 EA – “Advanced” outdoor RBS
Base Station	3AC36067AB	4070 IO - "Optimised" Indoor RBS
Base Station	3AC36081AA	4070 EO - "Optimised" outdoor RBS
Base Station	3BQ28005AB	4070 EA Ex outdoor Safe “Advanced” RBS

6. UBIQUITY : ASSOCIATION & INTEGRATION OF GSM HANDSETS

Abstract :

This "A4400 UBIQUITY Product Description" contains the following items :

- A4400 UBIQUITY services description
  - ONE NUMBER,
  - ONE IDENTITY,
  - ONE DIRECTORY,
  - ONE MAILBOX,
- A4400 network constraints,
- A4400 system resources,
- A4400 releases service level.

It does not cover the configuration aspects, fully detailed in FOP document issued by PSO.

6.1 INTRODUCTION

UBIQUITY is available on A4400 R2.1 onwards.

UBIQUITY is the combination of several key selling arguments, in order to integrate GSM phones as the other resources (fixed set, DECT set, Assistant) of any A4400 user and provide :

- ONE NUMBER service for all incoming calls,
- ONE IDENTITY service for all outgoing calls,
- ONE DIRECTORY to reach any extension of the company,
- ONE MAILBOX for all the sets of an individual.

For the mobile users, UBIQUITY aims to provide :

- a high level of accessibility, wherever they are, inside or outside the company,
- access to A4400 services, such as short dialling, call-by-name, review voice mail and remote programming of fixed set, all this right from the GSM set.

For the company using A4400, UBIQUITY aims to provide :

- a significant cost reduction on calls to GSM phones, most of the calls should end up in the user's voice mailbox,
- a significant cost reduction on calls from GSM phones, they can use all the corporate private network facilities in an optimised manner,
- a significant cost reduction for external callers, they are just charged for standard calls to fixed sets.

NOTE.
Throughout this document, "GSM set" applies to any PUBLIC, CELLULAR (GSM, DCS...) handset.

6.2 POSITIONING

UBIQUITY is a complementary service offered on A4400.
Within UBIQUITY, ONE IDENTITY, ONE MAILBOX and ONE DIRECTORY will be provided as a single implementation, whereas ONE NUMBER can have several.
We focus here on the preferred implementation of ONE NUMBER, but also give the necessary information to provide a basic solution.

UBIQUITY is a key selling argument for the global mobility response provided by A4400.
When UBIQUITY uses A4635 intensively, it is a means to :

- replacement of former A46xx systems,
- increasing the number of ports on running A4635 systems,
- promotion of Attendant Manager packages on A4635 with ONE NUMBER service

6.3 USER RESOURCES AND RULES

Inside the company, a single A4400 user may have the following resources :

- a fixed set,
- a DECT set,
- an Assistant,
- a voice mailbox.

UBIQUITY provides the A4400 user with the possibility to naturally integrate his own GSM set into his other resources.

Mandatory rules to using all UBIQUITY services :
In order to have all incoming calls going through ONE NUMBER, GSM number must not be promoted.
In order to have all outgoing calls placed under ONE IDENTITY, GSM operator must provide CLIP presentation.
In order to have ONE DIRECTORY service, every user in the company must have a mailbox.
In order to have ONE MAILBOX that collects all messages, GSM operator Voice Mail must not be used.

6.4 ONE NUMBER

When inside the company, network-wide, A4400 provides mobile users with the possibility of roaming without losing a single call.

When outside the company, A4400 allows any user to have the same level of service on his GSM set, but with giving him the choice of:

- either not accepting any incoming call to be forwarded to his GSM set,
- or accepting incoming calls to be forwarded to his GSM set.

To do so, the A4400 user may :

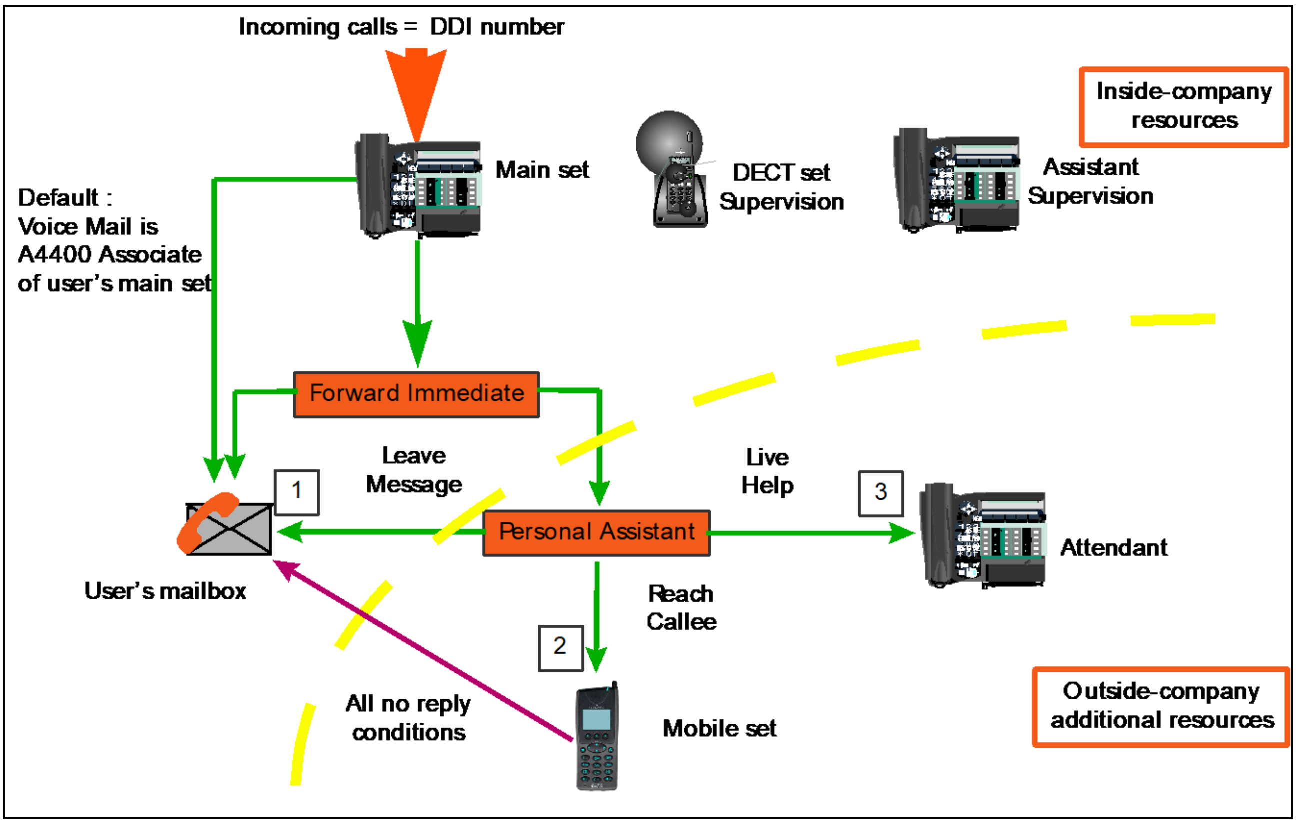
- leave his fixed set in standard configuration ; A4635 will take messages for him,
- use a Personal Assistant that invites callers to :
  - simply leave a message,
  - reach him on his GSM set,
  - call attendant/assistant.

Personal Assistant is core of ONE NUMBER :

- A4400 user chooses when to route callers to his Personal Assistant,
- callers choose to leave message, call GSM set or call attendant according to user’s instructions.

6.4.1 Service description

Below is a diagram of the ONE NUMBER service.



Some details about the ONE NUMBER service.

For A4400 user, the flexibility of ONE NUMBER service is at 2 levels :
his calls are routed to any of the following :

- his desk set,
- his Personal Assistant,
- his Voice Mailbox,

his Personal Assistant helps callers to reach any of the following :

- his Voice Mailbox,
- his GSM set,
- an Operator/Assistant

This flexibility allows A4400 user to select, at any time, the best routing scheme for his callers according to his own "availability".
ONE NUMBER service perfectly adds a new routing scheme for users who are already familiar with the usage of voice mail for call answering service.

Without ONE NUMBER service, or when user does not accept calls to be forwarded to his GSM set, configuration is as follows :
A4635 is Associate of desk set, ie A4635 will take all calls with ring-no-answer/busy conditions.
User records specific greetings for both ring-no-answer and busy conditions.
A4635 assists user to take all his calls when absent or already busy.

With ONE NUMBER service, configuration is as follows :

- user has a Personal Assistant Mailbox, in addition to standard mailbox,
- user records a specific greeting menu for callers, in order to :
  - leave a message,
  - call GSM set,
  - call operator (or assistant) for live assistance.

Transfers to GSM set/operator are supervised.
In ring-no-answer/busy conditions, the calls will be routed to user's standard mailbox.
Again, A4635 assists user by taking all his calls when absent or already busy.

By default, namely, when there is no action by caller, (i.e. case of no DTMF input), incoming calls are just forwarded to user's standard mailbox.

General advice.

Personal Assistant should be activated only in case of immediate forwarding.

In case of ring-no-answer conditions, (temporary situations, after hours time), it is not sensible to welcome callers with an invitation to reach callee on his GSM set ; personal mailbox will take messages for the user.
In case of busy conditions, there is no point in ringing user's GSM set when he is already on the phone.
In addition to this, Personal Assistant mailbox is not able to play different prompts/menus according to the actual status of user's desk set.

CONCLUSION.

ONE NUMBER scheme, as proposed in this chart, is a guideline.
It is what is considered the simplest but most powerful service.
ONE NUMBER service must be able to cope with customer specific requirements, such as additional destinations, (call by name mailbox, individual secretary per A4400 user,…).

6.4.2 One number implementation

All implementation details are given in FOP document issued by PSO.
The reader will find a summary of the necessary actions to be carried out below.

ONE NUMBER implementation requires per user :
- at system administrator level :
  - a virtual extension as the entry point of Personal Assistant, (standard A4635 addressing scheme),
  - a caller's menu mailbox, (type 31), Personal Assistant itself,
  - a virtual extension as the output point to GSM set, (this extension will be used for accounting)
  - a transfer mailbox, (type 34), which allows immediate transfer to personal mailbox on GSM call failures.
- at A4400 user level :
  - the recording of caller's menu of Personal Assistant,
  - the programming of his Personal Assistant destinations,
  - the recording of an appropriate greeting in his own mailbox.

6.4.3 One number usage

In the above description, we explained how to set up and use ONE NUMBER service.

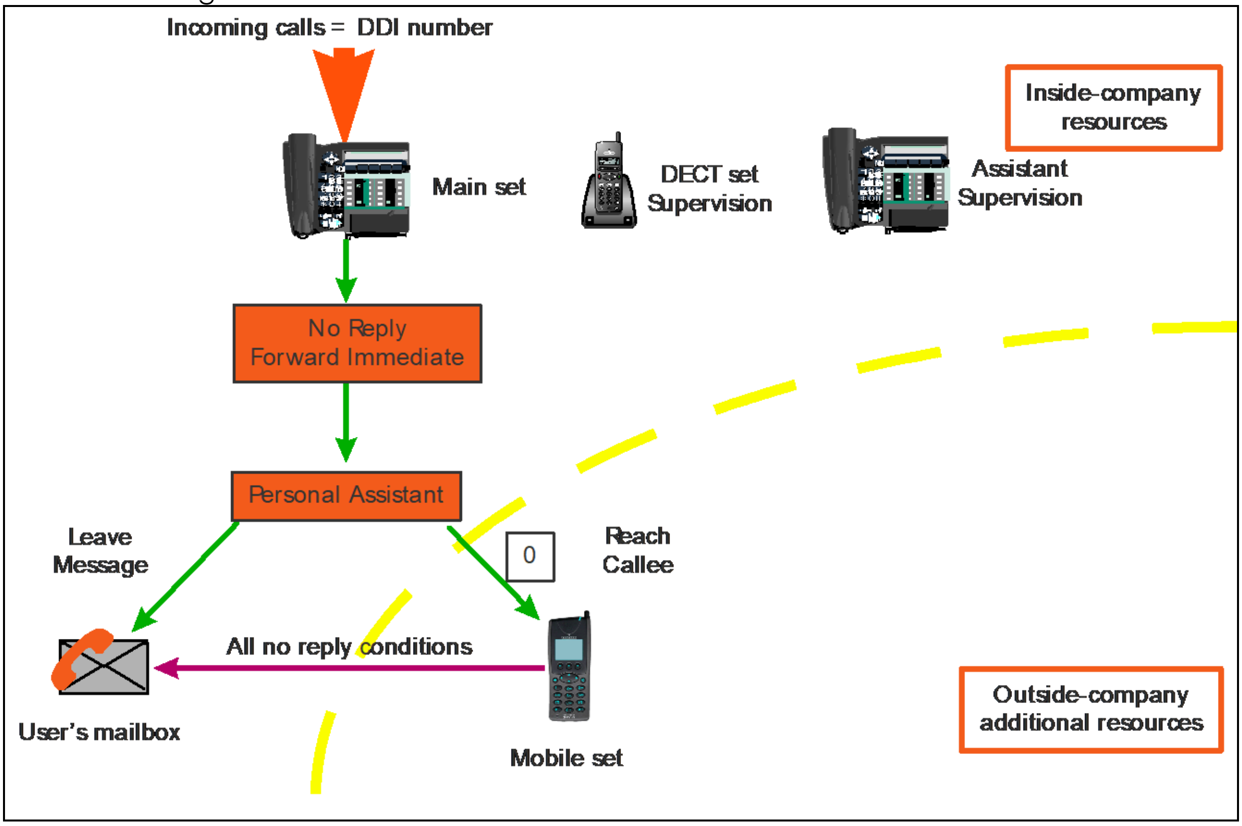
In order for the user to select the right way of accepting calls, either at his desk, roaming inside the company, network-wide, with his DECT set, or outside the company, with his GSM set, he should, set up the following keys/repertory entries on the telephone sets :
forward Personal Assistant,
forward Voice Mail,
cancel forwarding.

Wherever these actions are initiated, they all apply on user's desk set.

6.4.4 Basic ONE NUMBER service

Basic ONE NUMBER service is described here for information only.
Since it is not as powerful as the previous solution, there is no way for the user to accept or refuse his calls being forwarded to his GSM set.
The only advantage of Basic ONE NUMBER service is the ease of set up.

Below is a diagram of the Basic ONE NUMBER service.



Some details about the Basic ONE NUMBER service.

The incoming calls can be routed to Personal Assistant according to all the standard situations :

immediate forwarding,
ring-no-answer forwarding,
A4400 associate.

Personal Assistant of Basic ONE NUMBER service is nothing else but user's mailbox greeting.
We would advise a separate user's greeting for BUSY conditions.
There is no point in trying to reach callee on his GSM set when he is currently on the phone.

By default, incoming calls will be handled as a leave message in callee's mailbox.
On request of the caller, just by pressing one key, the call is routed to user's GSM set.

Whatever the no answer conditions of this call, the caller is asked to :

leave a message to callee,
leave a message to somebody else,
give up.

This ONE NUMBER implementation is the easiest to set up and requires :

a virtual extension as the output point to GSM set, (this extension will be used for accounting)
additional programming on user's mailbox by system administrator, namely virtual extension as zero-out value,

the recording of appropriate greetings (no answer/busy) by user himself.

This ONE NUMBER implementation has the following drawbacks :

there is no way of avoiding calls reaching GSM set !

there is no EASY way of driving callers to a local voice response inside the company.

6.5 ONE IDENTITY

ONE IDENTITY service allows GSM users to place outgoing calls with the same identity, name and extension as if they were using their desk set.

For the caller, ONE IDENTITY service provides the following facilities :

* access to all A4400 network resources, (numbering plan, LCR services)
* access to voice mail,
* access to remote programming of desk set,

For the callee, ONE IDENTITY service provides the following facilities :

* simple caller identification,
* all non answered calls appear with caller's A4400 internal identity,
* all voice messages are stamped with caller's A4400 internal identity to allow reply/call sender actions.

To have ONE IDENTITY service, GSM operator must provide CLIP presentation eg : for all outgoing calls or DISA sevices

DISA reminder.

DISA can be set up in the following ways :

* standard form, with or without password,
* automatic substitution, with or without password,
* standard form with password **AND** automatic substitution without password.

This last form will be used for ONE IDENTITY service, and named **PAM (Personal Access for Mobiles).**

It allows GSM users, when identified, to gain access to A4400 services, as if they were using their desk set. On the other hand, when roaming outside the coverage of the subscribed mobile operator, (international mobile operators agreements), they still have access to A4400 resources, provided they know the standard DISA password.

6.6 ONE DIRECTORY

When inside the company, users place their calls with the help of A4400 phone book.

Thanks to A4400-A4635 integrated management, A4635 handles an exact replication of A4400 directory, for all voice mail subscribers.

A4635 provides a Call-By-Name mailbox as part of the basic package.

Default factory setting number is **411**, with additional '4' leading digits up to the numbering plan length.

Call-by-Name mailbox gives access to all subscribers that are local to a single A4635 message server.

In a centralised voice mail configuration, (n A4400 nodes, 1 A4635 node), the whole A4400 population can be reached through Call-by-Name mailbox.

In a shared voice mail configuration, (n A4400 nodes, p A4635 nodes), only those subscribers with their mailbox on the same node as the **411** mailbox can be reached.

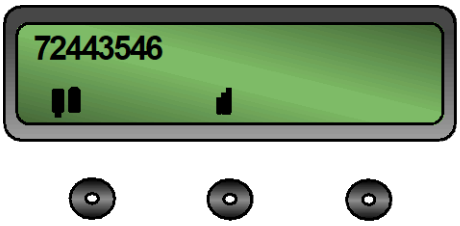
6.6.1 service description

This feature which is especially dedicated to GSM sets can also be used by Z sets.
Any set, UA, DECT, Z, outside the network, can also use this feature.

Warning.

411 mailbox can handle CCITT or ITU161 mapping, but only one mapping for whole system.
Preferred keypad mapping for UBIQUITY users is ITU161.

Usage example : Caller wants to reach "Lalo Shiffrin"



For the above name, user has to enter the following sequence :
- call PAM number,
- dial **411** mailbox,
- type in the name of the callee : 72443546 (or shorter if non ambiguous)
- caller hears callee's name with his own voice
- press "1" key for OK

In case of 'ring-no-answer/busy/forwarded to Voice mail' conditions, A4635 prompts caller to leave a message to callee.

6.6.2 A4635 ports usage

The description of the above feature shows that A4635 endorses all the actions, namely :
- prompt caller for Call-by-Name,
- interpret user key press,
- present caller with single callee or list of callees,
- call the selected extension,
- if callee answers, transfer the call,
- if callee does not answer, propose to caller to leave a message or hang up.

The whole sequence has an average time of 45 seconds, assuming that :
- 70 % of calls are answered, 30 seconds average time,
- 30 % of calls are not answered and caller leaves message, 80 seconds average time.

Figure below shows number of additional ports that are required on A4635 for Call-by-Name.
Given a pure Voice Messaging configuration, we provide the number of Call-by-Name calls that A4635 can handle during peak hour :
- with no additional ports,
- with an increase of 2 ports.

All data are computed with Grade Of Service P.02 (2 % loss calls probability).

We have chosen 2 different configurations :
- a small A4635, 200 mailboxes with heavy load, average usage 5mn/day/subscriber
- a large A4635, with 2 different populations :
  - ÷ 200 mailboxes with heavy load, 5 min/day/subscriber,
  - ÷ 800 mailboxes with light load, 1,5 min/day/subscriber.

Reader should remember that A4400 provides Call-by-Name for most of the Reflexes/DECT sets.
A4635 Call-by-Name is dedicated to external callers and internal Z set users.

	Nb of ports for VM only	D-b-N calls during peak hour with no additional A4635 port	D-b-N calls during peak hour with 2 additional A4635 ports
Small system	6	20	130
Large system	10	50	175

6.7 ONE MAILBOX

All UBIQUITY users have ONE MAILBOX.
This ONE MAILBOX is capable of the following features :

- serve as answering machine all calls presented to ONE NUMBER extension,
- serve as recipient mailbox for all voice messaging flow,
- provide Message Waiting Indication by LED on either desk or DECT set,
- provide Message Waiting Indication by outcall on any of desk, DECT and GSM sets,
- provide direct access from 2 out of all the sets (i.e. desk and GSM, DECT and GSM). This last feature uses “alias mechanism”, namely one single mailbox with 2 different numbers.

UBIQUITY user mailbox is created/modified thanks to A4400-A4635 integrated management for the following parameters :

- user name,
- user extension,
- mailbox type,
- mailbox COS,
- mailbox language,
- mailbox initial password.

All other parameters must be adjusted directly on A4635 System Management Terminal.

6.8 ACCOUNTING

Regarding accounting, a UBIQUITY user has 2 different numbers :

- actual extension,
- virtual extension associated to GSM set.

Actual extension is charged for all calls issued by :

- his desk set,
- his GSM set, when going through PAM,
- A4635 for all MWI by outcall to GSM

Virtual extension is charged for all calls issued by :

- A4635 when calling GSM set.

6.9 A4400 NETWORK CONFIGURATION

UBIQUITY has the following limits in a A4400 network configuration :

- PAM access must be located on the same node as user’s fixed set,
- Virtual extensions (Personal Assistant entry point and access to GSM set) must be defined on the same node as user’s A4635.
- A4635 directory can only address subscribers who have their own mailbox on the same system as

the **411** mailbox that was called.

6.10 SYSTEM RESOURCES

6.10.1 Public trunks

In a business environment, UBIQUITY does not require additional T0/T2 resources.
All calls to or from GSM sets use external lines, whatever the path they are using, (direct or through PAM for incoming calls, direct or through Personal Assistant for outgoing calls) and will require the same number or voice channels.

6.10.2 DTMF receivers

On the other hand, UBIQUITY requires that additional DTMF receivers handle incoming calls from GSM.

Assumptions.

1 call/GSM/hour.
15 seconds DTMF receiver allocation/call
Grade of service = 2%

UBIQUITY users	DTMF receivers
32	2
64	3
128	3
250	4
500	6

6.10.3 A4635 ports

Just like DTMF receivers, resizing of A4635 is required to endorse additional voice traffic.

Assumptions regarding extra voice ports usage for UBIQUITY users

50% heavy users ; 4mn/day extra voice ports usage
50% light users ; 2mn/day extra voice ports usage
Grade of service = 2%

Table below gives a guideline for the extra ports calculation.
For any of the UBIQUITY orderable items, it gives the necessary A4635 ports, either for a brand new installation (column 0 existing ports), or the extra ports when UBIQUITY is added on a system already running A4635 (columns 2-16 existing ports).

	Existing Ports	0	2	4	6	8	10	12	14	16
users										
32		2	2	0	0	0	0	0	0	0
64		4	2	0	0	0	0	0	0	0
128		4	2	2	2	2	0	0	0	0
250		6	4	4	4	2	2	2	2	2
500		8	8	6	6	4	4	4	4	4

6.11 A4400 RELEASES SERVICE LEVEL

All the services described above apply to A4400 R2.1 – version ID B3.515

First A4400 version to run UBIQUITY is B3.513.
This version provides the following mandatory feature :
Forward to Virtual set forwarded to Voice Mail.
Former versions cannot use "true" virtual sets, but require a Z extension without physical equipment.

The following restrictions apply to today's shipped version B3.513.

- Original CLIP presentation (available on release 3.1…)
  This service allows UBIQUITY users to identify original caller any time the call goes through his Personal Assistant. This feature is available according to the mobile operator specific rules.

- PAM service
  The right service is to provide :
  a PAM service without password for all identified callers,
  a simple DISA service with password for all other callers (or UBIQUITY users roaming outside the coverage of their mobile operator)

- Identity after automated substitution
  The identity presented for all calls after automated substitution is internal identity of caller.
  This allows :
  Simple call back,
  Accurate time stamp and sender identification for voice mail messages,
  direct access to Voice Mail, Associated set, etc with the same prefixes as on the fixed set, …

- Accurate accounting for all calls of UBIQUITY user
  This is possible, thanks to the introduction of a virtual extension to reach mobile set.
  Accounting scheme in this case is the same as for "human operator", namely :
  1 accounting ticket generated for the whole call, (A4635 call + transfer)
  everything is charged on virtual extension associated to UBIQUITY user.

6.12 EVOLUTION WITH RELEASE 3.1

6.12.1 Original CLIP presentation

With the release 3.1 of A4400, all callsre-routed to the "UBIQUITY GSM" by A4400 are presented to the mobile set (GSM) with the original CLIP

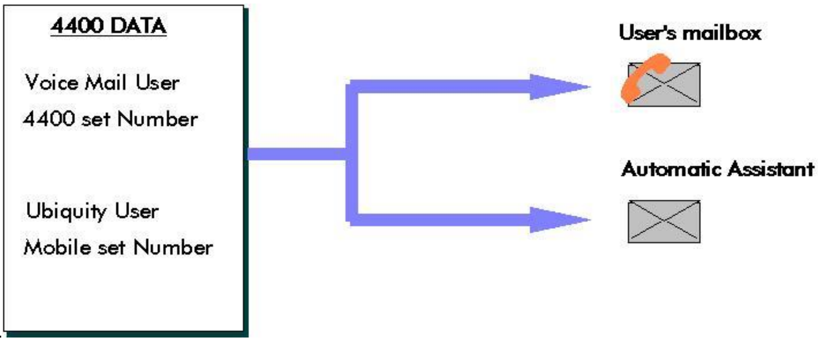
- The UBIQUITY user can better react to an incoming call and only accept really urgent calls
- Some GSM sets (e.g. Alcatel One Touch Pocket) store the CLIP number if the GSM handset is rung and call remains unanswered

Note : this feature is also available with the Release 2.1 "support"

6.12.2 Installation simplification

The installation and parameterisation of UBIQUITY application is greatly simplified as, from release 3.1 onwards, its management is fully integrated in Alcatel management applications :

- MGR
- 4730
- 4740
- 4755



The “User Information” are collected and processed within one single session. Both the user’s mailbox and automatic assistant are automatically configured. Installing and configuring UBIQUITY becomes as easy as ABC…

7. ALCATEL TELEPHONY ON PC : A4980 "PC PHONE"

Nowadays, telephone communication in itself is insufficient and businesses are expressing a new need, namely that of voice communication services on PC. Users are asking for more comfort and a wider set of features when office-bound. In short, modern enterprise telephony has come a long way since traditional telephony. E-mail paved the way for this increased tendency to use PC to communicate. We have first-hand knowledge of this evolution and of the need for PC and telephony convergence. We have done just this...

- the Alcatel 4400 release 3 proposes “Alcatel 4980 PC Phone Release 1”, a powerful real time voice/data communication PC Application designed to enhance end user productivity
- the Alcatel 4400 R3.1 proposes “Alcatel 4980 PC Phone Release 2” offering now hot new PC Telephony solutions **unique on the market** for our customers

- *The following document describes first the Alcatel 4980 PC Phone Release 1, then the second part describes what’s new in Alcatel 4980 PC Phone Release 2*
- *Note that Alcatel 4980 R2 is available with Alcatel 4400 R3.1 and is compatible with Alcatel 4400 R3 with some service restrictions (see “*Compatibility with 4400 R3*”). Of course Alcatel 4980 R1 is also compatible with Alcatel 4400 R3.1*

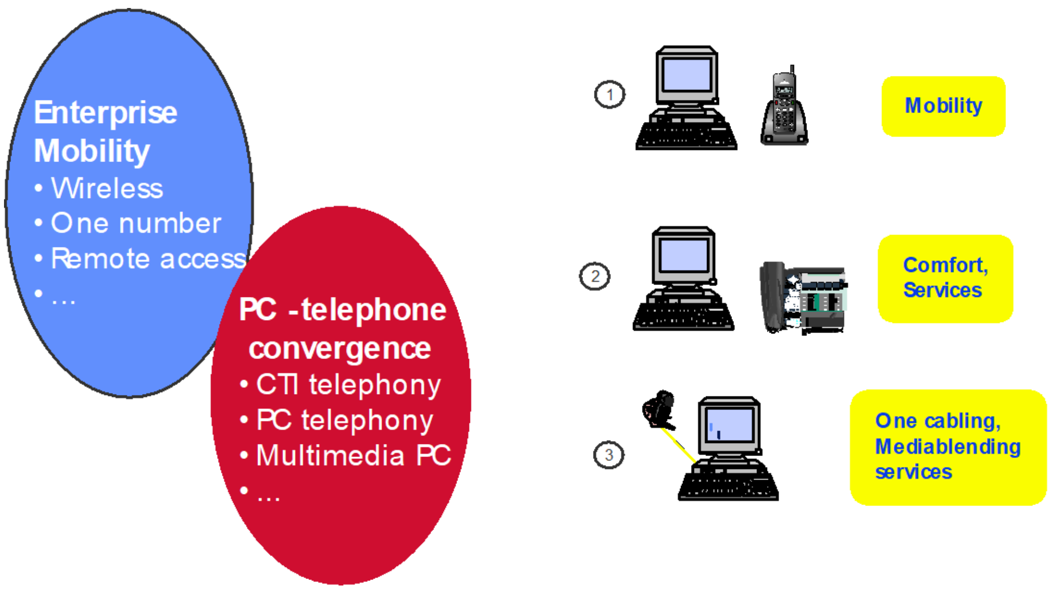
7.1 MAJOR TRENDS AND NEED IN THE ENTERPRISE COMMUNICATION WORLD

Today, the two major trends in the Enterprise communication world are obviously :

- Mobility
- PC and Telephone convergence

These trends will give rise to three basic user desktops with specific key advantages :

- mobile handset with associated PC : offers mainly "Mobility" to user
- desktop set with associated PC : offers mainly "Comfort & new Computer Telephony services" to user
- multimedia PC : offers mainly "One cabling" for the customer and "Mediablending services" to user



The complete range of DECT handsets brings mobility to everyone.
Of course, all the services and features available on a desktop terminal cannot be offered on a mobile handset. This is well accepted when on the move, but users ask for more when in their office :

- comfort
- extended set of features (group telephony, multiline, supervision, manager/secretary, new communication services of PC world)

A4980 "PC Phone" combines DECT telephony and PC to bring all this at the desktop. ***It's a "lightning" association offering a powerful desktop solution***

At any moment the user can handle the calls on PC or on a DECT handset :

- when the user is on his desktop : calls will be handled on PC and the DECT handset will be used for audio like a handset of a wired set
  (in this case there is a virtual 4035 phone set in the PC : softkeys, information on display, multiline, ...)
- when the user is away from his PC or in case of PC failure : the DECT handset will be used as a normal phone set (with the limitations of the DECT handset telephony)

More and more customers are interested in wireless systems

Users who only have a DECT set require the following features when they are in their office :

- easy call by name on PC
- screen pop up
- call log
- easy programming
- desktop integration with other Office Applications (Schedule, Outlook, Excel, …)

And of course this service is extended to other types of desktops (wired set, multimedia PC*)

* : multimedia PC running Alcatel 4980 PC Telephony is available for Tele-workers in Alcatel 4980 R2 and for future 4400 IP-PCX

7.2 MARKET NEEDS AND ALCATEL'S PC TELEPHONY OFFER

For Office desktop, the market need is a powerful real time voice/data communication PC App designed to enhance end user productivity :

- designed for **real time** person to person communication
- with **new, efficient, easy to use** voice/data communication services
- fully **integrated** with other desktop Apps
- for any desktop :
  - whatever associated voice device (Alcatel Reflexes, POTS, DECT, GSM, multimedia PC)
  - On Site, Off Site
- for any system : small, medium or large
- one powerful architecture for all needs
- easy pricing for all needs (Tele-worker, POTS, multimedia PC) and promotion package
- adapted for sales on installed base

The answer is Alcatel 4980 PC Phone

Additionally to the Office desktop services mentioned above, Alcatel 4980 "PC Phone" is a new client/server PC Telephony App offering :

- designed PC telephony for mobile users with DECT handset (mobile desktop, wireless PBX)
- designed PC telephony for Tele-workers with GSM or multimedia PC (in 4980 R2)
- the best of ergonomics at Desktop

- the feature rich Alcatel Telephony on PC
  - for all users with a telephone set, whatever the type of set
- reliability and performance thanks to Alcatel expertise and development of all software components (4400, CTI link, Telephony server and 4980 clients)
- a great user customization
  - PC screen layout, "One key" services, DECT handset programming, ...
- openness for App inter-working and developers

7.3 ALCATEL'S PC TELEPHONY CORE POSITIONING

Alcatel 4980 "PC Phone" is a product with a 3-core positioning :

1. it's the product of the future for your communication workplace

- full integration of : PC + phone set, providing a brand **new Enterprise Communication service**
- GroupWare*, Universal Call by Name*
- multimedia PC** (voice over IP)

2. it's a product really designed for mobile users

- On site : PC + DECT
- Off site : multimedia PC* (voice over IP), PC + GSM*

3. it's a powerful CTI product

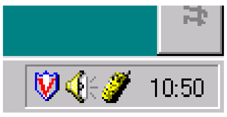
- Windows App inter-working : DDE, Notes*, 4635 VM*, 4400 UM*, TAPI*, MAPI*, LDAP*, ...
- OLE API with COM objects*
- Alcatel 4980 Client SDK for developers**

* : Those services are now available with 4980 in Release 2
** : Those services are not available with 4980 in Release 2

7.4 CLIENT APPLICATION LAYOUT AND PRESENTATION

<u>Three layouts are available on user's choice :</u>

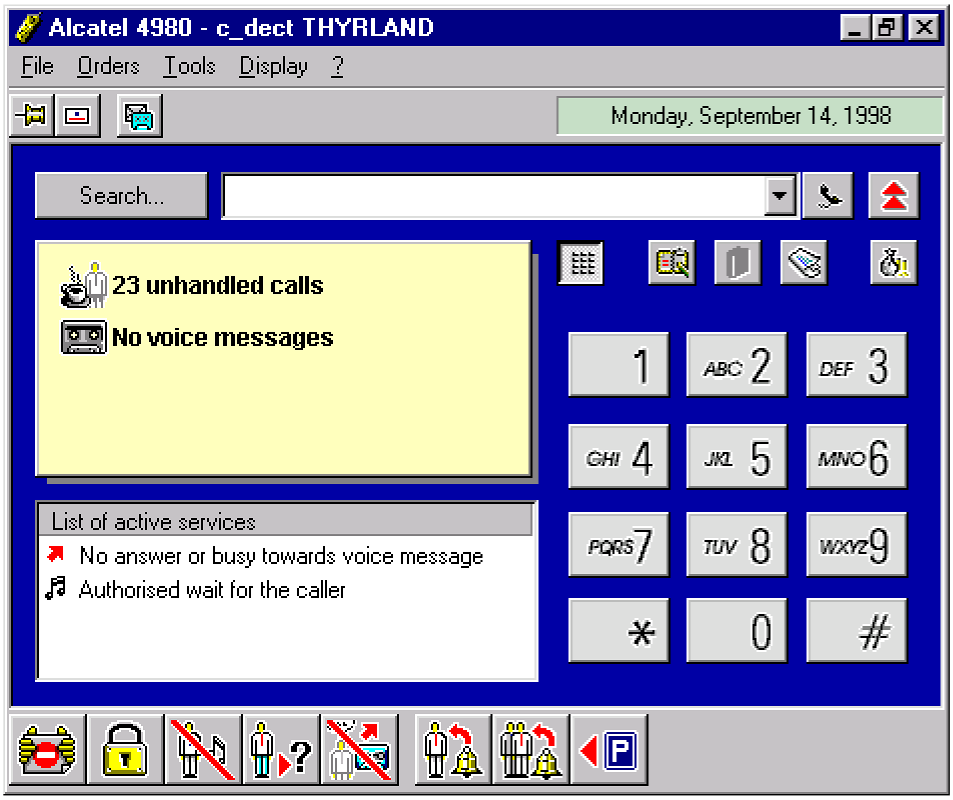
- the "PC Phone" Icon in taskbar of Windows desktop



- the "PC Phone" Toolbar (pop-up or permanent)

- the "PC Phone" Window

The client window snapshot below show the App in idle state



The basic Window contains :

- a visiting card area of caller or called party
  - in idle state (if no call) this area shows the mail notifications

- below the visiting card, a calls list area showing :
  - the services activated in idle state (Forwarding, Locking, ...)
  - the calls list when not in idle state

- a calling area with
  - a numeric keypad or speed dialing keys (on user choice)
  - a Dial by Name (or Number) field with associated Search key (left) and Dial key (right)
  - a Redial key
  - a Last Calls field (pop-up list in Dial by Name field)
  - a key to select the numeric keypad or speed dialing keys
  - a Repertory key
  - a Directory key
  - a Calls Logging key
  - a key to set-up a Private or Project call

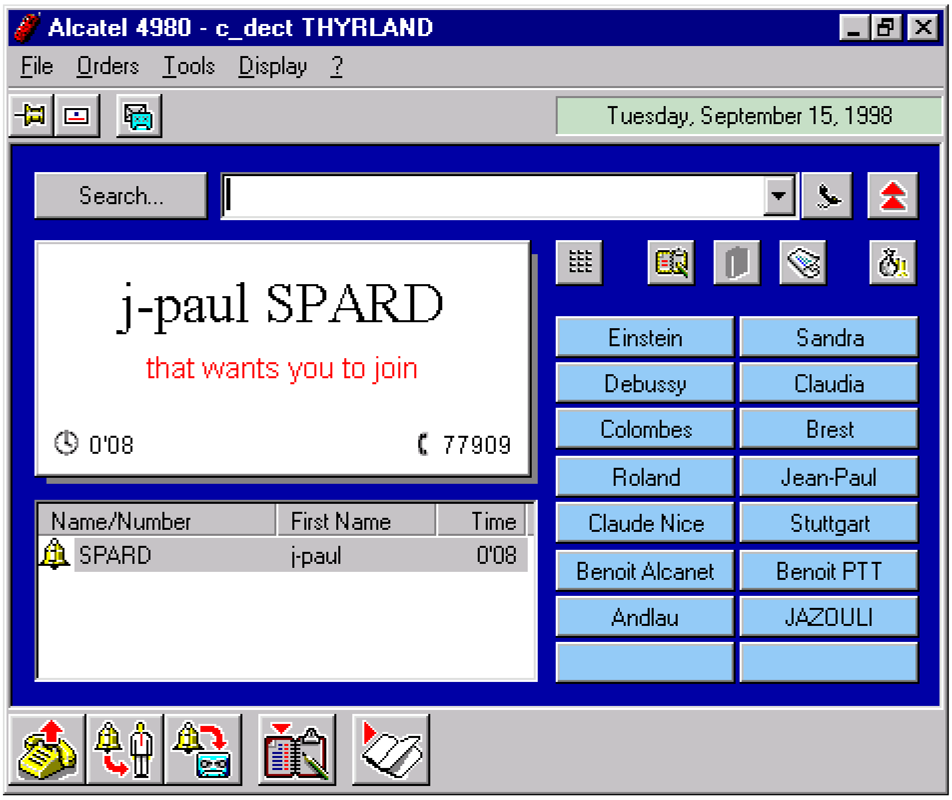
- a softkey toolbar to access dynamically the right services depending on communication state
  - in idle state, the content is fully user customized for the following services :
    - DND, Locking, Individual Pick-up, Group Pick-up, Call Park and macros (Example :
    - Forwarding with pre-programmed destination)

- an App toolbar with non communication services
  - permanent focus, hidden visitor card, access to voice mail, date display, ...

7.5 MAIN USER BENEFITS

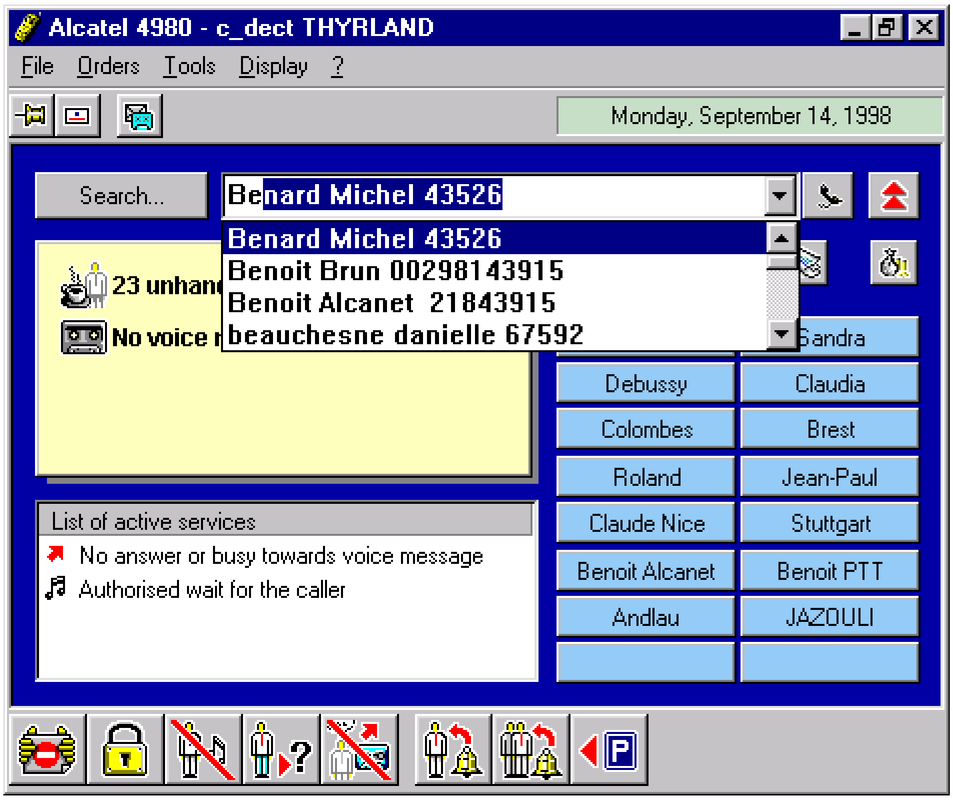
Screen pop on Incoming calls

- with Caller Name and number
- in conversation for second call consultation
- is available for user in any Windows Application
- fully user customizable : Icon, Toolbar or Window



Quick & friendly Outgoing calls

- Dial By Name in Company & Personal Phone Books
- Speed dialing keys, Last Calls list, Redial key, Calls Log



Audio

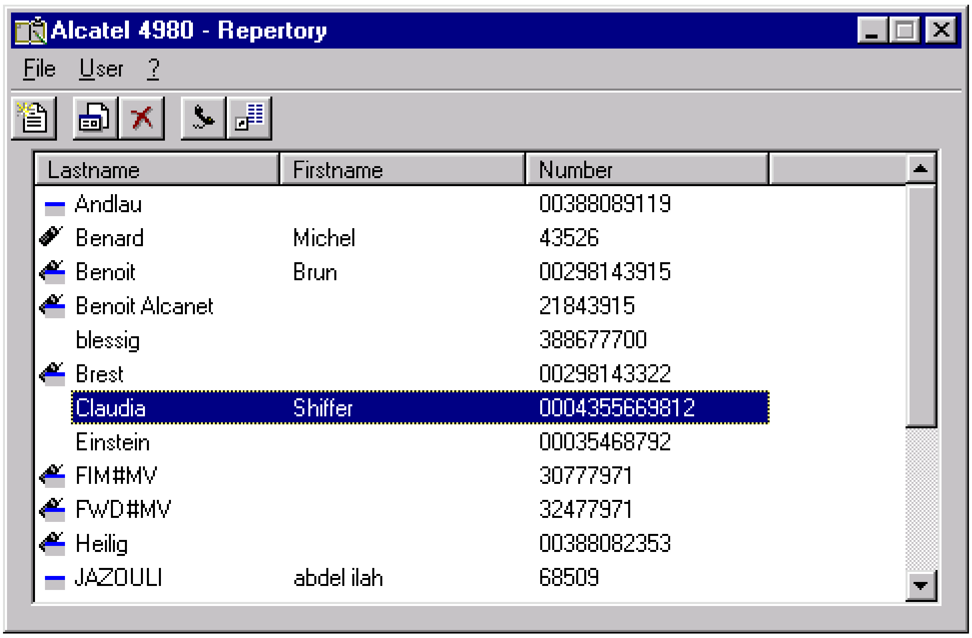
- Hands Free on DECT parking unit for 4074 GC
- Loudspeaker in handset for 4072

Easy "One key" access to basic features

- Forwarding, Voice Mail, Last Calls, Calls Log, Directory, ...
- fully user customizable for "One key" services choice

Personal Phone Book

- "Personal Phone Book" is an integral part of "4980" App
- Call set-up from "Third party Personal Phone Book" is available for :
  - Schedule for Office 95
  - Outlook for Office 97 & 98
  - Access, Excel, Word (Macros and procedure are in 4980 Help)
  - Notes (Lotus script and procedure are in 4980 Help)



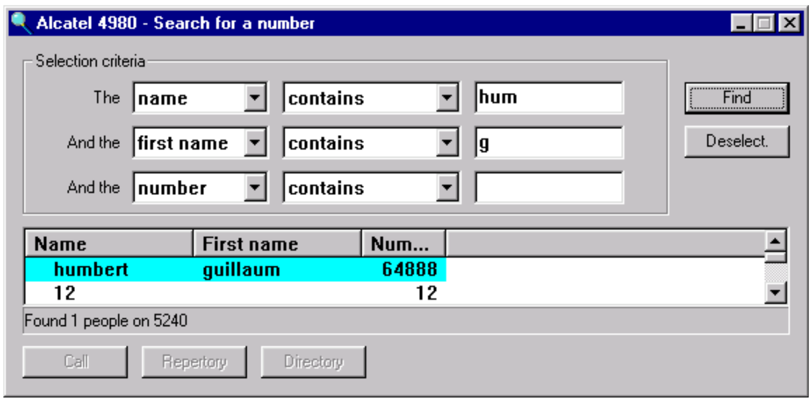
Alcatel 4000/4755 Company Directory/InfoCenter

- 4000/4755 Directory client is fully integrated with "4980" App (like for 4059 Multimedia)
- offers a complementary service for Consultation, multi-criteria search and absence information
- 4000/4755 Directory "Pop-up" will be available for specific calls
  - this service will be provided by Telephony Server thanks to inter-working with 4000/4755 Directory server

Search function in Company & Personal Phone Books

The search function in Company and Personal Phone Books is available on the three following fields :

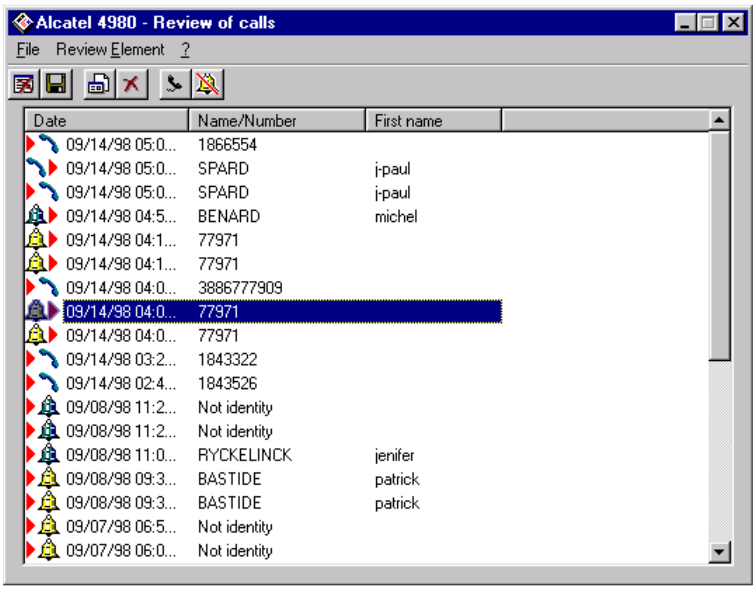
- Name
- First Name
- Directory Number



Calls logging (even if PC client is off)

On user choice the logged calls can be sorted as follows :

- incoming calls answered
- incoming calls unanswered
- outgoing calls answered
- outgoing calls unanswered



By pressing a toolbar key, the user can display the sort list or all calls in the Calls logging window

Calls logging is done at Telephony server level even if :

- PC client is off
- DECT handset is off

Note that no Call logging is available during Forwarding :
> this point should be solved in future with "Personal Call Routing function"

User customization

- screen layout, windows components and size, sounds, ...
- screen pop on incoming call, PC sound on incoming call, speed dialing keys, ...
- choice of some services display, depending on communication state :
  - in idle state, the content is fully user customized for the following services :
    - DND, Locking, Individual Pick-up, Group Pick-up, Call Park and macros (Example : Forwarding with pre-programmed destination)
  - on incoming call : macros (Call Diversion with pre-programmed destination)
  - during call : Directory pop-up, Netmeeting services

Transparent Phone/PC use

- the user can transparently handle the calls on PC or Phone Set (fully synchronized) whenever he likes

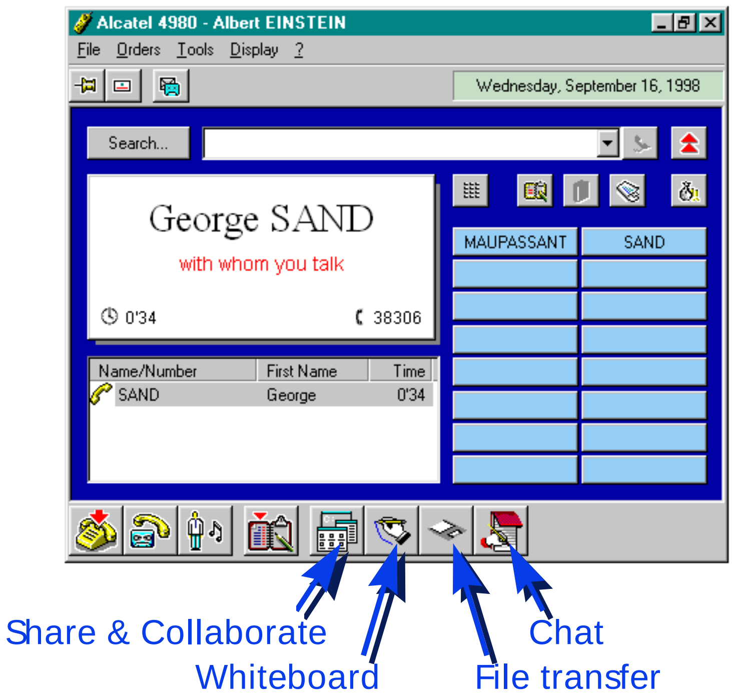
New feature rich Alcatel Enterprise Telephony on PC

- multiline Telephony
- Manager/secretary GroupWare Telephony
- New Enterprise Telephony
  - access to most 4400 Alcatel Enterprise Telephony services
  - access to many new communication services coming from :
  - 4400 Management, 4980 App, App inter-working (Netmeeting, Schedule, ...)

Netmeeting services

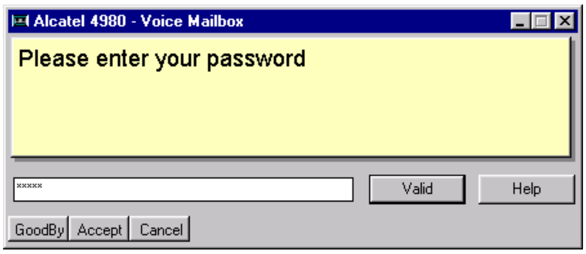
During conversation the user has direct access (softkeys) to the Microsoft's Netmeeting App services :

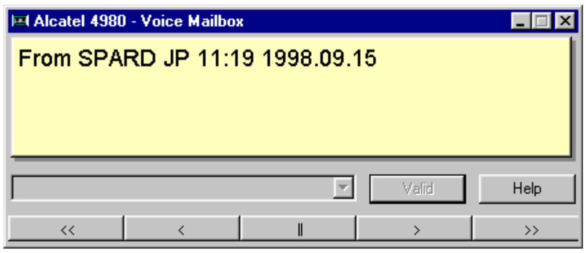
- Share & Collaborate
- Whiteboard
- File Transfer
- Chat

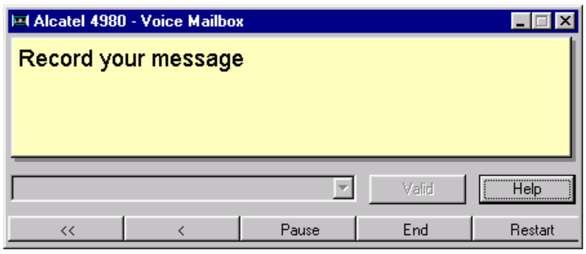


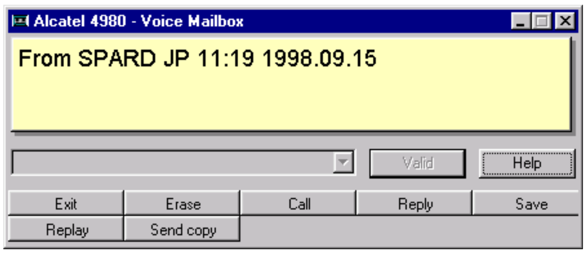
Visual Mailbox

- Voice Mail notification (4635)
- access to 4635 voice mailbox (like a set with softkeys)









Mobility

- Virtual office with "Alcatel 4980" Mobile services
- For DECT handset users, "4980" mobile services are available by login on any PC on the same node.

Inter-working with popular desktop Windows App

- TAPI Assisted Telephony (ex. : Notes, Schedule, Access, Excel, Word, ...)
  - after installation or launching, 4980 client will be automatically the TAPI dialler for any TAPI Assisted Telephony Make Call request
- inter-working with TAPI compliant Apps (Outlook, Act, Twixtel, …) for "Make Call"
  This allows to set-up calls from Outlook and call progress in handled by A4980 client

- DDE services : Make Call with number or name, ...
  (DDE services are only available in Alcatel 4980 Advanced)

For 4980 R1, DDE services are limited to "Make Call" with number or name.

- New services are available in 4980 R2 and will be provided depending on customer needs ...

7.6 LIST OF ENTERPRISE TELEPHONY SERVICES IN R1

- Incoming calls
  - Automatic Screen pop with Caller Name display

- Outgoing calls
  - Callee Name display, Dial by Name, Speed dialing keys, Personal Phone Book

- Call waiting consultation

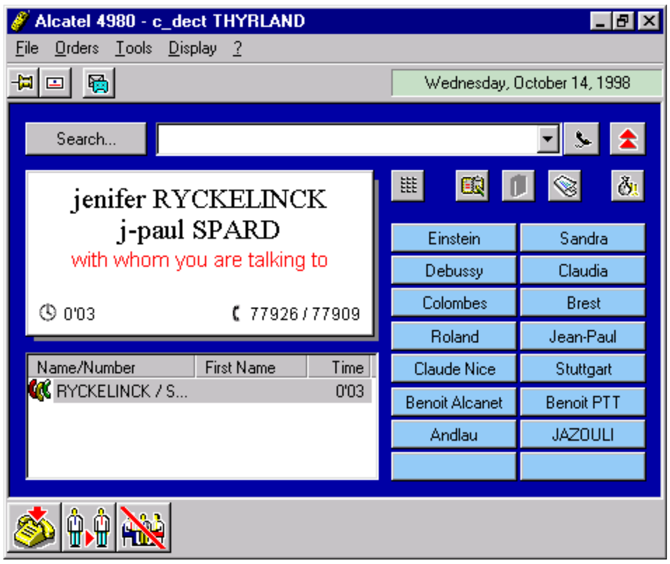
- Hold/Unhold (exclusive)

- Enquiry call

- Broker call

- Transfer

- Three party conference



- DTMF sending
- Business account code (by Prefix)
- Private call with PIN (by Prefix)
- Call-back request
  - only if busy (no notification on PC)
- Call pick-up
  - individual, group
- Call Forwarding
  - immediate, on busy or no reply
- Send message to pager (DTMF)
- Multiline mode

  Multiline mode is limited :
  - to line keys monitoring (no supervision keys monitoring)
  - to 5 simultaneous calls

  Note that monitoring of supervision lines is not necessary as the incoming calls are taken on the line keys with pick-up indication
  Alcatel 4980 multiline works also on Manager/secretary set (see below)
- Supervision of other sets in a workgroup
  - this service is not managed by Alcatel 4980 R1 (see GroupWare of Alcatel 4980 R2)
  - this service is only available on wired set in multiline mode when associated with 4980
- Manager/Secretary

  First of all, Manager/Secretary is only available with 4020/4035 in multiline mode (today's solution)
  Alcatel 4980 "PC Phone" is compatible with Manager/secretary phone set and offers following services :
  - services on PC & set :
    - supervision function of filtered calls is provided between Manager/Secretary
    - DSS/BLF function is limited to outgoing calls and monitoring (incoming calls are on normal line keys)

In case of CSTA monitoring, a DSS/BLF outgoing call initiated from PC or set will call the normal line keys on called party (Manager or Secretary)
– all filtering/absence activation are possible
* limitations on PC (but available on phone set) :
  – outgoing calls are limited to main Directory Number of the associated phone set
  – Manager/Secretary mini-messaging on screen is not available
  – secret listening is not available

Full Filtering tables feature is available (through CSTA extension) :
* common filtering tables (16 max in 4400)
* personal filtering on specific ISDN caller identity (CLI)

- Twinset
  Alcatel 4980 PC Telephony Application is compatible with "Twinset" function :
  * In case of « Twinset » configuration, Alcatel 4980 R1 Application works normally with the main set (desktop set).
  For example in case of Manager/secretary, this allows the Manager to use the 4980 App for the desktop set.
  What's not possible is to toggle the 4980 App between the two sets …

- Mail notification
  * Voice Mail notification (4635)
  * Non answered calls Logging (ISDN, Internal)

- Voice Mail
  * access to 4635 voice mailbox (like a set with softkeys)

- Softkeys
  Softkeys are dynamic service keys depending on state of communication.
  They offer a high level of ergonomics as at any moment the user has only the relevant services on screen.
  Alcatel 4980 App offers softkeys for many communication Applications :
  * 4400 softkeys for 4635 Voice Mail services
  * 4400 softkeys for Telephony services
    The softkeys presented by Alcatel 4980 App are synchronised and coherent with 4400 Call Handling : communication state, service availability and the user rights are checked
    For example : "Conference" softkey will never be proposed by "4980" Client if not available for this user at 4400 Call Handling level
  * "Alcatel 4980" softkeys for complementary services

For example : copy/paste caller/callee number to Personal Phone Book
  * Alcatel 4980 softkeys thanks to inter-working with other Windows App
    For example : during a call, Directory screen pop-up or Netmeeting services

- Call Barring

- Abbreviated Numbers
  * only common Abbreviated Numbers are managed
    - from 4980 Phone Book, Speed dialling keys or numeric keypad of PC
  * Abbreviated Numbers with post-dialling is not available from 4980 App (only from phone set)

- Hunting groups
  * a user with 4980 client can belong to a 4400 Hunting group (cyclic or parallel groups)

- Overflow on busy or no reply

- Set Lock/Unlock, personal user password

- Protection against
  * (e.g. : mail access, forwarding, intrusion, ..)

- Multi-node moving limited to DECT handset services

- Audio
  - Hands Free on DECT parking unit for 4074 GC
  - Loudspeaker in handset for 4072
  - no control from PC for DECT handsets

- DECT handset and Reflexes set programming
  - Repertory (name, number)
  - Password
  - Associate Number
  - UPK (User Programmable Keys)
  - Services in idle state : Lock, Forwarding, Overflow Number, ...

  DECT handset and Reflexes set programming is provided through 4400 Management App.

7.7 ARCHITECTURE

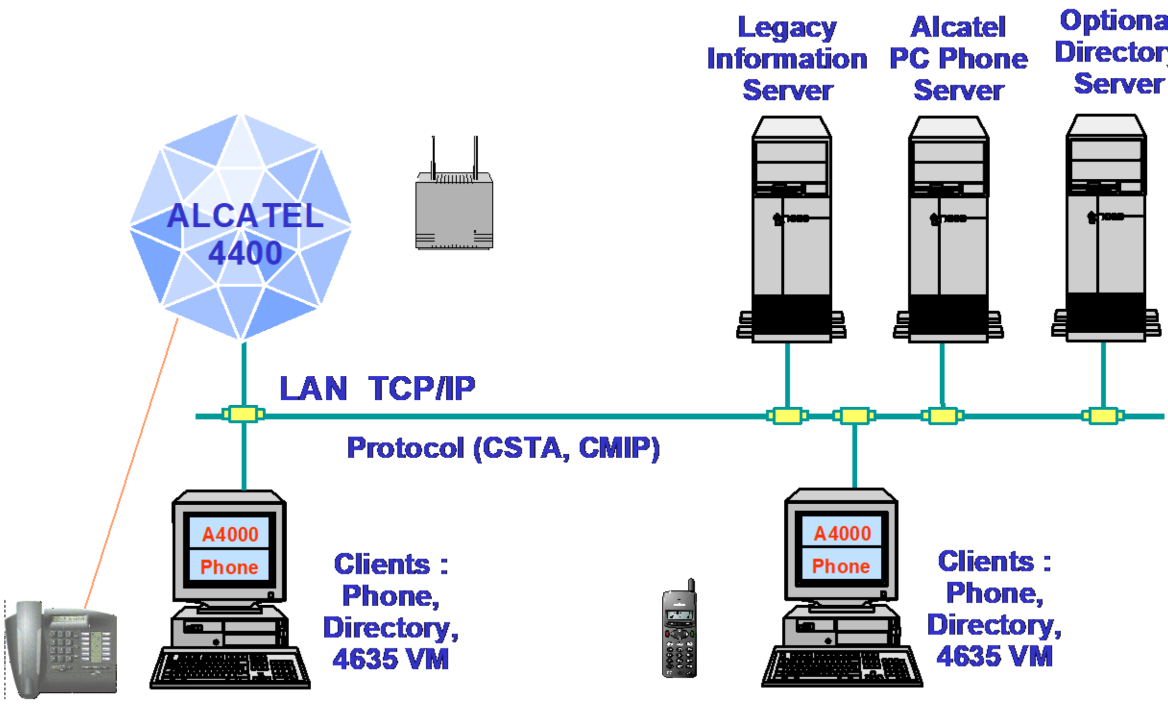
Alcatel's PC Telephony is based on a third party and client/server architecture

Server App runs on **Windows NT Workstation** or **Windows NT Server**

Client App runs on **Windows NT Workstation** or **Windows 95** or **Windows 98**

At "4980" server level :
no Windows NT client license will be necessary as connection with PC clients is made in "socket" mode

At client level : "4980" will be a 32 bits application :
this means no compatibility with Win 3.x; this is important for inter-working ability with other 32 bits MS Windows App



As shown in the figure above,

The 4400 system providing the Telephony services is connected to a "PC Phone" server via TCP/IP Lan.
The "PC Phone" server runs the 4980 App and provides the server function for the 4980 clients on TCP/IP Lan.
There is **no connection** between the phone set (DECT handset in the case shown here) and the user's PC.

Up to three Alcatel clients App can run on user's PC :

- 4980 "PC Phone" for voice communications services
- 4000/4755 Directory client for Directory services
  - this Directory client is fully integrated with 4980 client in R1
- 4635 Visual Messenger client for Voice Mail and Fax services
  - this client will be integrated with 4980 client in R2

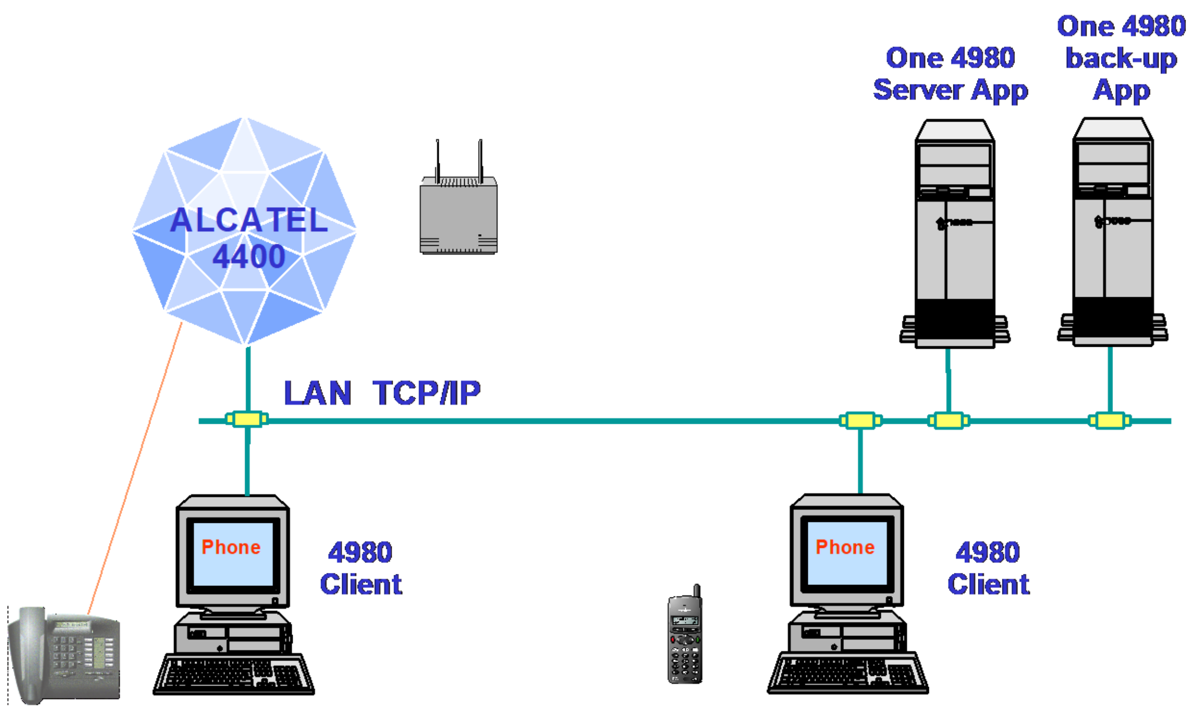
"Call by Name" and **Enterprise Phone Book** service is offered by "4980" server App. The first time, the Phone Book is downloaded from 4400 system (for new Phone Book entries) and then automatic real time update is provided in case of Phone Book entry creation or modification.
Integration of 4980 client with 4000/4755 Directory client offers additional Directory services from PC Phone (access icon, pop-up, automatic launching, automatic App switching, ...)
Directory 4000/4755 is an option

In case of 4635, **Voice Mail services** (like on 4035 phone set) are also offered on "4980" PC Client.
Integration of 4980 client with Alcatel 4635 VM Client on PC (called "Visual Messenger") is planned for R 2, in this case another "Access Gateway" will be necessary on LAN (not shown on the figure above)
This "Access Gateway" is an option

Alcatel's PC telephony offers an architecture based on standards : CSTA, CMIP protocols
CSTA protocol is used to access the 4400 voice services.
CMIP protocol is used to access the 4400 management services

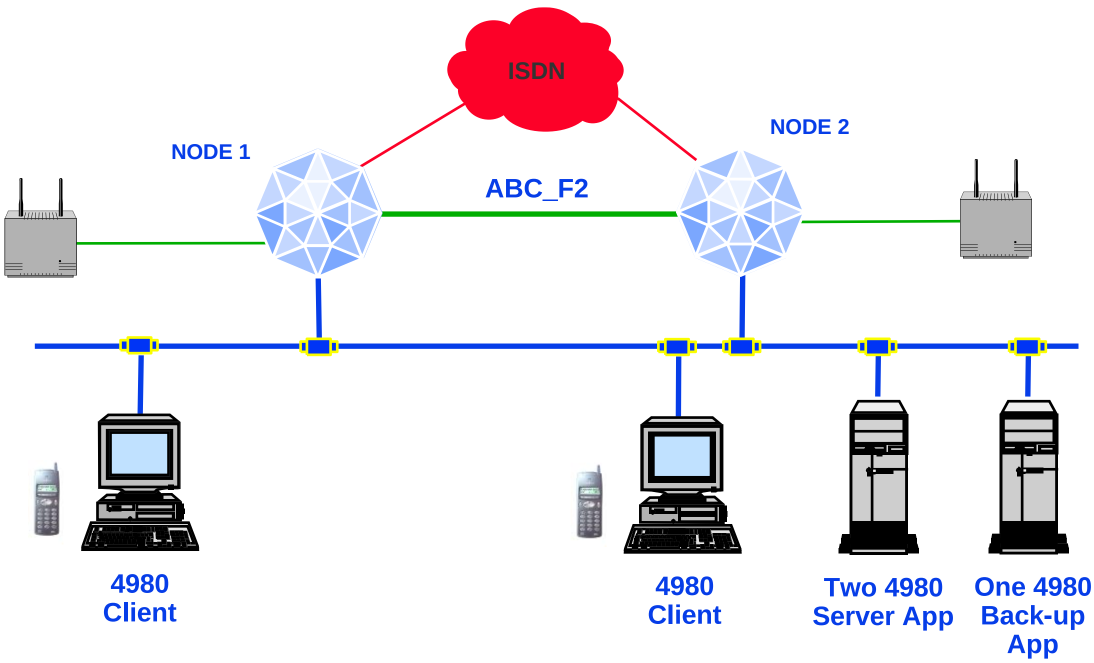
Client LAN requirements : TCP/IP protocols
TCP /IP stacks are native for Win 95, Win 98 and Win NT

One back-up server can run on another dedicated PC

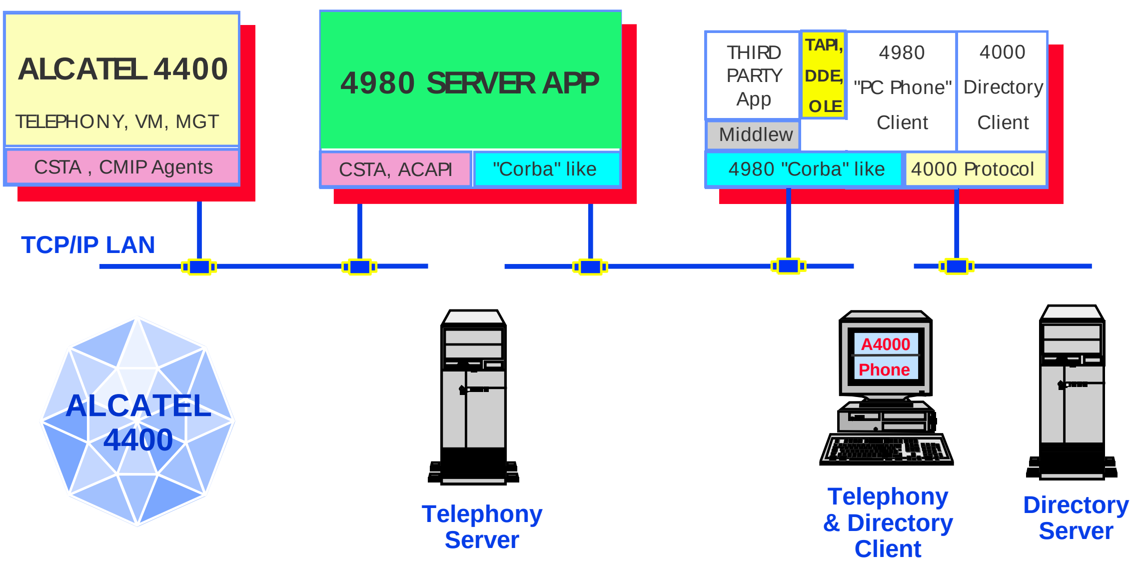


Networking : one "Alcatel 4980" server App is necessary per 4400 node

In case of small 4400 nodes in private ABC-F network (see figure below for the case of two small network nodes) :
several 4980 server App can run on the same PC
nevertheless, the LAN bandwidth between nodes must be sufficient
tests are in progress for LAN requirements
in this case, the servers require one back-up App server on dedicated PC



Alcatel's PC Telephony : functional architecture



As shown in the figure above,

Alcatel 4400 provides :

Telephony (4400 Call Handling), Voice Mail (VM 4635) and Management (MGT) Applications
CSTA Agent (for Telephony and Voice Mail services) & ACAPI Agent (for Management services)

Telephony server (Windows NT workstation or Windows NT server) provides :
"4980" App (based on 4400 Telephony + 4980 supplementary services)
server function for PC Clients

"4980" Client (Windows NT or Win 95 or Win 98) provides :
* presentation of 4980 "PC Phone" services
* Directory pop-up and integration with 4000/4755 Directory Client
* interfaces for third party App

For third party App, the openness provided is the following :
* TAPI Assisted Telephony (ex. : Schedule, Outlook, Excel, Word, ...)
* DDE services
* TAPI limited to "Make Call" and "Receive Call" services : available in Release 2
* MAPI : available in Release 2
* OLE automation : available in Release 2
* LDAP : available in Release 2
* 4980 Middleware with SDK for developers : planned in Release 3 (Alcatel Telephony SDK - with : OCX, Active X)

- note that this SDK will first be provided at 4980 client level

Standard based architecture : CSTA, CMIP protocols
* CSTA (with private extensions) offers access to the 4400 Call Handling and Voice Mail services (service access and synchronization)
* CMIP (with ACAPI on top) offers access to supplementary services like : Phone Book, set programming, Forwarding activation, ...

A proprietary "CORBA" like software is used to link the 4980 client App with the 4980 server App.
Another proprietary Directory protocol is used to link the 4980 client App and the Directory client App with the 4000/4755 Directory server App

Note :
CSTA extensions for 4980 are given by :
- CSTA private events (only for 4980 use), CSTA events with private data or CSTA escape services

7.8 MAIN 4980 PHONE SETS CHARACTERISTICS

Alcatel 4980 offers PC Telephony for all users but the Telephony service level depends on the type of associated phone set :

* "Alcatel 4980" for mobile users with UA & GAP DECT handsets (4074, 4072)
  - works in multiline or monoline mode
  - offers "Alcatel 4980" Desktop Mobility services

* "Alcatel 4980" for desktop users with 4010, 4020, 4035
  - works in monoline or multiline mode
  - works with wireless Reflex sets (i.e. equipped with 4097 CBL Plugware)

* "Alcatel 4980" for desktop users with 4004
  - works only in monoline mode

For DECT handsets, the multiline possibilities of the set will be enhanced at PC level :
- due to poor ergonomics (lack of keys or display) the user will encounter some multiline service limitations when moving, especially for GAP handsets
- the user has full multiline possibilities on PC when on desktop

For desktop sets, the multiline possibilities (number of lines) of the set will be the same at PC level

<u>DECT handsets can have up to 12 MPK (Management Programmable Keys) with :</u>

- line keys
- service keys : Mail, Forwarding, Call Deflection, ...
- DSS/BLF and Filtering keys for Manager/Secretary

The main targets for desktop sets are :

- 4980 "PC Phone" associated to 4035 or 4020 for full multiline
- 4980 "PC Phone" associated to 4010 for multiline on two lines
- 4980 "PC Phone" associated to 4004 for monoline

The range of 2G phone sets (4003, 4011, 4012, 4023, 4034) will also be handled by Alcatel 4980 App

<u>Notes :</u>

1) In 4400 Call Handling, supervision and Manager/Secretary services are only possible on multiline sets
2) "Redial" and "Transfer" keys are more or less available on all DECT handsets

Recommended Profiles for 4074 handsets

Monoline ***Multiline***

The figure above shows the recommended profiles for 4074 DECT handsets in monoline and multiline modes.

<u>Notes :</u>

Call Diversion service is only available in 4980 R2

Manager/Secretary service is not available on DECT handsets.
At desktop level Manager or Secretary will use a 4035 wired set and the PC Telephony services of "4980" will be provided for the desktop set.
When moving with a DECT handset, the Manager or Secretary call will be made through the handset Repertory.

7.9 ALCATEL 4980 NETWORKING AND BACK-UP

Alcatel 4980 is a real **Network-enabled Application**.
This means that :

- services like GroupWare, Post-It sending or data sharing (Netmeeting services) are available private network wide in an Alcatel 4400 homogeneous network
- management is centralized on one Alcatel 4980 server
  This administration server is chosen by Administrator configuration
- back-up is available for 4980 servers distributed in network

Alcatel 4980 integrates Server back-up function offering hot standby service as Alcatel 4400.
One back-up server can replace any of the other 4980 server in network.

7.10 PERFORMANCE, CAPACITIES AND LIMITATIONS

One Alcatel 4980 "PC Phone" server App can handle up to 2000 PC clients for normal voice traffic (at 0.16 Erlang)

A maximum of one Alcatel 4980 "PC Phone" server App with back-up can be connected per 4400 node

Today, Alcatel 4980 networking services (GroupWare, Management, …) are limited to 5 Alcatel 4400 nodes in homogeneous private network.

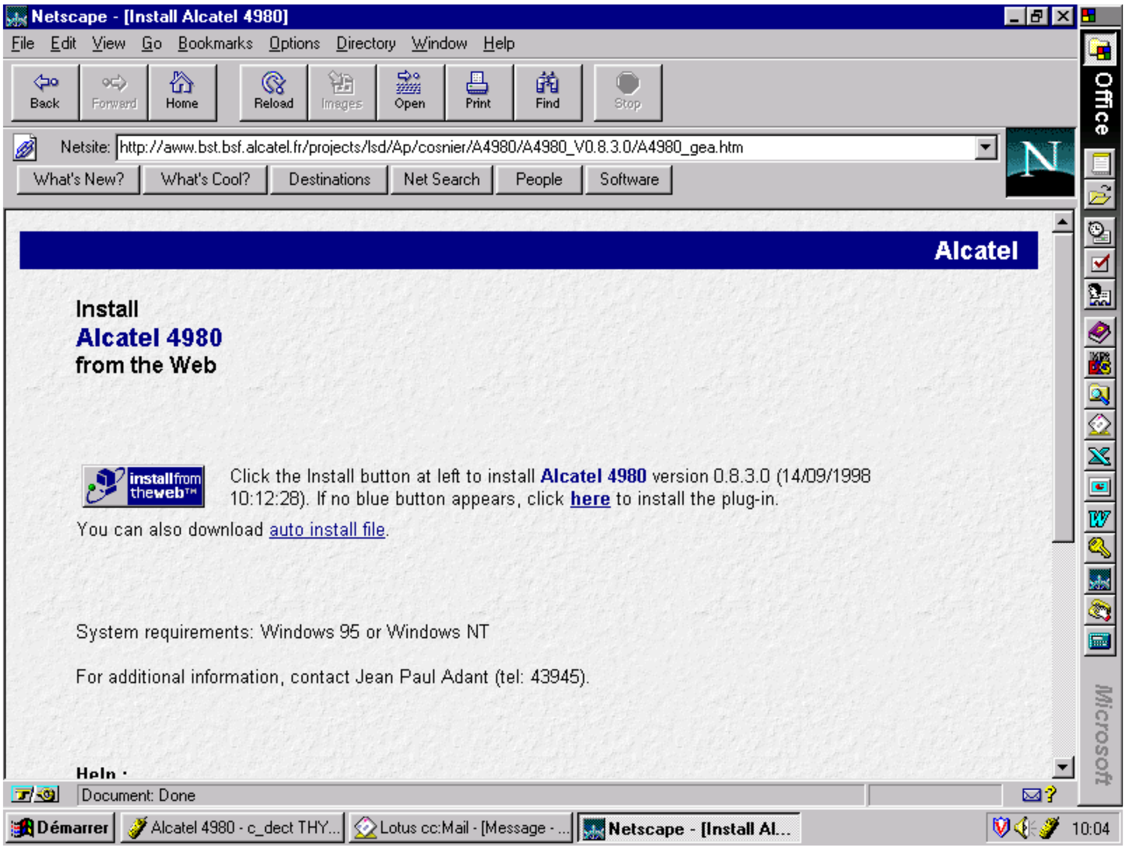
Today the 4980 server App runs on a dedicated NTserver PC
Nevertheless, this constraint should disappear in future after technical tests. The goal is to run simultaneously with other Alcatel or non Alcatel App on the same hardware server.

First of all, "Alcatel 4980" mobile services are limited to DECT handsets

Secondly, "Alcatel 4980" mobile services are limited to home node

Hence, multi-node mobility is limited to today's DECT handset services
(this should be solved in future evolution)

7.11 INSTALLATION AND UPDATE



Easy installation of 4980 Client

1. Server software is installed from CD-Rom

2. Client software is installed at server level from CD-Rom
allows configuration of the server software

3. Several possibilities to ease first time 4980 client installation :
– Centralized client installation by customer's software distribution tool like SMS (System Management Server)
– By set-up on file server (on a LAN)
– By download from 4980 WEB server of a .exe install file (5 Mbytes)

Automatic Client software update :

If any server software change needs a client update, the user will be informed by message on his PC.
Then an automatic download and installation will be performed.

"4980" server/client software update :

Each DU will have the possibility of downloading the "4980" software from Alcatel's Intranet WEB server : to be confirmed by PSO

7.12 ORDERING

<u>Two commercial products are available in Release 1</u>

Alcatel 4980 "PC Phone" offers all standard services for Telephony, Voice Mail, Customisation, App inter-working : TAPI Assisted Telephony, ...

Alcatel 4980 Advanced "PC Phone" offers additionally :

Netmeeting services

- Share & Collaborate
- Whiteboard
- Chat
- File transfer

Openness

- App inter-working : DDE interface

Notes :

- Alcatel 4980 R1 doesn't include OLE interface for App Inter-working
- Assisted Telephony will be available in both products allowing "Automatic Dialing" from 4000/4755 Directory or any third party Windows App (ex. : Schedule, Outlook, Excel, Word, ...)

Two possibilities for Alcatel 4980 "PC Phone" ordering :

"Alcatel 4980" software without server PC for all countries

- one CD-Rom with "4980" server & client software
- App server runs on Windows NT Workstation or Windows NT Server
- Client runs on Windows NT Workstation or Windows 95 (98)
- no Windows NT Client license necessary
- hardware : Pentium II at 266 Mhz, 64 M RAM

" Alcatel 4980" with server PC for : French, English, Spanish, German

- only Windows NT Workstation will be delivered
  - Pentium II at 300 Mhz, 128 M RAM
- 4980 software is installed and tested in Factory

7.13 ALCATEL 4980 SOFTWARE LICENSES

Alcatel's 4980 software licenses are based on :

- number of "Alcatel 4980" server access
  - price per seat dependent on number of server access
  - price is not dependent on phone set type

- level of services : "Alcatel 4980" or "Alcatel 4980 Advanced"
  - Alcatel 4980 : about 800 FF end user price per seat for 100 access (without server hardware)
  - Alcatel 4980 Advanced : about 1300 FF end user price per seat for 100 access (without server hardware)

8. NEW PC TELEPHONY SERVICES WITH ALCATEL 4980 R2

Alcatel 4980 Release 2 (synchronized with 4400 R3.1) offers hot new PC Telephony solutions for our customers.

First a brand new "**GroupWare service**" **unique** on the market, offering simultaneously new Data and Telephony real time communication features, specially designed to enhance workgroup productivity.

Secondly **Tele-worker solutions** offering a transparent access to 4980 enterprise communication services for nomadic workers and home workers equipped with multimedia PC (Voice over IP), GSM or 4052 Reflexes Extender plugware (MCK) for Reflexes Phone set.

Thirdly, **Universal Call by Name** which is a major improvement of 4980 "Call by Name" function offers now another new powerful service **unique** on the market. Call by Name search is now automatically performed is several Directories : Alcatel 4000/4755, Lotus Domino or LDAP Directories …

Fourthly, new **Desktop integration services**, to make life easier for any enterprise user when communicating : inter-working services with 4635 Visual Messenger or 4400 Unified Messaging for mails notification and consultation, with Lotus Notes for Directory and Workgroup services, with TAPI compliant Apps (Outlook,…) to make voice calls, with MAPI compliant Apps for E-mail (Outlook), with LDAP Directory servers, Calls planning, Post-It and Calls related Notes
Enhancements of the already existing "Call Logging" and "Personal Phone Book" functions brings to user desktop state of art services
New openness is available for developers with an API to access 4980 COM objects.

Last but not least, this new release now also offers the 4980 PC Telephony services for users equipped with ***analog phones***.

In addition to 4980 and 4980 Advanced we will have two new commercial products with specific licence and pricing :

- **4980 GroupWare** : 4980 + GroupWare
- **4980 Nomadic** : 4980 + GroupWare + Advanced services + Nomadic

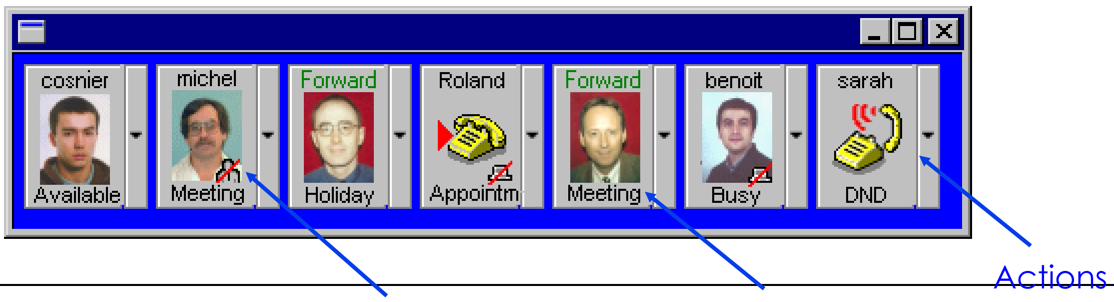
8.1 ALCATEL 4980 "GROUPWARE"

Market need for GroupWare

To work with efficiency in Office environment, we all need a PC application designed :

- **to show in real time my partners presence and availability**
- **for real time person to person communication (voice & data) with my partners**

The answer is Alcatel 4980 GroupWare

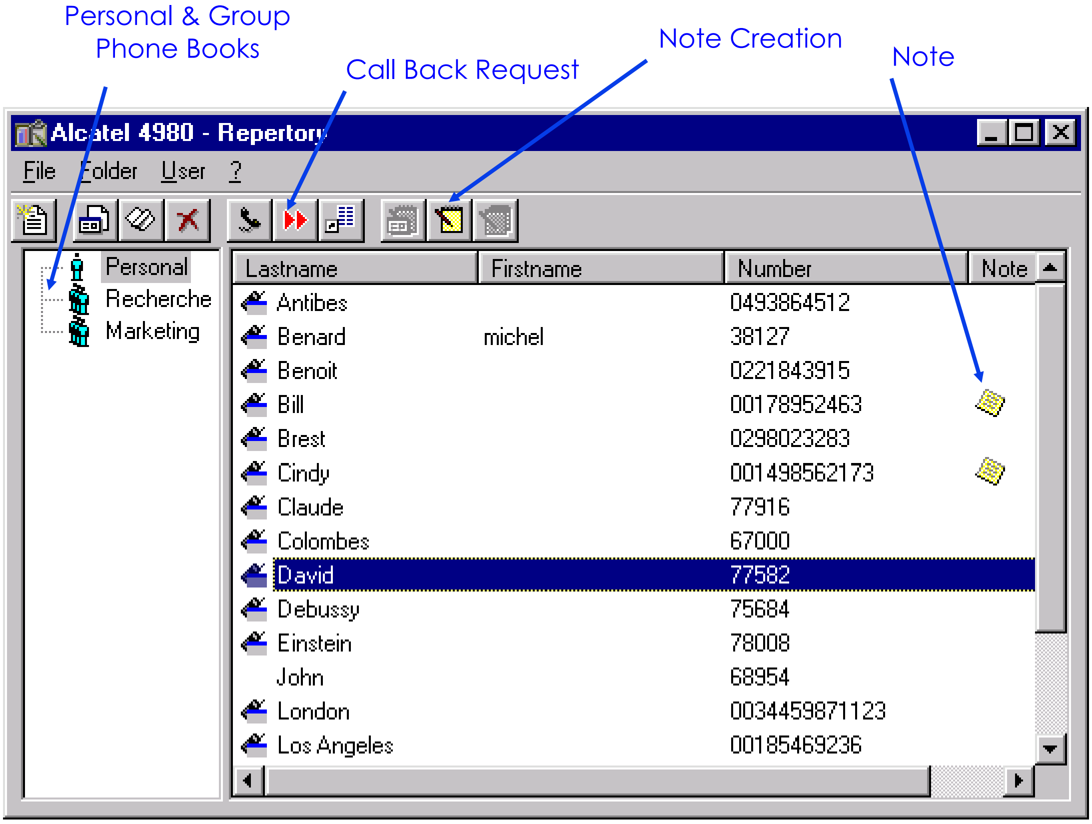


Alcatel 4980 "GroupWare" solution offers brand new workgroup services for 4980 users :

- **PC monitoring** shows in **real time** your partner availability :
  - partner absence/presence information (based on partner's activity on PC)
  - partner's Agenda state (from 4980 Agenda or Lotus Notes) shown by means of text below the photo/icons
  - data actions : Personal Agenda management, Post-it (by click on arrow on the right side of photo/icon in GroupWare toolbar)

- **Phone monitoring** shows also in **real time** your partner availability :
  - Partner Phone state (idle, ringing, busy, forwarded, DND) shown by means of icons
  - Voice actions : Call, Call Intercept, Call Back Request, Photo (by click on arrow on the right side of photo/icon in GroupWare toolbar)

- **Group Phone Book** :
  - one Phone Book is available per group
  - each group Phone Book is managed by all group members

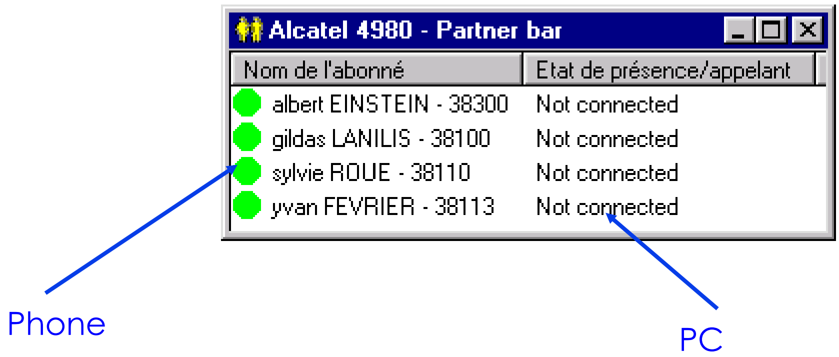


- **Agenda management from PC (4980 App)**
  - the user can manage his A4980 Personal Agenda with absence information (Meeting, Holiday, …) for workgroup display in GroupWare toolbar
  - otherwise the user can also manage his Personal Agenda in Lotus Notes
  - management by third party is also possible (to be confirmed) : the user can display or manage Agenda of another group member if authorized by him
    Typically, a Secretary can manage the Agenda of her Manager and display Agenda of workgroup members to have more information about their activities
  - Agenda management and consultation will be possible from 4980 App on PC

- **Post-It**
  - This service allows the 4980 user to select destination by click on arrow on the right side of photo/icon in GroupWare toolbar then to create a Post-It and to send it

- **User customization**
  - The user can change the layout of the GroupWare toolbar : order and place of toolbars, view in full or list form (see below), place of partners in toolbar (change by drag & drop)

Display of workgroup is possible in toolbar form (icons or photos with associated name and state) or in list form with small icons (like Windows Explorer)



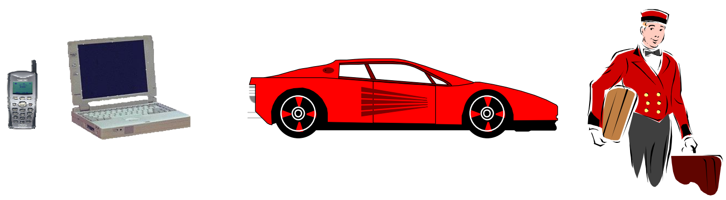
GroupWare services will be available private network wide (to be confirmed for Manager/Secretary)

<u>Provisioning level :</u>
- maximum number of 4980 users in GroupWare : 200
- maximum number of groups : 50
- maximum number of users per group : 20
- maximum number of groups per user : 3

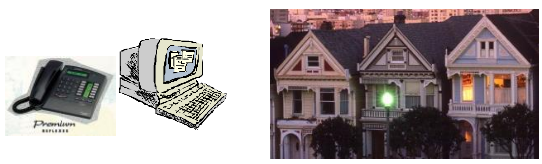
8.2 ALCATEL 4980 : TELEWORKERS SOLUTIONS

- **4980 with Multimedia PC (VoIP)**

- **4980 with GSM**
  (or analog set)



- **4980 with Reflexes phone set**
  (4052 Reflexes Extender)



Alcatel 4980 "Nomadic" is a solution offering transparent access to 4980 enterprise communication services for nomadic workers and home workers equipped with multimedia PC (Voice Over IP), GSM, analog phone set or home DECT :

- Main concept : while the remote worker is connected to his enterprise for data services he benefits transparently (simultaneously and for free) from the 4980 communication services (4980 signaling and VoIP)
- This solution provides a response for home workers who needs local mobility (home DECT) and full mobility (GSM)
- Nomadic base on GSM is fully complementary & integrated with "Ubiquity" offer (forwarding, one identity for outgoing calls, call by name, mailbox management)

At any moment (in idle mode or in conversation), the remote user can easily switch by softkey from VoIP mode to GSM mode. Reversal, the user can easily switch by softkey from GSM mode to VoIP mode in idle mode.

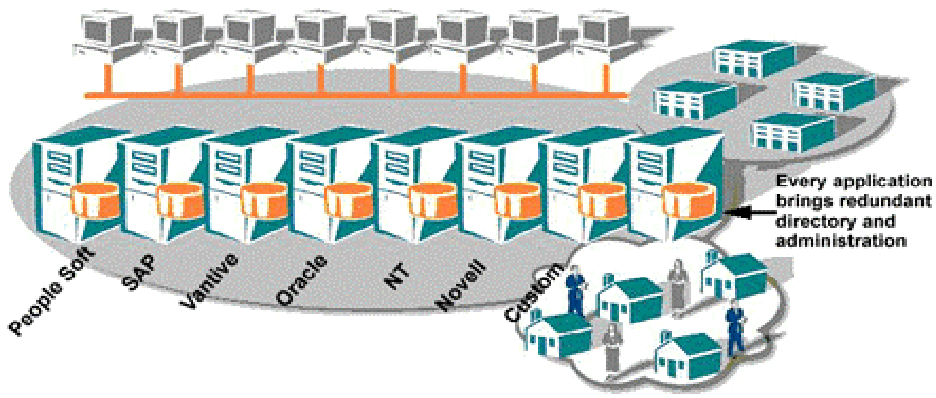
Another solution adapted for full home workers is to use **Alcatel 4980** (Standard, GroupWare or Advanced) for a Reflexes set connected via remote 4052 Reflexes Extender plugware.

Please refer to chapter "Tele-worker" for more detailed description.

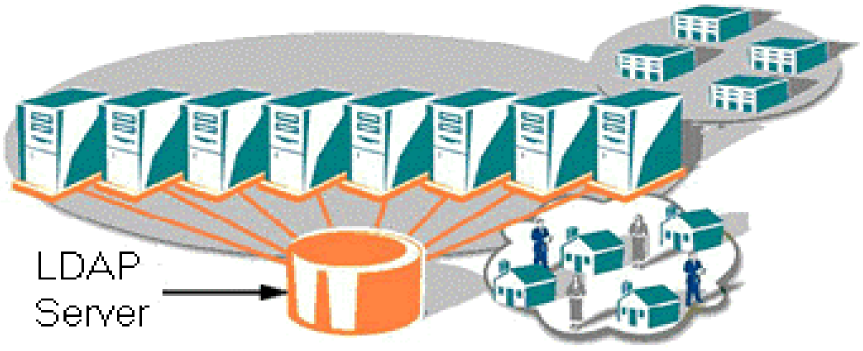
8.3 UNIVERSAL CALL BY NAME

8.3.1 LDAP Overview

Today, without LDAP Directory server



Tomorrow, with LDAP Directory server and LDAP-enabled Applications



LDAP v.3 (Lightweight Directory Access Protocol) is :

- The universal solution for Directory look up, Replication and developing LDAP-enabled Apps
- Designed to replace proprietary Directories : Novell, Microsoft, PBX, …
- Targeted at management applications and browser applications (Netscape/Communicator 4, Internet Explorer 4, Outlook 98)

- But today, no standard is available for :
  - Replication (LDUP)
  - Developing LDAP-enabled Applications (DEA, DEN)

DEA (Directory Enabled Application), DEN (Directory Enabled Network)

- RFC2251 from IETF (Internet Engineering Task Force)

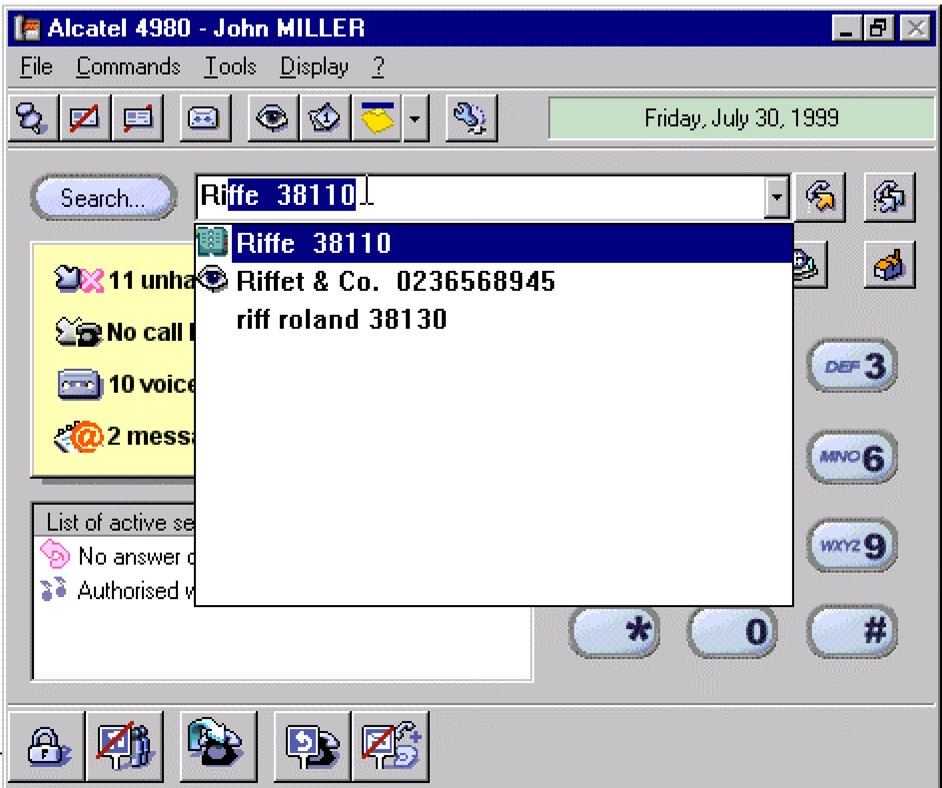
The main target for Universal Call by Name is Customers having their own Directories (LDAP, Domino Notes). For LDAP Directories, Netscape and Exchange are already tested with Alcatel 4980.

Alcatel 4980 **Universal Call by Name** is a unique feature allowing search in multiple Directory servers with automatic overflow in the following Directory types :

- 4980 Personal Phone Book
- 4980 Group Phone Books
- 4980/4400 Company Phone Book
- Alcatel 4000/4755 Company Directory
- Lotus Notes Domino Directory
- LDAP compliant Directory

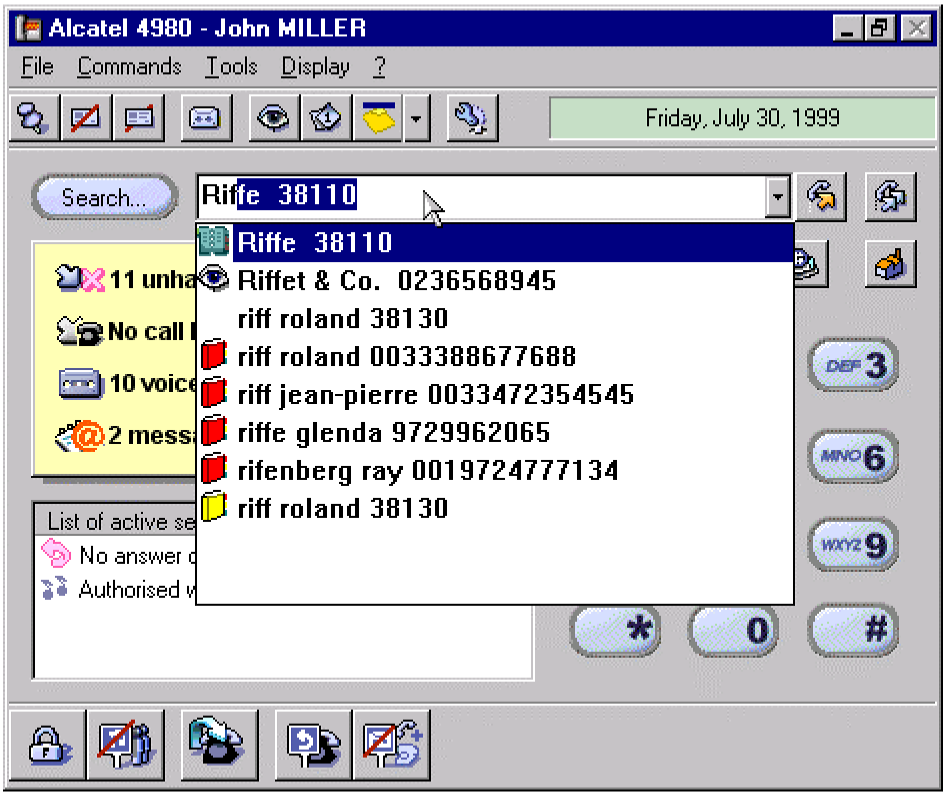
Basically, Call by Name search is always performed in the 4980 Phone Books. Overflow to other Directories (Alcatel, Lotus or LDAP) is performed on user choice by customization.

Lets see the following example were "Glenda Riffe" is the searched person. After entering 2 letters (Ri) in the 4980 Call by Name field we have directly the answers below :



As shown with/without the icons on the left side :
- the first answer comes from Personal Phone Book
- the second answer comes from Group Phone Book
- the third answer is from 4980/4400 Company Phone Book

Now, after entering 3 letters (Rif) in the 4980 Call by Name field we have directly the answers below



As shown with the icons on the left side, we have now additional answers coming from LDAP Directory (red icon) and Lotus Domino Directory (yellow icon)

<u>Complementary to Universal Call by Name, Alcatel 4980 offers also :</u>

- "Universal Name Display" in Visiting Card for any incoming/outgoing call
- "Universal Search function"
- "Universal Directory Pop-Up" information : see snapshot below

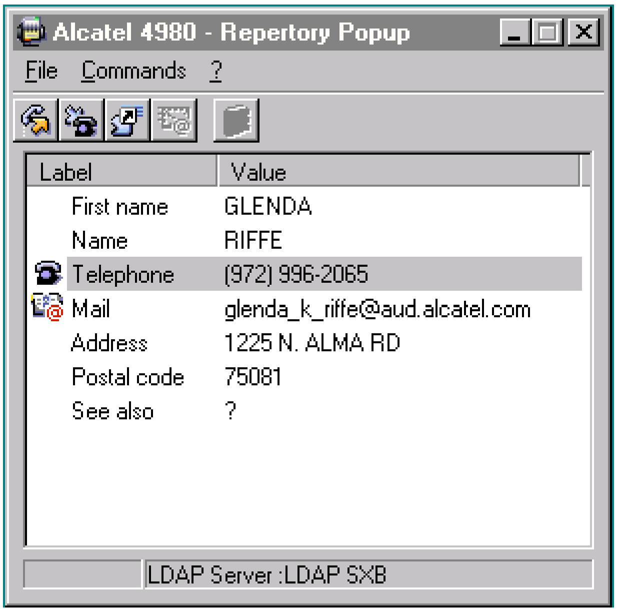
Pop-Up information is available in several cases :
- on incoming/outgoing call, with automatic/manual Pop-Up on user choice
- in Search function
- in Personal or Group Phone Books
- in Calls Logging
- for partner in GroupWare toolbar

Note that :

- the List of external Directory servers (Alcatel, Lotus, LDAP) is given by Administrator
- the searching order of the servers and the usage (Search function, Call by Name or Pop-Up) is given by user

In the following LDAP Directory Pop-Up snapshot example :

- the Telephone and E-mail field are signed (shown by associated icon) and therefore the user can directly activate them by mouse click in the Pop-Up window



8.4 DESKTOP INTEGRATION SERVICES

New Desktop integration services are now available for any enterprise user, they will really make his life easier when communicating.

8.4.1 Application Inter-working services : Messaging

Alcatel 4980 R2 offers new messaging Inter-working services with :

- ♦ Alcatel 4635 Visual Messenger
  - for voice&fax mail notification and consultation access directly from 4980 App
  - for Call Back sender from Visual Messenger with call handled by 4980 (OLE interface)
  - for Call Back sender from 4980 (MAPI interface)

  The inter-working services with VM are based on OLE or MAPI interface (if VM is integrated with Outlook)

  MAPI (Mail API) interface is at PC client level and is part of MS Windows OS
- ♦ Alcatel 4400 Unified Messaging
  - for voice, fax, E-mail and multimedia messages mail notification and consultation access directly from 4980 App

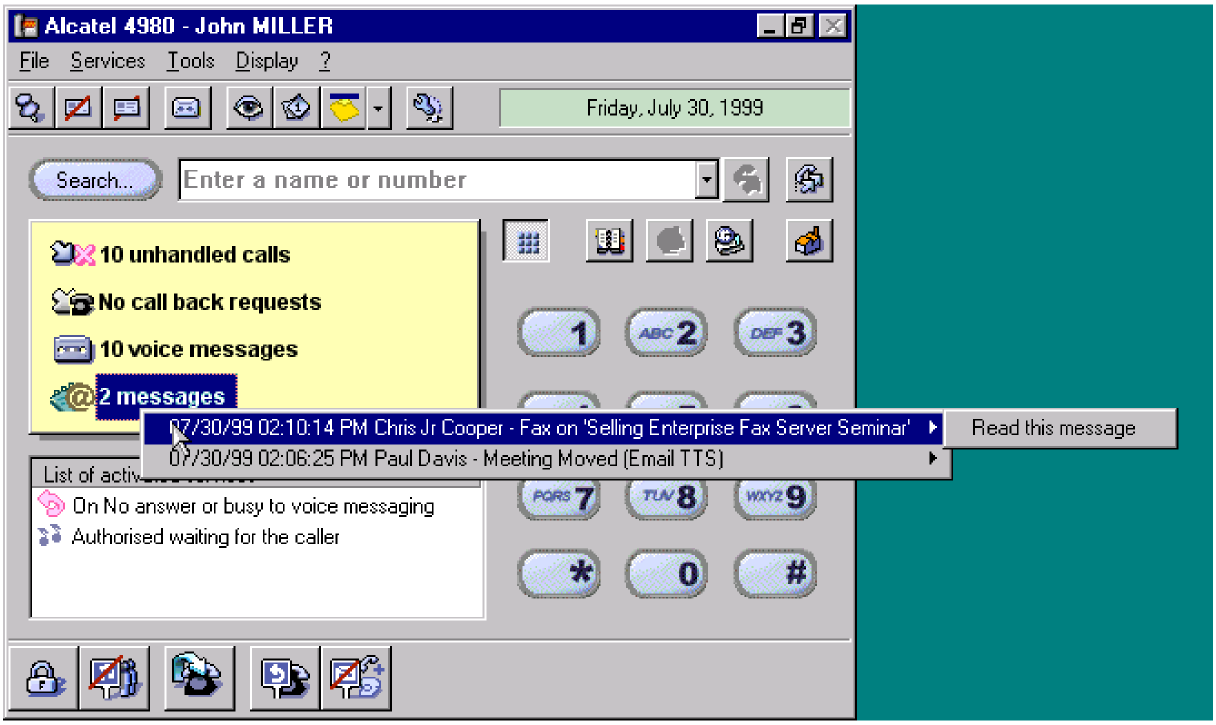
- for Call Back sender from 4980

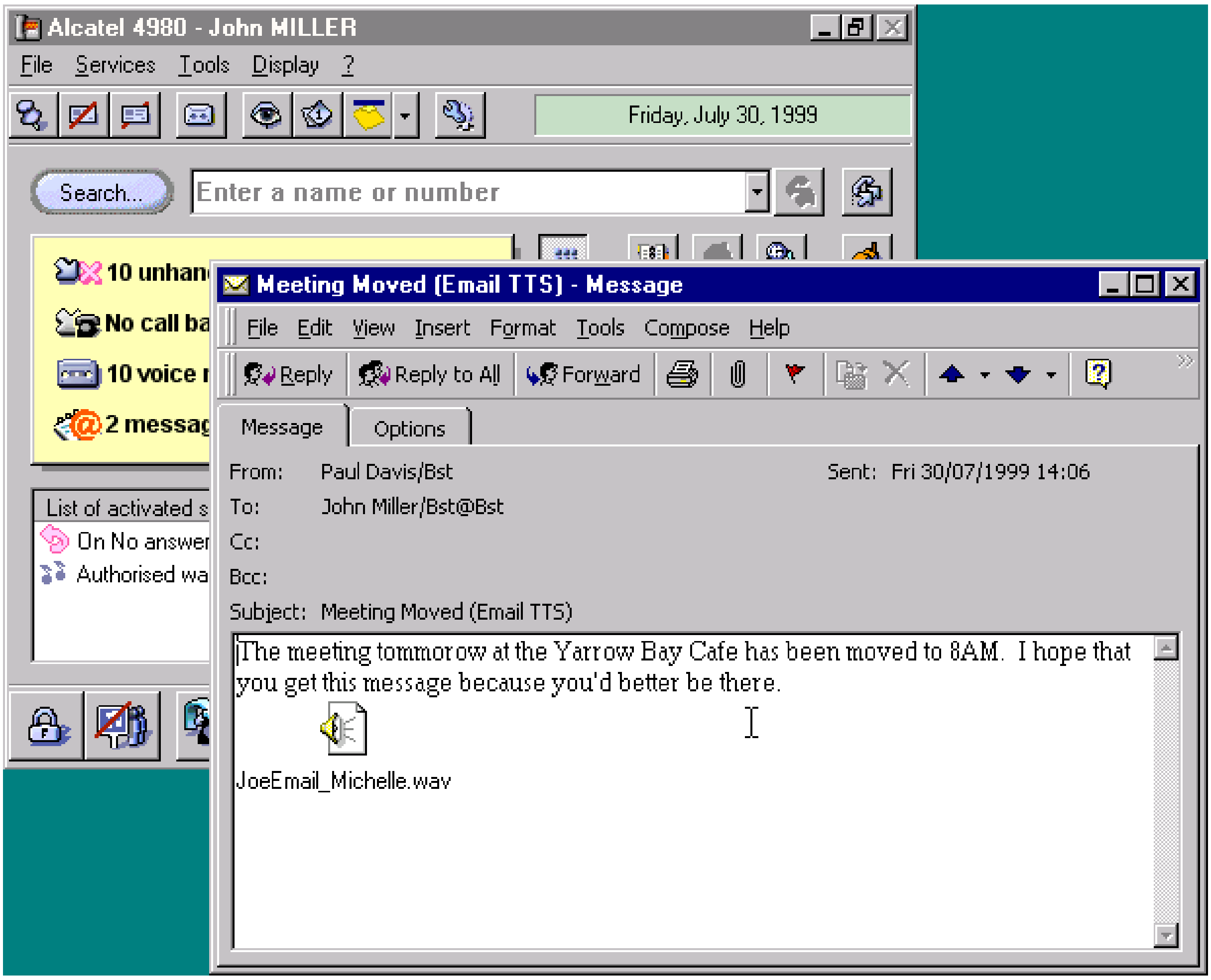
The inter-working services with UM are based on MAPI interface

- Exchange, Outlook, Notes, cc-mail, Netscape .. (MAPI compliant Apps)
  - for E-mail messages notification and consultation access directly from 4980 App
  - for Call Back sender from 4980

Note :
In case of MAPI interface, Call Back sender is available if the mail sender is known in the Personal or Group Phone Books

The two snapshots in the example below show the mail access with MAPI interface directly from Alcatel 4980





8.4.2 Inter-working services with Lotus Notes

The inter-working services with Lotus Notes are the following :

- E-mail notification and consultation access directly from 4980 App
  (as seen before in messaging inter-working services)
- 4980 Call by Name with search in Address Book of Notes
  (as seen before in Universal Call by Name services)
- Directory synchronization (4980/4400 Company Phone Book <> Address Book Lotus Domino). Personal Address Book of Notes is not synchronized with 4980 Phone Book.
- Call set-up from Notes (Address Book, Alcatel Address Book, Personal Address Book)
  This service is provided through TAPI Assisted Telephony function
- 4980 Directory Pop-up with information from Notes Address Book
  (as seen before in Universal Call by Name for LDAP Pop-Up service)
- Call Planning in Notes Agenda Base
- Agenda synchronization (Agenda 4980 <> Notes Agenda Base) for Agenda states and Call Planning

Except for E-mail notification/consultation access (MAPI) and Call set-up from Notes (TAPI Assisted Telephony), this inter-working is based on a **4980 server specific Lotus "Connector"**
In fact 4980 software offers a Lotus Domino "Connector" toolkit and documentation allowing to bring into operation the inter-working services with Lotus Notes. This must be done in accordance with the specific Lotus environment of our client and with Alcatel's PSO assistance (Partner Program service)

8.4.3 Application Inter-working services : TAPI, Directories

The inter-working services with TAPI Applications or Directories are the following :

- inter-working with TAPI compliant Apps (Outlook, Act, Twixtel, …) for "Make Call", "Release Call" and "Receive Call services". This allows typically for TAPI Directories or PIM to :
  - make a phone call automatically with 4980 App directly from any Windows TAPI App
  - release the phone call from 4980 client or TAPI App
  - have a Directory Pop-up synchronised with an incoming call on Alcatel 4980

- inter-working with **Alcatel 4000/4755** or **Lotus Domino** or **LDAP** Directory servers for Call by Name, name display on incoming/outgoing calls and Directory information display (Pop-Up) as seen before in Universal Call by Name

8.4.4 Alcatel 4980 Openness for developers

The Alcatel 4980 R2 offers now access to the 4980 services for developers :

- As seen before, inter-working with **Alcatel 4000/4755** or **Lotus Domino** or **LDAP** Directory servers is directly managed by Alcatel 4980. In order to provide the same services (see Universal Call by Name) with other Directory types, a universal Directory API is provided at client level.
  This API (DLL) allows the developer to create connectors to various Directory types by handling other specific protocols.

- Alcatel 4980 client is also an OLE server for OLE/COM compliant Apps by providing an API offering 4980 COM Objects for voice calls with basic Telephony services, Calls Logging services and Personal Phone Book services

8.5 ALCATEL 4980 FOR ANALOG PHONE SETS

Alcatel 4980 for analog phone sets offers new opportunities for sale as about 70% of installed base are analog sets

Alcatel 4980 is an Application bringing services and comfort for all types of sets.

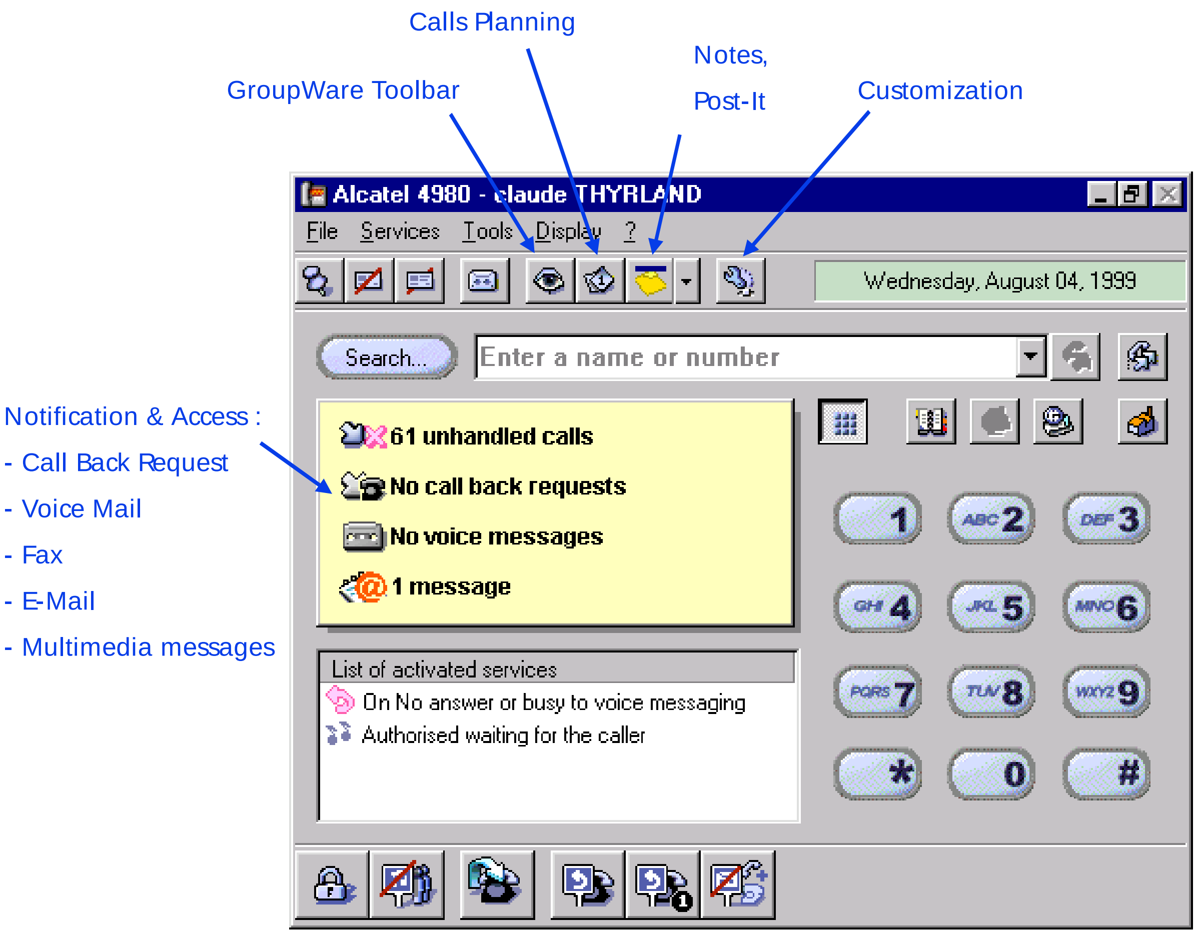
The list below shows examples of services provided for analog sets :
- 4980 GroupWare for analog sets
- Universal Call by Name (LDAP, Domino Notes, Alcatel 4755/4000)
- Calls Related Notes
- E-mail notification (Outlook, Notes, …)

Nevertheless it's important to notice that the Alcatel 4400 Telephony services for analog sets are unchanged. That means analog sets will work in monoline mode (no Manager/Secretary service is possible)

Finally, Alcatel 4980 is an Application at same price whatever the associated set/PC.

8.6 NEW PC TELEPHONY SERVICES

8.6.1 New look



The snapshot above shows the 4980 client with a new Look and Feel (new icons in 256 colours) of release 2 and the new services :

- In the toolbar : GroupWare display, Calls Planning, Calls Related Notes and Post-It, Customization keys
- In the messaging card : Call Back Requests - Voice messages, Fax messages, E-mail and Multimedia messages notification and access

8.6.2 New Telephony services

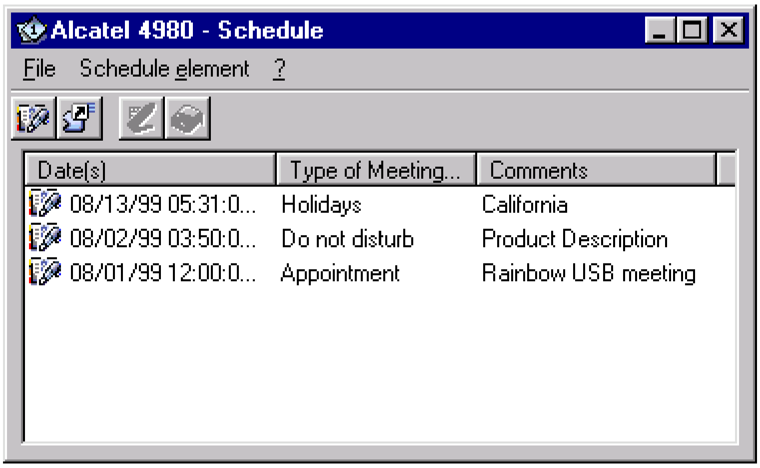
The new 4400 Telephony services available now on Alcatel 4980 R2 running with Alcatel 4400 R3.1 are :

- Call Back Request
- Call Diversion

Audio controls (loudspeaker volume, micro level) is available on 4980 softkey for multimedia PC (for Tele-worker with VoIP)

And as for the GroupWare toolbar seen before, it's now also possible to customize the Visiting Card with a Photo (at Administrator and User levels)

8.6.3 Calls planning



"Calls planning" is a new service of Alcatel 4980 R2
The 4980 user can pre-program calls and associated message or object in a calendar date/time form.
When the time is elapsed, a call notification is displayed in pop-up window on PC to remind the call object and to invite the user to set up the call.
In case of no user acknowledge at PC level, notification is also carried out at phone set level by ringing and associated message display. If the phone set is forwarded, notification is still done on the forwarded set and doesn't follow the forwarding.

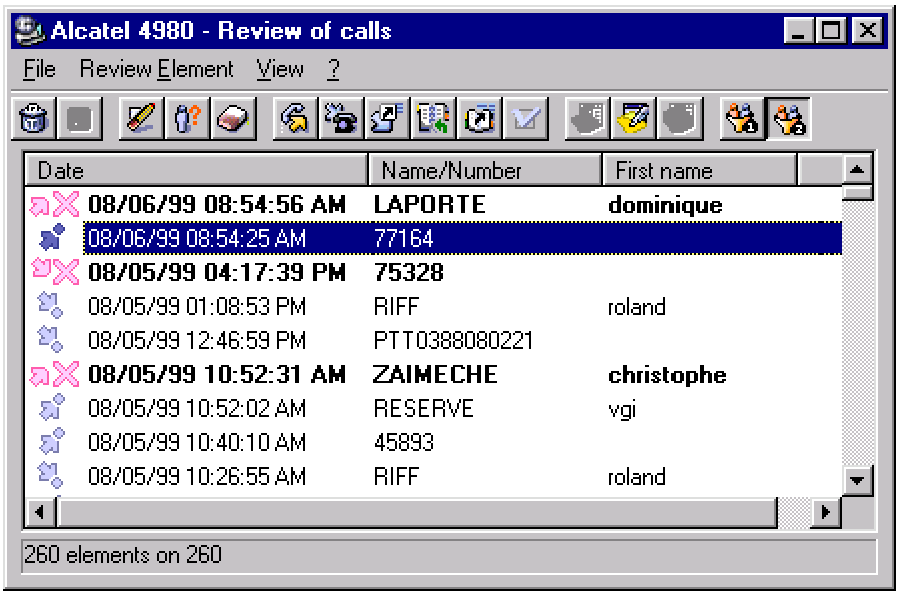
As seen before "Calls Planning" is also possible from Lotus Notes Agenda base (see : Inter-working services with Lotus Notes)

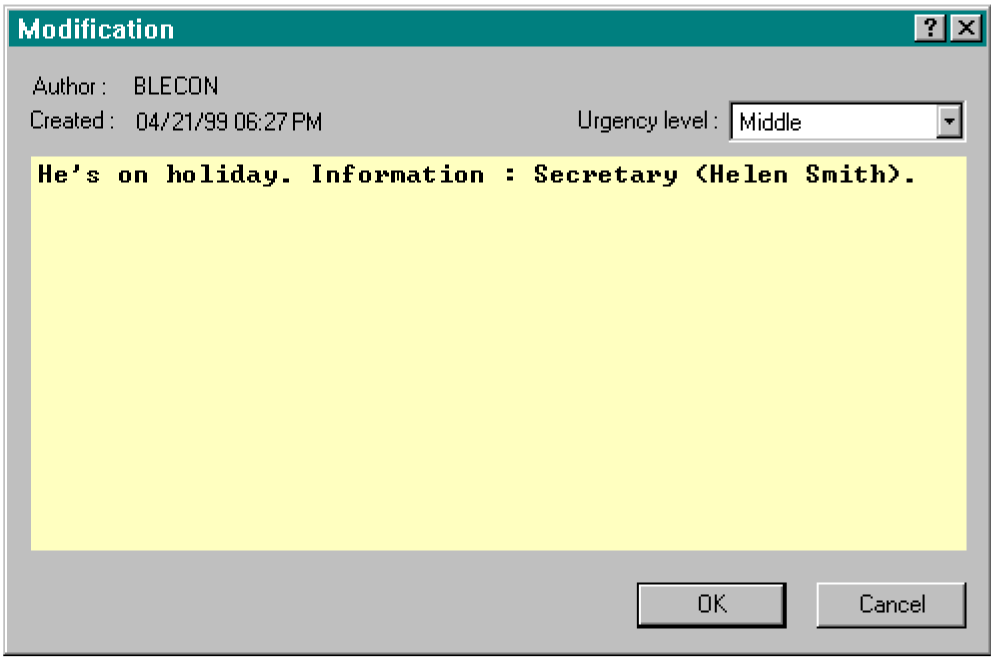
8.6.4 Calls Logging

This service becomes now "state of art" with :

- Callback Request, Copy/Paste services to Personal Phone Book
- Calls Related Notes associated to a call
- New types of calls display : business/private line, call with associated Note, overflowed call, call Intercept, voice mail call, ….
- Filtering **type of calls** and/or **specific caller/callee** for display (two profiles max)
  Note that filtering for storage will not be available for 4980 R2

- Data in text format for printing (Example : Excel)





This service allows the 4980 user :

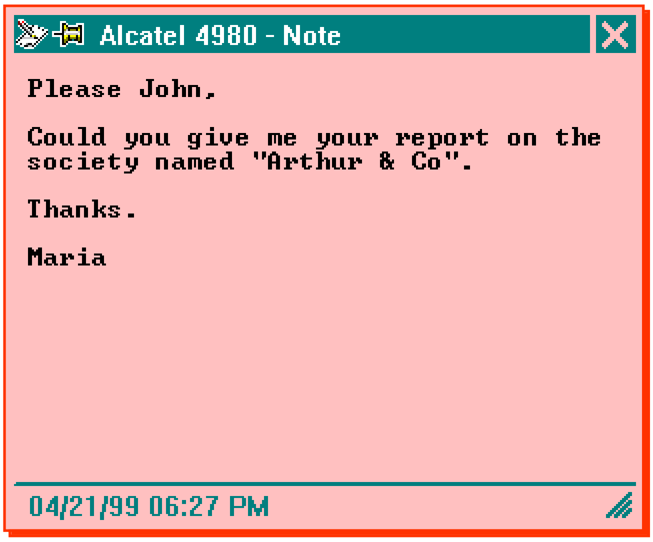
- to create a call related Note before, during or after a voice call with up to three urgency levels
- to associate personal notes to any voice call by store and retrieve in Call Logging tool
- to associate personal notes to any user by store and retrieve in Personal Phone Book tool

to have automatic or manual Note Pop-up on PC screen when calling or called by user with an associated Note. Automatic or manual Pop-up choice is done by user customization

Provisioning Level :

- maximum number of Calls Related Notes and Post-It per user : 500
- maximum number of characters in a Call Related Note and Post-It : 1000

8.6.5 Post-It



This new service is really designed for GroupWare real time communication, for example when :

- your partner is engaged in conversation on phone

- your partner's phone set is forwarded to voice mail (but he's present on PC)
- you need an urgent response for a question
- ….

This service offers to any 4980 user to :

- ♦ create a Post-It with up to three urgency levels
- ♦ stick a Post-It on his own PC Windows desktop for Reminder

Additionally, in case of GroupWare the 4980 user can :

- ♦ send and display in real time Post-It on PC screen of his partner
  - Sending is possible at any moment (even if destination user is not logged in 4980)
  - Display is done when destination user is logged in 4980
- ♦ Individual Post-it sending is possible by drag&drop : to be confirmed
- ♦ Post-It broadcast to all group members is also available
- ♦ send Post-It in E-mail form is also possible (if E-mail address and MAPI connection are available)

8.6.6 Personal and Group Phone Books enhancements

Compared to Alcatel 4980 R1, the Personal and Group Phone Books enhancements are :

- ♦ Print Phone Books
- ♦ Data import/export in text format (Example : Excel)
  Another proprietary format (arp) is more designed to save Phone Books and allows to keep the Phone Book hierarchy and the shortcuts to the speed dialling keys
- ♦ Calls Related Notes associated to a user (to be confirmed for Group Phone Book)
- ♦ Folders in Personal/Group Phone Book, allowing the user to manage Phone Book information by function or project
- ♦ Copy/Paste from Call Logging, Search window, Pop-up Information window
- ♦ Copy/Paste by drag&drop between personal Phone Books

8.6.7 Security

- ♦ automatic 4980 App locking after n log fails
- ♦ hide correspondent Name/number in visiting Card but also in Calls list

8.7 TABLE WITH SERVICES CONTENT IN ALCATEL 4980 COMMERCIAL PRODUCT

Services	Basic	GroupWare	Advanced
Inter-working : V Messenger, Unified Messaging	☺	☺	☺
Inter-working : TAPI, MAPI	☺	☺	☺
Inter-working : LDAP	☺	☺	☺
Calls Related Notes, Post-It	☺	☺	☺
Calls Planning	☺	☺	☺
Call Back, Call Diversion	☺	☺	☺
Groupware : Data Monitoring		☺	☺
Groupware : Phone Monitoring		☺	☺
Group Phone Book		☺	☺
Agenda for absence info		☺	☺
Inter-working : Groupware Lotus Notes		☺	☺
Netmeeting services		☺	☺
DDE, OLE server API			☺
Nomadic VoIP			☺
Nomadic GSM			☺
Nomadic Analog set			☺

8.8 AVAILABILITY AND COMPATIBILITY WITH 4400 R3

Alcatel 4980 R2 is available with 4400 R3.1
- Beta Test Objective : October 99
- Technical Release objective : Q1 2000

When 4980 R2 runs with 4400 R3, the following services will not be available :
- 4980 for Analog phone sets
- All 4980 Nomadic services
- Telephony service : Call Diversion

Licence compatibility
When 4980 R2 runs with 4400 R3 : the GroupWare services are available with 4980 Advanced license

8.9 ALCATEL 4980 COMMERCIAL PROMOTION

Alcatel 4980 commercial promotion will be launched by the following means :
- each 4400 R3 or R3.1 will offer three 4980 user licenses (GroupWare/Advanced) for free
- a commercial operation will be done by the DU in order to send the 4980 CD-Rom full evaluation version to all 4400 R3 customers

- an easy customer made installation will be offered

8.10 OPTIMIZATION : KILLING POINTS

PC server **platform optimization** is possible in several ways :

1. Alcatel 4980 server software can run on Win NT Workstation (no need of MS NT Server licence)

2. Alcatel 4980 server platform can be shared with other software :
   - Alcatel 4635 Visual Messenger, Alcatel 4740 or customer software : to be confirmed, tests are in progress

3. For private network configuration with small nodes, PC server platform can run several 4980 server software (1 per node)

4. There's no need of NT server back-up function : Alcatel 4980 offers server back-up function

5. TCO is low through :
   - ♦ Centralized (with the customer's software tele-distribution tools) or Automatic client update
   - ♦ Alcatel 4980 is an Auto learning App
   - ♦ Alcatel 4980 HelpDesk will be the same as for Alcatel 4400 (managed by Telecom Manager)

6. Alcatel 4980 offers one architecture for all needs (whatever the phone set, In Site/Off Site)

8.11 EASY INSTALLATION AND UPDATE

Alcatel 4980 server installation

- ♦ NT server installation is normally done by customer's IT Manager
- ♦ Alcatel 4980 App installation is normally done by Telecom Manager

Alcatel 4980 Client installation is normally done as for other customer clients

- ♦ With centralized client software installation tool like SMS (System Management Server) Alcatel 4980 is already tested with SMS process

Else, three possibilities are available by Alcatel 4980 to ease first time 4980 Client software installation

1. Install from network file server set-up
2. Download from 4980 WEB server and install
3. Centralized client installation by administrator : to be confirmed
   - Shell Kick-Start Windows NT4 (Resource kit NT4, Microsoft, for free)

In case of client or server software evolution, **automatic 4980 Client update** is provided.
Else like for installation, the customer's software teledistribution tool can also be used for update.

8.12 QUOTATION AND ORDERING WITH ACTIS

The following basic rules are in use for "ACTIS" tool :

- One 4980 server license is necessary per node
- Back-up server needs no license
- For 4400 R3.1, three licenses are available :
  - 4980 Basic
  - 4980 GroupWare
  - 4980 Advanced

  In order to ease the ordering, Actis tool will present the main services of each license ..

- as shown in the table above, licenses are organized on Russian Dolls model :
    - Advanced license offers also all services of GroupWare and Basic licenses
    - GroupWare offers also all services of Basic license

pricing

For each licence type (Basic, GroupWare, Advanced), the following pricing model is used for 4980 WPL :

First the customer needs to buy a package including 4980 server and 10 seats licences

- additional user licences are available per packages of 10 seats
- the price of licence is unique whatever the terminal associated to 4980 client software : Analog, Reflexes set, DECT handset, multimedia PC, GSM
- for Nomadic services :
  - no extra server is needed
  - no extra licence is needed when the user is in the Company
  - VoIP gateway for IP Networking can be used with the same pricing

8.13 NEW BUSINESS POSITIONING

4980 + any voice device

Now Alcatel 4980 is a powerful communication tool for sales :

- on any desktop :
  - On Site : UA, Z, DECT
  - Off Site : GSM, multimedia PC
- on any system : small, medium or large

Alcatel 4980 has strong killing points for sales :

- unique on the market
  - architecture, services for user productivity, for any desktop/system
- attractive market price, easy pricing
  - 4980 is an Application at same price whatever the associated set/PC
- adapted for sales on installed base (Z)
- promotion package

8.14 ALCATEL 4980 FAQ

Main Questions of a IT manager and Alcatel Answers
Alcatel 4980 PC Phone

25/06/99

By selling applications, at a certain stage of the project , the IT manager will be part of the buying center. Please find attached a list of main questions IT manager raise, when it comes to integrate these applications into their network.

	Alcatel Answers
• **How will the Application affect the LAN traffic?** .	• Only a low signaling traffic of about 5 Kbit/s in average is required between 4980 server and 100 Alcatel 4980 clients. A technical document with complete information about 4980 traffic tests is available • For clients with real time GroupWare monitoring, this traffic value will be higher (tests are in progress). • TCP/IP socket mode connections are used for the clients. • For Tele-workers with multimedia PC, additional low voice traffic (Voice over IP) is necessary but generally on different LAN segment. Thanks to H323 stack including G723.1 voice compression with silence suppression, VoIP traffic is about 14 Kbit/s.
• **How will this application affect day to day administration? (Cost/ help desk/ support people, training of support people…)**	• Alcatel 4980 App administration is normally done by Alcatel 4400 **Telecom administrator** • First, Alcatel 4980 user is simply created by adding a parameter when creating a 4400 telephone user with 4740 or 4755 management tools. • After 4980 server installation and configuration (user license, user password), day to day administration is low : - The 4980 NT server is not a domain server and needs no MS NT client/server licenses (CAL) - Alcatel 4980 client needs no user configuration at first installation - Alcatel 4980 is an **auto-learning** App : use of telephone services are available without training, like any office App training is only necessary for advanced usage. Alcatel 4980 is a unique PC Telephony App offering **softkeys** for the best of ergonomics - **permanence of telephone service** is available as the user can always phone on his set without Alcatel 4980 App - **Alcatel 4980 client update is automatic** - no specific Alcatel 4980 management is required for user moving • Alcatel 4980 Helpdesk or technical support is the Alcatel 4400 Helpdesk/technical support This avoid the need of dedicated help desk or technical support for end users. For those reasons we can assume that **Alcatel 4980 TCO is low**.
• **How complex is it to install the Alcatel 4980 in a**	• Alcatel 4980 PC Telephony server runs on NT4 Workstation or NT Server platform and integrates seamlessly into the existing customer network over a TCP/IP infrastructure. The 4980 NT server is not a domain server and needs no MS NT client/server licenses (CAL).

client/server environment? • **How complex is it to install the Alcatel 4980 in a client/server environment?**	• Generally, NT platform installation is done by customer's IT administrator. • Then **installation of 4980 server App is done by Telecom administrator, like for 4400**. First, 4980 server software installation done with a wizard is easy and needs to enter few parameters. Then, App configuration needs few 4400 knowledge. Nevertheless, this installation requires technical people who have attended an Alcatel 4980 training course. Alcatel is able to achieve this installation of course. **Alcatel Answers** • Interconnection with Alcatel 4400 PBX platform is performed via TCP/IP Ethernet LAN segment. • Alcatel 4980 PC Telephony server supports all network topologies: single site, multi-site, LAN/WAN, homogeneous or heterogeneous numbering systems, distributed servers; to satisfy all customer needs. • **Alcatel 4980 App server can also be pre-installed (BTCO) and configured (full BTCO) in Alcatel Factory**
• **What are the methods to install this application on the client stations?**	• Several possibilities to ease first time 4980 client installation : - Centralized client installation by **customer's software distribution tool** like SMS (System Management Server) - By set-up on file server (on a LAN) - By download from 4980 WEB server of a .exe install file (5 Mbytes) So no need to go on each PC to install the add-on software. **Alcatel 4980 client update is automatic**
• **Do you need to be NT administrator to install an 4980 client ?**	No, except for TAPI installation (Windows Registry, Windows Repertory)
• **What about the compatibility of this application with Windows 2000?** • **How does it influence other standard clients (GroupWare, Lotus/ Internet Explorer, SAP , ...)**	• **Alcatel 4980 PC Phone is 2000 year compliant**. We support Win NT 4.0, Exchange 5.5, Outlook 97, Outlook 98, Domino/Notes 4.5 & 4.6, Domino/Notes 5.0 (test in progress), I.E 4.01, Netscape 4.x. It will be of course fully compliant with Win 2000, Outlook 2000, I.E 5.0, Exchange 6.0 & Domino/Notes 6.0 when available. • There's no influence to other standard clients
• **What is the value of 4980 GroupWare and Agenda compared to existing softwares like Outlook and Notes ?** • **What is the screen refresh rate of 4980 GroupWare toolbar or list ?**	• **Alcatel 4980 GroupWare offers a brand new service compared to Notes or Outlook** • **Alcatel 4980 is a unique product on market offering real time GroupWare for high efficiency at work.** It shows in real time the presence and availability of your partners. Availability is based on PC monitoring (partner activity and Agenda) and simultaneous Phone monitoring. Actions by mouse click in GroupWare toolbar offer then **real time** communication with your partners. • Of course, 4980 GroupWare is integrated with Agenda Notes and will be integrated with Agenda Outlook in next release. • Screen refresh of 4980 GroupWare toolbar or list is done every 3 seconds.
	• Alcatel is updating its All In One catalogue of services to be able to provide a full range of technical services dedicated to 4980. This will be fully defined

• **What is the detail of your maintenance and support program for this application?**	in xx : to be defined These services will cover the following items: - Pre-installation - Installation - Post-installation support supports - Training
• **What's about reliability ?**	**Alcatel Answers** - Server Back-up function is provided for free by Alcatel 4980 App (one back-up for n servers) This is important for **cost optimization and low TCO** (no need of expensive NT back-up server)
• **What's about security ?**	- Security function is provided by 4980 App (password encryption, blocking after n login fails, control of password change, …) This solution offers user mobility controlled by 4980 App and openness to other platforms Nevertheless NT login can be used on user choice (automatic 4980 login with NT password) For Tele-workers, the existing RAS (Remote Access Server) and Firewall for data access will be used.
• **What are the PC hardware requirements ?**	• PC Client - OS : W95, W98 or NT4 - P 120 minimum - 32 M RAM for W95 or W98, 64 M RAM for NT4 - Disk space : 20 Mbytes • PC Server - OS : NT4 (Workstation or Server) - P II 300 MHz minimum - 128 M RAM - Disk space : 100 Mbytes (50 clients) to 550 Mbytes (2000 clients)
• **Do you need a dedicated server ?**	• No, Alcatel 4980 server App can run with 4635 Visual Messenger, Alcatel 4740 or any customer software provided that they're no compatibility problems and the **real time performance** are not affected. Hence tests must be done … Therefore, Alcatel 4980 server needs the following resources for 2000 clients : - CPU : about 10% in average - 4980 process : about 7,5 % in average - Memory : 38 M RAM - Disk space : 100 Mbytes (50 clients) to 550 Mbytes (2000 clients) Those values are for : P-II, 300 MHz, 128 M RAM, NT4.0 Workstation SP3 Software size : 3 M Complete information about 4980 resources tests is available in a technical document.

	This is important for **cost optimization and low TCO**
• **What are the resource needs for 4980 client ?**	Alcatel 4980 client needs the following resources : - CPU : less than 10% in average - Memory : about 2 M RAM (depends on Phone Books, Call Logging) - Disk space : 20 M Software size : 2 M
• **What are your development tools ?**	- C++ for server - Visual C++ for client This ensure high performance of 4980 real time App
• **What's about WEB and Java technology ?**	- Today, a windows client offers better performance and ergonomics. - Later on when more mature, WEB and Java technology will also be provided.
• **Do you provide API or SDK for developers ?**	• Today, Alcatel 4980 R2 client offers OLE server API with COM objects for : - Basic 4980 telephone services - Personal Phone Book services - Call Logging services • Later on in a future release, a SDK with COM objects to access all 4980 client services will be provided This will allow to develop specific client fully adapted for customer needs

9. ALCATEL'S TELE-WORKER SOLUTIONS

9.1 INTRODUCTION

Our society has changed from being blue collar to being white collar. This may change our attitude to work. It is not possible to deliver a steel mill to a cottage each day for the worker to use, but it is possible to deliver electronic paperwork to the cottage everyday. With the array of telecommunications products and services becoming available at ever reducing costs, the burden of change must be towards home-centering our lives or nomadic working manner. This evolution, known as 'teleworking', 'telecommuting', 'electronic homework', 'telecottage', is becoming a reality. In the same way law is changing to face this evolution and to take in account the change of the working place.

The result is that more and more companies are offering employees the option of working away from headquarters at home, in a regional telecommuting center, from a branch office or while not from a hotel room.

9.1.1 Tele-worker profiles

Who are the tele-workers ?

- ⇨ Office workers who spend one or more days a week working from home
- ⇨ Call Center Agents, Attendants or Secretaries working at home or in Tele-work center
- ⇨ Technicians in service jobs in the field with laptops who want to order parts or to write and transmit reports
- ⇨ Sales people who work from their laptops in the field or at home and occasionally come in their Company offices
- ⇨ Executives who communicate in a variety of electronic ways while travelling or at home

The two basic profiles of the tele-workers are :

- ⇨ The home-worker : which mainly works at home
- ⇨ The nomadic-worker : which mainly is on the roads

Furthermore, in most of case the Tele-Worker is also time partly an In Site Enterprise worker

9.1.2 What are the motivations and benefits?

For the company

- ⇨ Flexibility in accessing human resources (location and time)
- ⇨ Costs savings (office costs, overheads, ...)
- ⇨ Motivated employees
- ⇨ Better productivity (from 10% to 40% reported)
- ⇨ Tax incentives (in some countries)

For the employee

- ⇨ Save commuting time and costs
- ⇨ Less stress
- ⇨ More flexibility (time and location)
- ⇨ Better overall quality of life

For the Community

⇨ Improve road traffic conditions
⇨ Reduce unproductive hours
⇨ Reduce pollution
⇨ Impact on unemployment (professional mobility)

9.1.3 What are the expectations?

For the company

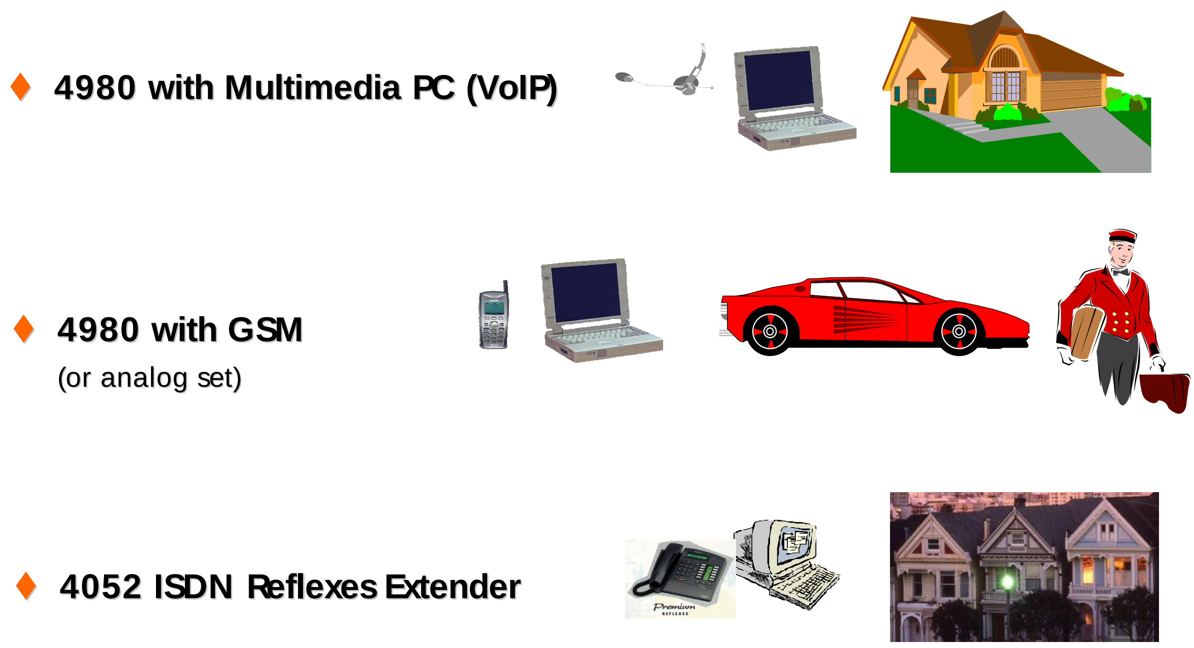
⇨ Quality of Service
⇨ Transparent operation for Calling Customers (voice quality, response time, Tele-worker empowered with Company's information base)
⇨ Tele-worker must have possibility to benefit from same facilities as locally by accessing all enterprise Applications (Telephone, Email, Directory, …)
⇨ Controlled/mastered Telecom costs
⇨ Telecom costs in Company's charging App
⇨ Optimised outgoing calls by Call-back procedure or break-in
⇨ Security by DISA or Call-back procedure

For the employee

⇨ A remotely transparent access to his Enterprise' for voice and data services
⇨ transparently accessible by internal (workgroup, secretary, boss, ...) or external callers (customers)
⇨ All costs supported by the Company (equipment, telecom costs,...)

9.2 ALCATEL'S TELE-WORKER SOLUTIONS

There are three basic tele-workers solutions :



As shown in the picture above, there are three basic tele-workers solutions :

♦ Alcatel 4980 with multimedia PC (Voice Over IP)

♦ Alcatel 4980 with GSM

♦ Alcatel 4052 ISDN Reflexes Extender
(Note that this solution works with or without Alcatel 4980)

Alcatel 4980 “Nomadic”
Alcatel 4980 “Nomadic” is a solution offering transparent access to 4980 enterprise communication services for nomadic workers and home workers equipped with multimedia PC (Voice Over IP), GSM, analog phone set or home DECT :

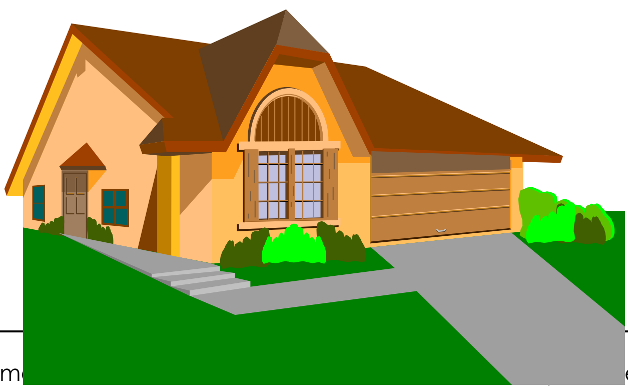
- Main concept : while the remote worker is connected to his enterprise for data services he benefits on the same single line, transparently (simultaneously and for free) from the 4980 communication services (4980 signaling and VoIP)
  Two TCP/IP sessions are established, one for 4980 signalling (about 50 bit/s) and another in H323 mode for VoIP (about 15 Kbit/s). We can notice than required data rate is low thanks to silence suppression
- This solution provides also a response for home workers who needs local mobility (home DECT) and full mobility (GSM)
- At any moment (in idle mode or in conversation with external calls), the remote user can easily switch by softkey from VoIP mode to GSM mode. Reversal, the user can easily switch by softkey from GSM mode to VoIP mode in idle mode.

Those Alcatel “Nomadic” solutions are available with Alcatel 4980 “Advanced” licence.

Another solution adapted for full home workers is to use Alcatel 4980 (Standard, GroupWare or Advanced) for a Reflexes set connected via remote 4052 Reflexes Extender plugware described below.

Alcatel 4051/52 ISDN Reflexes Extenders
Alcatel, by its new Reflexes Extenders family, will give the opportunity to remote workers to be connected to the corporate enterprise as easily as they were in their office. Remote workers use the same Reflexes deskset they use in their office, with transparent access to all the PBX functionality. In the same way, by using the same Reflexes Extender, Remote workers will have also data/LAN capability.

9.3 ALCATEL 4980 WITH MULTIMEDIA PC

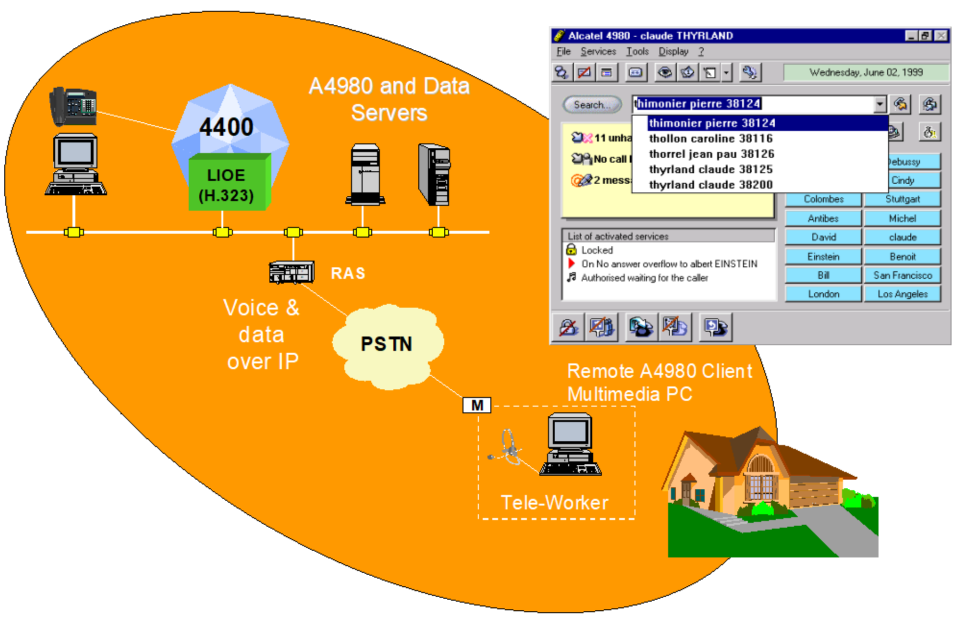


End User Service

With this solution, the Tele-worker has access to all 4980 services except for Netmeeting App sharing due to bad performance of the modem link (about 33 Kbit/s)

This solution is typically for Office workers or Sales people and offers :

- voice calls for free (VoIP) during Tele-Worker's data connection
- possibility to phone directly with headset on PC
- good voice quality
- mobility without GSM handset
- all powerful 4980 services :
  - GroupWare, Call by Name, Voice Mail, Call Logging, ...



Architecture

- Off Site : the Alcatel 4980 client runs on multimedia PC
- In Site : the 4980 server is connected to Enterprise LAN with other legacy data servers
- the workplace in the Company the tele-worker works transparently in/out of Company. This workplace can be virtual
- the 4400 is equipped with the VoIP Gateway
- the RAS (Remote Access Server) is necessary for remote access and security

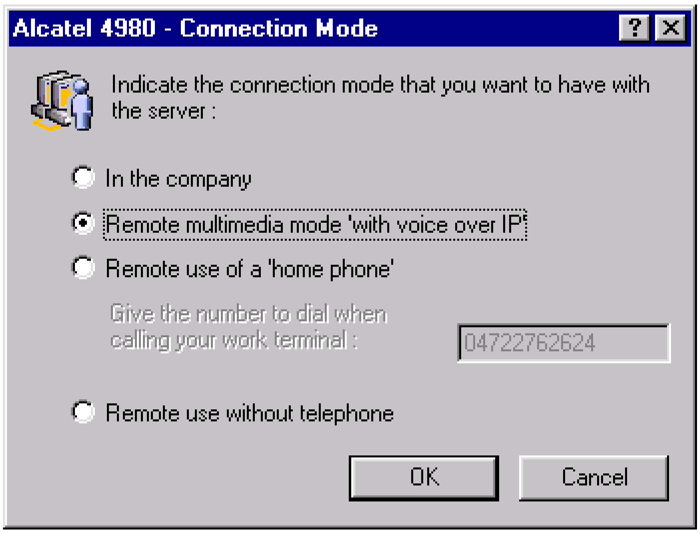
How does it work ?

- a data connection is established to the Company via the RAS
- Alcatel 4980 signaling and VoIP are transported on the same line
- we have a modem connection (Analog or ISDN modem for two B channels)
- we will have two IP sessions :
  - one for signaling to the 4980 server
  - another for voice to the VoIP Gateway

While connected for data needs, the nomadic user has the same 4980 services as if he was in his Enterprise Office :

- 4400 Call Handling is done for his local phone set in Enterprise

- after remote user login (see figure below), the phone set in Enterprise becomes unusable and voice is routed to remote location thru VoIP gateway



- workplace (phone set & PC) in Enterprise can be virtual
- the nomadic worker can transparently work in or out of Enterprise

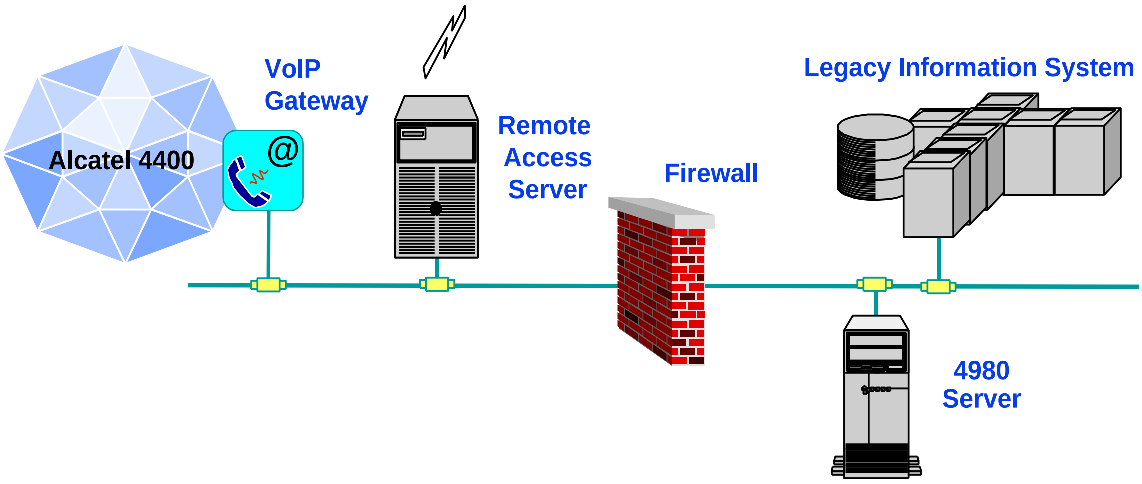
Tele-Worker's Telecom costs are

- controlled by the Company
  - all incoming/outgoing calls are handled by 4400 in the Company
  - Charging is done for his local phone set in Enterprise
- totally charged for the Company
- optimized for outgoing calls by 4400 Call-back procedure

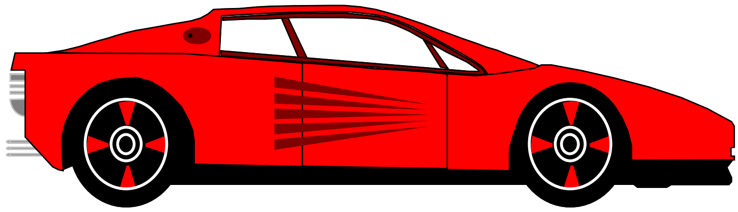
For mobility needs, easy switch over to GSM mode can be activated from 4980 through softkey in idle mode or in conversation
This allows to use VoIP mode for Telecom costs optimization and to switch in GSM mode only when needed for short time local mobility

Recommended Network Configuration for security

- Remote Access Server (RAS)
  - existing RAS for data access will be used
  - otherwise an Alcatel RAS offer should be provided : to be confirmed
- Firewall
  - can be used for signaling connection to 4980 server
  - has not to be used for IP voice connection to 4400 IP gateway


VoIP
Gateway
Legacy Information System
Alcatel 4400
Remote
Access
Server
Firewall
4980
Server

9.4 ALCATEL 4980 WITH GSM



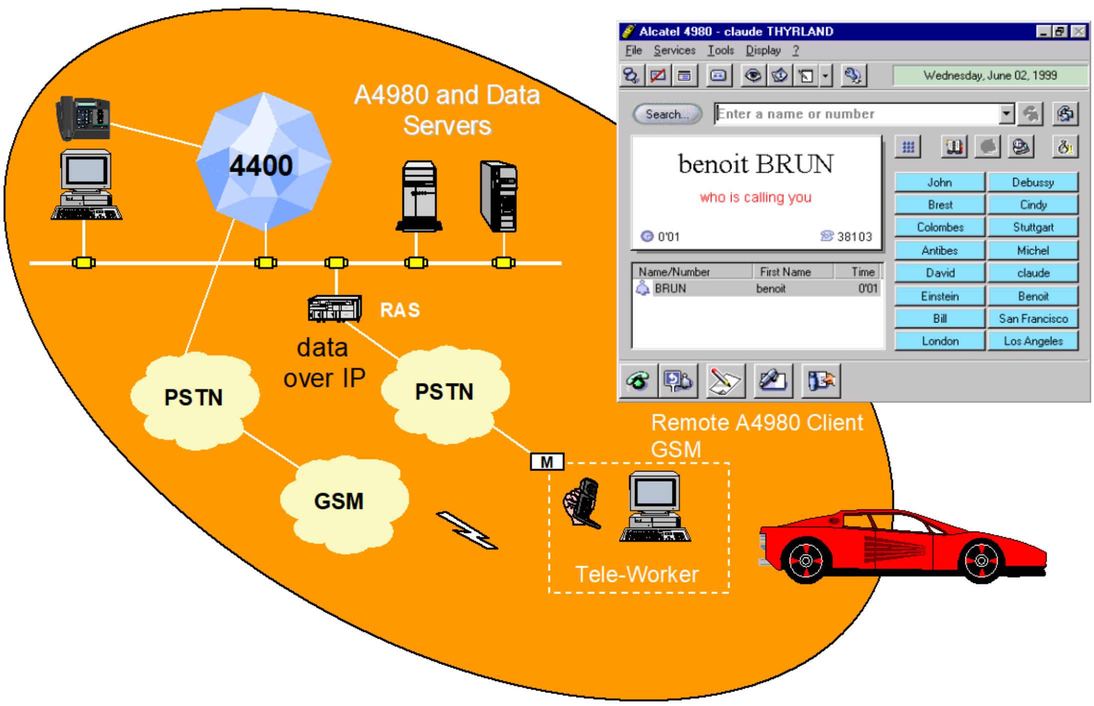
End User Service

With this solution, the Tele-worker has access to all 4980 services. In this case there is no impact on voice quality due on simultaneous data transmission.

•

<u>This solution is typically for Managers, Sales people or Technicians on field and offers :</u>

- ♦ GSM voice calls controlled and charged to the Company
- ♦ optimized outgoing calls (Call-back)
- ♦ mobility of GSM handset
- ♦ all powerful 4980 services with audio on GSM handset



Architecture

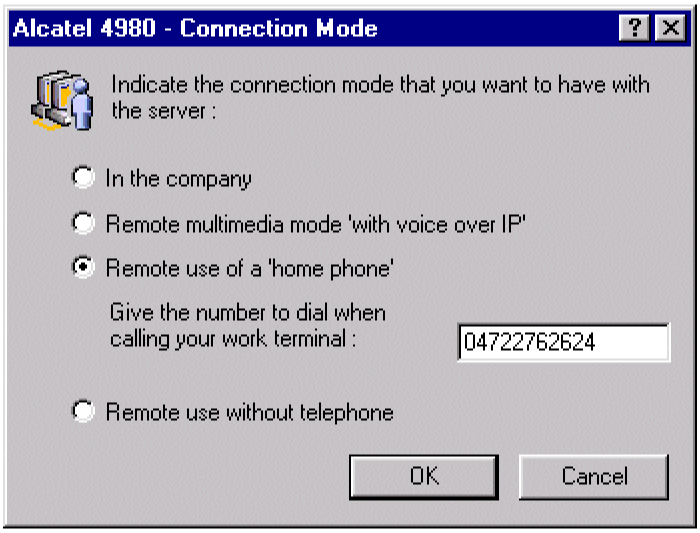
- Off Site : the Alcatel 4980 client runs on standard PC
- In Site : the 4980 server is connected to Enterprise LAN with other legacy data servers
- the workplace in the Company the tele-worker works transparently in/out of Company. This workplace can be virtual
- the RAS (Remote Access Server) is necessary for remote access and security

How does it work ?

- a data connection is established to the Company via the RAS
- Alcatel 4980 signaling is transported on the same line
- we have a modem connection (Analog or ISDN modem for two B channels)
- we will have one IP sessions for signaling to the 4980 server

While connected for data needs, the nomadic user has the same 4980 services as if he was in his Enterprise Office :

- 4400 Call Handling is done for his local phone set in Enterprise
- after remote user login (see figure below), the phone set in Enterprise becomes unusable and voice is routed to remote location through GSM network



- GSM calls are handled from PC with 4980 client and GSM handset is only used for off-hook and audio
- workplace (phone set & PC) in Enterprise can be virtual
- the nomadic worker can transparently work in or out of Enterprise (with GSM handset only or not)

Tele-Worker's Telecom costs are

- controlled by the Company
    - all incoming/outgoing calls are handled by 4400 in the Company
    - Charging is done for his local phone set in Enterprise
- totally charged for the Company

What's about "Ubiquity" ?

When moving, the Tele-worker will use 4400 GSM "Ubiquity" services :

- One Number service for incoming calls
- One Identity service for outgoing calls
- One Directory (of 4635 mailbox subscribers)
- One Mailbox

When connected to his Company, the Tele-worker has advantage to use the powerful 4980 services :

- One Number service for incoming calls
- One Identity service for outgoing calls
- GroupWare
- Call by Name in Company Directories
- Easy Voice Mail handling
- Call Logging
- E-mail notification
- ...

9.5 PC CLIENT REQUIREMENTS FOR ALCATEL 4980 “NOMADIC”

- Hardware
  - minimum : PC 120 MHz, 32M for W95/98, 20M disk space
  - multimedia PC if VoIP with full duplex sound card and optional PC headset/handset
  - modem or S0 connection
    - minimum : 33,6 K for modem
- Software
  - Win 95, 98 or NT
  - Remote Network Access to access Enterprise RAS
    - available in standard with Windows
  - Netmeeting 2.1
    - only necessary for VoIP and Netmeeting services
    - especially powerful PC and link is required for App sharing
  - 4980 Client software

9.6 KILLING POINTS FOR ALCATEL 4980 “NOMADIC” SOLUTIONS

- One license for all needs (In Site/Off Site, whatever the associated phone set/PC)
  - all Tele-workers solutions (multimedia PC, GSM or analog set) are available with Alcatel 4980 Advanced
  - no extra license for Tele-worker is needed

- One architecture for all needs (In Site/Off Site, whatever the associated phone set/PC)
  - no specific server or Application is needed
  - VoIP gateway for IP Networking can be used

- Attractive market price, easy pricing
  - 4980 is an Application at same price whatever the associated set/PC

9.7 ALCATEL 4051/4052 ISDN REFLEXES EXTENDER

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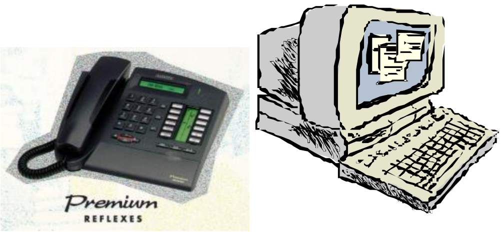

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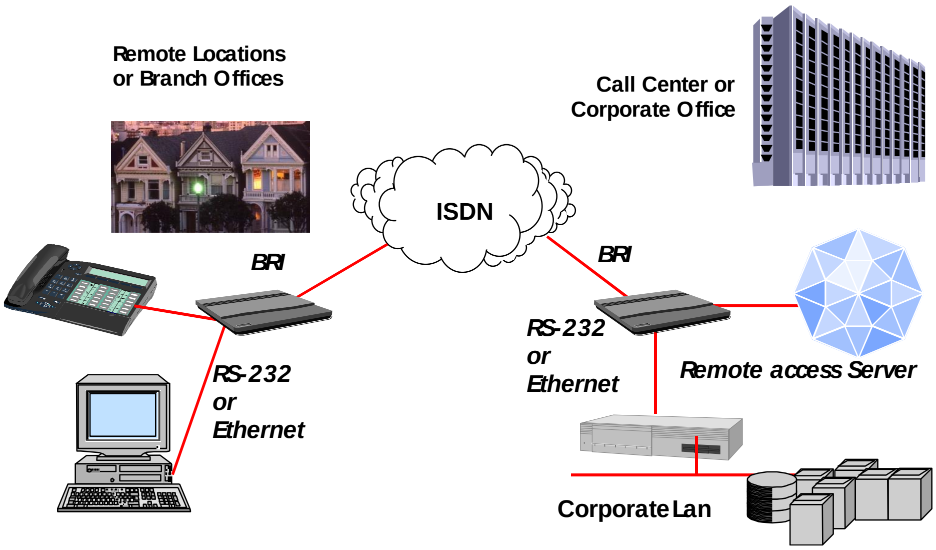
9.7.1 Introduction

Alcatel, by its new Reflexes Extenders family, will give the opportunity to remote workers to be connected to the corporate enterprise as easily as they were in their office. Remote workers use the same Reflexes deskset they use in their office, with transparent access to all the PBX functionality. In the same way, by using the same Reflexes Extender, Remote workers will have also data/LAN capability.

The ISDN Reflexes Extenders will be the first in a family of UA line extenders. It will provide an integrated terminal adapter and three different data connectivity options. By using this ISDN Reflexes extender, remote workers could be connected to corporate PBX or LAN, as they were located in the corporate building.

Remote worker environment will be connected to the corporate enterprise by using the ISDN network.



Voice, UA signaling and data are multiplexed on one or two B channels depending on the option used.

Concerning UA link, Reflexes Extender box in PBX side, re-generates exactly the same stimuli coming for the set. In the same way, Reflexes Extender box in remote side, re-generates exactly the same stimuli coming for the PBX. The PBX doesn't see that the Reflexes terminal is connected remotely.

Remark: only one of UA B channels (voice channel) and the corresponding UA signaling channel are handled by the ISDN Reflexes Extender on the UA link. Therefore, concerning sub-devices, it is only possible to connect a MAC/PC sub-device and a CTI plugware behind the Reflexes set.

As said before, three different data connectivity options are available:

- S option
  Voice, UA signaling, and data are multiplexed on the first B channel.

- E option
  Voice and UA signaling are multiplexed on the first B channel and Ethernet data on the second B channel.

- T option
  Voice and UA signaling are multiplexed on the first B channel and RS-232 Windows dial-up networking data on the second B channel.

These three variations, all of which are user selectable using the same hardware platform, provide better data performance, increased ease of use and improved manageability to remote workers.

9.7.2 main product benefits

The ISDN Reflexes Extenders offer telecommuters new levels of functionality, providing higher speed data, easier installation and setup, the choice of Ethernet data connectivity and a built in ISDN line interface. The main benefits of Reflexes Extender are:

Remote workers are provided access to the same resources as in-office workers
Users get to keep their proprietary Reflexes digital sets with the same level of feature as they were directly connected to the PBX. In the same way, they could connect their PC to the corporate LAN.

Single ISDN BRI line solution
Only one common solution for both voice and data aspects.

Highly stable voice functionality
Voice is handled independently from data communications. If the PC at the remote site or the LAN at the switch goes down, all telephone functionality remains operational

Three data connectivity options
Data and voice in or not in the same B channel, RS232 or Ethernet
Simple to use
Simplified user set-up programming by using the Reflexes set in the remote site

Security
On the voice connection, security is provided via password authentication.

Running costs
Two levels of Call on demand reduce connection charges (see chapter 6).

Software upgrades
Flash ROM allows downloading of software upgrades

9.7.3 Reflexes terminal Functionality

The Reflexes Extender Switch and Remote modules communicate by transmitting signals between the switch location and the remote location across a single ISDN BRI B-channel. Speech is coded to 32Kbps ADPCM or G729A at 8Kbps formats and is combined with the UA signaling for the Reflexes set on the first B channel.

Remark: choice between ADPCM and G729A is only offered with S option (only ADPCM for E and T options).

9.7.3.1 PBX functionality available on the set

The Reflexes Extender series will be compatible with ALCATEL 4200 and 4400 PBXs.

Users get to keep their proprietary Reflexes digital sets with the same level of feature as they were directly connected to the PBX.

Remark: only sub-device MAC/PC (2G terminal) and CTI plugware (3G terminal) interfaces are fully operational.

9.7.3.2 Reflexes Extender functionality available on the set

Configuration
Some parameters have to be configured, like ISDN number, speed, and so on. Using software keys on the Reflexes terminal configures the remote module.

ISDN line management

The ISDN communication between the two modules is managed by using a free UPK (user programmable key). This UPK will be chosen and selected by the user

9.7.4 Data Connectivity Options

The Reflexes Extender's has three data connectivity options, all of which are user selectable using the same hardware platform. The three variants, which are described below, are the S, E and T options. Each provides the same voice functionality.

9.7.4.1 Reflexes Extender : S option

The S option of the Reflexes Extender provides multiplexed voice and data on the first B channel. In this case, RS-232 data is combined with voice on the first B channel for a maximum data throughput of 38.4Kbps in the absence of voice and 19.2Kbps (38.2kbps with G729A compression) with voice.

At the switch site, data is split off from the voice traffic by the switch module and is connected to the LAN via a terminal server. At the remote site, data is split from the voice traffic by the remote module and is sent to the PC via an RS-232 connection.

When using for data connectivity, the user has two options for connecting their data:
Via terminal emulator software
Via Windows dial-up networking (by selecting a null modem dial-up adapter).

Remark: while all data connectivity on the option S must be routed through the switch module and then to a terminal server, the E and T versions give users two different methods for connecting data at the central site: via the switch module or directly to the RAS (see below)

9.7.4.2 Reflexes Extender : E option

The E option provides voice and phone signaling only on the first B channel. The second B channel of the ISDN BRI line is used for Ethernet data connectivity at speeds of 64Kbps, plus data compression, regardless of voice traffic. In this case there are two possible user data configurations:

⇨ Via Switch Module- In this configuration, the remote user connects their digital Reflexes phone and their PC into the Remote module. Their voice and phone signaling travels over the first B channel to the Switch module, which hands this traffic to the ALCATEL PBX. The remote user's Ethernet data is sent over the second B channel to the switch module, which passes it through to the LAN via an Ethernet connection.

  Note: this configuration is recommended for small networks only (roughly 50 MAC addresses or less, but dependent on LAN traffic) as heavy network traffic may overload the switch modules.

⇨ Direct to RAS (Remote Access Server)- In this configuration, the remote user connects their Reflexes phone and their PC to the remote module. Voice and phone signaling is sent to the ALCATEL PBX, via the Switch module, over the first B channel. The Ethernet data is sent directly to the corporate RAS via the second B channel, and never passes through the Switch module. In this case, the data call on the second B channel will dial into a B channel that is terminated by the RAS at the switch site. When data is connected directly to the RAS, and not through the switch module, the RAS does not need to be at the same site as the PBX.

Because all the Ethernet data is sent over the second B channel, the Reflexes extender is able to provide 64Kbps data regardless of voice traffic.

The option E of the Reflexes Extender acts as an intelligent bridge to establish the Ethernet data connection between the remote and switch sites. No routing capabilities are include.

9.7.4.3 Reflexes Extender: T option

The option T is similar to the option E in that it uses the first B channel strictly for voice and phone signaling traffic, and sends all data over the second B channel at 64Kbps. However, on this version all data is sent via an RS-232 connection using Windows Dial-Up networking.

On the second B channel, the option T acts as a limited functionality terminal adapter, providing a PPP connection using a subset of the AT Command Set. The remote user connects their PC to the Remote Module using an RS-232 cable, and then sends and receives data (over the second B channel of their BRI line) using windows dial-up networking on their PC just as if they were connected to an ISDN terminal adapter. The option T emulates a generic TA, in that it uses a basic subset of the AT Command Set.

The option T can be used in the same two basic configurations as the option E (see above), but also provides the option of connecting their data directly to an Internet Service Provider (ISP)

The option T provides no data compression. However, option T works with software based data compression on the PC, including the Stacker LZS data compression software that is built into Windows.

9.7.5 Additional Features

The Reflexes Extender series have a number of different features, as described below.

ISDN Line Interface

The products include an ISDN U Interface for use in the North American marketplace and support ISDN BRI lines both behind the PBX and off the CO switch.

Call On Demand (COD)-

All option provide Call On Demand for the voice call on the first B channel. When an Extender is in Call On Demand mode (a user selectable option), the ISDN line will automatically disconnect after as user determined interval of inactivity, in order to save on ISDN line usage charges. As soon as the remote phone goes off-hook or has a keypress, or as soon as there is an incoming call, the ISDN line is reconnected and the remote phone returns to normal operation. The ISDN line will disconnect again after the user determined time interval of inactivity. Call on Demand applies to the first B channel only and does not affect use of the second B channel.

Two levels of COD will be proposed:
- ⇨ a minimum level, where only the basic call is operational (PBX want to ring the set or to update the Message Notification LED on the set – user pushes any key or hang-up)
- ⇨ a maximum level, where all services are available on the set (any action or event coming from the PBX or the user are taken into account)

Data Call On Demand (DCOD)

This feature is found on option E units only. When an 8912E remote module is in DCOD (a user selectable option) the second B channel is monitored for data transmission activity. After a user-defined period of inactivity, the B channel will be disconnected to save on usage charges. As soon as the Remote user attempts to send data, or as soon as data is sent to the remote user from the central site, the B channel will be brought back up and data transmission will occur normally.

Dialback

All options support Dialback mode (user selectable option). When Dialback mode is selected, a remote user who attempts to go online will pick up the phone, which will then establish a connection with the switch module and immediately be disconnected. The switch module will then re-establish the connection with the remote module, but all long distance charges from that point on will be billed to the central site, rather than the remote user's location.

Software Upgrades

The product includes Flash ROM that allows new software versions to be downloaded directly into the units when upgrading is required. This means that software upgrades can download directly into the units.

9.7.6 Security

On the voice connection, security is provided via password authentication. Remote users are required to enter an 8 to 10 digit password each time they attempt to log onto their PBX and establish a connection. The LAN that the Extenders are connecting data to provides data security.

With the E and T options, PAP and CHAP authentication by the RAS are supported.

9.7.7 Technical characteristics

UA terminal range supported
- ⇨ 2G : Alcatel 4034, Alcatel 4023
- ⇨ 3G : Alcatel 4035 (Advanced)

Sub-devices supported
- ⇨ 2G : MAC/PC interface, keyboard, add-on modules (20 or 40 keys)
- ⇨ 3G : CTI (TAPI), keyboard, Add-on modules (20 or 40 keys)

Speech compression
- ⇨ ADPCM at 32Kbps or G729A at 8kbps

Data rate

- ⇨ Option S: up to 38.4Kps (19.2Kbps in case of ADPCM compression)
- ⇨ Option T: 57,6Kbps fix allocated
- ⇨ Option E: 64kbps

ISDN interface

- ⇨ ETSI NET 3
  Remark : can connect to BRI from Telecom or from BRI in back of PBX

Power supply

- ⇨ 12V DC supplied by a wide range 90 to 264V AC adapter

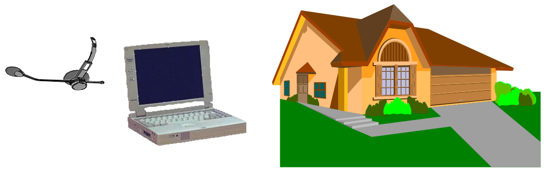
9.7.8 Package definition

Two commercial references (one for switch module , one for the remote module):

- ⇨ Alcatel 4052 Reflexes remote extender : 3BA57289 AA
- ⇨ Alcatel 4051 Reflexes local extender : 3BA57290 AA

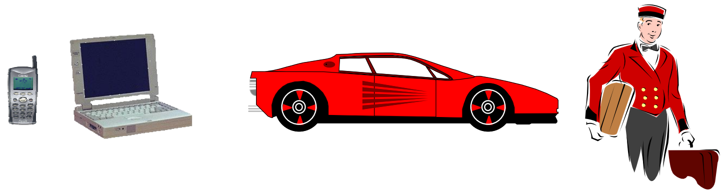
9.8 WHICH TELE-WORKER SOLUTION ?

♦ **4980 with Multimedia PC (VoIP)**



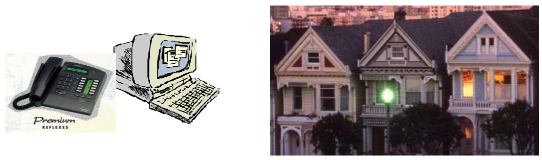
♦ **4980 with GSM**

(or analog set)



♦ **4052 ISDN Reflexes Extender**

(with/without 4980)



9.8.1 The parameters for solution choice

User profile and adapted voice communication tools

- ⇨ Home Worker : Reflexes Extender, VoIP (modem or S0)
- ⇨ Nomadic Worker : VoIP (modem) or GSM
- ⇨ Home Worker with phone mobility : GSM

Voice QoS must be high for Home Worker

- ⇨ Decreasing quality : Reflexes Extender, GSM, VoIP
- ⇨ but QoS of GSM is very depending on environment, VoIP with modem/S0 is equivalent
- ⇨ S0 (2B) is recommended for QoS or simultaneous voice/data

Local or Full mobility :

- ⇨ Home DECT or GSM

Security needs

- ⇨ same as for data access (RAS, Security Dynamics Server)

Equipment costs

- ⇨ depending on number of remote users

Telecom costs (8 hours/day)

- ⇨ Analog or S0 with 1 B channel (~1700 FF/month)
- ⇨ S0 with 2 B channels (~3100FF/month)
- ⇨ GSM (~6000FF/month for trafic of 0.2 E)
- ⇨ Analog (~450FF/month for trafic of 0.2 E)

Enterprise Estate Office costs are more or less equivalent

- ⇨ about 1500 FF/month

9.8.2 Conclusion : the recommended choices

Home Worker (connected 8 hours/day)

- ⇨ Reflexes Extender on S0 for voice quality

- ⇨ Telecom costs /month : ~3100 FF(2B)

Home Worker & Nomad (connected 2 to 3 hours/day)

- ⇨ Reflexes Extender device or VoIP on S0 for voice quality
- ⇨ GSM for mobility
- ⇨ Telecom costs : ~650 FF(1B) + ~550 FF(2B) or ~2300 FF(GSM)

Nomad (connected 2 hours/day)

- ⇨ VoIP on modem for free voice calls
- ⇨ GSM for voice quality or if low traffic
- ⇨ GSM for mobility
- ⇨ Telecom costs / month : ~450 FF(1B) + ~1500 FF(GSM)

10. ALCATEL 4635

This section provides an overview of generic functions of Alcatel 4635 voice and fax processing system and explains the unique characteristics and features of the product.
Alcatel 4635 product is a fully integrated voice and fax messaging system for the Alcatel 4400 PBX. The system is physically and logically integrated with the Alcatel Crystal Technology (ACT™). This integration offers additional features and unparalleled :

- user friendliness for the subscriber of the system,
- ease of installation and operation for the administrator of the system.

Hardware integration described below provides the smallest foot print of all Voice and Fax messaging systems on the marketplace.

Interested readers will profitably refer to the detailed Alcatel 4635 product notes, namely :

- A4635 Product Note 1 - Generic Feature Description
- A4635 Product Note 2 - Fax Applications
- A4635 Product Note 3 - Voice and Fax Networking Solutions
- A4635 Product Note 4 - Visual Messenger

All these product notes are available on-line on VPC web site.
http://aww.vpc.aut.alcatel.at/proj/a4635

10.1 A4635 R2.0 INTRODUCTION

This section of A4635 Product Description intends to provide the reader with a description of the main evolution provided by R2.0 of the product as well as customer benefits.
These are :

- Visual Messenger, a desktop application that allows A4635 users to create/send/manipulate voice and fax message directly from their Windows-based workstation,
- IP OctelNet, an evolution of OctelNet, the richest and most convenient voice messaging network protocol, now available over IP for lower running costs and higher quality, speed and security,
- Global Message Redundancy, a voice and fax storage policy that allows full mirroring of disks for system and user information.

10.2 VISUAL MESSENGER

10.2.1 Overview

As voice and fax messaging become widely accepted productivity tools, users request better tools to manage the flow of messages. A desktop-based management tool for the Alcatel 4635 mailbox can dramatically improve the ease and value of voice and fax messaging. Experienced “power users” as well as new mailbox users can benefit from this tool. Alcatel Visual Messenger gives users visual access to their mailboxes from their personal computers.
Alcatel Visual Messenger is a multimedia messaging application that lets users easily review and send voice and fax messages with simplicity and convenience. Visual Messenger takes advantage of users’ familiarity with Microsoft Windows-based applications. The basic elements of today’s personal computers — a monitor, keyboard, mouse and (optionally) speakers — provide a useful complement to the telephone, giving access to standard messaging features plus providing new capabilities as well.
Visual Messenger allows new users to learn how to use their mailboxes quickly, with little or no training. Experienced users gain additional convenience and flexibility in managing their voice and fax messages. Visual Messenger displays a “window” on the user’s personal computer so they can view all their messages, including incoming faxes.
Visual Messenger simplifies message management with features such as folders, local storage and a “recycle bin”. Users view their faxes on the desktop. And message addressing is

simplified, allowing users to "browse" and "drag-and-drop" addressees' names for individual messages and to create and edit personal distribution lists.
Visual Messenger has been designed and developed for use as an enterprise-wide PC-based solution for voice and fax messaging. It operates on most Windows PCs and within most networks, allowing many organisations to provide their users with PC-based management of messages using the *existing* computing environment.

This section will cover the following aspects :
- Workstation – Telephone association,
- Mailbox outlook,
- Reviewing voice and fax messages,
- Sending voice and fax messages,
- Mailbox management,
- Architecture,
- Installation and configuration.

Visual Messenger description will end with a quick reference guide, icon and associated service.
This guide highlights, at-a-glance, the power and ease of use of Visual Messenger.

10.2.2 Workstation – Telephone association

This association is done during User Logon.
Visual Messenger logon endorses the following functions :
- user identification, (name and mailbox number)
- mailbox access
- selection of operation mode for the Visual Messenger session
    - workstation - phone set association, telephone is used to play/record voice
    - work off-line with multimedia workstation, this mode can be used to review messages that were downloaded during any previous sessions.

Remote Access to Visual Messenger

Visual Messenger can be used successfully by remote users. If an employee visits a different work location with wide area access to the organisation's data network, the user only needs to enter the telephone number of the temporary location in the extension field, when logging on to Visual Messenger. The message server will then instruct the 4400 switch to place Visual Messenger outcalls to the phone at the temporary location.

Note that all calls initiated by Visual Messenger supersede any screening/forwarding on the telephone set associated to the session.

Remote access can also be established from the user's home or any other location with two direct inward dial phone (DID) lines. One phone line is used for a network connection to the user's primary LAN using Remote Access Services (RAS) or point-to-point protocol. The second line is required for message playback and recording. Even without the second line for voice communications, a single line provides a remote message waiting indicator and a view of the message queue. It also permits viewing faxes and downloading messages for local storage. Another attractive remote access alternative is to use an ISDN connection, since it typically offers a data and a voice connection simultaneously on the same line.

Figure 1 : Visual Messenger logon screen.

10.2.3 Mailbox outlook

Your messages can be viewed on your PC.
The global organisation of Visual Messenger main screen is the now very common one in Microsoft (and Web) environments :

- top is commands area, 1 icon per function
- left is folders area, ORGANIZE all messages according to subject/activity/...
- right is information area, PRIORITIZE, FOCUS

Folders are also offered with Visual Messenger, which clearly help in organising your voice and fax messages after review.
Folders manipulation is as easy with Visual Messenger as it is with Windows Explorer.
Folders can contain sub-folders which can contain sub-folders which can contain…
Creating/Naming/Moving/Deleting folders is up to Visual Messenger user.
Just like messages, deleted folders are moved to recycle bin, and erased either on request or at the end of the session.

The columns display information about each message: type of message (voice, fax, compound fax/voice, broadcast), urgent/non urgent (red exclamation), private or not (lock), sender's name or phone number (for systems supporting CLIP, even numbers from outside callers can be displayed), date and time, length (in sec. or pages), status (lock means saved), and expiry date, which is the scheduled date to purge the message according to the user's class of service.
With a single mouse click Visual Messenger users can sort messages by any of the message-summary criteria: type, time and date sent, expiration, etc… in either ascending or descending order. Messages can also be organised in folders. Folders allow users to establish a message organisation specific to their personal needs. To provide meaningful associations with message content for future reference they can also add a text "note" next to a heard message.

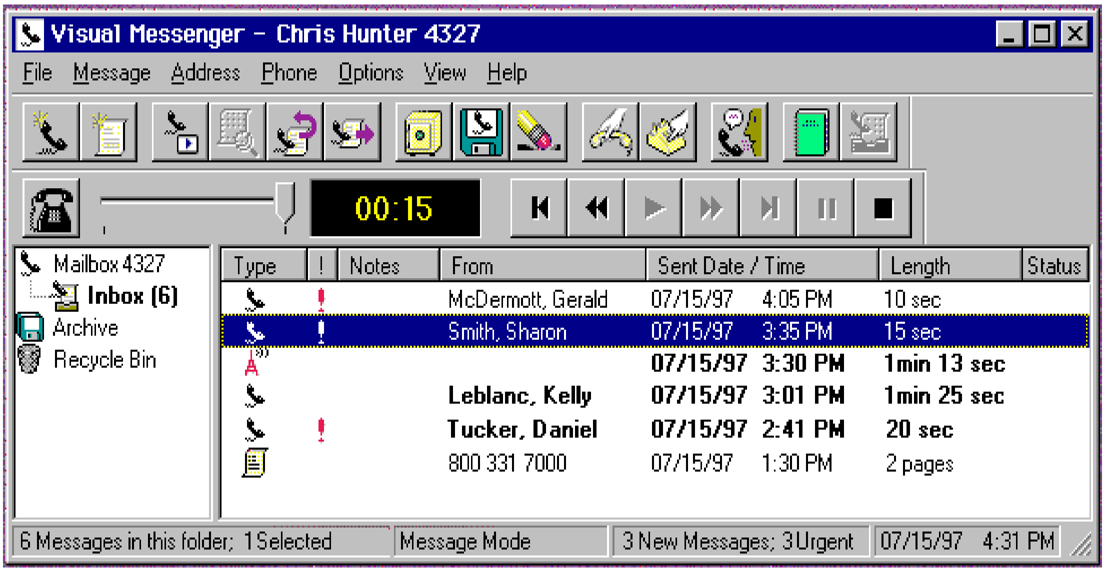


Figure 2 : Visual Messenger showing Inbox.

10.2.4 <u>Reviewing voice and fax messages</u>

As you look at your messages, you may select any message in any order for playback.
As you review messages, they are played over the phone, (quality, privacy & minimal LAN impact)
The default phone extension is defined at mailbox creation time and can be modified by the user at login time or during the session.
With this feature, he can roam throughout the company, and keep the ability to use Visual Messenger, with messages played on the phone set close to his workstation.

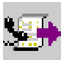
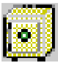
During message review and at the end of listening to a message, Visual Messenger's onscreen player controls give users various options. Users can navigate through a message during review using:

- ***Jump to a specific point in the message*** - with the use of the slide bar and timer (see Figure 3), the user can select any point in the message to begin playback, a convenient feature when replaying a message segment containing financial figures or a telephone number.
- ***Restart from the beginning, or rewind by 6 seconds***
- ***Jump forward by 6 seconds or jump to the end***
- ***Pause/resume***
- ***Stop***

Figure 3 : Visual Messenger Player Controls.

Users can also handle the message by using these additional options at any time:

-  ***Reply by sending a message*** - Users can quickly and easily respond to a message from another user by choosing the reply icon and recording a message. With Visual Messenger, users can add other recipients to the reply message, a feature that is not available on the telephone user interface (TUI).

-  ***Reply by calling the sender*** - After hearing certain messages, users may want a live conversation with the sender. If the "call sender" feature is enabled, the user will be transferred to the sender's extension.
-  ***Forward a copy*** - the recipient of a message may want another user either to hear it or take action on it. He or she records introductory remarks and sends a copy of the message to one or several users or to a personal distribution list. All messages can be forwarded except messages originally marked "private."
-  ***Erase*** - If a message requires no further action, the user can erase it. Erased messages initially move to the "recycle bin". The user can retrieve an erased message from the "recycle bin" by dragging it back to the Inbox with the mouse, retaining the message on the server drives. This "undelete" feature is available through the Visual Messenger, but not through the TUI. Messages in the "recycle bin" are counted as "previously heard" messages if the user logs on directly from a telephone rather than using Visual Messenger. Users can empty messages from their own "recycle bins" at any time. Messages that were in the "recycle bin" when the user exits Visual Messenger are automatically emptied and are permanently erased from the message server's drives.
-  ***Save*** - a user can save a message if it requires action or will be needed for reference at a later time. Saved messages become "archived" in the message server and are identified as "archived messages" when the user logs on to the mailbox directly from a telephone.
-  ***Local Storage*** - Visual Messenger's Local Storage option permits a user to save a message for a long time or permanently as a ".wav" file on the PC. Once stored locally on the PC a message can remain indefinitely and is no longer subject to the message server's automatic deletion controls assigned in a user's class of service. A display of locally stored messages is accessible from the Visual Messenger main window (see Figure 2). Users can review the contents of a locally stored message by reading the "note" and by playing the message through the PC speakers. Locally stored messages may be dragged or copied back to storage on the A4635 message server. Restored messages appear as "new, unheard" messages, and the original envelope information is replaced by the time and date when restored. Local storage not only permits saving key messages for a longer time, but also frees up disk space on the A4635 server.

10.2.5 Viewing Fax Documents

Users enjoy significant convenience and privacy due to their ability to view fax documents using Visual Messenger. Users with classes of service allowing fax can receive fax documents directly into their mailboxes. With Visual Messenger, users have the ability to view these faxes in the privacy and convenience of their own office without the need to print them to or retrieve them from a fax machine (see Figure 4). If a printed copy of the fax is needed, the user can print it to any fax device or to any printer accessible to the personal computer — including those on the network. Printing faxes from Visual Messenger also makes it possible to print to high-speed, high-quality laser printers providing a cleaner document in less time. If the fax document is received in an orientation or type that is inconvenient to view, it can be rotated and zoomed within the fax viewer.
Faxes can also be saved as either .bmp or TIFF files from the viewer.

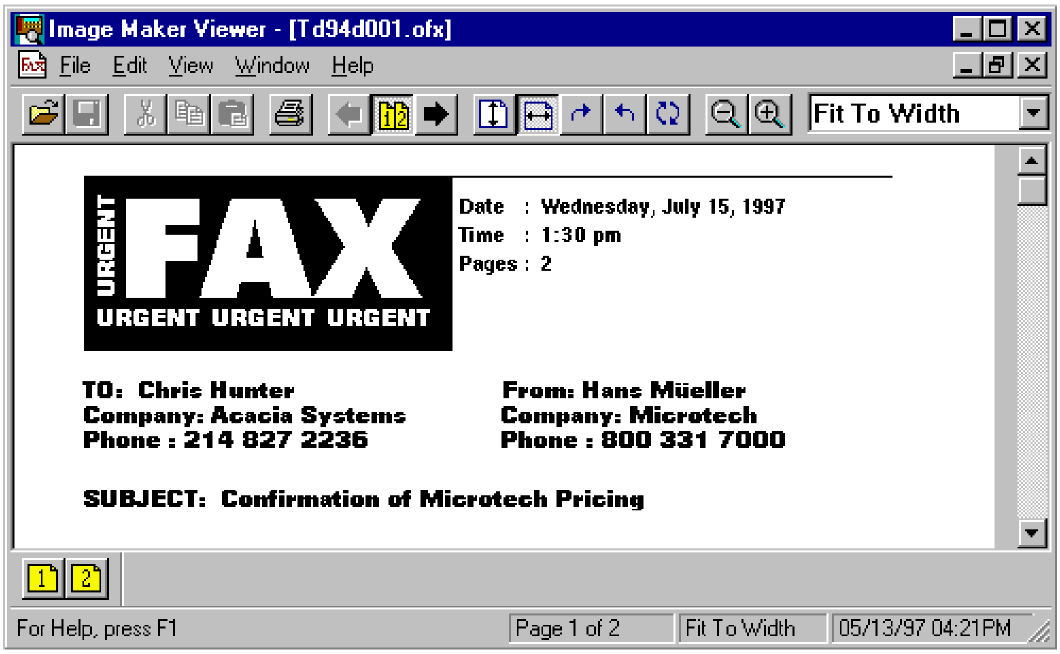


Figure 4: Visual Messenger allows users to view faxes received in their mailboxes.

10.2.6 <u>Sending voice and fax messages</u>

Recording voice messages

The second main function of the mailbox is sending messages. With Visual Messenger, users select the "record" icon on screen and, if their telephone is not already off-hook, the message server calls them at their telephone extension to begin recording a message. Visual Messenger's player controls make it easy for users to pause during the recording of a message, review the message from the beginning or append an additional recording at the end. With these capabilities, users can edit a message before sending it.

Addressing messages

When recording is complete, the user can send the message to a wide variety of destinations — another user, several users, a pre-determined list of users or even to his or her own mailbox. Users address messages simply by entering mailbox numbers, by typing recipient names on the keyboard or by browsing the system's directory. With the browse function, the user selects entries by double clicking on them or by dragging and dropping the name to the recipient list. If a user is uncertain if the name selected is the intended recipient, they can play the addressee's spoken name. (Users can also select remote destinations on servers equipped with networking software. For further information about networking features, please refer to the *A4635 Networking Solutions section*.)

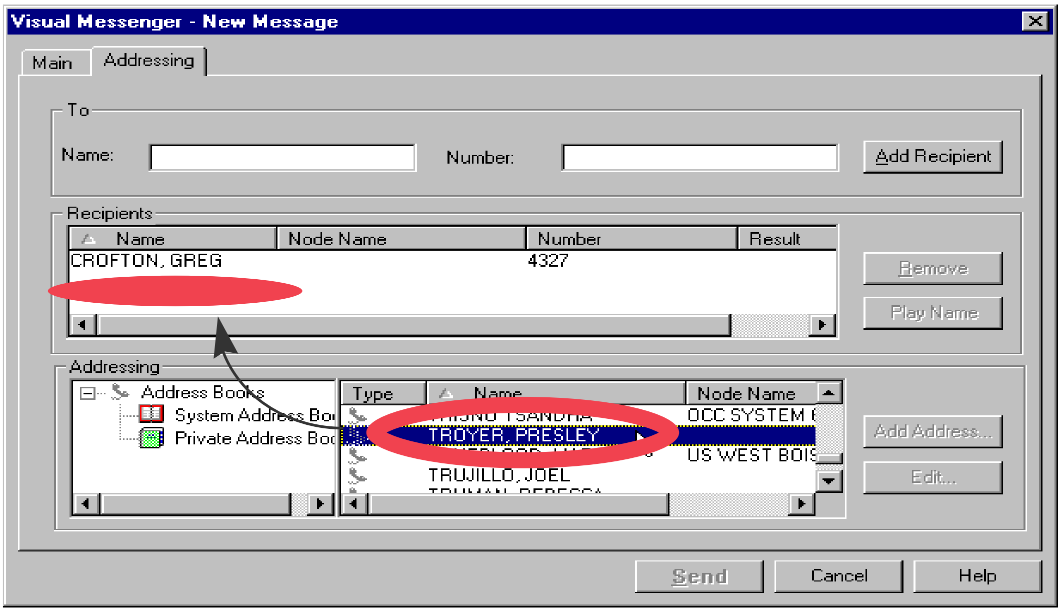


Figure 5: Visual Messenger - Addressing screen.

Delivery Options

Before sending a message, a user may select one or more delivery options, including private, urgent, message receipt notification and future delivery, using the Send Options checkboxes (See Figure 6 for an example) in the New Message Dialog box.
If a user marks a message *private*, the recipient sees and hears that the message is private and is prevented from forwarding or archiving copies of the message.
Users may mark messages for *confirmation of receipt*, which provides notification to the sender when recipients have reviewed the messages. A notice is sent to the message originator's mailbox when each recipient has reviewed the entire message; this notice will appear in the sender's main message queue.
Visual Messenger can deliver messages marked *future delivery* at a specified time in the future (up to 365 days in advance). The sender chooses the date and time for delivery of the message, using his or her PC keyboard.

10.2.7 Outgoing Fax Documents

With Visual Messenger, sending a facsimile can now be accomplished by simply addressing and "printing" the fax. Users can send an image of any file on their personal computer by identifying the file and addressing the facsimile (see Figure 6). Addresses can be selected from either a System Address Book or a Private Address Book maintained by the user, or the user can specify any fax number or fax-enabled mailbox by entering it in the Number field of the New Fax Dialog box.

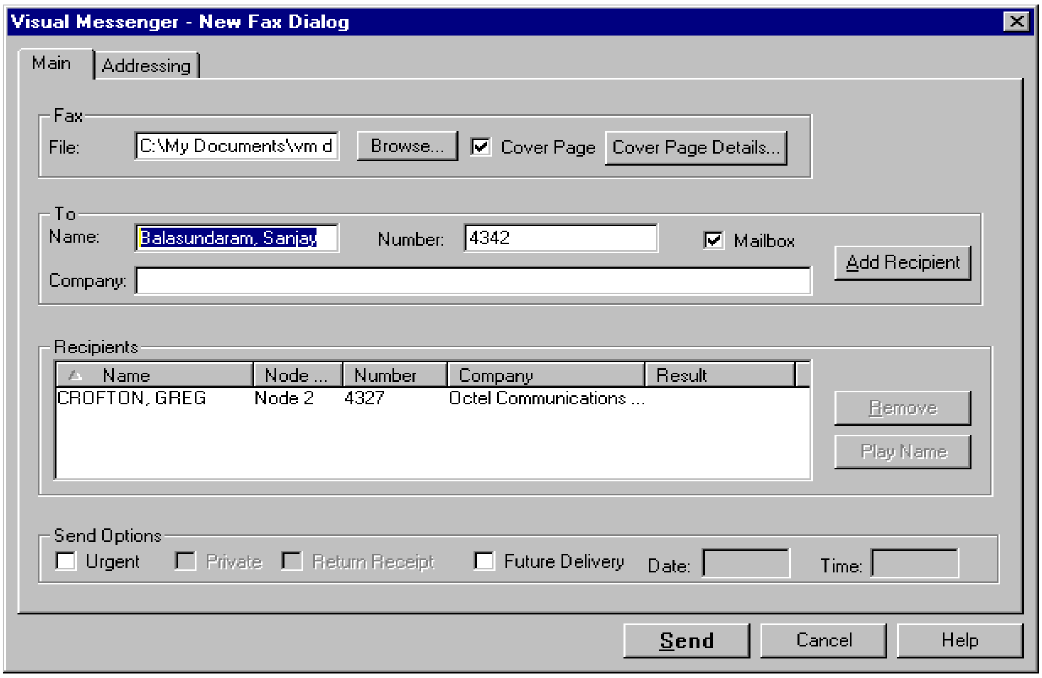


Figure 6: Addressing a new fax message.

Users are also given the choice to include a cover sheet and to specify delivery options for their fax messages. Fax cover sheets can contain sender and recipient details, a subject and a memo. Users can select from a choice of cover sheet styles and can specify a company logo from those added to the system by the system manager. Users can also create a scanned image of their own signature to include at the bottom of the memo section of the cover sheet. The user can preview these fax cover sheets onscreen to ensure they will appear as desired.

Faxes sent with Visual Messenger use the Alcatel A4635 server as a fax server to deliver documents to subscriber mailboxes and to other fax devices. Senders will receive either a notification of successful delivery or an error message for any destinations for which delivery was unsuccessful. Error messages are displayed in the Result column next to the addressee identification in the New Fax Dialog box, as shown in Figure 6. A user's voice mailbox must have a fax-enabled class of service to use the facsimile function.

For added convenience, Visual Messenger supports the ability for users to send a fax directly after creating a document in another program, such as Microsoft Word. Figure 7 shows how easily a document can be turned into a fax and sent from within virtually any application. In the example shown, an outbound fax is sent directly from Microsoft Word by virtue of selecting "Alcatel Fax" from the print command menu. This action automatically launches the Visual Messenger application if it isn't already running, requests the user to log in, presents the New Fax dialog box from which to select the recipient(s) and then uses the Alcatel A4635 server to deliver the fax.

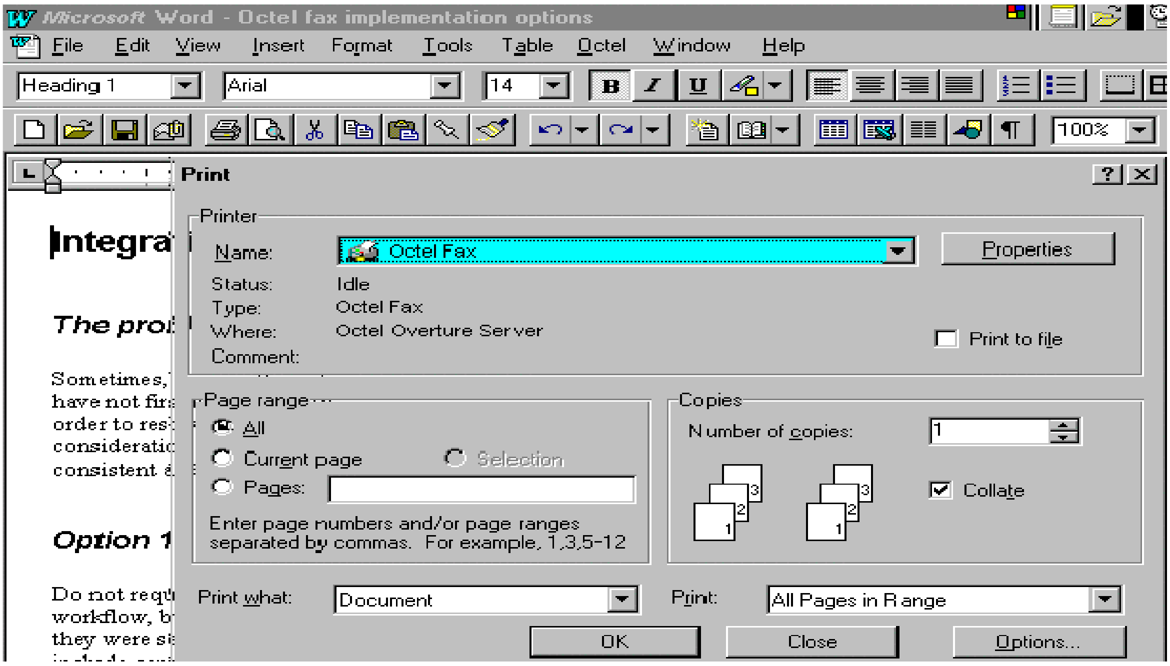


Figure 7: Sending a new fax directly from Microsoft Word.

10.2.8 Administrative Options

In addition to providing enhanced messaging functions, Alcatel Visual Messenger allows users to perform standard administrative tasks on their Aria mailboxes.

Multiple Mailbox Access

A user may need access to more than one mailbox when calling the server — a personal and departmental mailbox, for example. Visual Messenger users may easily specify another mailbox to log into via a menu command, allowing for an easy transition from one to the other.

Names and Greetings

Visual Messenger provides a convenient way to manage the mailbox name and greetings. As shown in Figure 8, a user can record or re-record a mailbox name, personal greeting or extended absence greeting. Users simply select their choice of greeting from the drop-down box and use the recording tools to record or re-record the greeting.

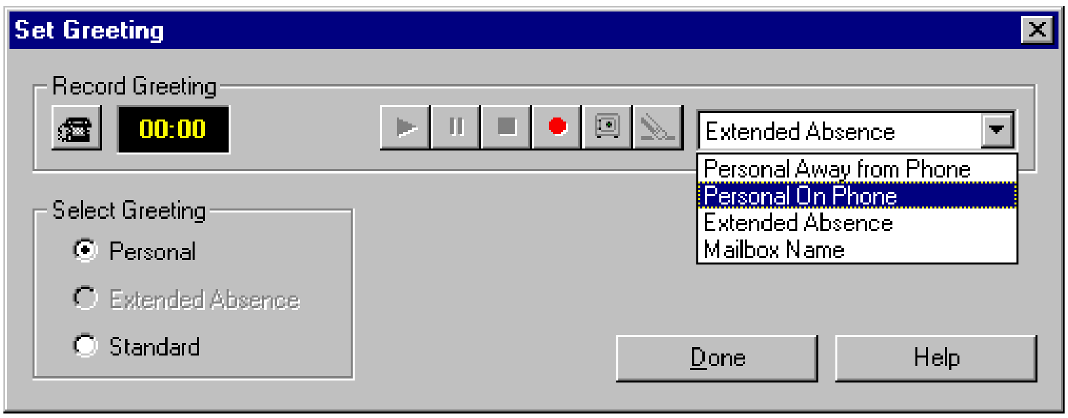


Figure 8: Set greetings and mailbox name.

Personal Group Lists

The power of voice processing is enhanced when users can record a single message and send it to a group of mailboxes simultaneously. With Visual Messenger, each user can create 15 personal group distribution lists onscreen with up to 25 destinations each. Users can easily create, edit or delete personal distribution lists with the computer keyboard and mouse. The user can simply begin keying in a name in the Edit Personal Address Group dialog box, and Visual Messenger provides the closest match (see Figure 9). Visual Messenger also supports recording a "spoken name" for the list, which is played when addressing a message to the group list through the TUI. Users can review the destinations on lists and the names of the lists themselves either from Visual Messenger or through the telephone.

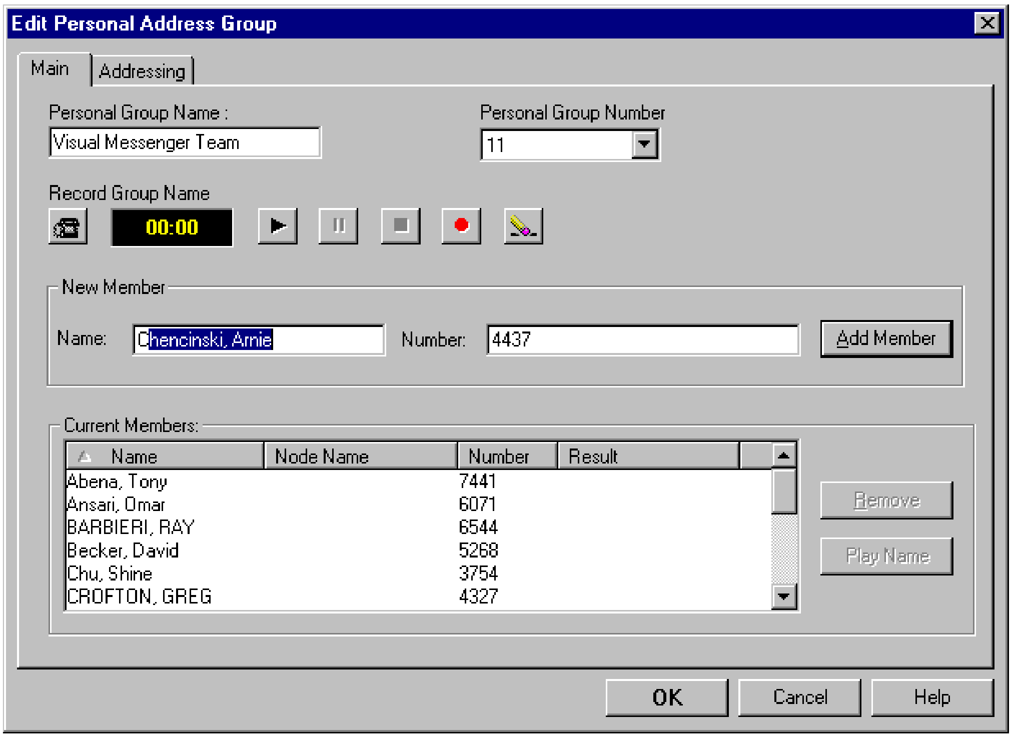


Figure 9 : Drag-and-drop editing of a personal group list.

Personal Passwords

Users can change the personal password to their mailbox with Visual Messenger. This password controls access to the mailbox both through the telephone and through Visual Messenger.

10.2.9 Architecture and components

Introduction

A4635 Rel 2.0 architecture provides the foundation for Visual Messenger. A4635 Rel 2.0 is the marriage of telephony and information technology in a manner that capitalises on the benefits of both. A4635 Rel 2.0 architectural model consists of three components:

- Alcatel 4635 voice and fax server,
- Alcatel 4635 Access Server software,
- Alcatel Visual Messenger Client software (see Figure 10).

The first architectural component, A4635 server, is a telephony-grade voice and fax server. A4635 servers provide high performance, outstanding reliability and a full range of call answering, messaging and voice/fax processing capabilities.

The second component, Alcatel Access Server software, runs on a high-performance Windows NT-based LAN server. Alcatel 4635 Access Server software provides a number of key technical functions that enable and facilitate communications between A4635 server and the data network. In the A4635 Rel 2.0 architectural model, A4635 server is used to communicate with the telecommunications environment; the Windows NT-based PC running A4635 Access server software is used to communicate to the data network environment.

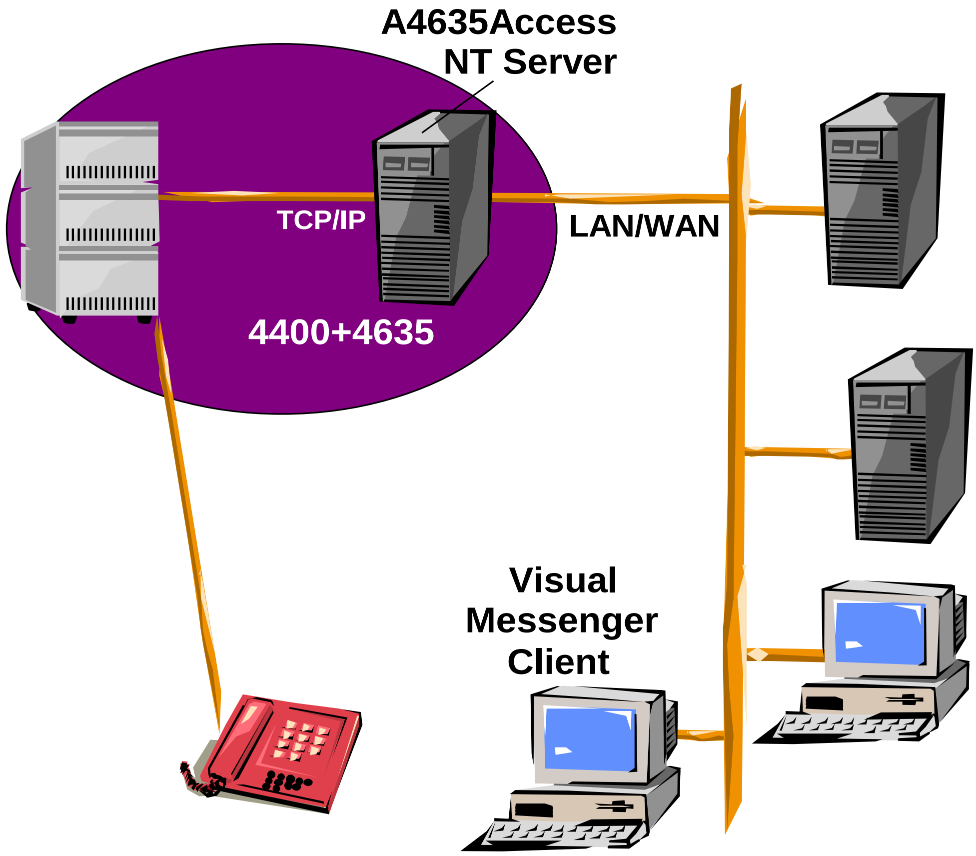


Figure 10: Visual Messenger client software utilises the A4635 Access Server as a bridge between A4635 server and the data network.

Alcatel 4635 Message Server

VPM35

- Pentium CPU at 133 MHz (Q1 '99 - 200 MHz)
- 32 MB memory
- One IDE/ATA disk onboard, up to 3 additional IDE/ATA disks on MSBI
- Embedded and 10BaseT Ethernet
- up to 64 Voice and Fax ports on SPA3 boards

VPS35

- i386EX CPU at 33 MHz
- 32 MB memory
- One IDE/ATA disk onboard
- Embedded and 10BaseT Ethernet
- up to 8 Voice Ports onboard

Alcatel 4635 Access Server

Alcatel Access Server opens A4635 over the LAN.

Alcatel Access Server is the gateway between 4635 and the Visual Messenger clients, and endorses the following services :

- Communication Services
    - Multiplexes Visual Messenger requests to/from A4635
    - Translates Microsoft .WAV format to/from A4635 coding
- Security Services
    - Authentication of clients
- Administration Services
    - Configuration
    - Operation
    - Trace logging & error reporting

Alcatel Access Server description

- Standard Intel-based PC
- Ethernet interface
- HW requirements
    - Pentium 200
    - 128 MB RAM
    - 1GB disk space
- SW requirements
    - Windows NT 4.0
    - Service Pack 3
    - TCP/IP support

Visual Messenger Client

- Standard Intel-based PC
- Ethernet interface
- Minimum HW requirements
    - i486, 33 MHz
    - 24 MB RAM
    - 15 MB disk space
    - Optional HW
    - Sound card,
    - microphone/loudspeaker
- SW requirements
    - Windows '95
    - Windows NT 4.0

10.2.10 Installation and configuration

Visual Messenger Authorisation Requirements

Visual Messenger is licensed software that is enabled through A4400 authorisation procedure. The authorisation determines how many users on a message server may use Visual Messenger. The Client Access License parameter must be enabled in the class-of-service assigned to Visual Messenger users. The fax viewing and sending features require a fax enabled mailbox that is authorised with the Fax Manager option.

Installation Options

Visual Messenger features many flexible installation options for the system administrator. During installation, administrators set telephone switch and network connectivity options and establish access privileges to administrative functions. End users can choose whether to install Visual Messenger for execution from a file server or from their own personal computer.

System Administration

There is very little ongoing administration necessary for either the A4635Access NT server software or Visual Messenger. Once the application has been installed and configured, administration primarily involves verification downloads of the System Address Book and troubleshooting, if and when necessary.

System Address Book

A valuable component of Visual Messenger is the System Address Book, which provides a directory of all subscribers with mailboxes on the Alcatel A4635 message server and includes all subscribers contained in the optional NameNet™ directory. The system manager can download this directory through a simple command and can establish a schedule for automated, unattended directory refresh. Each time a user logs in after the system address book has been updated, the user will be notified and offered the option of automatically updating his or her personal copy. The directory is an invaluable component of Visual Messenger that allows users to address messages quickly and easily.

Troubleshooting

A4635 Access server software provides tools that assist troubleshooting by both the system administrator and Alcatel customer service agents. Event logging, called tracing, can be enabled to provide a very detailed log of activity for A4635 Access Server and Visual Messenger client.
A4635 Access trace includes records of all transactions.
Visual Messenger trace includes every user action including keystrokes and mouse clicks.
These trace logs enable in-depth troubleshooting. A4635 Access Server software also includes a peg file that indicates usage levels of Visual Messenger.

Mailbox Security

A4635 software protects the integrity of the server and the information contained in users' mailboxes and messages by offering the most extensive security capabilities available.
Users create personal passwords for their mailboxes. This password is required for logon both through the telephone and through Visual Messenger. The personal password, which can also be changed either through the telephone or onscreen, prevents others from entering the mailbox and accessing messages.
For greater security, A4635 software can require users to change their passwords periodically, and can check to be sure that they are random numbers, that they are not based on the mailbox number and that they are not the same as the previous password. Use of Visual Messenger is subject to these same password restrictions.

Online Help

Visual Messenger includes an extensive online facility that is indexed by topic and is context-sensitive to the specific Visual Messenger feature being used at the time of the request. The user must simply click on the help button on any dialog box or press F1, and relevant help will be displayed. Or, users may select Visual Messenger help topics from the help menu. The help menu allows users to browse through an index of help topics, or use the "find" command to search for the topic of interest. For convenience in reviewing in printed form, individual help topics can be printed.

User Preferences

When installing Visual Messenger, users can customise Visual Messenger to their own preferences. They can select whether to install the application on a file server or directly on their personal computer. They also select the storage location of their private directory and archived messages. A wide variety of other preferences can be established for using Visual Messenger, including the choice of message notification, dialling options, including the use of a calling card, and confirmation of copies, moves and deletions.

10.2.11 Visual Messenger Quick Reference Guide

Alcatel Visual Messenger on your computer offers many advantages over the traditional telephone user interface:

- See all the messages in your mailbox and access them randomly
- Sort messages
- Add message notes
- Organise messages in folders that you name
- View faxes on your computer with privacy Send faxes from your computer directly from Windows applications
- Recover deleted messages
- Use the dialler to place calls
- Manage your group lists visually
- Copy messages to your computer
- Browse system directory and private address book

Also using your computer you can:

- Record and change greetings
- Change personal password

The main icon bar includes the following functions:

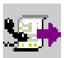
 New Voice Message

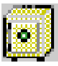
 New Fax Message

 Play Voice Message

 View Fax

 Reply to Message

 Forward Message

 Save Message

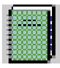
 Download Message

 Delete Message(s)

 Call Sender

 Dial

 Set Greeting

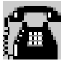
 Open Address Books

 Go to Inbox

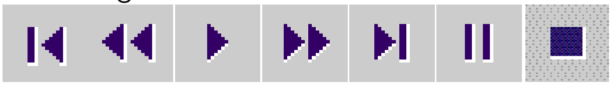
Associated telephone states

 Disconnected,

 Ringing

 Connected

Recording controls include:



Rewind to Beginning/Previous Message
Rewind
Play Voice Message
Fast Forward
Fast Forward to End/Next Message
Pause
Stop

In the left window you'll find :

 Recycle Bin

10.3 IP OCTELNET

IP OctelNet is the new name of former Digital Networking.

10.3.1 Overview

Alcatel's voice and fax mail networking solutions allow organisations with multiple message servers to easily and cost-effectively exchange voice and fax messages between locations. Many Alcatel customers have the need to offer a crisp, clear communication path for voice and fax messages between remote locations spread around the globe. This path should be readily available, and it should be economical, secure and easy to manage. In addition, Alcatel believes that it should be as easy for users to send a voice or fax message to another location as it is to send a message to another user on the same server. With Alcatel's full-featured networking solutions, we have delivered on that belief. A user on Alcatel 4635 server can quickly and easily send a voice or fax message to one or thousands of users located in countries all over the world, using all of the message delivery options that are available in sending a message to an employee in the next office. Organisations that extend messaging to all locations experience increased productivity, decreased telephone bills and higher customer satisfaction.
The benefits of networking between multiple offices of the same organisation can easily be extended to other affiliated entities with message servers, such as suppliers, important customers, strategic partners, the investment community, educational institutions, law firms, accounting firms and more. In this way, the significant benefits realised from voice and fax messaging within a single location can be extended not only enterprise-wide, but beyond, to improve *all* of your important communications.

Alcatel offers three voice mail networking solutions:

- **Enhanced Networking,** based on the **OctelNet**™ specification- for full-featured analog networking of voice and fax messages between voice processing servers which support it (Alcatel 4635, Octel Aria, Serenade message servers)[1]
- **IP OctelNet** - for full-featured TCP/IP digital networking of voice and fax messages between voice processing systems, which support it (Alcatel 4635, Octel Aria message servers).
- **AMIS Analog** - for basic analog networking of voice messages between Alcatel servers and other vendors' AMIS-equipped systems.

[1] *For readers, knowing the Octel Aria products. The networking solutions on A4635 do not need the additional Aria Network Interface Card (NIC). Authorisation of networking services (Network Manager Packages) is different from Aria. SNMP access to message traffic information is not implemented on A4635.*

OctelNet and IP OctelNet products and AMIS Analog networking operate independently of each other. An Alcatel server can have OctelNet, IP OctelNet or AMIS Analog only, or all three networking solutions on a single server.

This section will provide very limited information regarding AMIS Analog and OctelNet. Interested readers will profitably report to dedicated product note ; see reference and location in the introduction of this document.

10.3.2 <u>OctelNet/NameNet basics</u>

An OctelNet network ensures that the full benefits of voice and fax messaging are extended across the enterprise.
OctelNet is a proprietary software solution of Lucent Technologies Octel Messaging Division (Octel) for analog networking between voice processing servers. It is available on the Alcatel 4635 server, as an optional software package. With OctelNet an organisation can tie all of its employees together on one cost-effective voice and fax messaging network that is easy to use and manage. Users with mailboxes on one Alcatel server can easily exchange voice and fax messages with users on other Alcatel (or Octel) servers, from the largest Alcatel 4635H serving a headquarters location to the smallest Alcatel 4635J located in a small office. Up to 1000 servers may be linked in a single network.
Enhanced Networking based on OctelNet is used to link remote locations of a single organisation, as well as to link a business enterprise with other organisations, such as its vendors and customers who use Alcatel (or Octel) servers. In large enterprises, it can also be used to send messages between two or more 4635 servers at a single location (private 4400 network).

Feature Availability and Ease-of-Use

OctelNet provides feature-rich voice and fax messaging functionality. Most features of same-server messaging are available to users when sending and receiving network messages. For example:

- OctelNet messages can be marked urgent, private and future delivery.
- Message confirmation, including confirmation of receipt or notification of non-receipt, can be obtained.
- Extended Absence Notification and Extended Absence Block.
- Network mailboxes can be included in system and personal distribution lists.
- Message envelope information is available.
- Users can reply to messages received from a remote server by using the same command used to reply to a message on a local server.

The only features not available over the network that are available on a local server are Check Receipt and Call Sender.

NameNet Directories

The NameNet feature of OctelNet can create self-maintaining network directories that require little administrative effort. NameNet is also flexible enough to allow the system manager to control and modify a directory if desired.
A Network Directory is a directory of remote users stored on a local server. A user entry in a Network Directory consists of the node number, the mailbox number (and alias address, if applicable), the ASCII name and the spoken name. User Network Directory entries can be created automatically and maintained based on message traffic.

The maximum size of the Network Directory is 40,000 entries.

Secure Transmission

OctelNet provides secure, reliable and confidential message transmission. A “handshake” between servers is automatically performed each time a network call is placed, and encrypted security codes are exchanged before actual message transmission begins. Line quality is also checked before messages are transmitted. If line quality fails to meet

prescribed standards, the server will automatically retry transmission at intervals specified by the system manager. Should the line drop during transmission, partially received messages are erased and re-transmitted completely in the next call. Once all messages are transmitted, the servers automatically disconnect.
In addition, destination addresses can be verified through NameNet. When a remote user's name exists as an entry in a sending node's Network Directory, the mailbox number and ASCII name of the remote user as known by the sending node are transmitted along with the network message. The remote node matches this information with its ASCII name for that mailbox number. If there is no match, because the name has changed or the mailbox has been deleted, for example, the message is returned to the sender. At this time, the entry, whether permanent or usage-based, is automatically deleted from the sending node's Network Directory. This function increases message security by preventing misdirected messages.

OctelNet Key features and benefits

Enhanced Networking based on **OctelNet**	
Feature	**Benefit**
Multiple, geographically dispersed voice processing servers are connected in a single voice and fax messaging network	• Extends voice messaging benefits across an entire organisation and/or between an organisation and its vendors and customers
Up to 1000 A4635 servers can be linked together	• Supports large corporate and government networks
Full-featured messaging using standard user interface	• Provides ease-of-use • Ensures that full benefits of voice messaging are extended across entire organisation
Operates on all Alcatel 4635 servers	• Protects customer investment in Alcatel equipment
Dial-by-Name addressing across the network	• Provides ease-of-use
Spoken-name confirmation for network messages	• Provides confidence that sensitive messages are being delivered to the right person
Extended Absence Notification across the network	• Increases messaging convenience for users
Extended Absence Block across the network	• Provides important information to message senders
Classification of network users as priority, standard, or network access prohibited	• Permits more efficient management of the network by allowing the system manager to apply the appropriate grade of network access to different levels of the organisation

Enhanced Networking based on **OctelNet**	
Feature	**Benefit**
Outcall delivery transmission schedules	• Provides flexible options for efficient management and prioritisation of network traffic
Separate transmission schedules for weekdays and weekends/ holidays	• Minimises transmission costs • Provides greater flexibility and cost savings in managing network traffic
Self-maintaining network directories	• Requires little or no administrative effort • Provides flexibility to control or modify network directory as desired
Send unlimited number of messages during a transmission session	• Controls transmission costs
One copy of message sent and stored for multiple users on remote server	• Saves transmission time and disk space
Bi-directional messaging sessions	• Eliminates set-up charges and security identification exchange for second call
Ports can be specified exclusively for network transmission	• Maximises efficiency and guarantees available ports for networking
Security code checking	• Ensures secure, confidential message transmission
Line-quality checking	• Assures high quality network connection • Assures high quality voice transmission
Automatic erasure of partially received message and re-transmission of entire message following a line drop during transmission	• Ensures reliable, complete message transmission
Automatic retry schedules for busy, ring-no-answer and failure to meet line quality standards	• Saves system managers additional work
Destination address verification	• Provides confidence that sensitive messages are reaching their intended destinations

10.3.3 IP OctelNet description

IP OctelNet allows customers to connect their message servers directly to their corporate data networking infrastructures in order to transmit voice and fax messages directly over existing wide area networks (WANs). Using any public or private network that supports TCP/IP, organisations can connect up to 1000 Alcatel 4635 servers worldwide for full-featured message exchange. (See Figure 4.) IP OctelNet is cost-efficient, fast, easy to use and offers the same flexible sending features as analog networking.

IP OctelNet Topography

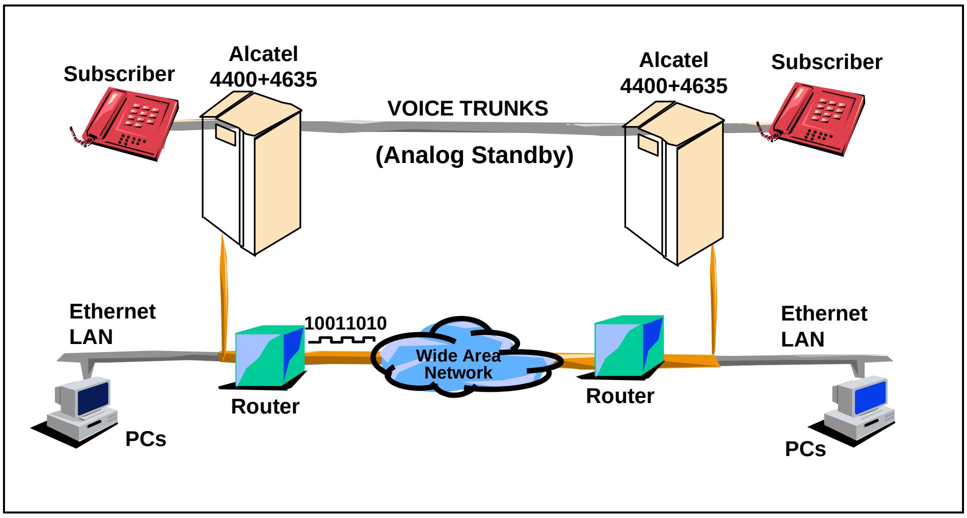


Figure 1

IP OctelNet uses the same delivery schedules, network addressing and user interface that are employed with analog networking, making it both easy to use and administer. Digital transmission of the messages is generally faster, more accurate and higher quality than analog transmission.
With IP OctelNet, the Alcatel server is connected to the customer's digital network via A4635 LAN Interface. The sending server directs voice and fax messages addressed to a digital node to the unique IP address of the remote server, as programmed by the system manager in the node profile. Each Alcatel server on the network is given its own IP address. Multiline IP OctelNet is not supported.

Analog Fallback Message Delivery

IP OctelNet works in conjunction with OctelNet analog networking to ensure the transmission of messages in the event the digital network is unavailable. System managers set up a system retry schedule for IP OctelNet, defining the interval between unsuccessful attempts and number of times the server will attempt to send messages digitally. If the system manager has enabled Analog and the IP OctelNet retry schedule has been exhausted, the server will automatically "fall back" to analog transmission to deliver the messages in the queue for that node. The server will also fall back if any single message in a node queue is unable to be transmitted over the digital network. When this message and any others remaining in the queue are transmitted, the server automatically reverts back to IP OctelNet for its next delivery attempt.

Advantages of IP OctelNet

- **Lower communication costs**. IP OctelNet utilises the customer's existing high-speed, cost-effective data network infrastructure, eliminating the use of low-speed, higher-cost analog transmission facilities and providing cost efficiencies and economies of network facilities.
- **Perfect transmission quality**. Voice messages are transferred as digital data directly from one server to another with no intermediate conversions from digital to analog and back again. This eliminates the loss of sound quality and the introduction of line noise that may be experienced during an analog transmission over a typical analog telephone connection, resulting in crisp, clear transmissions.
- **Improved delivery speeds**. IP OctelNet uses high-speed data network facilities to deliver messages in seconds, not minutes – even during peak traffic periods. The improvement depends on the speed of the line connecting the message servers. For example:
  - A 256 kbps line transfers a minute of voice data in just a few seconds.
  - A 10 Mbps Ethernet (direct connection) could yield a speed increase of up to 20 to 1. That is, the voice portion of a one-minute message could be transferred in less

than 3 seconds. Actual results will vary due to overhead in connection set-up/close times, number of messages sent in one connection and the current networking load.

- **Server port traffic conservation.** IP OctelNet utilises the A4635 LAN Interface for the network connection, eliminating the need to utilise server voice ports for network traffic, and thereby freeing those ports for other applications.
- **Improved message security.** IP OctelNet provides a higher level of security for networked voice traffic because all messages are encoded with a proprietary algorithm.

Impact of Using the Data Network

The impact of adding voice and fax message traffic to a data network will depend on the amount of networked traffic that is generated. In order to ensure sufficient bandwidth exists, customers can estimate the impact of adding IP OctelNet traffic to the network. Alcatel 4635 servers with Rel 2.0 software digitise voice at a maximum rate of approximately 24K bits per second. To digitally network voice and fax messages, approximately 33% overhead is added to this figure, to account for addressing, data checking, etc. This results in 32 Kbits per second, or 240 Kbytes of data per minute of voice. In other words, a 4-minute message would constitute less than one Megabyte of data. If an OctelNet analog network is currently in place, a customer can calculate the amount of data that will be added to the network by multiplying the current analog messaging traffic in minutes by 240 Kbytes/minute.
It is also important to examine the impact of adding IP OctelNet traffic to the customer's routers. To calculate this impact, one must take into consideration the router's throughput. Using an example of a 256Kbit per second line and the Ethernet's theoretical 10 million bit per second capability, 1/39th of the Ethernet's capacity (256,000/10 million), or 2.5%, would be used. The A4635 voice servers are designed to send no more than 20 minutes of voice message traffic in 1 minute, which would equate to utilisation of a maximum of 6.5% of the Ethernet's capacity.
Using a data network to transmit voice and fax messages can also dramatically improve the message transmission speed. Using a 256 Kbits per second WAN as an example and a 32 Kbits per second message data rate, sending messages would use approximately 1/8th of the WAN's capacity (32/256) per second of message. Therefore, to send a 1- minute message would take approximately 7.5 seconds (1/8th of 1 minute), a significant improvement over analog transmission time. Similarly, a 64 Kbits per second WAN would take 30 seconds to transmit a 1- minute message (32/64, or ½, of 1 minute). Actual results will vary based on network traffic and efficiency.

IP OctelNet Key Features and Benefits	
Feature	**Benefit**
Connectivity to existing LAN/WAN	• Takes advantage of existing data infrastructure • Lowers transmission costs • Faster transmission times • Improves voice port throughput for other applications • Improved message quality
Analog fallback	• Ensures message delivery
Message encryption	• Improves message security

10.3.4 AMIS/OctelNet comparison

AMIS Analog Networking provides interoperability between Alcatel and non-Alcatel voice mail servers.

AMIS Analog networking allows Alcatel servers to exchange voice messages with users on systems from other vendors. This optional networking feature is available on all Alcatel 4635 servers. The AMIS Analog networking implementation on the Alcatel 4635 fully conforms to an industry standard, Audio Messaging Interchange Specification (AMIS) analog protocol. The protocol defines a basic level of voice messaging functionality—sending, receiving and replying—that must be adhered to by anyone offering an AMIS analog networking product.

Comparison of OctelNet and AMIS Analog		
Function	**OctelNet**	**AMIS Analog**
Send	Y	Y
Receive	Y	Y
Reply	Y	Y
Fax Messaging	Y	N
Dial-by-Name	Y	N
Spoken-Name Confirmation	Y	N
Delivery Options (private, urgent, future delivery)	Y	N
Line Quality Testing	Y	limited
Bi-directional Message Sessions	Y	N
Maximum Message Length	30 minutes	8 minutes
Messages per Network Call	unlimited	9
Multiple Mailbox Destinations per Message Transmitted*	Y	N
Operating Cost	less	more

* *Messages sent to AMIS destinations are transmitted once for each destination mailbox, even if to multiple mailboxes at the same destination.*

AMIS Analog Key Features and Benefits	
Feature	**Benefit**
Interoperability between Alcatel servers and other vendors' systems	• Extends basic voice messaging benefits across or between organisations with different vendors' systems
Alcatel user-interface features	• Provides ease-of-use • Promotes user acceptance of messaging
Operates on all Alcatel 4635 servers	• Protects customer investment in Alcatel equipment
No special hardware Compatible with low-cost analog lines	• Ensures easy installation with no inconvenience to users • Available at all customer sites

10.3.5 Network Addressing plans

An important aspect of planning and implementing a voice and fax messaging network is the network addressing plan. A4635 provides a high degree of flexibility in establishing a network addressing plan that is convenient and easy to use. Four approaches are commonly implemented:

Simplified Network Addressing minimises the number of digits users need to enter as a network prefix when addressing messages. This method is ideal for organisations that have only a few network nodes. For example, if a company has message servers in Paris, Wien and Berlin, the system manager could assign the prefixes 1, 2 and 3 to these nodes, respectively.

Simplified Network Addressing				
Name	**Office**	**Node Prefix**	**Mailbox Number**	**Network Address**
Zidane	Paris	1	8255	**1**8255
Herzog	Wien	2	7241	**2**7241
Köpke	Berlin	3	295	**3**295

Direct Distance Dialling (DDD) addressing uses long-distance area codes as network prefixes. With this plan organisations can use the existing telephone directory as an OctelNet mailbox directory.

Direct Distance Dialling Network Addressing					
Name	**Office**	**DDD Telephone Number**	**Node Prefix**	**Mailbox Number**	**Network Address**
Zidane	Paris	(331) 3938.8255	331	8255	**331**8255
Herzog	Wien	(431) 29121-7241	431	7241	**431**7241
Köpke	Berlin	(491) 7901-295	491	295	**491**295

Uniform addressing plans also take advantage of existing telephone directories by mirroring the existing PBX dialling plan. In these plans, users enter the same number when addressing a network message as when dialling using the internal telephone.
2 different examples are given below.
Last one shows a configuration where any mailbox of the network can be reached locally or from a remote node with the same numbering plan.

Uniform Numbering Plan Network Addressing					
Name	**Office**	**Telephone Number**	**Node Prefix**	**Mailbox Number**	**Network Address**
Zidane	Paris	3938.8255	3938	8255	**3938**8255
Herzog	Wien	29121-7241	29121	7241	**29121**7241
Köpke	Berlin	790-1295	7901	295	**7901**295

Uniform Numbering Plan Network Addressing					
Name	**Office**	**Telephone Number**	**Node Prefix**	**Mailbox Number**	**Network Address**
Zidane	Paris	68255	6	68255	**6**8255

Herzog	Wien	77241	7	77241	**7**7241
Köpke	Berlin	8295	8	8295	**8**295

All-Digit addressing allows users to address messages using the complete DID number of the recipient. With this plan organisations can use the DID listings in an existing telephone directory as an OctelNet mailbox directory, as well.

All-Digit Numbering Plan Network Addressing					
Name	**Office**	**Telephone Number**	**Node Prefix**	**Mailbox Number**	**Network Address**
Zidane	Paris	(331) 3938.8255	331	8255	**3313938**8255
Herzog	Wien	(431) 29121-7241	431	7241	**43129121**7241
Köpke	Berlin	(491) 7901-295	491	295	**4917901**295

System managers can use any combination of addressing plans to create a customised scheme that best meets the needs of the organisation. Up to 200 dialling prefixes can be associated with each node on a network.

10.4 GLOBAL MESSAGE REDUNDANCY

Global Message Redundancy (GMR) involves duplicate storage of all messages on 4635H.
This software-based capability uses mirrored drive pairs (drive 2 & 3 - message drives) to store messages redundantly. If a drive fails, the system seamlessly switches to the redundant drive and messages are available without interruption.

Some rules about Global Message Redundancy :
- Two system drives (drives 0 and 1) provide system drive redundancy,
- Messages are not stored on system drives, therefore use drives 2 and 3
- Everything stored on the message disks is made redundant by software

Voice storage comparison

Table below shows the voice storage capacity ; all values expressed in hours

	Standard message storing	**Global Message Redundancy**
Disk 0	40	0
Disk 1	40	0
Disk 2	210	210 (mirror of disk 3)
Disk 3	210	210 (mirror of disk 2)
Total	**500**	**210**

10.5 A4635 R3.0 INTRODUCTION

This section of A4635 Product Description intends to provide the reader with a comprehensive document giving the main evolution provided by R3.0 of the product as well as customer benefits.
These are :

- Visual Messenger evolution, now open the leading clients in electronic messaging, namely Microsoft Outlook and Lotus Notes. Visual Messenger for Outlook/Notes now provides the e-mail user with the ability to handle voice and fax messages as well as A4635 mailbox configuration.
- Visual Messenger integration with A4980,
- Web Messenger, an alternative to Visual Messenger for businesses who widely deploy web-based solutions for global messaging to seamlessly handle voice, fax and e-mail,
- Integration between
- ultimate A4635/A4400 networking solutions to provide large enterprises with a transparent network management for all voice flows, telephone as well as voice and fax messaging,
- last but not least, A4635 Hotel offer evolution, to meet the demands for multiple languages as well as advanced usage of voice messaging by guests.

After product evolution, reader will have a look to :

- A4635 SW delivery policy and installation guidelines,
- A4635/A4400 network release description and dependencies,
- A4635 orderable items
- Migration examples.

10.6 DESKTOP MESSAGING SOLUTIONS - SETTING THE SCENE

A4635 is a successful voice and fax messaging solution for A4400 for several years.
By end '99 installed base is 7000 systems, 35000 ports and around 2 million users.

Some leaders are sharing most of the market of electronic messaging desktop SW, either with a client-server architecture, Outlook from Microsoft and Notes from Lotus, or with a web-based architecture.

Users, now experts in both domains, voice and electronic messaging, are asking for a unique tool to handle all their messaging flows.

On the other hand, infrastructure renewal cannot be done as a snap of fingers. In many cases, recent investments in A4635 and electronic messaging backbones will not be thrown away in the short term.

A4635 Desktop messaging solutions combine all services, for all flows on user's desktop.

In other words, this allows A4635 Desktop messaging solutions to :

- maximise level of services,
- maximise adherence of IT manager and users,
- minimise training of users (down to zero),
- minimise infrastructure impact.

Along with the renaming of 4400 IP-PCX into OMNI 4400, brand name for A4635 desktop messaging solutions will be :

OMNI MESSENGER

10.7 VISUAL MESSENGER FOR OUTLOOK

10.7.1 Microsoft Outlook market

Outlook is the client SW for Microsoft Exchange, but can be used outside Exchange e-mail infrastructures such as :

- HP OpenMail
- Netscape
- any other POP3/IMAP4/SMTP mail server, namely most of recent e-mail systems

Today, Outlook is bundled with Microsoft Office and is the right candidate to serve as the client of choice for various e-mail engines.

European market end '97 (source IDC)
- 2.8 M users for Exchange
- 1.2 M users for HP OpenMail
- 1.0 M users for Netscape

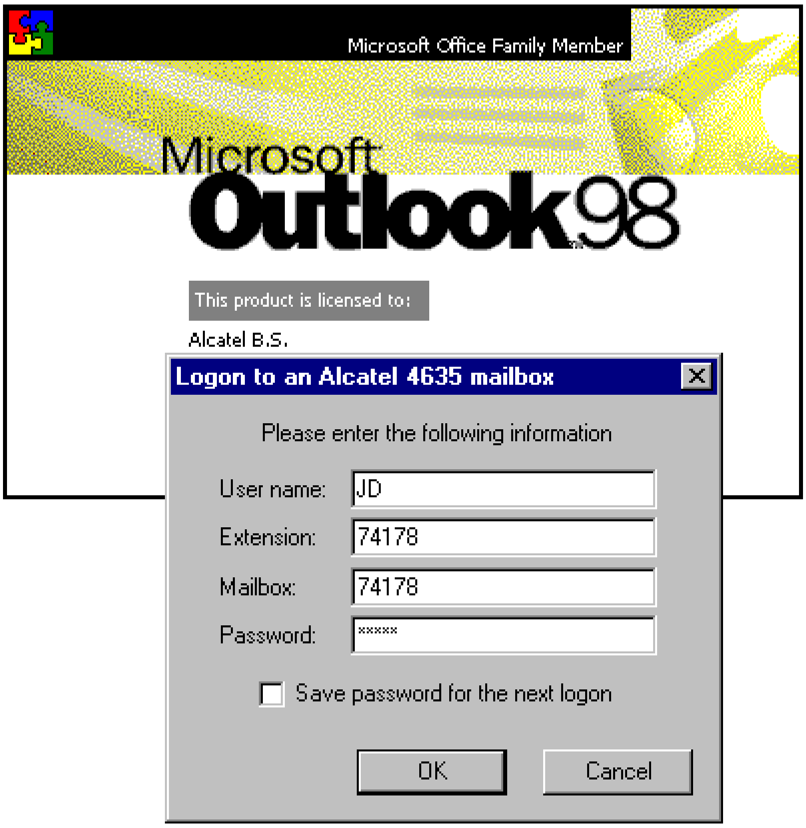
10.7.2 Services description

Logon procedure

Logon procedure has 2 roles :
- start Visual Messenger services from Outlook,
- attach a telephone extension, either internal or external (GSM), to the desktop session.

The logon procedure is system dependant. It is defined as follow:
By default, the user should enter the password each time Outlook is launched.
- On a Windows NT machine, the user has the possibility to save the password with the help of the check box. The password is encrypted and is written in the registry. Logon information are NT user dependant. Therefore, each NT user has its own configuration.
- On a Windows 95 and 98 machine, the logon procedure is the same as Windows NT. However, each user will share the same configuration.

User has the possibility to skip password entry request by clicking the box
"Save password for next logon"
This is the recommended usage to have Outlook silently starting Visual Messenger services.



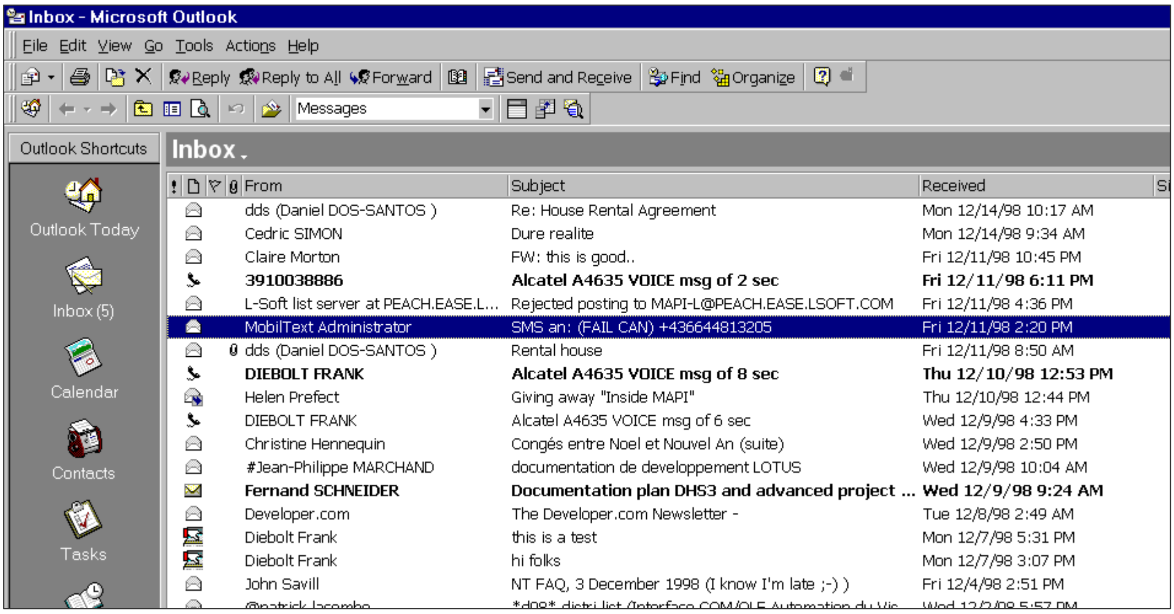
Single Inbox VIEW

Visual Messenger for Outlook was developed in the full respect of Outlook rules :
- ergonomics,
- look and feel,

- client / server work-split (actions, storage, possible multiple directories)
- extensions facilities.

Voice/Fax messaging with 4635 is welcome and quite natural in this environment.

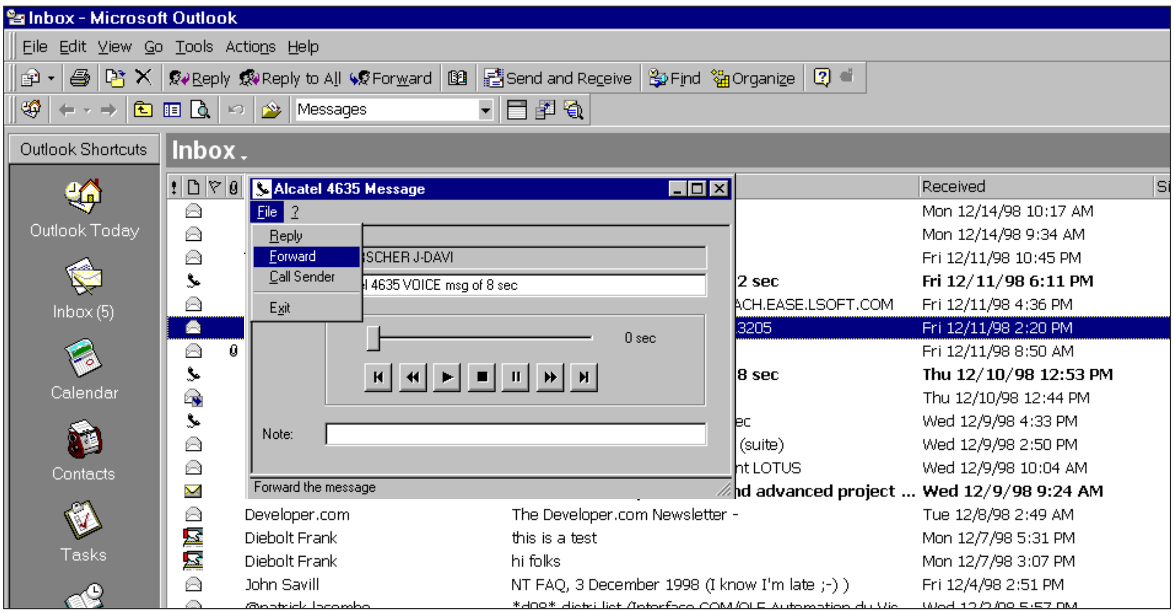
All messages, voice, fax, and e-mail messages are VIEWED in the same inbox.
Sorting is possible according to any criteria, type of message, sender, time stamp, subject,... in order to organise messages and actions upon them.



Voice/Fax message control

PURE Outlook ergonomics : an illustration of the golden rules of VM for Outlook.
Voice messages ARE messages.
No matter the content, no matter the repository, A4635 messages are manipulated (played, viewed, sorted, searched, folded) with the same ergonomics as any other message.

Just like other messages, voice messages can have their SUBJECT and NOTE fields updated by user. This feature facilitates further message manipulation/storing/retrieval.
Of course, all what applies to Voice messages also applies to Fax messages.



Actions upon voice/fax messages

Actions are divided in 2 categories :

- message creation,
    - New Voice Message,
    - New Fax Message,
- actions upon messages stored in user's mailbox,
    - "Reply",
    - "Forward",
    - "Call Sender"

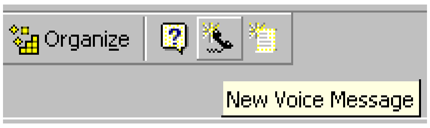
Notes.

Outlook standard actions "Reply" and "Forward" are overridden to allow the user to reply or to forward a voice/fax message the same way as he does for e-mail messages.
Outlook "Reply to All" action is not relevant for A4635 message and therefore disabled.
All actions listed above use the standard Visual Messenger dialog boxes.

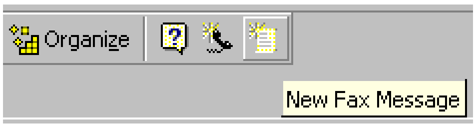
Actions are possible through :

- Outlook client "Actions" menu,

- dedicated toolbar button for New Voice Message



- dedicated toolbar button for New Fax Message



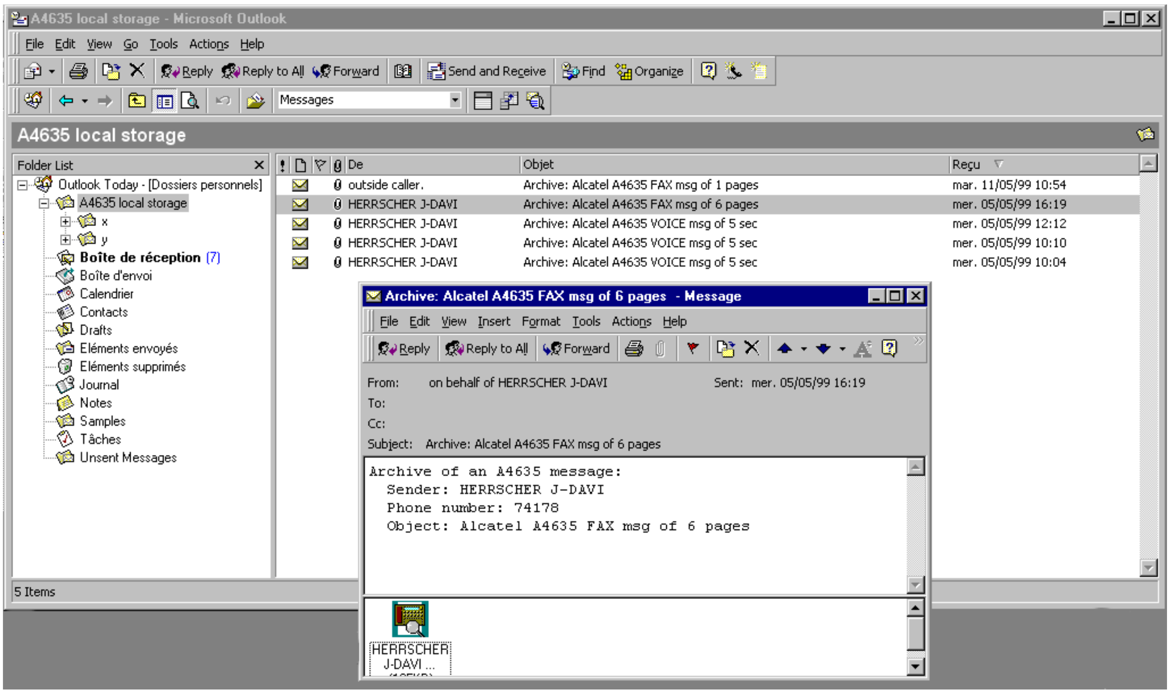
Export voice/fax messages to e-mail environment

To export a 4635 voice/fax message, drag and drop the 4635 message in any Outlook folder other than Inbox and Recycle Bin. In the destination folder, a mail message will be created with a WAVE/TIFF file attached.
According to the 4635 message type, the e-mail message will contain :

- a WAVE file for a 4635 voice message.
- a TIFF file for a 4635 fax message.
- a WAVE file and a TIFF file for a 4635 compound message.

Once exported, this message is manipulated as any other message from the e-mail environment.
For instance, this allows mailbox owner to forward a incoming message, left in Answering/Recording mode, to a recipient which is outside the corporate network, but can be reached through his I'net/X.400 Mail address.

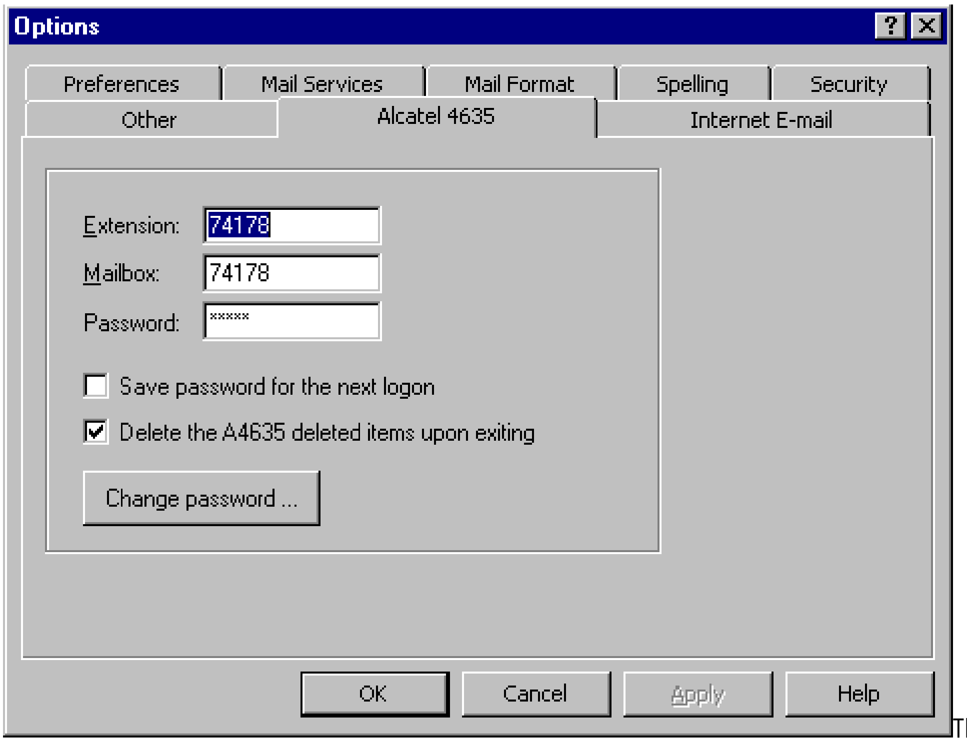
Note
Exporting messages creates a copy of A4635 message into e-mail environment.
User has still the possibility to play/print the 4635 message from the phone.
It's user policy to actually delete A4635 message from Inbox and therefore from message server.



Mailbox configuration - 1

The menu Tools/Options… displays the Outlook settings with Alcatel 4635 property sheet.
User can configure :

- Extension attached to the session, this can be done to use a particular set during the Outlook session, for instance a GSM set when on the move,
- A4635 mailbox to review during the session,
- A4635 mailbox password,
- Save password for next session ; this allows a silent start-up of Visual Messenger services,
- A4635 recycle bin policy

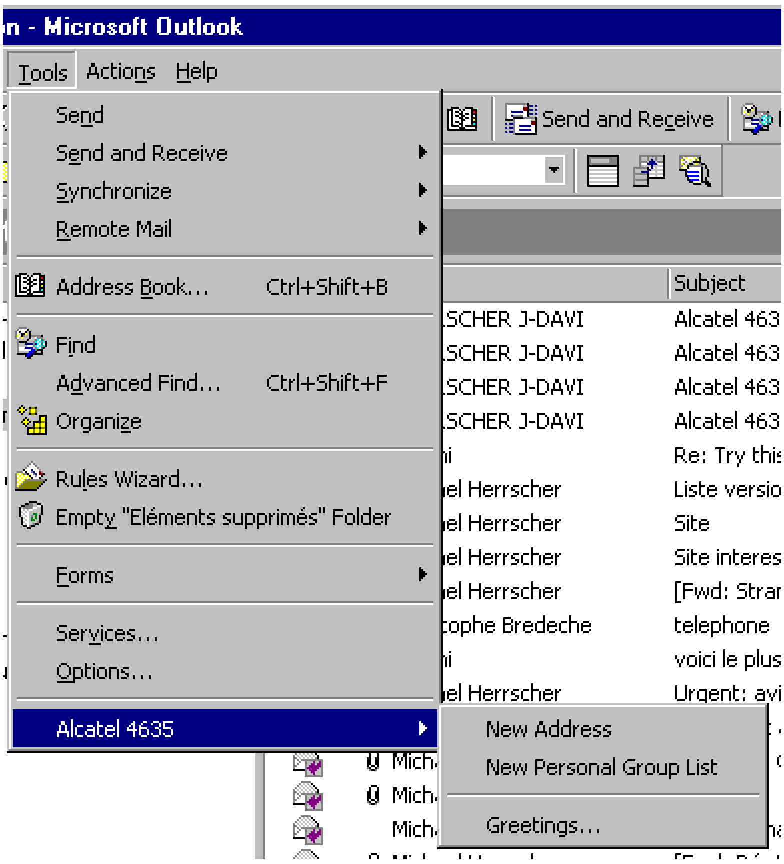


The user can add

Mailbox configuration - 2

The menu Tools/Alcatel 4635 allows the user to configure :

- New address in personal address book,
- New personal group list,
- Mailbox greetings



Help about extension

The Alcatel 4635 Help menu entry provides an Outlook extension specific online help.
Visual Messenger contextual help will continue to be available from each Visual Messenger screen.
Moreover, a link to the full Visual Messenger help will be available from the Outlook extension help.
The About Alcatel 4635 menu entry displays an about box with the version of each Visual Messenger components.

10.8 VISUAL MESSENGER FOR NOTES

10.8.1 Lotus Notes Market

Lotus Notes/Domino is choice for messaging and groupware for many corporates :

- Air France
- Alcatel
- Arthur Andersen
- Bayer
- Danone
- Eli Lilly & Co
- General Motors
- Henkel
- Lloyds
- Novartis
- Renault
- Société Générale
- Vivendi
- …

European market end '97 (source IDC)
5.1 M users

Lotus Domino, the server for Lotus Notes clients, is available on a wide range of platforms, starting from Linux or WNT PCs up to mainframes.

10.8.2 Services description

Logon procedure

Logon procedure has 2 roles :

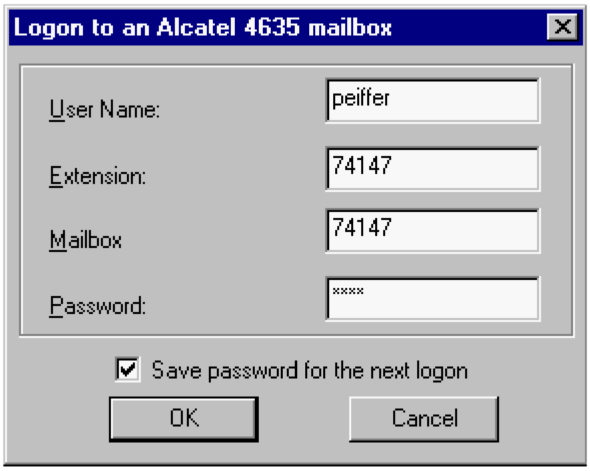
- start Visual Messenger services from Notes,
- attach a telephone extension, either internal or external (GSM), to the desktop session.

The logon procedure is system dependant. It is defined as follow:

By default, the user should enter the password each time Notes is launched.

- On a Windows NT machine, the user has the possibility to save the password with the help of the check box. The password is encrypted and is written in the registry. Logon information are NT user dependant. Therefore, each NT user has its own configuration.
- On a Windows 95 and 98 machine, the logon procedure is the same as Windows NT. However, each user will share the same configuration. In other terms, the logon procedure will be activated with the data of previous logon procedure.

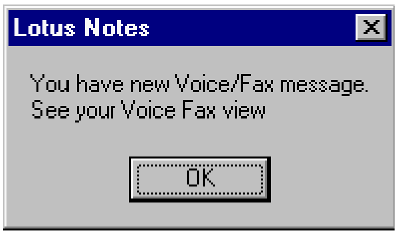
User has the possibility to skip password entry request by clicking the box
"Save password for next logon"
This is the recommended usage to have Notes silently starting Visual Messenger services.



Message waiting indication

Each time a new message arrives in A4635 message server, Visual Messenger for Notes extension
writes a voice/fax document in user's mail database. This document holds all information about the new voice/fax message. Voice/fax document holds also a link with the real voice/fax document stored into the A4635 message server.

On new voice/faxes messages Visual Messenger for Notes displays the following dialog box



Separate flows of messages – separate views

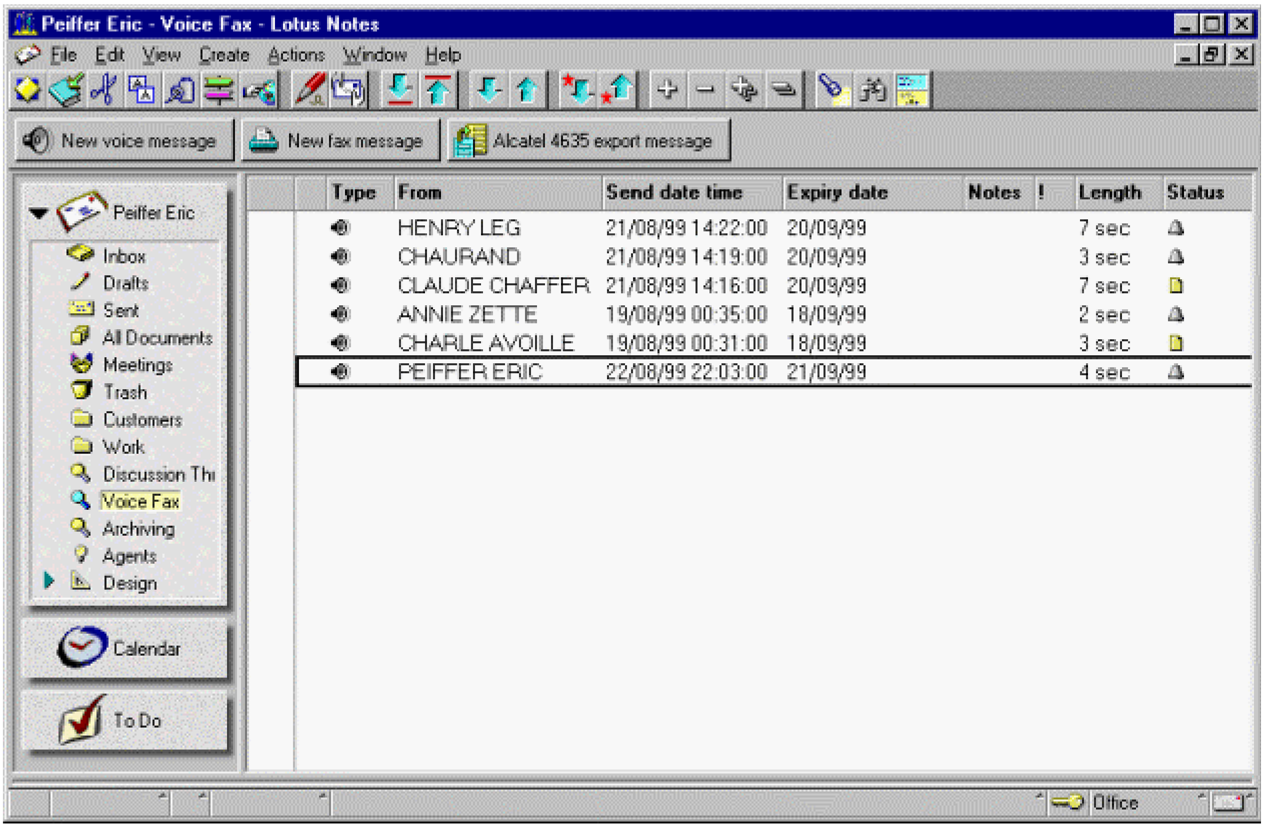
Visual Messenger for Notes was developed in the full respect of Notes rules :

- ergonomics,
- look and feel,
- client / server work-split (actions, storage, possible multiple directories)
- extensions facilities.

Unlike Visual Messenger for Outlook, Visual Messenger for Notes provides access to voice/fax messages through a dedicated VIEW.
This policy is natural in Notes environment where dedicated views are handling separates flows (messages, documents, workflow...)
Movements from one view to another is possible and will apply to voice/fax msg.
Remember that voice/fax messages ARE messages.

Voice/Fax view also allows the user to initiate following actions :

- New Voice Message,
- New Fax Message,
- Export message, see detailed description below.



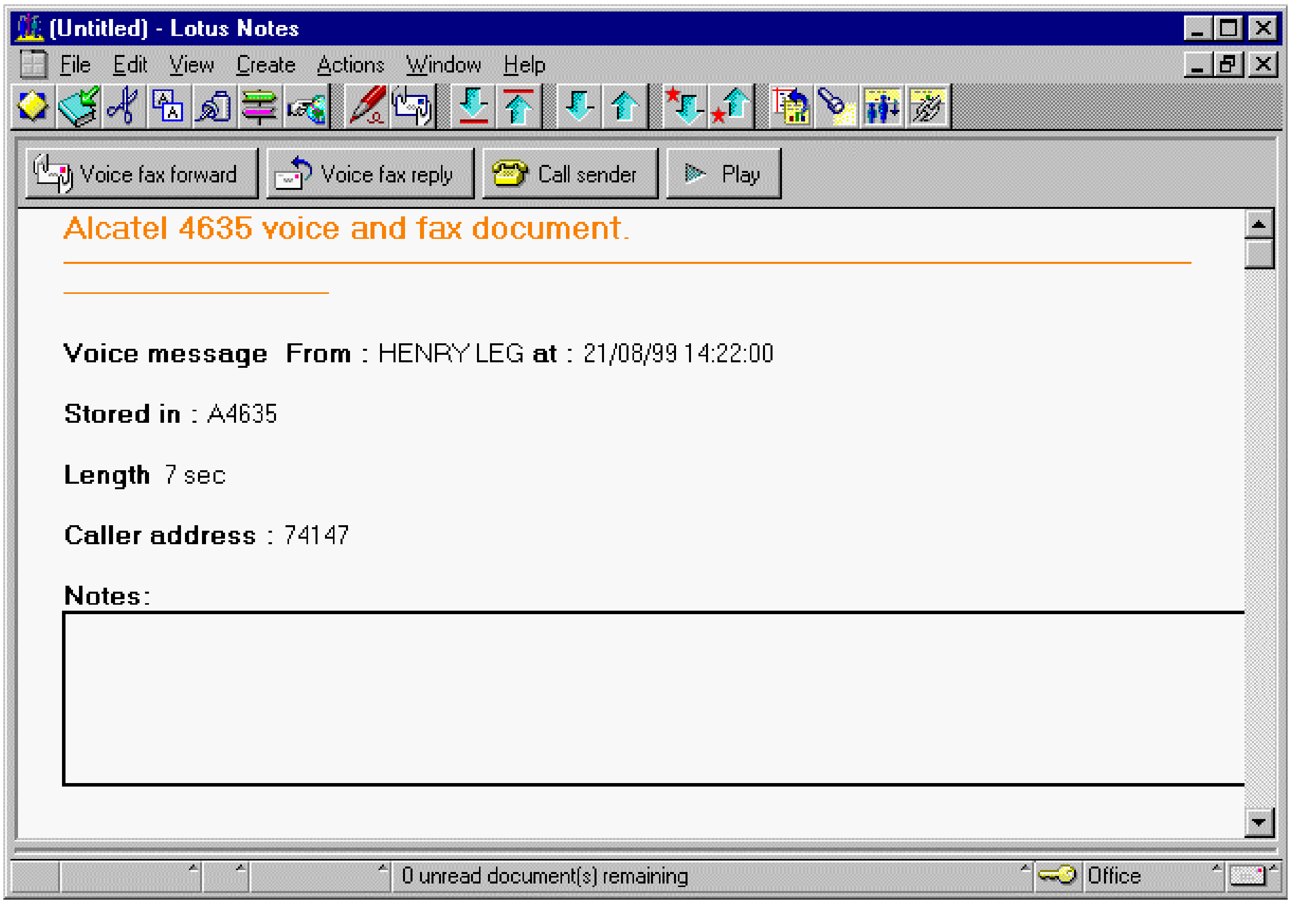
Voice/Fax Message control

PURE Notes ergonomics : an illustration of the golden rules of VM for Notes.
Voice messages ARE messages.
No matter the content, no matter the repository, A4635 messages are manipulated (played, viewed, sorted, searched, folded) with the same ergonomics as any other message.
Double-click on a voice message opens the document.
Most common actions are possible with single-button in the message toolbar.

Just like other messages, voice messages can have their SUBJECT and NOTE fields updated by user. This feature facilitates further message manipulation/storing/retrieval.
Of course, all what applies to Voice messages also applies to Fax messages.

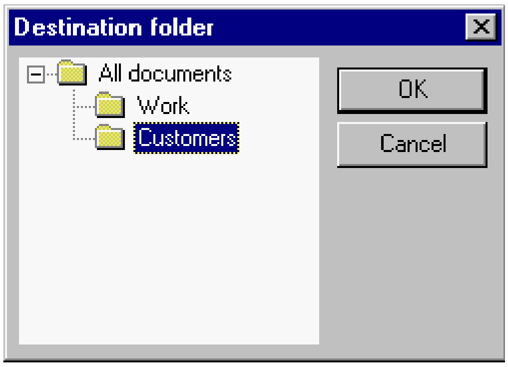
Voice/fax document view allows the user to perform the following actions :

- “Play”, (respectively view for a fax body part),
- “Reply”,
- “Forward”,
- “Call Sender”



Export voice/fax messages to e-mail environment

To export a message, user presses "Export Message" in the Action bar of the Voice/Fax view. Following dialog box is then displayed



User expands folders tree, selects actual folder where to export document and presses OK button.
In the destination folder, a mail message will be created with a .wav/.tif file(s) attached.
According to the 4635 message type, the e-mail message will contain :

- a .wav file for a 4635 voice message,
- a .tif file for a 4635 fax message,
- a .wav file and a .tif file for a 4635 compound message.

Once exported, this message is manipulated as any other message from the e-mail environment.

For instance, this allows mailbox owner to forward a incoming message, left in Answering/Recording mode, to a recipient which is outside the corporate network, but can be reached through his I'net/X.400 Mail address.

Note
Exporting messages creates a copy of A4635 message into e-mail environment.
User has still the possibility to play/print the 4635 message from the phone.
It's user policy to actually delete A4635 message from Inbox and therefore from message server.

Example of exported document

Mailbox access and configuration

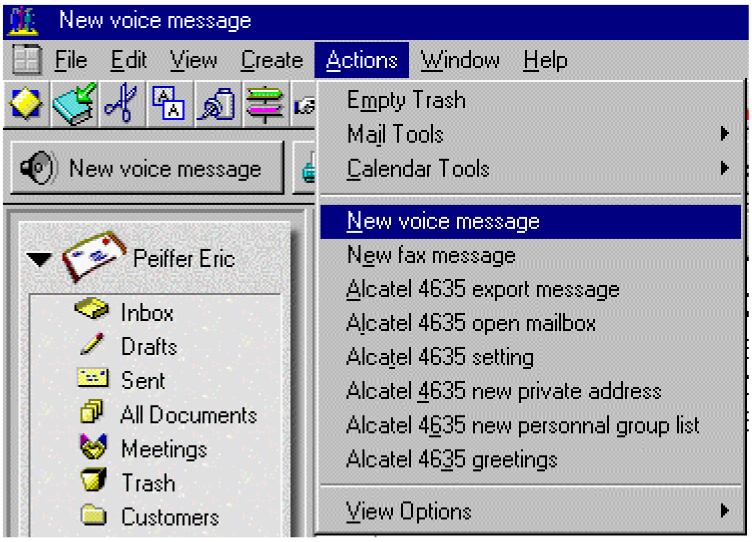
Mailbox access and configuration is done from "Actions" menu in Voice/Fax view.
Mailbox access allows user to :

- open a specific mailbox,
- modify extension attached to the session, for instance a GSM set when on the move

Mailbox configuration allows user to configure :

- New address in personal address book,
- New personal group list,
- Mailbox greetings,
- Mailbox password.

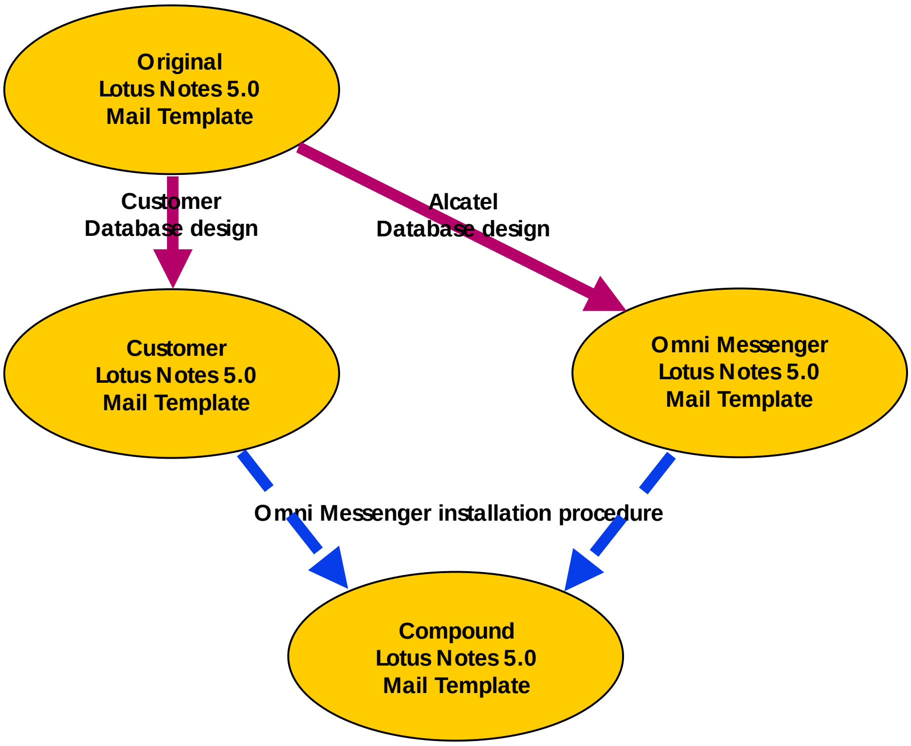
All actions listed above use the default Visual Messenger dialog boxes.



10.8.3 Mail templates concept

Lotus Notes is a database management system, with various usage, out of which electronic messaging, workflow and groupware.
Lotus Notes is shipped with a default database template, that can be tailored to customer needs.
Omni Messenger makes possible voice and fax services into Lotus Notes environment, by modifying the default Notes database template and adding the specific structures and semantics into it.

Omni Messenger is shipped with installation procedure that will allow on-site merge of both Customer and Alcatel designs into a single database.
This database will be applied to all Enterprise users which are Omni Messenger enabled.



Reader will refer to Omni Messenger installation section for all details of the procedure.

10.9 WEB MESSENGER

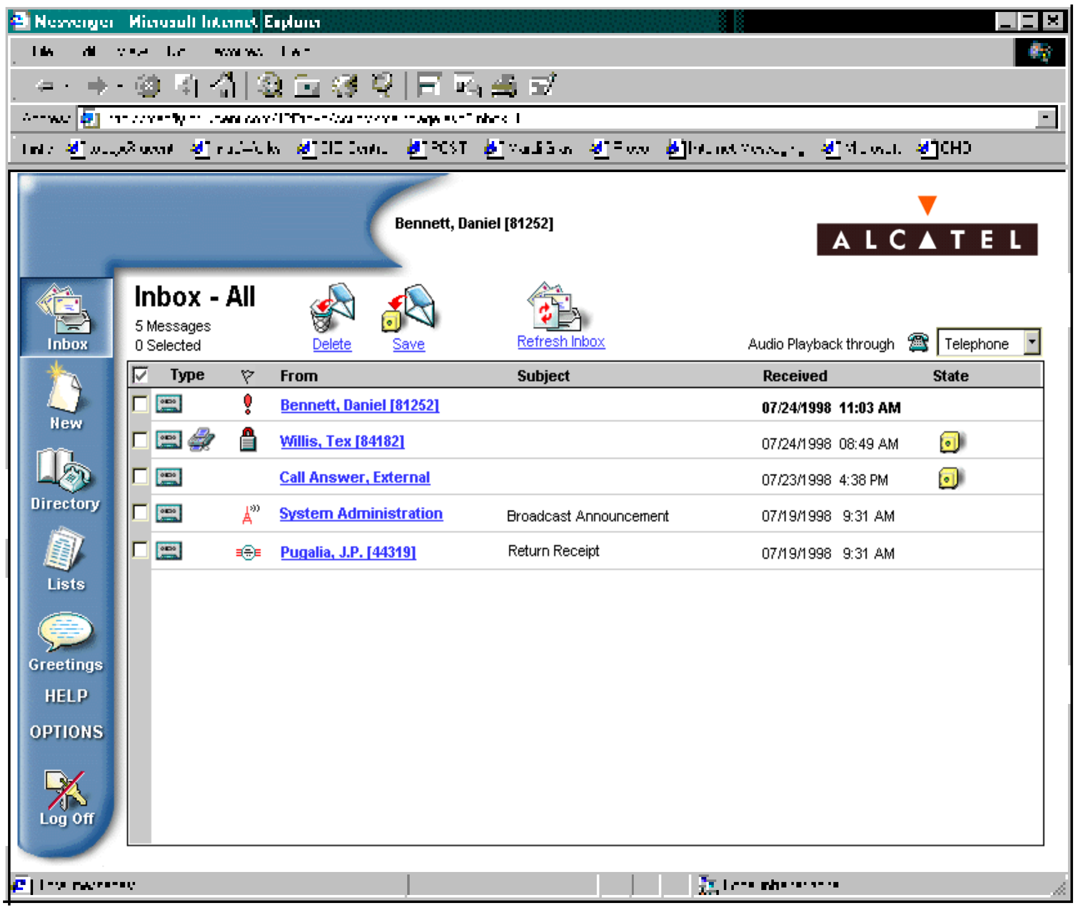
10.9.1 The WWW phenomenon

Web Messenger is Alcatel solution for companies who are deploying the usage of "thin clients" as the e-mail interface on the desktop.
Web Messenger allows users from these companies to have voice and fax services in addition to their e-mail.
Web Messenger is the appropriate solution for businesses who have not (will never ?) fully move to Microsoft infrastructure (e.g. research departments, universities, … and various UNIX addicts).
Web Messenger is the best solution for users on the move, using various workstations to retrieve their messages, either throughout the company or at home.
Any PC, Mac, Unix workstation and Alcatel WebTouch are web-enabled terminals.
Web Messenger makes logistics simpler for IS manager as far as there is no additional SW on users' workstations, (install, update, …)

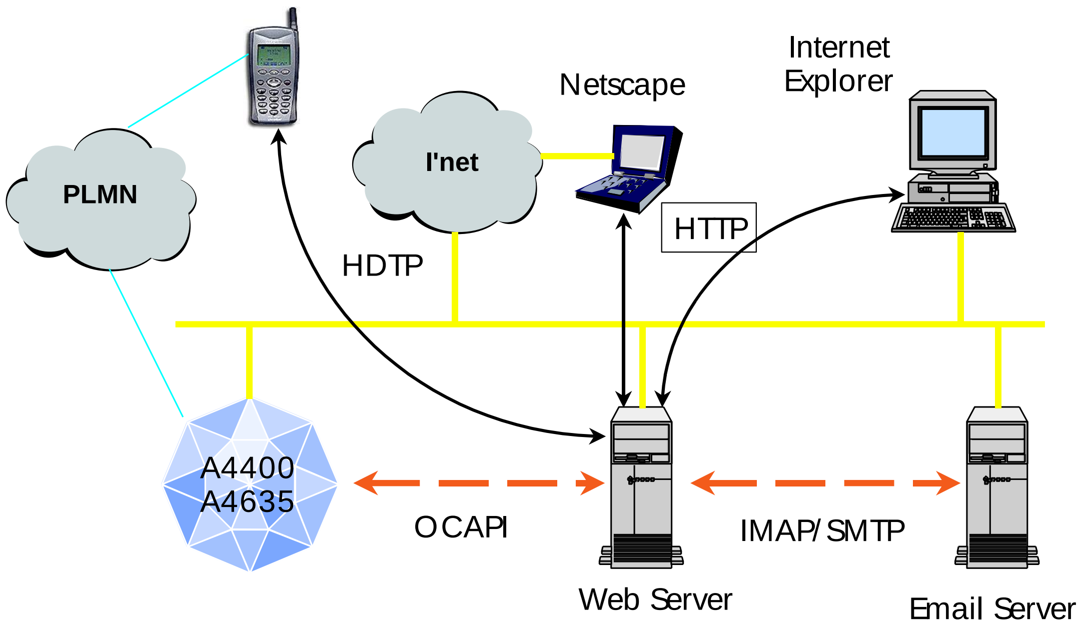
10.9.2 Services description

Web Messenger provides the user with the following services :

- Universal Messaging Client
- Display of Voice/Fax inbox
- Playback / record using multimedia PC
- Playback / record using the telephone
- Reply to voice message
- Call Sender
- Send a new voice message
- Mark message private to prohibit forwarding
- Voice messaging directory
- Set greetings
- Set mailbox password



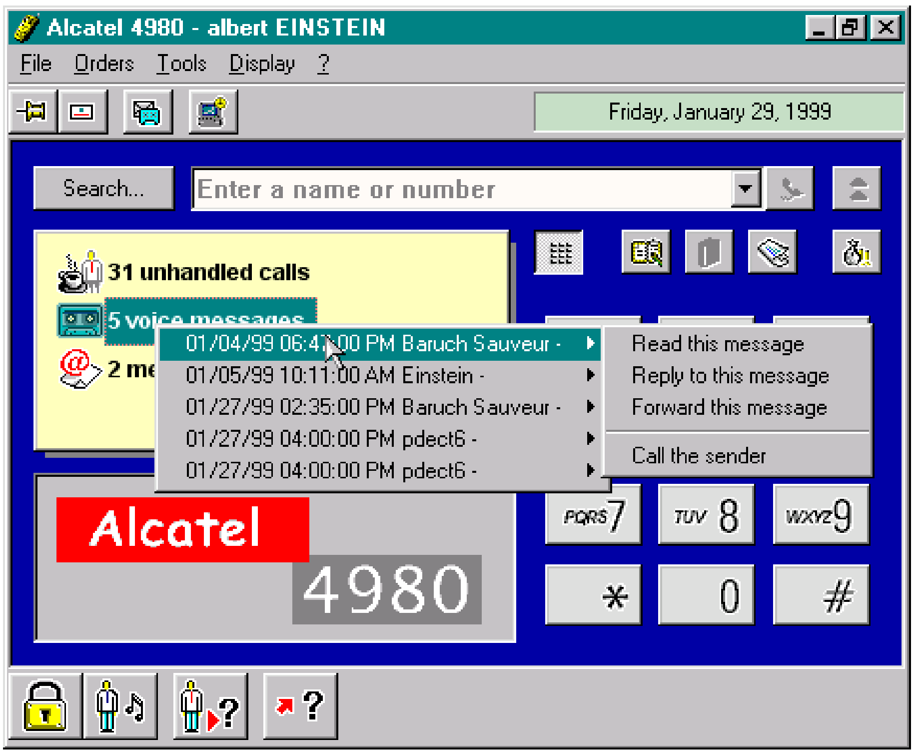
10.9.3 Architecture overview



10.10 VISUAL MESSENGER FOR 4980

Visual Messenger for 4980 targets customers with legacy e-mail or no e-mail integration needs.

Integration highlights
- login into 4980 silently starts Visual Messenger
- provide access to user's inbox from A4980 at-a-glance
- basic actions upon messages are done within A4980 client,
  - play/view message,
  - reply
  - forward
  - call sender



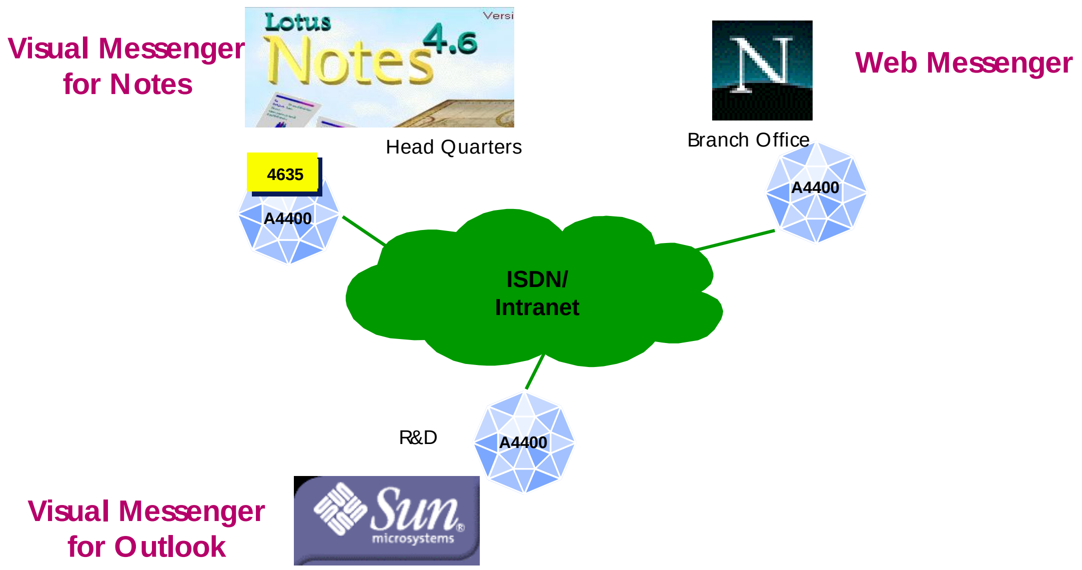
10.11 A4635 DESKTOP MESSAGING SOLUTIONS SUMMARY

Figure below highlights the services that are provided to users as well as environment constraints.

	Visual Messenger for Outlook	Visual Messenger for Notes	Visual Messenger for 4980	Web Messenger
Services				
Single login	Y	Y	Y	N
Single inbox VIEW	Y	N	N	N
Basic messaging functions				
Create/send	Y	Y	Y	Y (Voice only)
Review	Y	Y	Y (*)	Y
Reply	Y	Y	Y (*)	Y
Forward	Y	Y	Y (*)	Y
Call Sender	Y	Y	Y (*)	Y
Basic mailbox organisation functions				
Sort	Y	Y	Y	Y
Delete/Undelete	Y	Y	Y	Y
Fold,	Y	Y	Y	N
Drag 'n drop Voice & Fax messages into e-mail	Y	Y	N	N
Environment				
E-mail	MAPI/POP3/IMAP4	Lotus Notes	Any	Any
Desktop	WNT, W9x	WNT, W9x	WNT, W9x	WNT, W9x, iMac, Unix

(*) These actions are performed within 4980

4635 desktop messaging solutions can be combined over the Enterprise.
The example above shows a single 4635 (centralised voice mail topology) serving users from the whole 4400 network.
According to the needs and local e-mail infrastructure over the different nodes, IT manager can select the best 4635 desktop messaging client for every user.

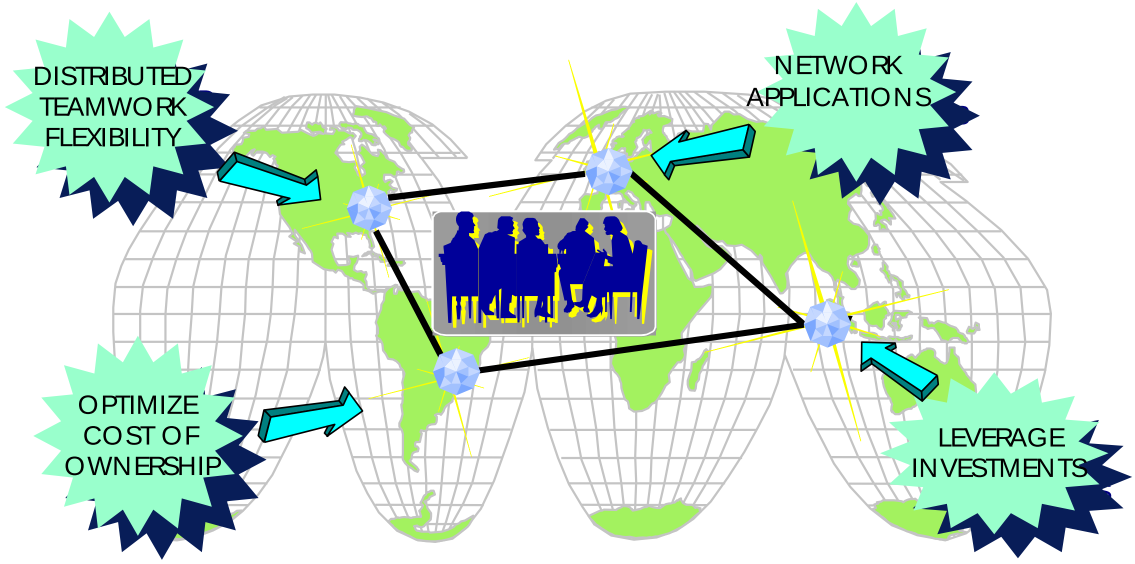


10.12 NETWORKING – DOMAINS

10.12.1 The needs

A voice mail network transparent to the company organisation

- single point of administration is 4400 one
- single directory is 4400 one



Leverage investment – Optimise TCO
4635 Domains is a response for large systems demands as well as existing systems that are necessary to grow in size, namely more than 64 ports. 4635 Domains will provide a transparent migration from a single system to a distributed network, seamlessly. A4400/A4635 integrated management avoids duplication of tasks and guarantees, up-to-date directories, networkwide.

Distributed teamwork flexibility
4635 Domains provides the highest transparency over the network with distributed teams. No matter where are located all persons of a team, 4635 Domains makes all of them appear local.

Network Applications
4635 Domains is the expected response to Global Automated Attendant.
Deploying Ubiquity over a network may have been creating some frustration to users when calling dial-by-name mailbox to reach somebody. Up to now, a mobile user could only reach subscribers located on the same node.
With 4635 Domains, this is no more a barrier.
Ubiquity user can call any dial-by-name mailbox in the Domain, 4635 will put him through the right person, regardless of his physical location.

10.12.2 Overview

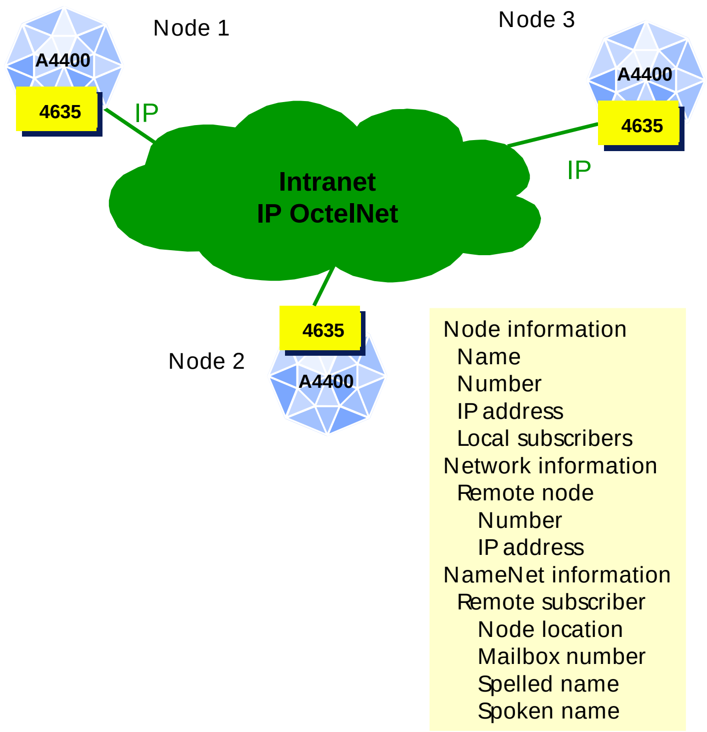
A4635 Domain solution offers organizations the ability to connect multiple servers while providing a single system appearance to end users. A4635 Domain extends voice messaging network and reaches an unsurpassed transparency of the underlying network to subscribers and external callers.

A4635 Domain provides mailbox-number addressing to users on a different server in the domain, without the need for an additional "network prefix." Any user can send a message to any other user in the domain and receive name confirmation, as if they were on the same server.
A4635 Domain also allows users to reply/call sender to telephone answering messages left by users from any server within the domain.

The fact that users reside on multiple servers is transparent to outside callers. In call answering mode, A4400 takes callers to the right mailbox on the right server. When using automated attendant, callers are routed to any extension on the A4400 network or any mailbox within the domain.

10.12.3 Architecture and implementation rules

A4635 Domain uses NameNet to locate users on other systems within the domain. Each A4635 server within the domain node is configured with the node number of every other server within the domain. NameNet entries for users on all other domain servers are stored as permanent entries within the NameNet directory of each domain node regardless of the user's class of service.
To locate a user within the domain for the purpose of sending a message or performing a transfer, the software first searches for local subscribers in local Node Information for a match. If no match is found, the software searches for remote subscribers in the NameNet Information that are associated with domain nodes. The system then performs the necessary operations to provide a single-server appearance.



Implementation rules :

- IP OctelNet strongly recommended, see performance section below
- Homogeneous numbering plan required,
- Unstructured numbering plan supported,
- 6 message servers per Domain, equivalent to 384 ports
- 40 000 NameNet directory entries.

Reader will note that Centralised 4635 makes Domains applicable to a very large scale.

10.12.4 Aria Domain Features

Mailbox Number Addressing and NameNet Confirmation

Users follow the same steps to send messages within a domain as they do to send messages on a single server. They record the message and enter the mailbox number or name of the recipient. If there are multiple users on different nodes with the same name, the sender hears a menu of choices (up to 9). In addition, NameNet can play spoken-name confirmation in the voice of the addressee (regardless of whether the message is addressed by spelling the name or by mailbox number). Spoken-name confirmation is also available when adding any domain member to any group list.

Reply/Call Sender to Messages Left in Telephone Answering

A4635 Domain allows any user in the domain to reply/call sender to a message left in telephone answering mode by any other user within the domain. A4635 accomplishes this task by detecting and storing the caller identification that is passed from A4400. This also allows the recipient to hear the caller's name and mailbox number as part of the message envelope information.

Automated Attendant

In an A4635 Domain, automated attendant applications, (including basic "call-by-name" mailbox), are supported across the entire domain. If the caller reaches the automated attendant and wants to dial an extension, the system performs the transfer as normal. A4635 Domain assumes that the called person's extension is the same as his or her mailbox number (NameNet tracks only mailbox
numbers, not extension numbers).
Voice directory Ubiquity service, now extends to the whole enterprise, domain-wide.

10.12.5 Administration

Administration of A4635 Domains is :

- Full set-up of IP OctelNet, including NameNet,
- set Domain attribute for each node in the network configuration

User creation/modification/deletion, is achieved through A4400 integrated management. Adding a user on one 4400 node, will generate associated mailbox creation on the appropriate node, and A4635 directory updates on all A4635 directories of the voice mail network.
A4635 directory, spelled names, will always be in line with A4400 directory, networkwide. Spoken names will be broadcast Domain-wide at creation time and after any modification.

10.12.6 Performance

A4635 Domain requires IP OctelNet for high-performance message delivery. Performance is dependent upon the network topology used to transport messages across the data network.

A4635 domain servers can send and receive messages at a twenty to 1 rate provided that the slowest transmission speed between the two systems is at least 10 Mbps, and congestion on the network is minimal. This allows a 1-minute message to be sent in about 3 seconds. See IP OctelNet description and examples above.
In the event that the data networking connection is lost between servers within the domain, analog fallback is supported. However, Alcatel does not recommend implementation of analog fallback because of the impact to message delivery performance between domain servers.
Analog message delivery is up to 20 times slower than digital message delivery.

10.12.7 Requirements

To create an Aria Domain, there are a number of configuration requirements that must be met:

- The mailbox numbers must be unique across the domain, and all mailbox numbers in the domain must be the same length. Extensions must match or be a subset of the mailbox number.
- If any applications exist in the domain, all mailboxes for any one application must be on a single server. If an application is needed on multiple servers, it must be duplicated on each server.
- System and bulletin broadcast must be issued on each server separately.

10.12.8 Limits of A4635 Domains

A4635 Domain offers a full-featured, easy-to-use, cost-effective very large system solution, and by extent, Enterprise-wide solution. However, it is important to know that there are still some differences between A4635 Domain and a single server.

- Each A4635 is on different A4400 and therefore has his own hunting group. This means when a user on a A4635 message server outside of the domain sends a message to a user in a A4635 Domain, the network address entered must distinguish on which server the mailbox resides. This can be simplified by using a special numbering plan or, more convenient for the user, addressing recipient with dial-by-name.
- There is no real-time extended absence greeting when addressing a message to a user on another A4635 in the domain. This situation is handled as with networked servers. The message is sent and then the originator receives a notice that the recipient's mailbox had an extended absence greeting.
- Just as with networked message servers, there is no ability for a user to utilize the Locate Message Sent feature on a message sent to another server in the domain. However, users can still take advantage of confirmation of receipt and notification of non-receipt.

10.13 NEW HOTEL OFFERING

Please refer to A4400 Hospitality Services section.

10.14 OMNI MESSENGER SOFTWARE DELIVERY POLICY

Omni Messenger Software delivery consists in a single CD–Rom which contains :

- A4635 Access Server software and documentation,
- A4635 Administrator software and documentation,
- Visual Messenger client software and documentation, (1)
- Outlook extension dynamic link library and the Form application software,
- Notes extension dynamic link library, Notes mail templates with voice/fax capabilities for the R5.0 an R4.6 versions, the supervisor program,
- Web Messenger server SW and documentation (2)
- All installation/uninstallation programs

Notes.

(1) Visual Messenger, without any additional component suits for integration with A4980 R2.0

(2) Web Messenger software and documentation is not part of SW delivery as of 01/09/99.

The SW delivery - ONE single CD-ROM containing all server and client SW, and pricing policy - ONE single price/user - of 4635 desktop messaging solutions, is the best guarantee for IT manager to deploy the right tool at the right time for the right user.

Try n' Buy package, 5 licenses free, with all 4635 servers

The try n' buy package is useful to create awareness of the value of 4635 desktop messaging solutions. IT manager will immediately see the value of voice/fax messaging for users, and understand the limited effort for a global deployment.

10.15 OMNI MESSENGER SOFTWARE INSTALLATION

10.15.1 Installation rules and recommendations

Access Server SW is supported on :

- Windows NT4 Workstation, standard Alcatel shipment when delivered with PC-Server
- Windows NT4 Server

Access Server SW can run concurrently with A4980 and A4740.
Alcatel recommends not to install Access Server with non-Alcatel SW.

Client SW is supported on :

- Windows NT4 Workstation,
- Windows 95,
- Windows 98

System administrator :
- must install Access Server SW,
- optionally install Web Messenger Server SW,
- can deploy any combination of client SW.

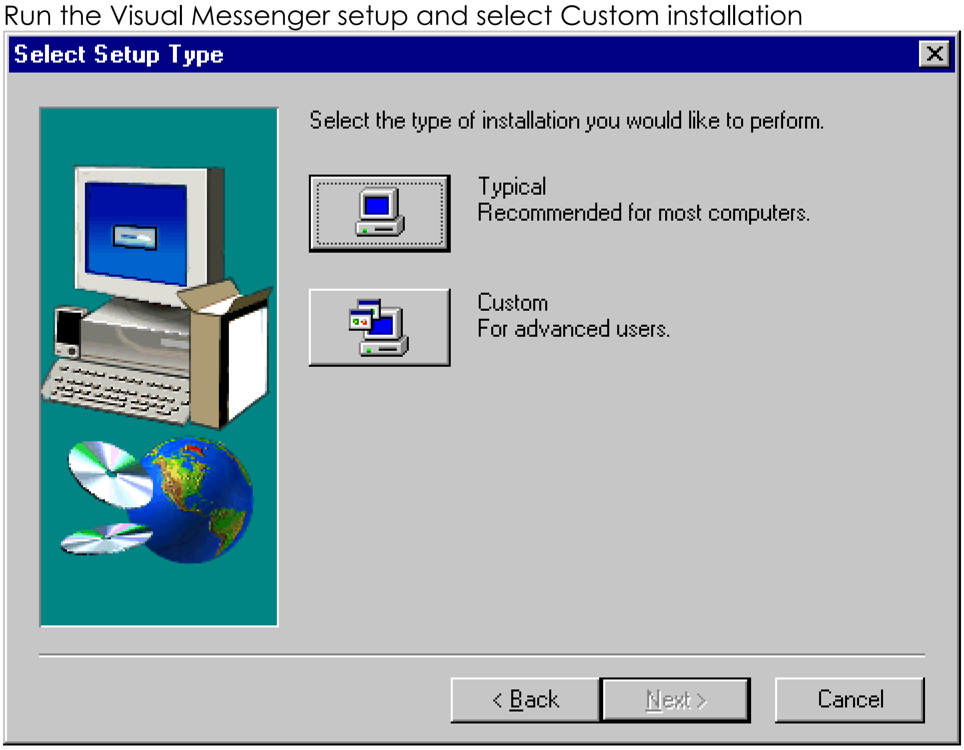
Alcatel recommends for client SW :
- local installation on every laptop PC, to allow mobile users to work off-line.
- remote execution mode, directly from network file server, for desktop PCs

10.15.2 Example of Visual Messenger for Notes installation

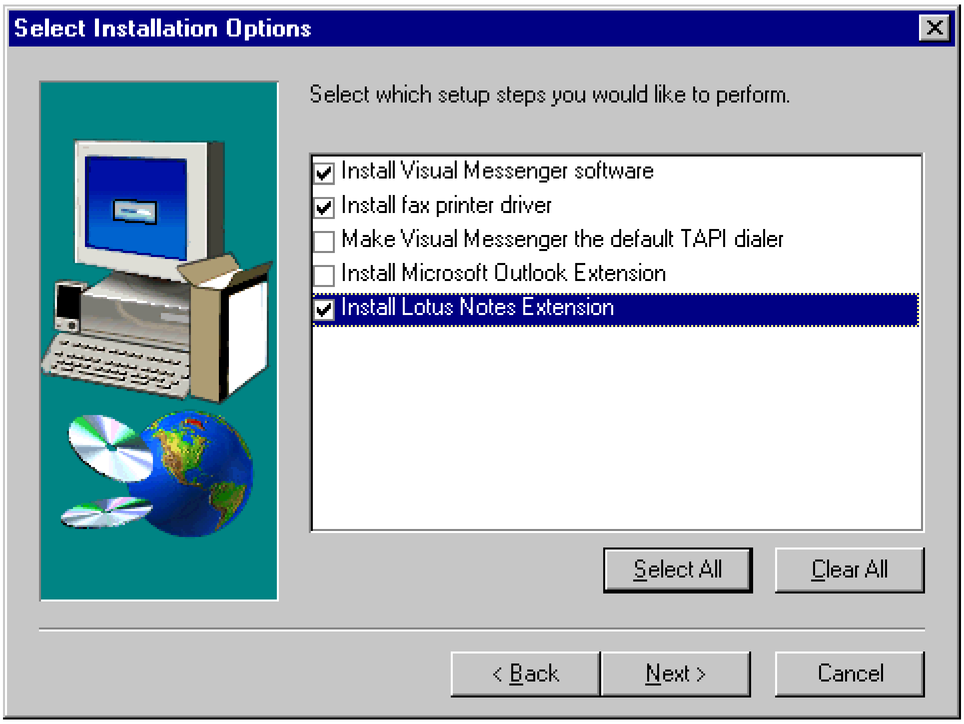
Client installation rules
- Under WNT4, user must have local administrator rights,
- Notes R 4.6 or Notes R5.0 must be already installed

Actual installation

Run the Visual Messenger setup and select Custom installation

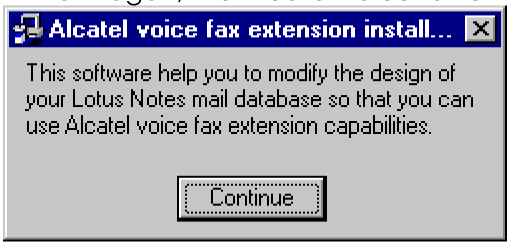


Click Visual Messenger, fax printer driver and Lotus Notes extension

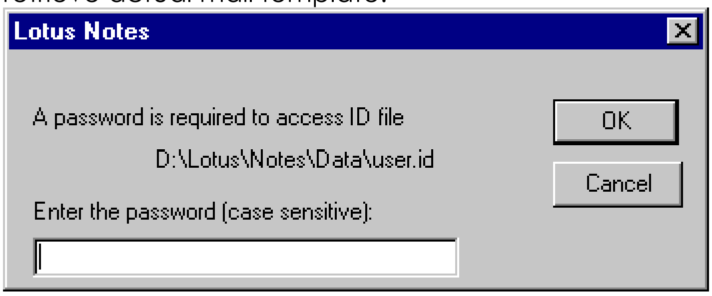


To complete the installation of the Notes Extension, PC must be restarted.

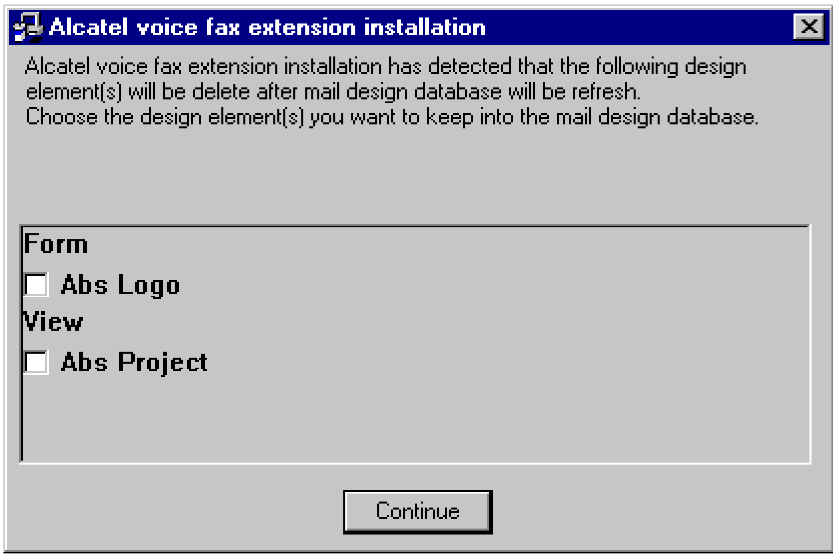
At next logon, the Alcatel voice fax extension installation program starts up



This program prepares the Notes mail database to be inherited by the Alcatel mail template. Enter the Notes user password, so that installation program can open Notes database to retrieve actual mail template.



Installation program checks if some design elements will be deleted after mail design database will be refresh. Set the checkbox corresponding to the design element you will keep in the design
and press **Continue**.



When installation program is completed, a new database template is created.
The last step you have to do is to refresh the design of the mail database:
Start Notes, select the command **Refresh Design…** in the menu File – Database.
The Design of the mail database will be refresh with design of the new inherited template.

User now has voice and fax services within his Notes client.

10.15.3 A4635/A4400 NETWORK RELEASE

The table below shows the feature-level provided by combination of various components.

Service	Visual Messenger	A4635	A4400
Omni Messenger services			
Visual Messenger for 4980	2.003	2.06	3.0 (C1.520)
Visual Messenger for Outlook	2.003	2.06	3.0
Visual Messenger for Notes	2.003	2.06	3.0
Network services			
Domains	Not relevant	3.0.1	3.0
Hotel services			
8 languages	Not relevant	2.2.1	3.0
Japanese language	Not relevant	3.0.1	1.x
New mailbox ergonomics	Not relevant	3.0.1	1.x
Room move	Not relevant	3.0.1	3.1 (C1.702)
Multiple wake-up	Not relevant	3.0.1	3.1

10.16 ORDERABLE ITEMS

All orderable items where introduced with A4635 R2.
A4635 R3 brings new services out of the existing references.

Hardware base packages

3BA 00199 AC **4635J Base H/W**
Including VPS35 Board, IDE Hard Disk, V24 Cable & Ethernet Cable
3BA 00237 AA **4635H Base H/W**
Including VPM-35 Board, SPA3 Board, IDE Hard Disk, V24 Cable & Ethernet Cable

Hardware spare parts

3BA 23118 AB **4635J VPS35 Board w/o HD**
Voice Processing Board for A4635J
3BA 73012 AA **4635H VPM35 Board w/o HD**
Voice Processing Board for A4635H

Software licenses

3BA 09234 AA **4635 Omni Messenger BSL 32 u**
3BA 09235 AA **4635 Omni Messenger ASL 32 u**
Base and Additional Software Licence 32 users
3BH 11470 AA **4635 Omni Messenger BAS**
CD-ROM including desktop and server software & documentation of Omni Messenger, namely, Visual Messenger, basic, Outlook/Notes version and Web Messenger
3BA 09236 AA **4635 IP OctelNet BSL 2 Ports**
3BA 09237 AA **4635 IP OctelNet ASL 2 Ports**
IP OctelNet requires OctelNet
3BA 09238 AA **4635 Global Message Redundancy**
System license, applicable to 4635 H only

10.17 MIGRATION

10.17.1 Existing configurations

A4635J with R1.x
- VPS35, 16 MB, IDE disk drive on board

A4635J with R2.x
- VPS35, 32 MB, IDE disk drive on board

A4635H with R1.x
- VPCPU, 16 MB
- MSB with SCSI disk drives
- SPA3

A4635H with R1.x, (introduction Q1 '98)
- VPCPU-1, 16 MB
- MSB-I with IDE disk drives
- SPA3

A4635H with R2.x
- VPM35, 32 MB, IDE disk drive on board
- MSB-I with IDE disk drives, when more than 1, one disk in total
- SPA3

10.17.2 Migration examples

From 4635J Rel1 to 4635J Rel2 (1)

Reference	Description	Qty
3BA27020AA	DRAM 16MB	1
3BA28110AA	4635 Ethernet Cable	1

From 4635J Rel1 to 4635H Rel2

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA28110AA	4635 Ethernet Cable	1
3BA73006AA	SPA3 Board	1

From 4635H Rel1 with VPCPU-1, IDE Hard Disk and MSBI Board to 4635H Rel2

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA28110AA	4635 Ethernet Cable	1

From 4635H Rel1 with VPCPU, 1 SCSI Hard Disk and 1 MSB Board to 4635H Rel2 (2)

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA27013AB	IDE Hard Disk	1
3BA28110AA	4635 Ethernet Cable	1

From 4635H Rel1 with VPCPU, 2 SCSI Hard Disk and 1 MSB Board to 4635H Rel2 (2)

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA27013AB	IDE Hard Disk	2
3BA73010AA	4635H MSBI Board w/o HD	1
3BA53047AA	ATB2 Card	1
3BA28110AA	4635 Ethernet Cable	1

From 4635H Rel1 with VPCPU, 3 SCSI Hard Disk and 2 MSB Board to 4635H Rel2 (2)

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA27013AB	IDE Hard Disk	3
3BA73010AA	4635H MSBI Board w/o HD	1
3BA53047AA	ATB2 Card	1
3BA28110AA	4635 Ethernet Cable	1

From 4635H Rel1 with VPCPU, 4 SCSI Hard Disk and 2 MSB Board to 4635H Rel2 (2)

Reference	Description	Qty
3BA73012AA	4635H VPM35 Board w/o HD	1
3BA27013AB	IDE Hard Disk	4
3BA73010AA	4635H MSBI Board w/o HD	2
3BA23158AA	ATB3I Card	1
3BA28110AA	4635 Ethernet Cable	1

Notes.

(1) The new A4635J boards, ref. 3BA 23118 AB, are delivered with 32MB

(2) VPCPU supports SCSI disk only, VPM35 supports IDE disks only. VPCPU-1 supports all formats. To convert a system from SCSI disks into IDE disks it is mandatory to use a kit VPCPU-1 + MSBI.

11. ALCATEL 4615

11.1 INTRODUCTION

A4615 Product Description intends to provide the reader with a comprehensive document giving the main features of the product as well as customer benefits. Reader can be any of :

- a A4400 distributor, (Alcatel DSU, independent distributor, Telecom Operator distributing A4400)
- a A4400 pre-sales engineer, to serve as a model for request for proposal,
- any A4400 expert, to learn more about the brand new voice messaging companion

A4615 Product Description is organised as follows :

- product positioning explains the place of A4615 in the A4400 messaging solutions range,
- general overview gives the fundamentals of the product,
- features description section is a complements to the official document, (also applies to provisioning level section). To illustrate ease of use of A4615, this section contains :
- Caller's Telephone User Interface flow
- Subscriber's Telephone User Interface flow
- Default (factory installed) Automated Attendant flow
- A4615/A4400 integration section will go through the following items :
- A4400 software delivery policy,
- mailbox management and audit
- localisation concerns,
- A4615 orderable items

11.2 PRODUCT POSITIONNING

The purpose of A4615 is to complement the **A4400 Compact** commercial offer of voice messaging solutions in the low–end market, to broaden voice messaging acceptance.
For easy read purposes, A4400 Compact will be named A4400 below.

A4615 at-a-glance :

- positioned as entry level product in the A4400 market segment in addition to the already existing voice messaging systems A4635H/A4635J
- addresses mainly customers who need '24 Hours Call Answering / Call Completion' functionality,
- available for stand-alone (no redundancy) CPU3-2 based 4400 systems,
- available for 4400 configurations with numbering plan up to 4 digits,
- provides either 2 ports, (recommended up to 30 subscribers,) or 4 ports above. Selection of the number of ports is done through capacity-on-demand licenses with Actis.
- supports up to 128 mailboxes,
- 320 min recording time,
- multilingual voice guidance up to 4 simultaneous languages,
- Automated Attendant and Information Service (Audiotex) are available in the basic package.
- Ultimate integration with A4400. A4615 is a daughter board of A4400 CPU3-2.Beside direct connection to PCM bus full digital interfacing via 'host interface' with A4400-CPU through ABC-A.
- A4400 integrated management for mailbox handling (create/modify/delete and audit). This applies to 4615 in hotel configurations and provides automated check-in/check-out.

- Easy and remote maintenance, in-line with A4400 concept. A4615-SW, voice prompt languages, country default settings are fully part of the global A4400 SW delivery and stored on hard disk. Software and language updates can be downloaded from A4400 to A4615,
- voice–guided management from a set, in 4 languages
- A4615–MMC : PC based management through a graphical interface over A4400 V24 interface :
- global system customisation,
- mailbox management,
- simple and full configuration of Automated Attendant and Audiotex applications,
- statistics about mailbox usage and system memory management,
- faults and diagnostics.

4615-4635J Comparison matrix

Features / Provisioning level	4615	4635J
Telephone Answering	Y	Y
Voice Messaging	Y	Y
Automated Attendant	Y	Y
Hotel	Y	Y
Record-on-Line		Y
Networking		Y
Centralised Voice Mail		Y
Reflex sets integration		Y
Ubiquity capabilities		Y
LAN backup		Y
Desktop interface / e-mail integration		Y
CPU	CPU 3	All
Numbering plan	4	10
Ports	2 - 4	2 - 8
Mailboxes	128	15000
Voice Storage (global / per message)	320 mn / 4 mn	40 h / 3 h
Simultaneous languages	4	8

11.3 GENERAL OVERVIEW

A4615 R 3.0 is physically located on a daughter board of A4400 CPU3 Step 2, VMU-OBCA, which combines two functions : VMU (Voice Mail Unit) and OBCA (Optimised B-Channel Access).
The VMU–OBCA daughter board provides :

- 4 voice mail accesses (VMU)
- 3 full duplex channels at 64 kbps (as OBCA today). It allows remote management and offers access to applications inside the A4400-CPU 3.

Only one A4615 is possible in a A4400 system.
A4615 cannot be used in addition to any other voice messaging system (4635/4630, 4620 or other VPS-interfaced system).

Main hardware characteristics of A4615 can be summarised as follows :

- A4615 software is running on two Motorola 56156 DSP :
- main DSP handles the Real Time Operating System (Spox), voice compression, file system, and application SW (Voice Messaging, Automated Attendant and Audiotex)

- slave DSP handles tone detection (DTMF, silence, fax carrier tones), communication with A4400 CPU over ISA bus, namely ABC-A interface and download of software, voice prompts and CDD (country dependant data)
- A4615 Voice storage is done on-board with NAND Flash memory (32 MB)

OBCA will not be described in this document as far as it is already fully documented in various A4400 literature, (Product Description, Reference Product Description, Technical documentation).

11.4 FEATURES DESCRIPTION

This section gives the product features at-a-glance.
It is organised as follows :

- Voice Messaging,
- Automated Attendant,
- Information Service (Audiotex)

Several features will be given more details.
Section ends with caller/subscriber and default automated attendant interface flow.

11.4.1 <u>Feature list</u>

- Voice Messaging
- Mailbox types
- Standard mailbox,
- ‘Answer only’ mode mailbox,
- Guest mailbox (Hotel/Office),

- Standard mailbox offers:
- Direct access to mailbox via consultation key
- Direct access to mailbox from anywhere
- Distribution lists
- Broadcast message
- Leave, erase and re-record messages
- Leave and replay a message
- Pause (continue) during message recording
- Send message to a distribution list
- Check receipt of messages
- Review, replay, erase, archive new messages
- Fast forward, pause and rewind during message review
- Review, replay, erase archived messages
- Send copy of message with introduction
- Send copy of message with introduction to a distribution list
- Reply a message
- Time stamp
- Notification by message waiting LED (if supported by subset)
- Notification by outcalling (2 modes): Message signalling or message delivery
- Notification call to pager
- Dial by name
- Personal options:
- Record / update personal greeting(s) / select 3 personal greetings
- Record / update name
- Modify password
- Create distribution lists and edit list members
- Select language for voice guidance
- Program notification settings for out-calling (telephone number, schedule, turn on/off)
- Switch mailbox in ‘Answer only’ mode and record 'Answer only' greeting
- Common mailbox
  (caller hears as greeting the ‘common mailbox announcement’ if no personal greeting is

recorded for this mailbox and the mailbox is configured for functions short cut, default function for AA or function for non-existing mailbox)

- Access to internal Info-service (Audiotex)
- Statistics related to voice mailbox accesses
- Answer only mode:
- Caller can listen to the 'Answer only' greeting, but no message recording is possible.
- A standard mailbox can be switched in this mode by mailbox administration or by the mailbox owner (personal options).
- Guest mailbox Hotel:
- Easy to consult & use (= no password from room phone), only for consult from other or external phone; message automatically played; reduced options only like skip, replay, delete, get time stamp information).
- Check-in: Guest mailbox automatically allocated to room phone, same password & language as allocated to phone; mbx. ready for leaving messages, Guest may only record his name
- Check-out: Guest mailbox closed for leaving messages and de-allocated (= deleted) if no more messages available. If messages available, Guest mailbox also closed, mailbox can be consulted at least one more hour.
- Guest mailbox Office (e.g. for temporarily staff):
- Like Guest mail box Hotel, but two distinctions: mailbox has to be configured; mailbox owner has to enter his password during first mailbox access, he may record his name
- Easy to consult & use:, no password needed to consult from own phone, only for consult from other or external phone; message automatically played; reduced options only like skip, replay, delete, get time stamp information, no personal options.
- Automated Attendant:
- 2 level AA (main and sub menus). Each level with up to 9 (+Operator) choices per menu
- Default AA with default announcements
- 2 separate company greetings / tree structures for business respectively after business hours, switch-over: time-dependent and manually (via voice guided administration menu)
- Interactive multilingual voice guidance in up to 4 languages
- Dial by name
- Main menu and sub-menu: digit association free customisable:
- Transfer to operator
- Transfer to subscriber (to a predefined extension)
- Infobox (transfer to a predefined infobox / audiotex)
- Leave a message (mailbox number has to be entered)
- Free dialling (extension number has to be entered)
- Transfer to mailbox (to a predefined mailbox)
- Release (Good bye)
- Submenu (applicable only in main menu)
- Fax switch (transfer incoming fax to a fax extension)
- Statistics related to Automated Attendant
- Info-Service (Audiotex)
- Information service for external callers, infoboxes accessed through Automated Attendant,
- Information service for internal use, company confidential information accessed from subscriber mailbox main menu
- Chaining of infoboxes
- Statistic for number of infobox accesses (via voice guided administration menu)

11.4.2 Distribution lists

Distribution lists in A4615 are managed as follows :

- general distribution list - 1 per system
  automatically updated at each mailbox creation/modification/deletion,
  available to broadcast messages to all subscribers,
- standard distribution lists - 50 per system
  A pool of 50 DL is available to all users.
  Distribution list management.
- DL owner
  DL owner is the one user to create it,

DL owner rights are :

- record name,
- add/mod/del users inside DL,
- remove DL
- DL usage

all A4615 users, including DL owner, can use a Distribution List.

11.4.3 Busy greeting

Busy greeting will be played by A4615 to inform caller that the person he tries to reach is currently on the phone and not away from his desk.
Busy greeting intends to limit the number of calls diverted to voice mail without message left.
Busy greeting on A4615 is available for all subscribers and does not require them to record a personal greeting for this condition. Existing system prompts, (with optional subscriber recorded name) are played to callers.

The busy diverted call is announced to caller with :

"The called extension is busy. You have reached the mailbox "123". Please leave your message after the tone and press # (hash) when you have finished"

Or with :

"The called extension is busy. You have reached the mailbox of "Lalo Shiffrin". Please leave your message after the tone and press # (hash) when you have finished"

if the name for mailbox 123 (=Lalo Shiffrin) is recorded.

11.4.4 'Answer-only' greeting

'Answer-only' mode can be selected by A4615 subscriber, any time he considers he cannot take and act upon messages.
In such conditions, A4615 prompts user to record a specific 'Answer-only' greeting prior to toggle mailbox from Answer/Record to 'Answer-only' mode.
The first time subscriber uses this feature, he MUST record a specific 'Answer-only' greeting.
When subscriber toggles back from 'Answer-only' to Answer/Record mode, previous active greeting is retained.
When 'Answer-only' greeting already exists, and user want to go 'Answer-only' mode, A4615 prompts him to keep it or discard and re-record a new greeting.

11.4.5 Dial-by-Name and keypad mappings

Dial-by-Name feature is used by callers and subscribers to :

- leave a message to a subscriber,
- send a message to another subscriber,
- reach a person

when mailbox or extension number is not known.

Dial By Name feature uses the numerical keypad with its alphabetical relation based on DTMF codes (i.e.: digit 2 for letter A,B,C – digit 3 for letters D,E,F and so on ...) to enter at least the first 3 letters (and maximum 8 letters) of the user's name for the mailbox/extension he is looking for.
A4615 confirms the mailbox names matching to with the current letters entered with the keypad.
As soon as the A4615 finds a unique name, the user has to press the # (hash) key to confirm the selection.

An unique intelligent algorithm is implemented in order to manage simultaneously ITU E.161 and European keypads.
This algorithm avoids to dedicate dial-by-name to a single keypad configuration.

11.4.6 Wake–up

Wake–up feature in A4615 is slightly to the implementation of A4635.

The user interface for the Alarm Call is the same as for A4400/A4635. The announcements are different due to lack of existing phrases on the A4615.

Alarm Call setting

The extension owner dials the alarm call programming prefix.
He is prompted by the A4400 for wake–up time and destination phone number.
The caller is connected to A4615, which plays the confirmation announcement :

- ticking watch ... (long)
- announcement of the defined time (in the caller's subset language)

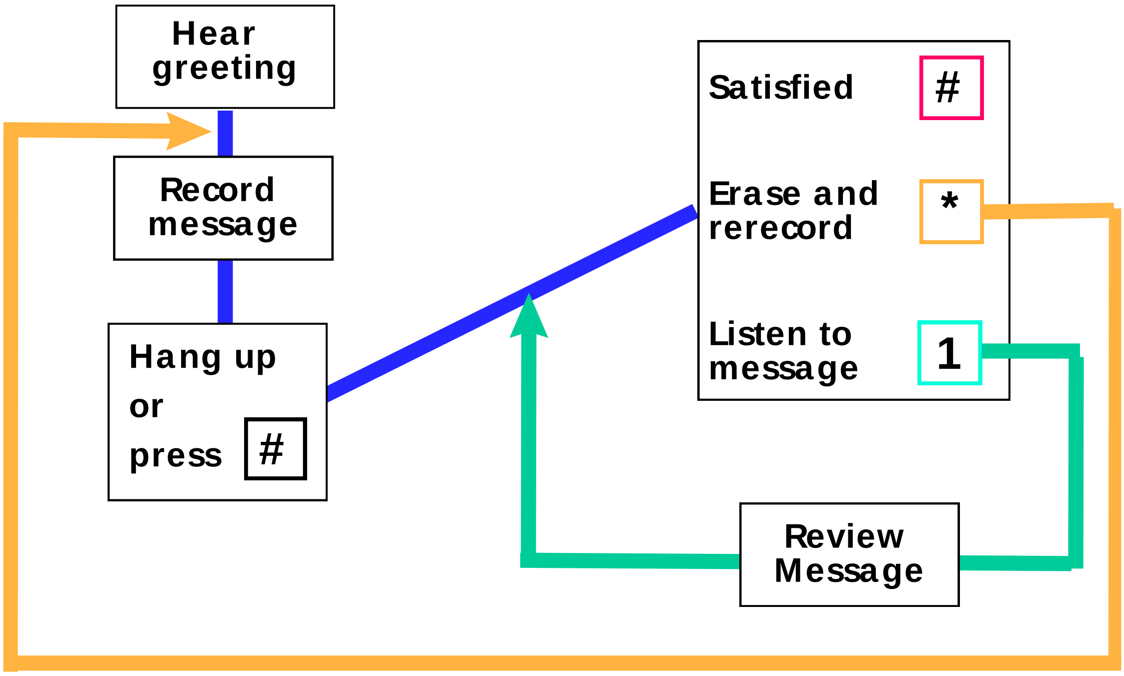
Actual Alarm Call

At the time for alarm call, A4400 establishes a connection to A4615 (all the alarm calls for the same time and with the same language are connected to the same voice mail port).
When callee goes off hook, A4615 plays the following announcement three times (defined by A4400) or until the user hangs up :

- ticking watch ... (short)
- ringing watch
- announcement of the defined time (in the subset language)

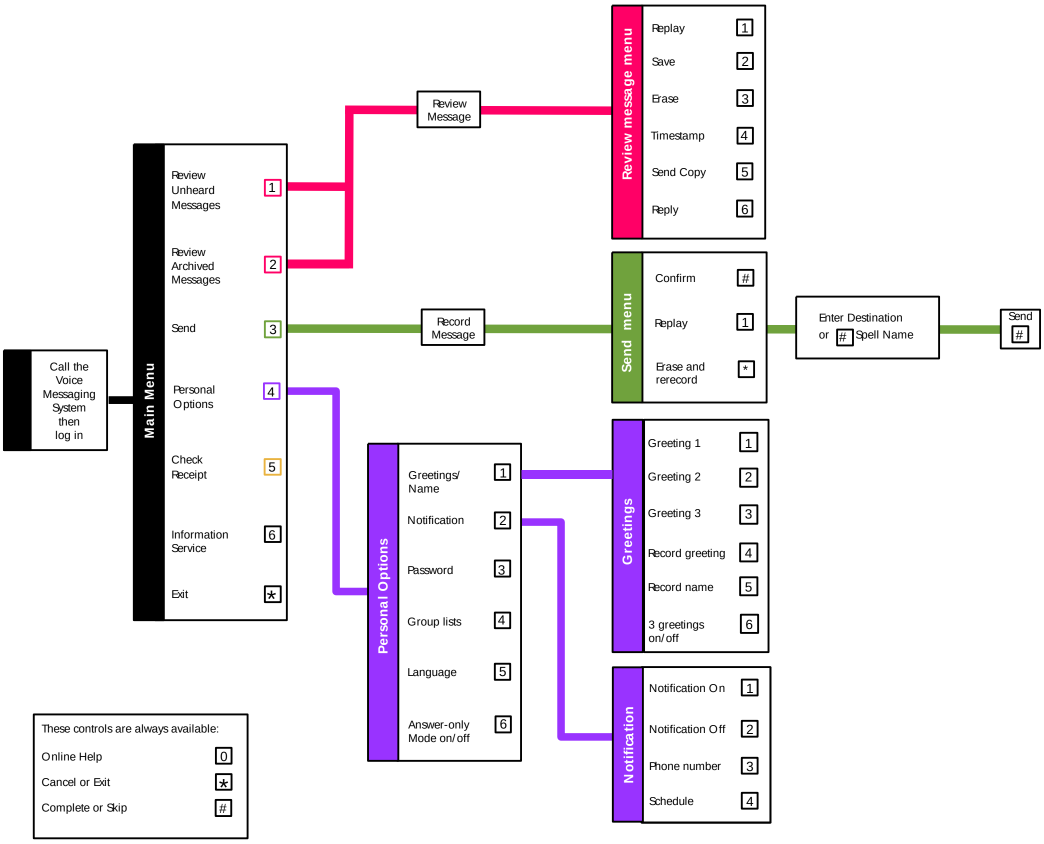
Note : A4615 wake–up feature is not possible in network. This means only subsets which are in the node where the A4615 resides can use A4615 wake–up feature.

11.4.7 Telephone Answering – Caller Interface

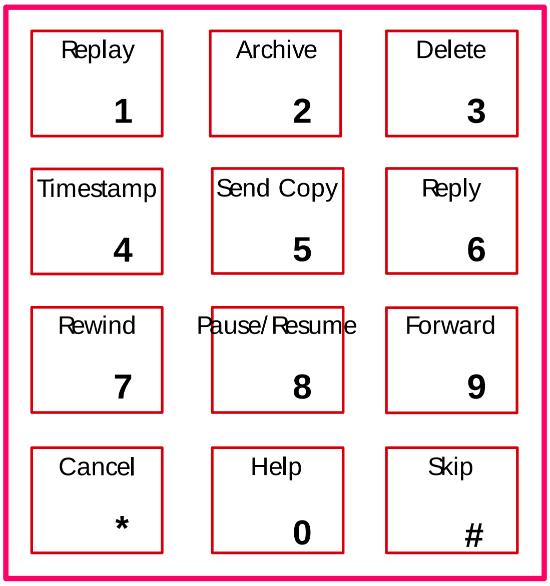


11.4.8 Subscriber interface

For ease of use, only main functions are described in the figure below.
More details are available in user's manual.



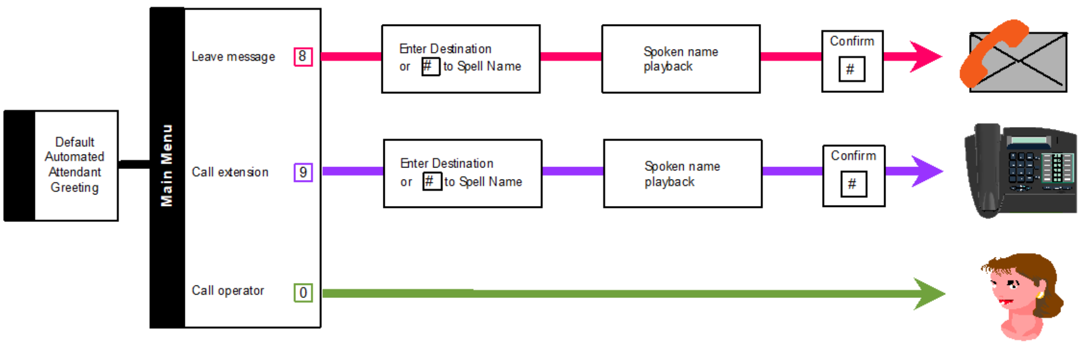
11.4.9 Keypad controls while/after listening message



General hints while listening to a message.

"*" cancel current operation
"0" Playback control menu (Restart, Rewind 10s, Pause/Resume, Forward 10s)
"#" Skips current message and goes to next

11.4.10 Default Automated Attendant



Notes.

Default Automated Attendant menu is country dependant.
Leave message function is always "8".
Call extension function is either "9" or "0".
Call operator function is either "0" or "9".

It is possible to system administrator to modify these default settings without having to record specific Automated Attendant greeting. System prompts are always played according to actual menu keys.

On all no-reply conditions on the called extension, 4615 takes the caller to subscriber's mailbox and plays the appropriate greeting, namely :

- Personal greeting for ring-no-answer and immediate forwarding to voice mail,
- Busy greeting for overflow to voice mail on busy condition,
- Answer-only greeting, when subscriber selected this mode (extended absence situations).

11.5 PROVISIONING LEVEL AND SIZING

A4615 voice mail system provides :

- 2 or 4 ports : number of ports is defined by capacity-on-demand through Actis licenses
- up to 4 simultaneous languages for voice prompts. The 4 languages used by 4615 are the 4 first languages defined within A4400. See 4615 localisation section below for details about 4615 and 4400 languages mapping
- up to 128 subscriber mailboxes (business or hotel configuration)
- up to 50 Information mailboxes (Audiotex), thereof :
- 45 public, for external callers,
- 5 private, for subscribers only,
- up to 4 minutes maximum length for a voice mail message
- up to 255 messages for one mailbox
- up to 8000 messages in total for all subscriber and information mailboxes,
- up to 320 minutes (more than 5 hours) of total recording time, see figure below for details

Number of simultaneous languages	Total recording time (subscribers and applications) Values are given in number of minutes
1	320
2	308
3	296
4	284

- automated attendant available in 2 modes (business hours, after hours).
  Each mode can be configured with a 2-level tree (one main menu and sub–menus). Main menu as well as sub–menus may have up to 9 (+ Operator) entries each. Recommended structure is to have up to 4 entries per menu.

Add examples of typical configuration, number of subscribers, Automated Attendant and Audiotex usage and the necessary ports.
The message to highlight is to restrict a 2-port 4615 system for less than 30 subscribers.

11.6 SYSTEM MANAGEMENT

A4615 can be configured in different ways :

- A4400 integrated management. Applicable to mailbox management only (create/modify/delete). Guarantees full coherence between A4400 and A4615 databases. Applicable with or without A4615 system up and running. Audit mechanisms will update A4615 database asynchronously, see dedicated section below.
- Voice guided administration from a telephone set.
  All configuration aspects are available. Does not guarantee A4400 and A4615 database coherence when used for mailbox management.
- Graphic interface administration from A4615-MMC.
  A4615–MMC is a PC–based configuration tool running under Windows 95/NT.
  All configuration aspects are available. A4615-MMC not guarantee A4400 and A4615 database coherence when used for mailbox management.
  A4615-MMC is connected through a transparent V24 connection to one of the four V24 CPU connector. Remote access maintenance to A4615 via RMA is a possible alternative.

11.6.1 A4400 integrated management

Following user attributes are passed from 4400 to 4615 for mailbox management :

- **Type**, defines the mailbox behaviour :
- Standard, suitable for all business subscribers,
- 'Answer only', not recommended for an individual
- Guest, suitable for all guests in Hotel,
- Automated Attendant, this mailbox is **unique** for one system, one AA entry point,
- **Number**, same as extension number,
- **Name**, same as user name,. Mailbox name will be used for call-by-name, address-by-name functions within A4615. Up to 8 characters are analysed for these features.

- **Language**, same as user phone set language
  Note.
  A4615 only supports 4 languages, whereas A4400 can support up to 8. If user language rank is above “4”, system default language will be used as mailbox language for this user.
- **Password**, will be used for Guest mailboxes in hotel environment only. Not relevant in business environment.
- **Notification**, defines the notification policy that will apply to the mailbox :
- No outcall notification,
- Signalling only,
  A4615 calls a number defined by mailbox subscriber.
  On A4615 voice prompt, mailbox owner presses the “*” key, then system hangs up.
  To retrieve his messages, mailbox owner has to call back A4615.
- Access to mailbox,
  A4615 calls a number defined by mailbox subscriber.
  After password authorisation, mailbox owner can review his messages.

11.6.2 Voice guided administration from a telephone set.

The interface between the voice mail administrator and the A4615 is based on the keypad of DTMF analog subset, or keypad of digital subset and voice guidance.
To access to the subset configuration menu, the administrator has to enter a password.
Voice guided administration from a telephone set summary :

- General
- information about installed software version
- select language for recording of “goodbye”, outcall notification, Automated Attendant and information mailboxes (Audiotex) announcements
- Voice messaging
- create/modify/delete mailbox
- create/modify/delete/review distribution list
- Automated Attendant and Audiotex :
- automated attendant number declaration (only one AA entry directory number possible)
- define business/after hours,
- business/after hours AA tree configuration
- duplicate business hours menu to after hours AA tree configuration
- recording of main menu announcement
- recording of company greeting announcement
- sub–menu functions and digit association/modification
- recording of sub–menu announcements
- “press star key” message notification on/off
- “transfer to operator type” declaration/modification (semi–supervised / supervised)
- “language choice question message” on/off
- fax number declaration
- infobox announcement (Audiotex) recording/deletion and function definition/modification
- statistics :
- mailboxes counters (2 counters) : “mailbox reached”, “message left”
- infobox (Audiotex) counter : “listened to infobox”
- statistics counter reset
- statistics information counters about the Automated Attendant (number of call, call duration sum, automatic distributed call number [means call distributed without caller DTMF digit sent]
- system global activity information. Maximum period covered by this history information is one week. For each hour of the day the following information is stored: free memory (in minutes of voice storage), seizure duration of A4615 ports (percentage), actual free memory (in minutes of voice storage)
- hit–list about the 3 mailboxes occupying the most system resources with time used for new and archived messages

11.6.3 Graphic interface administration from A4615-MMC

A4615–MMC is a PC–based configuration tool running under Windows 95/NT. Its graphical user interface provides a powerful and quick solution to manage the A4615 configuration.

A4615–MMC provides all the capabilities of Voice guide administration with a telephone set and in addition gives access to :

- system configuration file including country dependant data
  This actions is performed by :
- uploading data into A4615-MMC,
- changing parameters,
- downloading new settings to A4615 itself.
- database save/restore,
- upload of pre-recorded voice messages (Automated Attendant, Information mailboxes).

A4615-MMC is recommended to configure complex Automated Attendant structures.

11.7 DATABASE SYNCHRONISATION - AUDIT

This section discusses database synchronisation mechanism between A4400 and A4615 voice mail server. Database synchronisation avoids entering configuration data for a “voice mail enabled” subscriber twice. It also supports a “plug and play” feature for the voice mail, when a A4615 meets the first time an installed and configured A4400.

In out of service situations of the voice mail server A4615, configuration changes done during the voice mail downtime can be updated during the next database synchronisation process.

Three operation modes for database synchronisation between A4400 and A4615 are defined:

- **Plug and Play Mode**: A first time only process to convert existing mailboxes. This mode supports the A4400’s “init_mevo” command. A mailbox, which has been created either by MMC or by a phone set will be converted to a mailbox “created by A4400”.
- **Configuration Mode**: Integrated management commands during normal operation. This mode supports normal configuration changes either done with A4740 or with A4400’s internal “mgr” tool. This is the default mode.
- **Synchronisation Mode**: Regular database synchronisation updates, useful after out of service conditions. This is needed to support the “audit” command of A4400. At the end of a synchronisation mode a clean-up phase will remove all A4400 created mailboxes not yet synchronised.

A state variable inside the A4615 server will indicate the current mode of database synchronisation. The default mode after booting the A4615 server is always “Configuration Mode”.

The database synchronisation works roughly in the following sequence:

- A4400 initiates a data base synchronisation
- A4615 prepares a database synchronisation and acknowledges the request
- Multiple database update commands are sent from A4400 to A4615
- For each database update command, A4615 creates a new mailbox, or modifies an existing mailbox, or refuses the command (in error cases). The success or failure of the command is confirmed back to A4400. There is no specific need to delete mailboxes during database synchronisation.
- A4400 finishes the database synchronisation
- A4615 finishes all open requests. Now at the end of the database synchronisation mailboxes left-over, which has been managed by PBX previously, may be deleted by the voice mail server A4615.

11.8 NETWORKING ASPECTS

11.8.1 Voice Messaging network features

Voice messaging network features are all the services that allow a voice messaging system to exchange messages back and forth with another voice messaging system.
Voice messaging protocols are either "legal" standards, such as AMIS or VPIM, or "de-facto" standards, such as OctelNet.

A4635, which is the high-end voice messaging solution for A4400 implements all of these standards to cover the needs of businesses with homogeneous or heterogeneous voice messaging systems infrastructure.
A4615, which is dedicated to stand-alone 4400 configurations, does not provide voice messaging network capabilities.

11.8.2 A4615 and A4400 network

Unlike A4635, A4615 is not designed to be installed in a PABX network context as a centralised voice mail server.
Nevertheless, A4615 could be used locally in a node in addition of a A4635 in a network, provided that the following rules are respected :

- not more than one Voice Messaging system (46x5) per A4400 node
- A4615 mailbox owners must have their extensions on the 4400 node that hosts A4615 system.

11.8.3 A4615 and A4400 network roaming

A DECT user, titular of a A4615 voice mail, can use the roaming feature.
However, when attached to a visiting node of the A4400 network, user will be considered as any outside caller, and therefore will loose all A4615 integration services, namely :

- no Message Waiting Indication (MWI),
- no Mail key,
- no information about the count of messages,
- no direct access to personal mailbox.

11.8.4 Numbering plan

The numbering plan supported by A4615 is from 2 up to 4 digits wide.
Other 4400 configurations will require A4635 or Alcatel Unified Messaging systems.

11.8.5 Conclusion

For efficiency reasons, (centralised management, easy access to personal mailbox when roaming, single user interface for callers, single training for subscribers), we **do not recommend** to deploy a combination of heterogeneous Voice Messaging systems over a single 4400 network.

11.9 A4615 SW DELIVERY

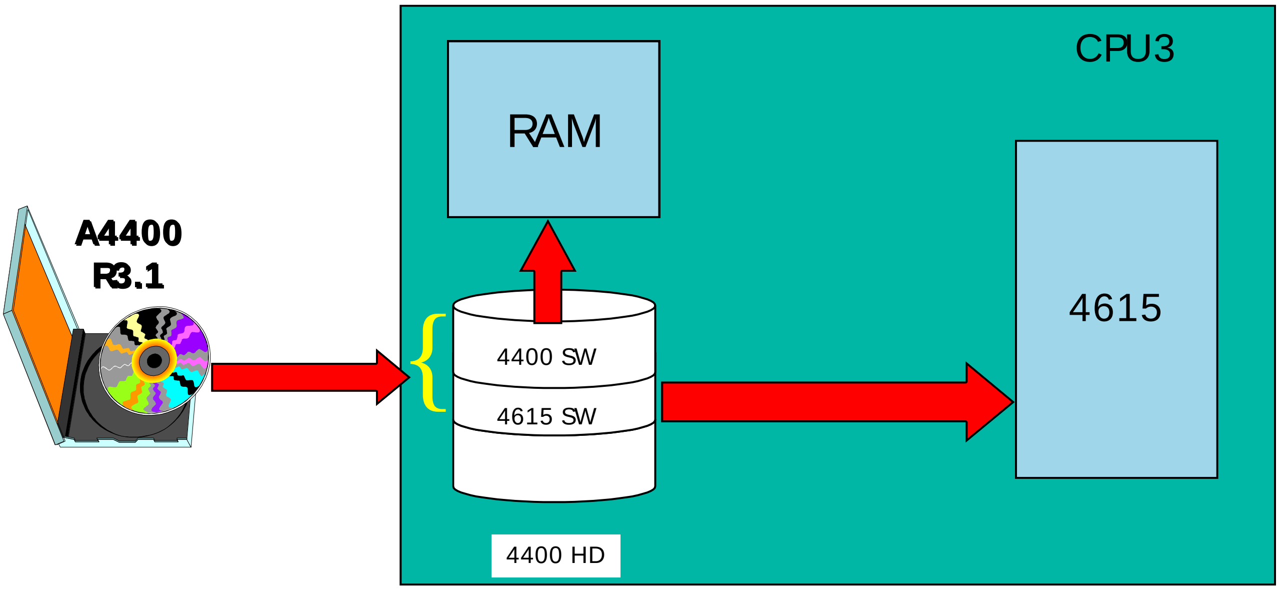
A4615 Application SW, voice prompts and country dependant data are part of 4400 SW delivery.

A4615 files consist of :

- application SW : boot-loaders and application programs for main and slave DSPs 1 000 KB
- country dependant data 4 KB
- voice prompts : one file per language (21 languages available on 01/09/99) 1 000 KB

COMPOUND PCM LAW

A4615 software is available in A-law as well as in Mu-law. Depending on the country parameter selected by the A4400 administrator during the global A4400 software installation, the A-law or Mu-law A4615 DSP software is copied from the delivery CD–Rom to the A4400 hard disk.
Language files are not depending on A-law or Mu-law.



11.10 A4615 LOCALISATION

A4615 localisation process is a conjunct action held by DSUs and VPC, with the co-ordination of the A4400 launching team.

Below is a summary of A4615 localisation process regarding "Voice translation and recording". Please refer to VPC web site (aww.vpc.aut.alcatel.at) for a detailed description.

Following languages are released with A4615 R3.0 (01/09/99)

- English,
- French,
- German,
- Spanish,
- Portuguese,
- Italian,
- Dutch,
- Flemish,
- Danish,
- Norwegian,
- Swedish,
- Finnish,
- Czech,
- Slovakian,
- Hungarian,
- Turkish,
- Latin American Spanish,
- Brazilian,
- Catalan,
- Lithuanian,
- Icelandic.

Periodic updates will be distributed when new languages will be released.

11.11 ORDERABLE ITEMS

A4615 is orderable through the following items :

- 3BA 53176 AA OBCA & A4615 Base Hardware

- 3BA 09290 AA A4615 Base for about 30 users (2 ports)
- 3BA 09291 AA A4615 Additional License for up to 128 users (2 ports)

12. NEW UA TERMINAL RANGE

12.1 NEW UA 3G LINE "*REFLEXES*" SETS

<u>In R3 this new range of UA 3g sets can be handled in two modes :</u>

- 2g emulation mode (emulation of 2G set)
- full 3g mode

This choice is possible set by set.

- For previous 4400 releases, the 2g emulation mode is mandatory
- ***Full 3g mode will be the default mode in R3***

<u>Some advantages of full 3g mode :</u>

- better SPK (System Programmable Keys) layout for coherence between sets and default keys (Forward, Redial)
- control by 4400 for :
  - timing control
  - new ergonomics (new navigator mode)
  - new services possibilities for future releases (ex. : Loudspeaker Announce during conversation, waiting beep in Loudspeaker or Handset, external ringing command)

The 2g set range is still managed in R3 (with the same SPK profiles as before)...

12.1.1 A4004 "*First*" Reflexes : 3G and 2G mode

4004 First REFLEXES

Monoline mode

Consult	Broker
Mail	Forward
Redial	Ring Tone
Volume -	Volume +

This set works only in **monoline** mode

This set has only one led for "Mail" notification (no display or led associated to the keys except for mail)

This set has 4 MPK (Management Programmable Key) :

- Redial, Forward, Consultation Call & Broker

Nevertheless, it is recommended to keep "Consultation Call" & "Broker" which are necessary in monoline mode

"Redial" & "Forward" services (shown in different colour on the figure above) are pre-programmed by default on the MPK

"Ring Tone" key is used for user choice of ringing melody
"Volume- and Volume+" keys are used for adjusting ringing level and audio level in handset

12.1.2 A4010 "*Easy*" Reflexes

12.1.2.1 4010 "*Easy*" Reflexes : 3G mode

4010 Easy REFLEXES

Multiline mode

Line 1	Line 2
Repert	Transf
Mail	ISDN
Redial	St/ Redial

Monoline mode

Consult	Broker
Repert	Forward
Mail	ISDN
Redial	St/ Redial

This set works in **multiline** or in **monoline** mode

This set has one led for "Mail" notification & only 6 Keys with associated led (Redial & Store/Redial are without led)

This set has 3 MPK in multiline mode :

- Transfer, Line 1 & Line 2

Nevertheless, it is recommended to keep the 3 MPK which are necessary in multiline mode

This set has 3 MPK in monoline mode :

- Forward, Consultation Call & Broker

Nevertheless, it is recommended to keep "Consultation Call" & "Broker" which are necessary in monoline mode
"Forward" service (shown in different colour on the figure above) is pre-programmed by default on the MPK

12.1.2.2 A4010 "*Easy*" Reflexes : 2G mode

Multiline mode

St/Redial	
Mail	ISDN
Repert	Transf

Monoline mode

Broker	
St/Redial	
Mail	ISDN
Repert	Consult

This set works in **multiline** or in **monoline** mode with the SPK key profiles shown above

12.1.3 A4020 "*Premium*" Reflexes

12.1.3.1 A4020 "Premium" Reflexes : 3G mode

Multiline

Forward	ISDN
Redial	St/ Redial

Monoline

Consult	Broker
Forward	ISDN
Redial	St/ Redial

This set works in **multiline** or in **monoline** mode

This set has one led for "Mail" notification and 12 Keys with associated custom display

This set has 3 SPK, 1 MPK and 8 UPK (User Programmable key) in **multiline** mode
"Forward" service (shown in different colour on the figure above) is pre-programmed by default on the MPK

This set has 3 SPK, 3 MPK and 6 UPK in **monoline** mode
It is recommended to keep "Consultation Call" & "Broker" on the two other MPK because they are necessary in monoline mode
"Forward" service (shown in different colour on the figure above) is pre-programmed by default on the MPK

12.1.3.2 A4020 "*Premium*" Reflexes : 2G mode

Multiline

St/ Redial	
ISDN	Redial

Monoline

Broker	
St/ Redial	Consult
ISDN	Redial

This set works in **multiline** or in **monoline** mode with the SPK key profiles shown above

12.1.4 A4035 "*Advanced*" Reflexes

12.1.4.1 A4035 "*Advanced*" Reflexes : 3G mode

Multiline or Monoline

Mail	ISDN
Redial	St/Redial

This set works in **multiline** mode or in **monoline** mode

This set has one led for "Mail" notification and 24 Keys with associated custom display

This set has 5 Softkeys, 4 SPK and 20 UPK (User Programmable Key)
The four SPK (System Programmable Keys) are shown on the figure above

All specific keys for multiline mode (Transfer) or monoline mode (Consultation Call & Broker) are provided on Softkeys.

12.1.4.2 A4035 "*Advanced*" Reflexes : 2G mode

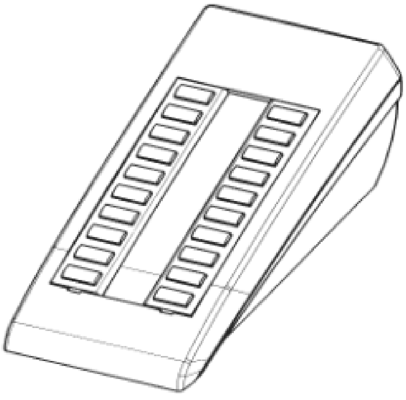
Multiline or Monoline

St/Redial	
Mail	ISDN
Redial	Mute/Int

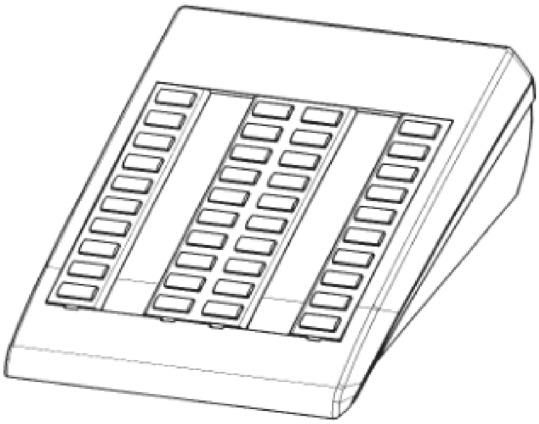
This set works in **multiline** or in **monoline** mode
The five SPK (System Programmable Keys) are shown on the figure above

12.1.5 ADD-ON MODULES (optional equipment for A4020 & A4035 sets)

A4090 M (medium)



A4090 L (large)



- They allow the user to have more programmable icon keys. (MPK/UPK)

- They are supplied with the associated set.

- The maxi number of keys on AOM is : **60 keys**

-> 3 x Alcatel 4090 M
or -> 1 x Alcatel 4090 L + 1 x Alcatel 4090 M

12.1.6 Recapitulative table

	FIRST	EASY	PREMIUM	ADVANCED
				
CHARACTERISTICS				
VISUAL				
Display		1x20 characters	1x20 characters	2x40 characters
Caller ID, date, time, length and cost of call			÷	÷
Message led	÷	÷	÷	÷
KEYS				
Programmable keys	8	8	12	24
Fixed keys		5	10	7
Diodes/icons associated to keys		6 diodes	12 icons	24 icons
Number of lines	single line	2 lines	multi-line (12 lines max.)	multi-line (24 lines max.)
AUDIO				
Loudspeaker		÷	÷	÷
Hands-free			÷	÷
Secret			÷	÷
Interphony			÷	÷
FUNCTIONS				
PROGRAMMATION				
Pre-programmed functions	up to 8	6 to 8	4 to 8	4 to 6
Programmable functions	up to 2	up to 3	7 to 9	19 to 20
Attendant station				÷
Manager/secretary station			÷	÷
Hunting group	÷	÷	÷	÷
DIRECTORY				
Personal	10 to 12 correspondents	10 to 12 correspondents	10 to 12 correspondents	30 to 45 correspondents
Collective	÷	÷	÷	÷
Integrated alphabetical keyboard			÷	÷
Call-by-name				÷
Text messages				÷
PHYSICAL CHARACTERISTICS				
SPECIFICATIONS				
Wall mounting	standard	standard	kit	kit
Ergonomic handset	standard	standard	comfort	comfort
Headset	÷	÷	÷	÷
DIMENSIONS				
Width	177 mm	177 mm	246.5 mm	285 mm
Height	59 mm	59 mm	86.5 mm	79.5 mm
Depth	196 mm	196 mm	219.5 mm	196.5 mm
Weight	600 g	630 g	880 g/1090 g	980 g/ 1190 g

12.2 NEW SUBDEVICE RANGE (TSC : **T**ransparent **S**ystem **C**onnectors**)**

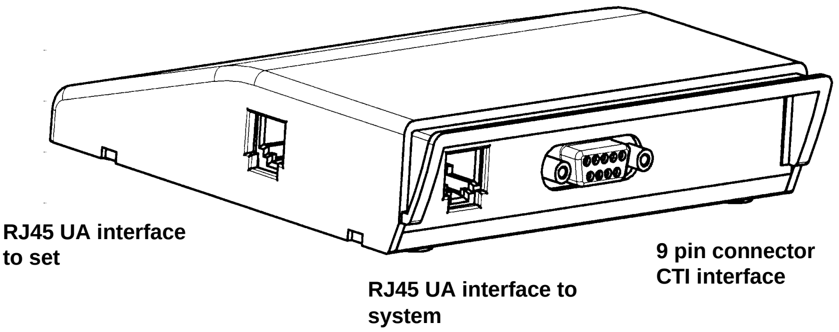
- With **Alcatel Plugware**, upgrading the installation becomes easy. By means of the **T**ransparent **S**ystem **C**onnectors **(TSC)** which are suitable for the whole range of Reflexes, it is possible to enhance the telephone services with flexible desktop solutions such as computer telephony applications, data transmission, analogue/ISDN peripheral connectivity, wireless system/network connectivity.

- These subdevices can be lodged inside A4020 & A4035 sets.
They can also be connected, as remote stand alone device, to all sets of the **3G** range and, in the same way, to all sets of the **2G** range (A4001 to A4034)

- Today several variants of subdevices are available:
  * **A4091 CTI:** offers Computer Telephony applications connection
  * **A4093 ASY/CTI:** offers a fully V24 asynchronous port and a CTI connection
  * **A4094 ISW:** offers a full S0 compatible bus with power feeding
  * **A4094 ISW/CTI:** offers a S0 compatible bus and a CTI connection
  * **A4095 AP:** allows the connection of a standard analog device
  * **A4097 CBL:** provides Cordless functions to UA sets and offers a CTI connection

12.2.1 ALCATEL 4091 CTI:

- Thanks to this interface, the user will be able to fully integrate most Computer Telephony applications compliant with TAPI rel. 1.4 protocol and UA telephony. This will allow the full featured UA telephony to be enriched by the power of PC & applications (Intra and Internet servers).



Transmission parameters

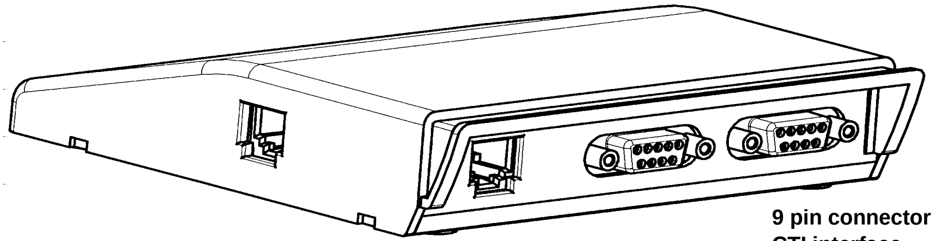
- Transmission rate: 9600/19200
- Format: 8N2 - Echo: no - Full duplex: no - Data rate adaptation: no - Flow control: specific.

Connectors:

- 2 x RJ 45 female (UA link & UA set), 1 x ISO 2110 female 9 pts (CTI)

12.2.2 ALCATEL 4093 ASY/CTI

- This interface offers:
  * <u>a fully V24 asynchronous port</u> for connection of PCs, PDAs, printers, plotters and so on....As needs evolve towards higher transmission speeds, even in the asynchronous mode, the A4093 ASY-CTI is offering a V14 extended mode, allowing the use of speeds up to 57600 bps.
  * <u>a CTI connection (see 8.2.1.)</u>



Transmission parameters

- Transmission rate: 9600/19200 for V24-CTI

- Two mode are supported by V24-ASY :
  -> V110
  -> V14

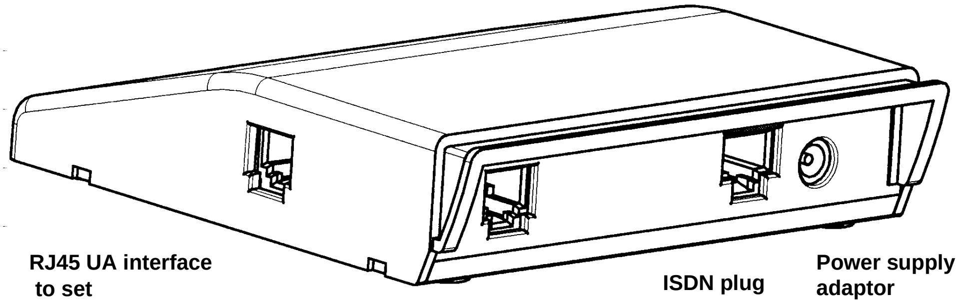
- The different bit rates supported by the V24-ASY are :
  -> in V110 mode: 50, 75, 150, 300, 600, 1200,
  2400, 3600, 4800, 7200, 9600, 14400, 19200 bits/s
  -> in V14 mode : 57600 bits/s

- The different formats supported by the V24-ASY are :
  - 1 start bit, 5 data bits, no parity, 1 or 2 stop bits
  - 1 start bit, 5 data bits, 1 parity bit, 1 or 2 stop bits
  - 1 start bit, 7 data bits, no parity, 1 or 2 stop bits
  - 1 start bit, 7 data bits, 1 parity bit, 1 or 2 stop bits
  - 1 start bit, 8 data bits, no parity, 1 stop bits

Connectors:
- 2 x RJ 45 female (UA link & UA set), 2 x ISO 2110 female 9 pts (V24/ASY & CTI)

12.2.3 ALCATEL 4094 ISW

- This interface allows the user to benefit from a full S0 compatible bus. The S0 bus can be configured
in one of two standard modes: point to point or short bus. This will allow the connection of standard S0 devices such as G4 faxes, PC with ISDN board, etc.
This bus allows to connect up to four ISDN sets which are power fed from the subdevice.
To see the different S0 buses, see § 3 BRA board description.



Transmission parameters:

- I 400 Recommendation compliant; protocol I 430.

Connectors:

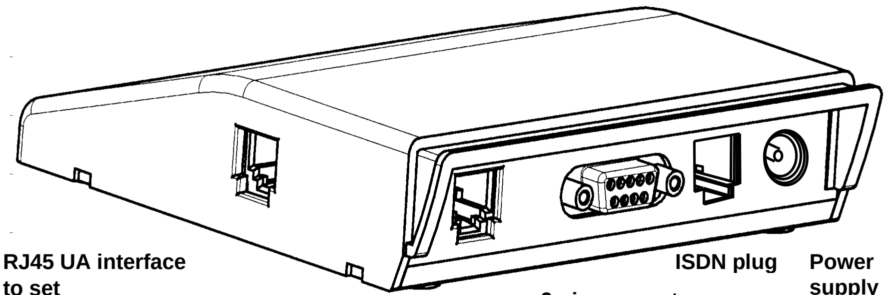
- 3 x RJ 45 female (UA link & UA set + S0 bus), 1 x coax plug for external AC/DC power supply device.
  (bus feeding)

12.2.4 ALCATEL 4094 ISW/CTI

- This interface offers:
  * a full S0 compatible bus. The S0 bus can be configured in one of two standard modes: point to point or short bus. This will allow the connection of standard S0 devices such as G4 faxes, PC with ISDN board, etc.

  This bus allows to connect up to four ISDN sets which are power fed from subdevice.
  To see the different S0 buses, see § 3 BRA board description.

  * a CTI connection (see 8.2.1)



Transmission parameters

- Transmission rate: 9600/19200 for V24-CTI

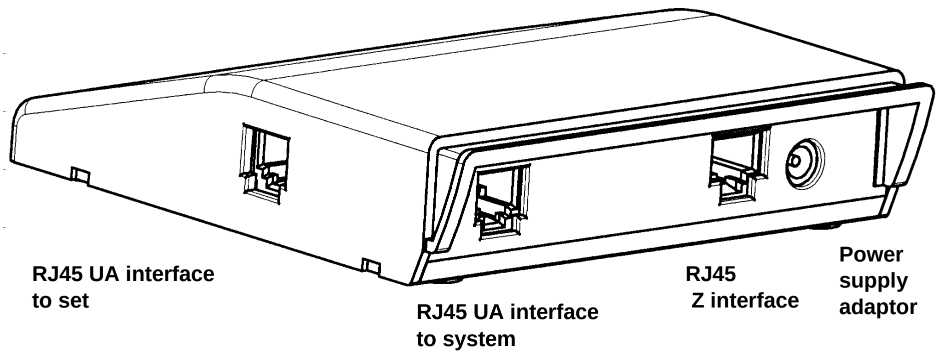
- I 400 Recommendation compliant; protocol I 430.

Connectors:

- 3 x RJ 45 female (UA link & UA set + S0 bus), 1 x ISO 2110 female 9 pts (CTI), 1 x coax plug for external AC/DC power supply device.(bus feeding)

12.2.5 ALCATEL 4095 AP

- This subdevice can connect the following analog devices:
  - ° an analog set (Z Proprietary interface)
  - ° a Minitel
  - ° a fax group 3
  - ° an answering machine
  - ° a modem
  - ° an external alarm (like behind a Z board)
  - ° a PC with modem



Transmission parameters:

- Compliant with analog 2 wires data terminals (TBR21)

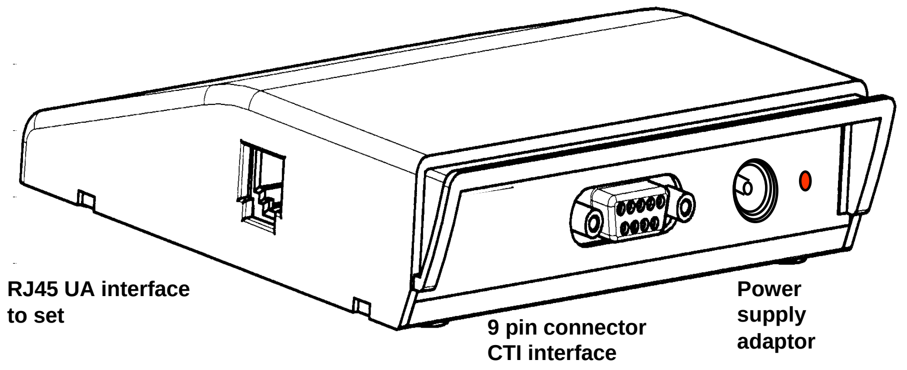
Connectors:

- 3 x RJ 45 female (UA link & UA set + a/b wires for analog terminal), 1 x coax plug for external AC/DC power supply device. (50Hz->30V DC/130mA)

12.2.6 ALCATEL 4097 CBL

- The 4097 CBL subdevice is a DECT UA link which enables any Reflexes™ subsets to become a DESKTOP WIRELESS extension keeping the user benefit of full A4400 systems services.

- It also provides a CTI interface (see 8.2.1).



Transmission parameters:

- Transmission rate: 9600/19200 for V24-CTI

- DECT Radio protocol (the antenna is embedded in the 4097 box).

Connectors:

- 1 x RJ 45 female (UA link), 1 x ISO 2110 female 9 pts (CTI), 1 x coax plug for external AC/DC power supply device (power feeding for CBL box & UA set).

- 1 x red LED indicating the good running of the CBL

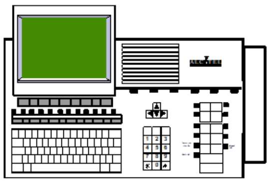
12.3 COMPATIBILITY ISSUES

	A4400 R2	A4400 R3
UA 2G Sets UA 2G Subdevices UA 2G Terminal Adapters	Compatible Compatible (4034/4012) Compatible: all 2G subdevices	Compatible Compatible (4034/4012) Compatible: all 2G subdevices
UA 3G Subsets	Compatible:"2G emulation" 4035 = 4034 4020 = 4012 + EAK 4010 = 4011 (reduced) 4004 = 4003 (reduced)	Compatible:"3G optimised mode" (minor differences)
UA 3G "TSC" subdevices	YES (CTI,CTI-V24,S0,S0-CTI,AP) Also compatible A4012 & A4034	YES (CTI,CTI-V24,S0,S0-CTI,AP) Also compatible A4012 & A4034
TSC on all 3G sets	NO (only 4035/4020)	YES (all 3G sets)
TSC standalone operation	YES	YES
Daisy chaining < 2 TSCs	NO	NO
3G TSC-DECT (including 4021 & 4036)	NO	YES (all 3 G sets NOT compatible with 2G sets

13. ATTENDANT SERVICES

13.1 ATTENDANT CONSOLE RANGE IN R3

4035 "Basic Attendant Console"
(can be used as existing 4034)



4048 "Flat Bed Console"

4059 "Multimedia Attendant Console"
(will replace existing 4058)

We have two new Attendant Consoles in R3 :
- 4035 in place of 4034
- 4059 "Multimedia Attendant Console" in place of 4058 (SBC)

Note that 4034 is still R3-compatible but not 4058 (an update is necessary as shown later on in this document)

The existing 4048 "Flat Bed Console" remains in R3 but will also be replaced in a further release

For 4059 "MAC" (Multimedia Attendant Console), the best configuration is a PC tower associated with TA MMK (TA MMK : Terminal Adapter for Multimedia Keyboard) under the operator desk table (not shown in the figure above)

For 4035 : no service evolution compared to 4034 (it's only a new phone set)

For 4059 "MAC" : it's a brand new product compared to 4058
- at hardware (physical phone set/PC keyboard integration) and software levels
- this important new product will be described in the following pages

New provisioning level in R3.1 :
- the number of Attendants is now increased to 50 Attendant Consoles per group and per node

13.2 THE NEEDS

The Attendant needs new <u>ergonomic</u> tools :

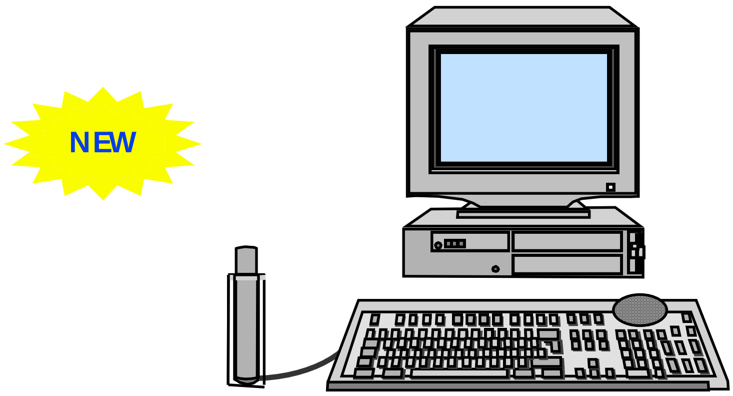
- To efficiently handle the incoming calls
  - need of an ergonomic workstation

- To efficiently manage the company greeting :
  - Higher degree of service to customers : who to find - and quicker
  - Ability to receive messages by Attendant
  - Ability to inform caller when absent person will return

- To give him opportunity to perform other tasks :
  - Visitor registration
  - Agenda
  - Booking of conference rooms
  - E-Mail

13.3 THE 4059 "MAC" OFFER IN R3

Firstly, the new 4059 "Multimedia Attendant Console" offers the "best of" user ergonomics thanks to an Alcatel multimedia PC keyboard with audio capabilities



<u>The Multimedia Keyboard (MMK) is basically an extended PC keyboard with audio capabilities :</u>
- extended PC keyboard with specific communication keys & softkeys
- audio capabilities with loudspeaker, hands free, handset/headset connection

Secondly, 4059 "MAC" will be an <u>open</u> Win NT4 or Win 95 multiapplication workstation :

<u>Able to run other Alcatel App designed for Attendant needs :</u>
- Alcatel 4000/4755 Directory/InfoCenter Client
- 4000/4755 Directory Pop-up
- 4059 BLF (Busy Lamp Field)
- 4059 MGT (Management)

<u>Able to run **selected & tested** third party App of the market like :</u>
- PIM, E-mail, Agenda, Scheduler, Directory, Infocenter

Win NT is recommended for real time performance and reliability

A basic level of **TAPI compliance** is offered for third party App : we will see the description later on in this presentation

13.3.1 Highlights

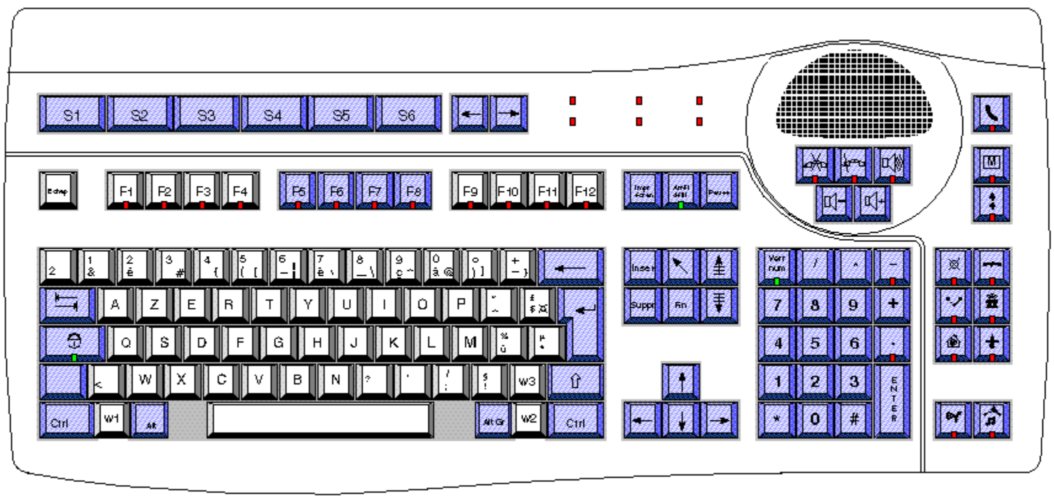
4059 "MAC" will offer the "best of" user ergonomics through :

* adapted multimedia keyboard
* new simplified screen layout
* window resizing
* 3D Ms Windows look
* Win 95, Win NT
* Performance : good time response (new performance of 4400 to 4059 link)

The new link for better performance is only available in R3.

* Ease of use : Windows log-in & password, automatic Directory launch/exit, auto App switching, auto server connection, strings are in plain language (not abbreviated as today), user Name is shown in the title bar, ...
* User customization :
  - 4059 MAC, 4059 BLF, Directory Client : "User Profile" for window size and position
  - 4059 "User Profile" for window layout size and position, background color, optional graphical traffic display ...
* Multi-user profiles by Windows session login & password
  Each profile is automatically selected by user Windows session log-in & password
  This feature is available for 4059 "MAC" and 4059 "BLF"
  For Directory Client, multi-user profiles is also available but at App level (not Windows session)

The Multimedia Keyboard



The multimedia keyboard offers :

* high level user ergonomics :
  - specific communication keys can be used while the user remains in any Windows App (example : incoming call seizure, audio setting, ...)
  - one specific key for easy App switch : 4059 <-> App (ex. : Word)
  - desktop integrated workstation through physical phone set integration
  - full PC keyboard compatibility : up to 13 keyboard types are available
  - extended PC keyboard adapted for user communication
  - audio keys
  - one hand level call control keys for main call control services
  - softkeys for other call control services

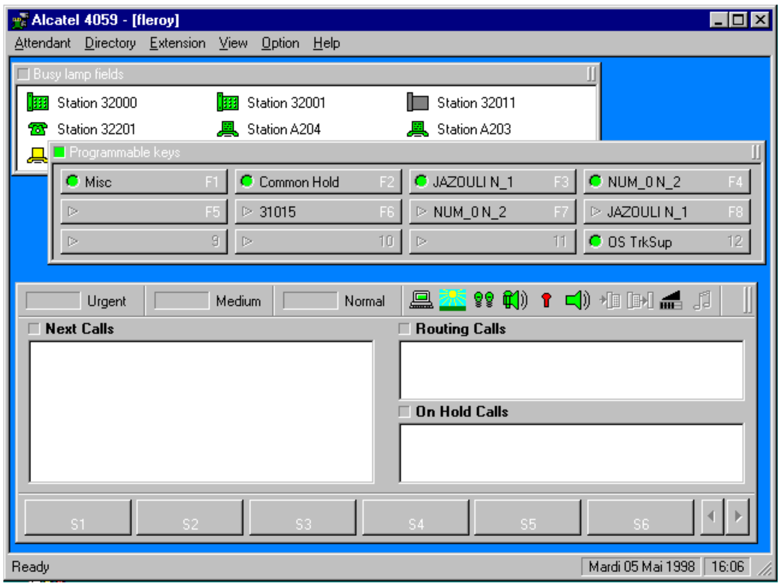
On the previous "multimedia keyboard" figure we can see :

* Specific communication keys :
  - P1 : Application switching key
  - 10 Fixed keys with Icons for main functions of the A4058 application
  - S1 to S6 : Softkeys for secondary functions of the application
  - " * " , " # " : telephone keys added on numeric keypad
* Modified PC keys :
  - F1 to F12 : a Led is added per key
  - "-" : a Led is added
  - "+" : the key is smaller
  - "." : the key is not on the same place
  - "0" : the key is smaller

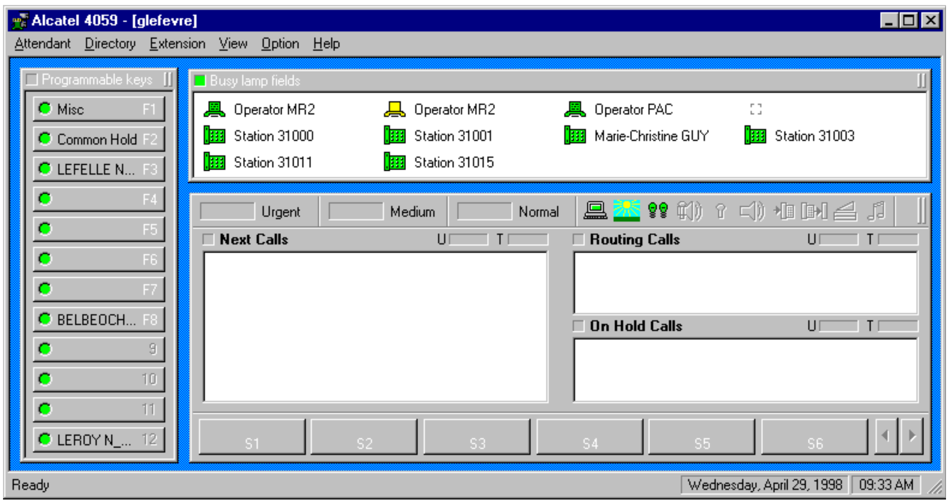
The multimedia keyboard is compatible with the following 13 PC keyboard types :
* French, German, UK, Spanish, Portuguese, Swedish/Finnish, US, Belgian, Swiss, Italian, Danish, Norwegian, US Cyrillic

New screen layout

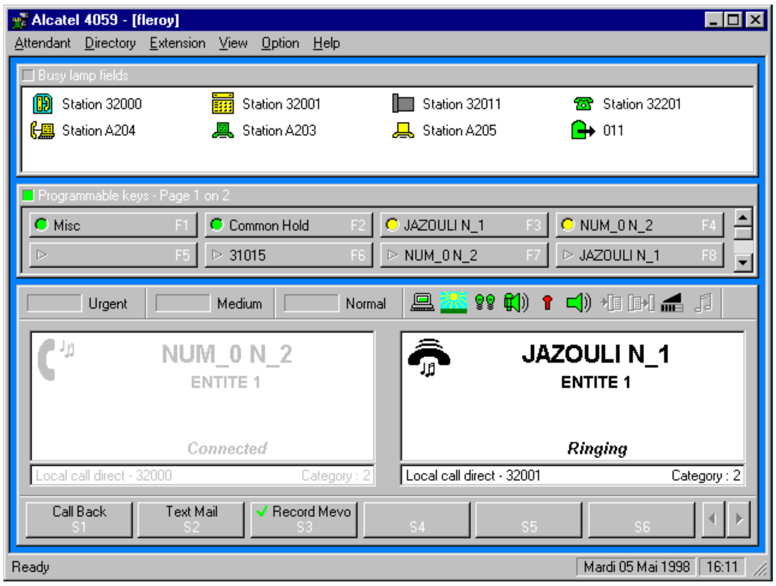
Screen Snapshot in Idle State : example 1



Screen Snapshot in Idle State : example 2



Snapshot During Call Handling



Alcatel's Attendant Communications App Suite

∗ High performance Attendant Telephony : 4059 "MAC"

∗ High performance and ergonomic Supervision : 4059 BLF

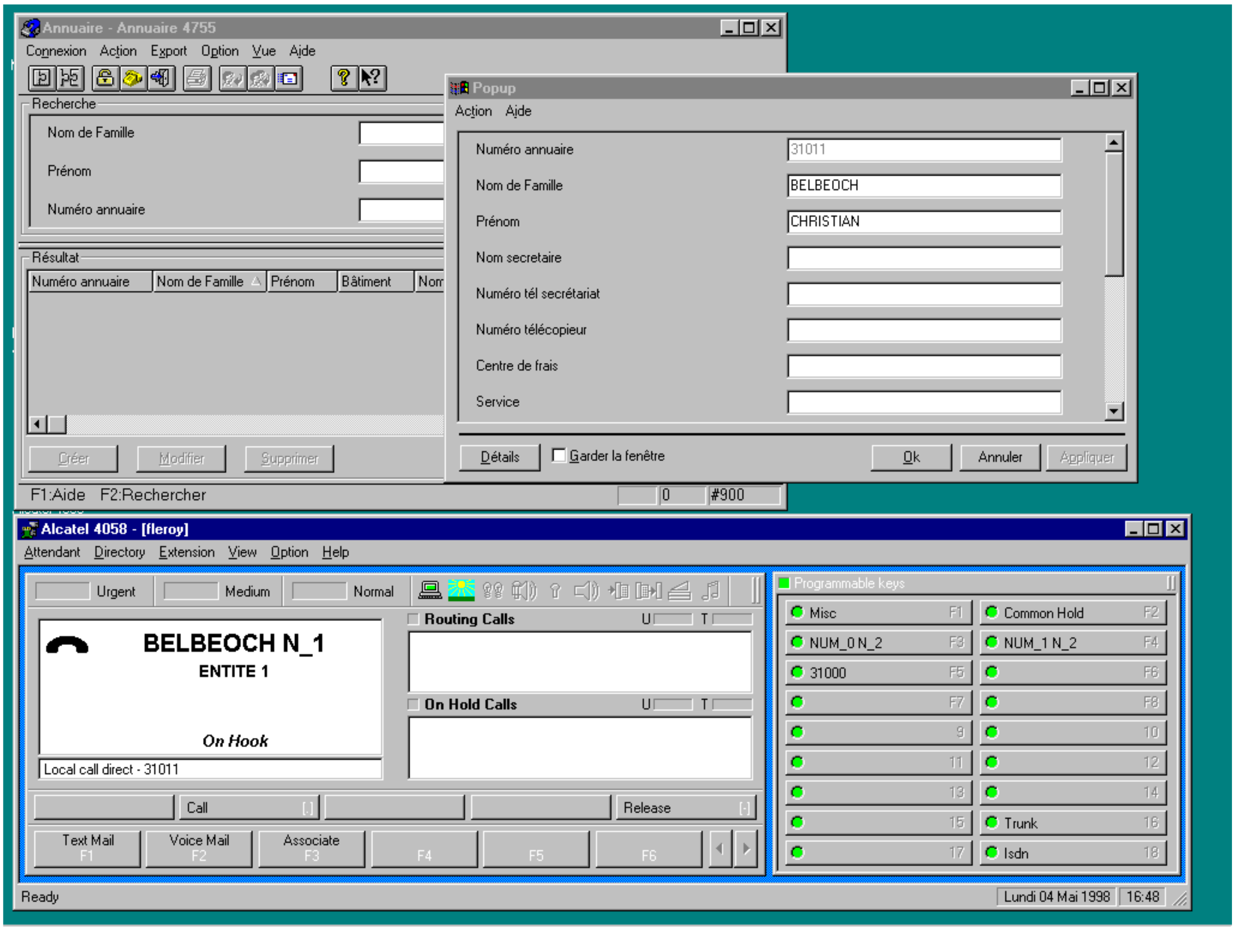
∗ Attendant Management : 4059 MGT
System & Subscriber management is done with 4059 "MGT"

∗ Corporate Directory (4000/4755 Directory/InfoCenter) with :
- ÷ Call by Name
- ÷ Directory Pop-up function
- ÷ InfoCenter services

In previous releases (R2, R1.5, R1.4) the SBC management was done at 4034 phone set level

Now the attendant Console management is fully integrated in 4059 "MAC" with additional services and good ergonomics (before we only had a basic 4034 set emulation interface)

Multi-App : Screen Snapshot (4059 + Directory client + Directory pop-up)

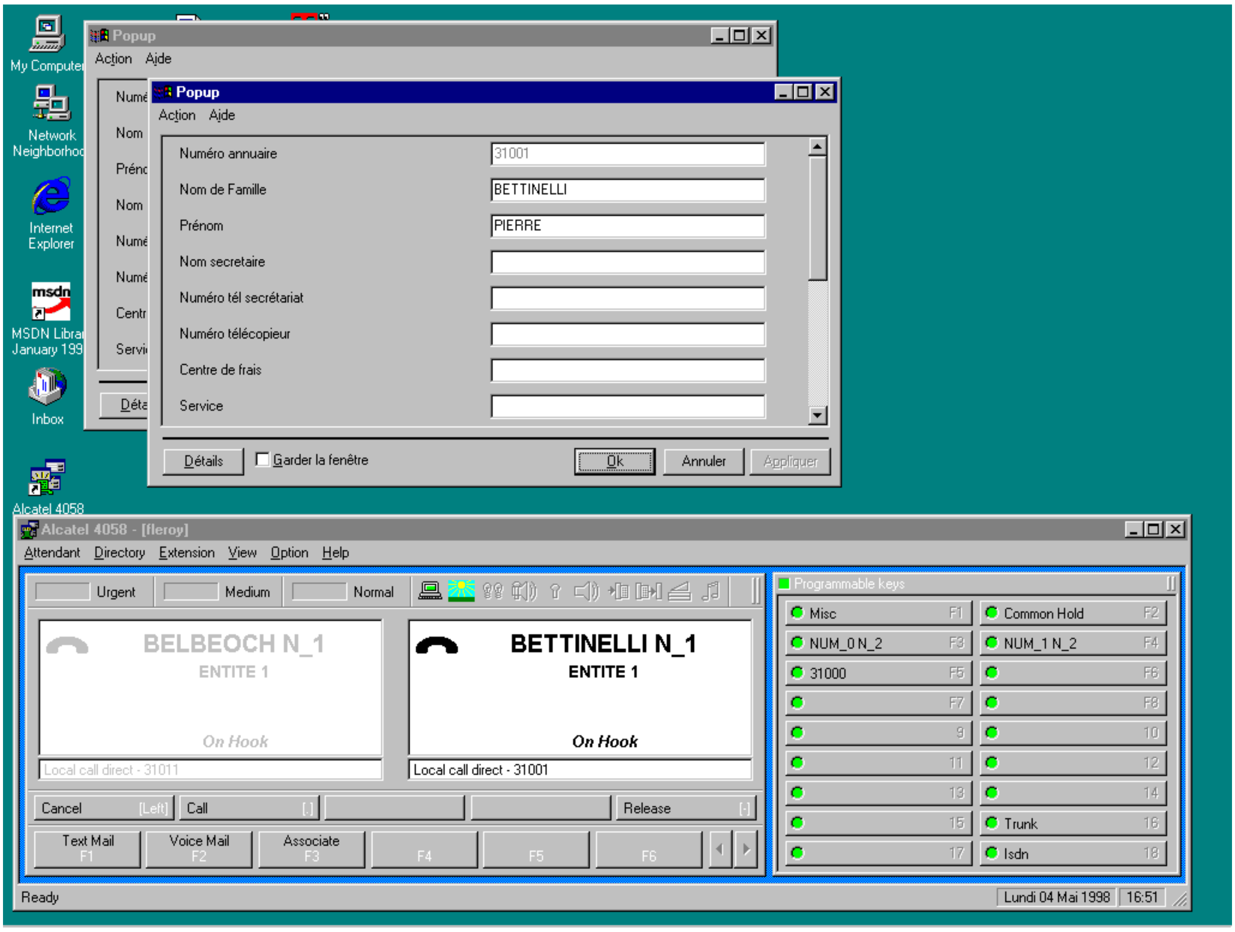


In the previous screen layout example we can see :

* in the lower part : a window for the Attendant application A4058
* in the upper part : the Directory Client window & a second window for the pop-up information about the call in conversation with the Attendant

Note that the Busy Lamp Field application 4059 BLF is not shown here

Multi-App : Screen Snapshot (4059 + Directory pop-up of two correspondents)



In the previous screen layout example we can see :

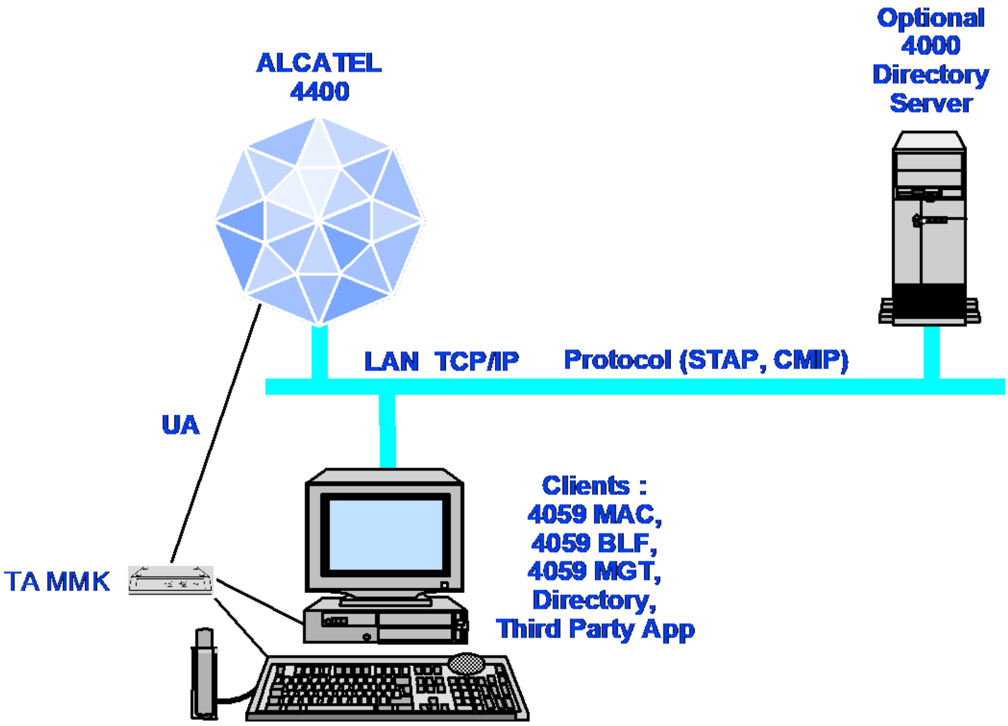
∗ in the lower part : a window for the 4059 Attendant application

∗ in the upper part : two windows for pop-up information about the call in conversation with the Attendant

Note that the Busy Lamp Field application 4059 BLF is not shown here

13.3.2 Architecture

General architecture



"Call by Name" is done in 4400 Phone Book or with Directory (on Attendant choice)

Hence, 4000/4755 Directory is optional for the "Call by Name" function but offers complementary services if needed.

TCP/IP LAN is used for :

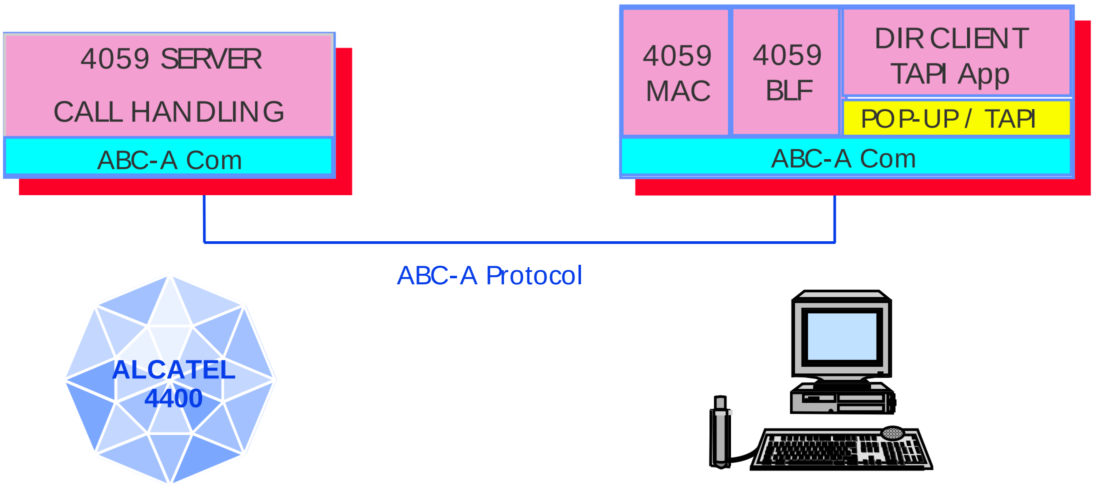
- Directory Client/server connection
- automatic dialing with "STAP" protocol between 4400 & Directory server
- 4059 MGT (Management) connection with 4400 (CMIP protocol)

UA link is used for :

- 4059 MAC connection with 4400
- 4059 BLF connection with 4400

TA MMK : means "Terminal Adapter for Multimedia Keyboard"

Software architecture



It's a Client/server architecture with :

- 4059 Call handling App in 4400
- call handling presentation and additional new services in 4059 MAC and 4059 BLF App

A proprietary "ABC-A com" middleware is used to link the clients with the servers.
ABC-A protocol transport is done on UA link from 4400 to TA MMK and V24 link (PC Tel or V24 data) from TA MMK to PC

On top of ABC-A middleware we find :

- a pop-up interface for Directory Client or TAPI Directory
- a TAPI interface for TAPI App

Hence 4059 is a TAPI server for third party App offering :

- Assisted Telephony (ex. : Schedule, Outlook, Excel, Word, ...)
- TAPI interface limited to "Make Call" and "Receive Call" services

Planned evolution :
OLE automation

Security

Standby mode is provided by multimedia keyboard in case of PC failure.

Openness

Alcatel 4059 "MAC" offers the following openness :

- Inter-working with any TAPI compliant CTI App
  - TAPI limited to "Make Call" and "Receive Call" services
    ex. : TAPI compliant Directories like "Twixtel"

- Inter-working with popular desktop Windows App
  - TAPI Assisted Telephony (ex. : Schedule, Outlook, Excel, Word, ...)
  - "Parlando" Info Center for Nordic countries : see following screen snapshot
  - OLE automation (future evolution)
    ex. : Excel, Outlook, Lotus Organizer, Act!, ...

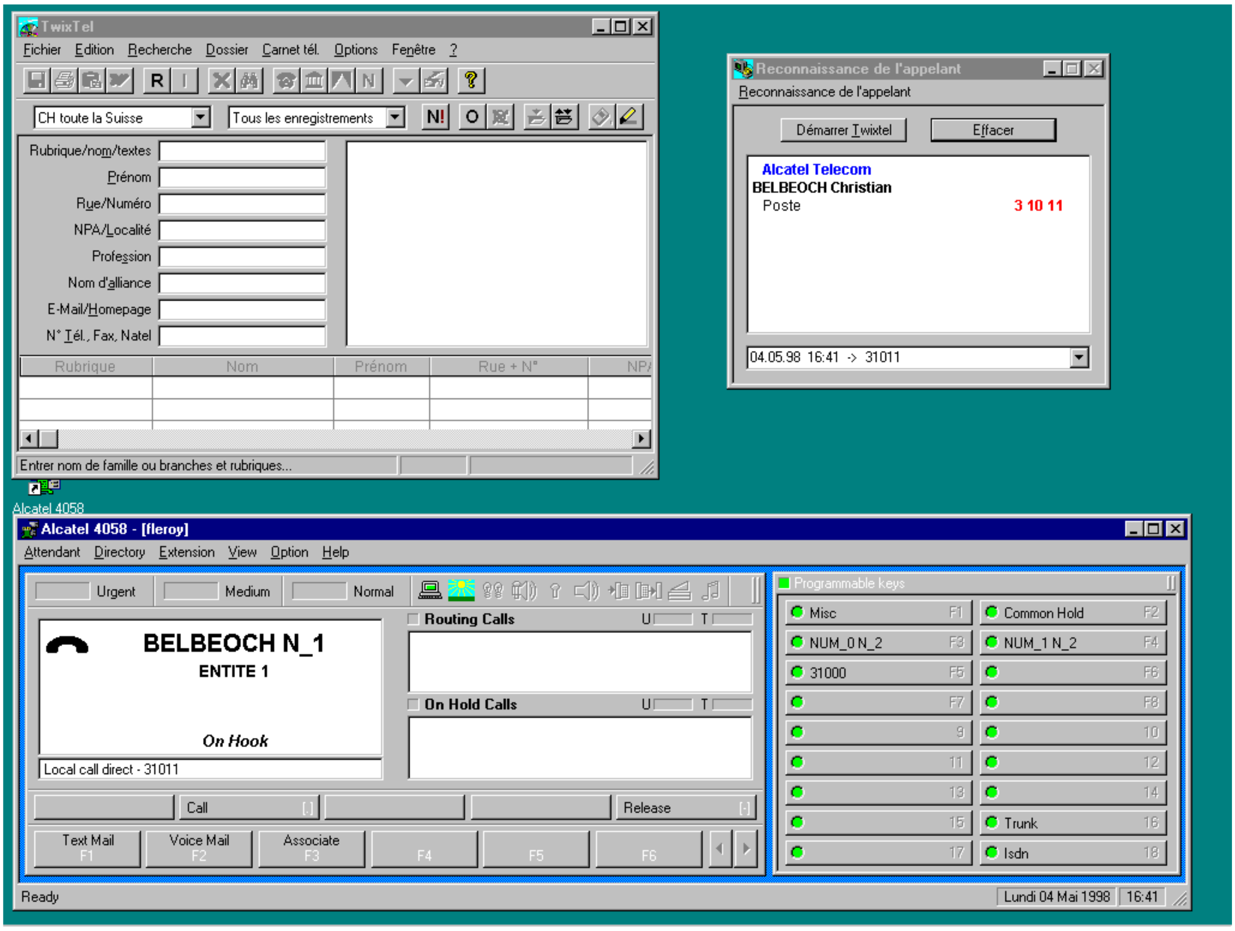
TAPI Assisted Telephony : same macros (Excel, Word) as for 4961 TAPI product or 4980 "PC Phone" are used and are available in 4059 help function.

TAPI compliant App : see following screen copies of public Directories : Twixtel (Switzerland) & Herold (Austria)

4059 MAC provides a TAPI 2.1 server on both platforms : Win 95 and Win NT
Normally TAPI 2.1 server is compatible with any TAPI 1.4, TAPI 2.0 or TAPI 2.1 application ...

This TAPI interface (limited to "Make Call" and "Receive Call") is linked to the pop-up function and software license.
Therefore TAPI services can also be available for the other Attendants Consoles on associated PC with pop-up license .... : to be confirmed

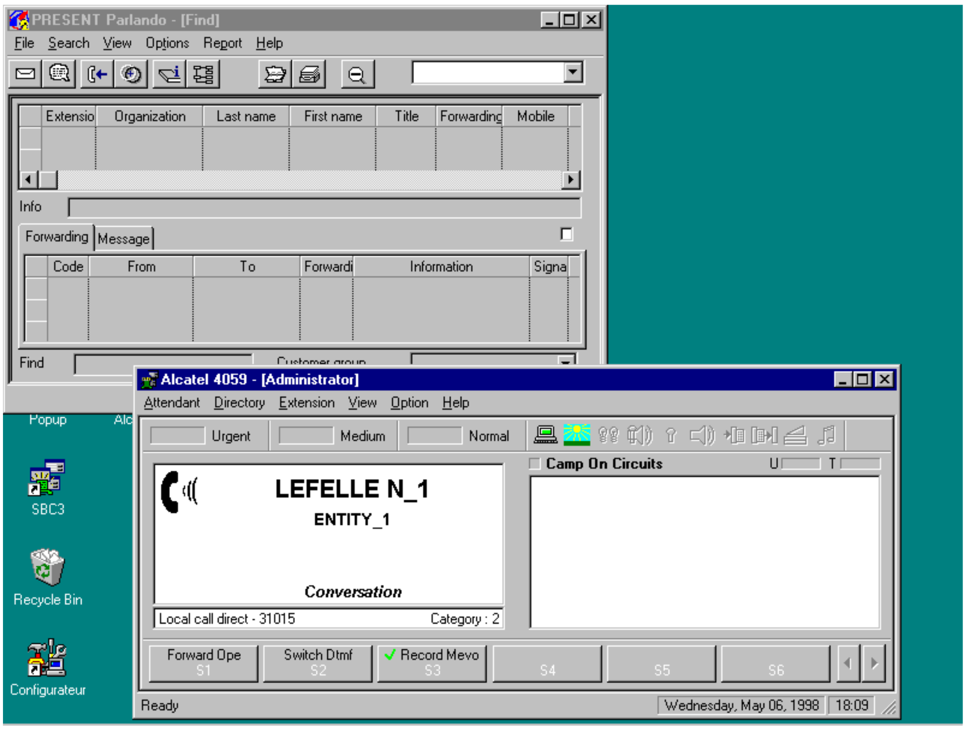
4059 and Twixtel : Screen Snapshot (4059 + Directory client + Directory pop-up)



In the lower part : 4059 MAC

In the upper part : Twixtel Directory search window (left) & Directory Pop-up window (right)

4059 and Parlando : Screen Snapshot (4059 + Directory client and pop-up)



13.3.3 Other evolution compared to 4058

Other evolution compared to existing 4058 :

- The right Directory and Personal Phone Book tools :
  - 4058 Local Directory is not suitable and therefore suppressed
  - Company Directory is 4000/4755 Directory or a third party Directory (ex. : Parlando in Nordic countries, see snapshot above)
  - Personal Phone Book is available in two forms : it is an integral part of 4059 App or can be also a third party App

- The right Attendant Management tool :
  - hence, Management on 4034 set is replaced by an optional 4059 Management Windows App

- The best of ergonomics for the integrated VIP "BLF" :
  - The VIP BLF function of 4059 MAC offers now the best of ergonomics :
    - no scrolling by limitation to 8 devices monitoring in 4059 BLF sub-window
    - call set-up (for direct call or transfer) by mouse click on screen

  - This change offers a clear positioning towards 4059 BLF which will be used in the following cases :
    - more than 8 devices monitoring per Attendant
    - network monitoring
    - need of specific 4059 BLF functions : mail per device, reading of absence info, …

Notes :
The "Personal Phone Book" in 4059 is linked with "Store & Redial Phone Book" function

The third party "Personal Phone Book" targets for integration with 4059 are :
- Schedule for Office 95
- Outlook for Office 97
- Why not also "Excel" in TAPI Assisted Telephony mode ?

Compared to previous management on 4034 set level, 4059 MGT offers :
- more services
- network wide management
- user friendliness

13.3.4 4059 Management Services

13.3.4.1 4059 "MAC" : Attendant Console management services

The following services are available in "Guide mode" and in "Service mode" :

Guide Mode
- Auto Answer mode
- Auto Transfer mode
- Ringing mode
- Language

Service mode
- Charging services : counters, reports, monitoring
- Traffic services : overflow, threshold
- Routing Table status : Attendant group, entity

13.3.4.2 4059 "MGT" : System & Subscriber management services

Alcatel 4059 "MGT" offers System & Subscriber management with the following :

* Easy to use & ergonomic
  ÷ only the relevant objects are shown
  ÷ two profiles are available :
    ⇒ Basic
    ⇒ Advanced

* Efficient and complete
  ÷ New objects available compared to previous 4058 Management :
    ⇒ Subscriber, Abbreviated dialing, Phone Book, DECT, ...

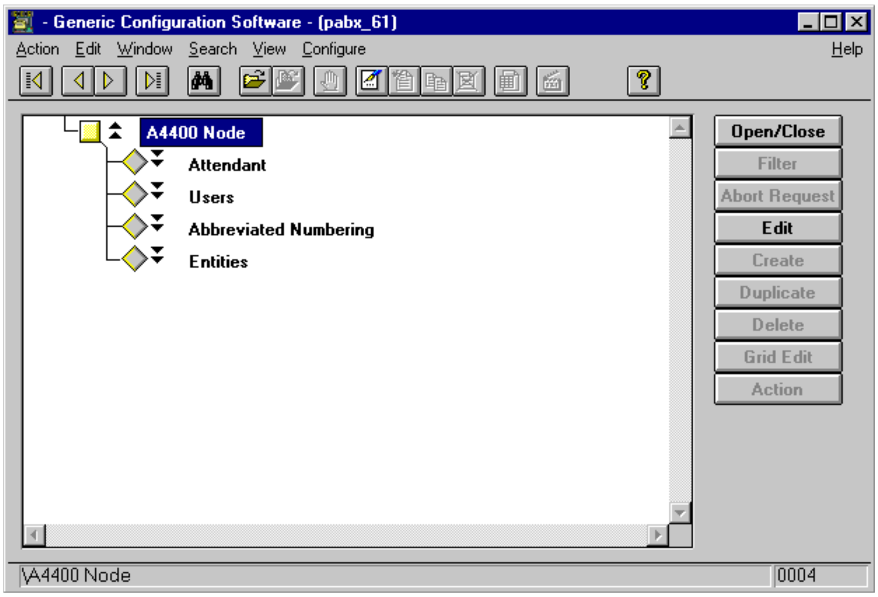
* Works private network wide

Compared to previous management on 4034 set level, 4059 MGT offers :

> more services
> network wide management
> user friendliness

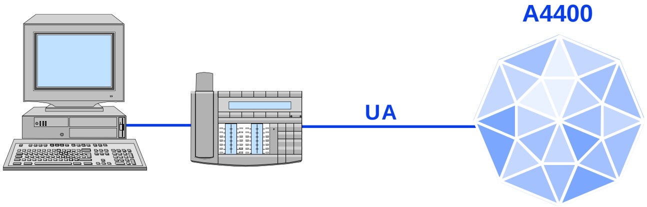
Up to 10 profiles are possible in 4059 MGT (the maximum of rights is given by default in Advanced profile)

4059 Management : Screen Snapshot

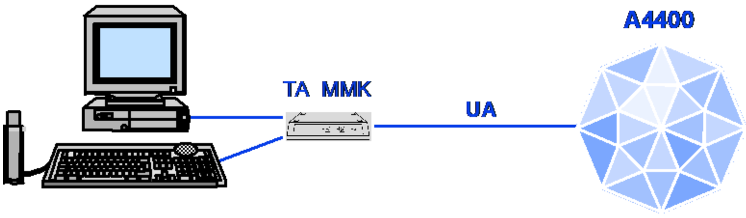


13.3.5 Availability

For 4400 R1.4, R1.5 and R2 : 4058



For 4400 R3 : 4059 MAC



For 4400 R1.4, R1.5 & R2 : 4058 R2.12x, 4812 R1.12x are compatible

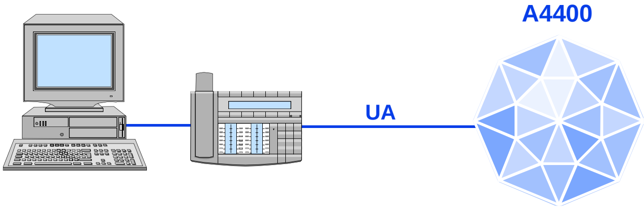
For 4400 R3 : 4059 MAC R4.x, 4059 BLF R4.x, 4059 MGT R4.4 are compatible

13.3.6 Update of 4058 to 4059

Alcatel 4058 is no longer functional in 4400 R3

Two 4058 update possibilities are offered :
- 4058 software > 4058 update software
- 4058 product > 4059 "MAC" product

Note that as for Alcatel 4048, 2 or 4 UA positions are required for 4059



What needs updating ? :
- 4058 software > 4059 software
- 4059 MGT software : if needed
  - Management services at 4034 set level are no longer available
- 4812 software > 4059 BLF sofware if associated with 4059
- PC ?
  - minimum is PC Pentium 166, 32 M, running Win95/Win NT4

One CD-Rom with the 4 new softwares is available :
 * 4059 MAC, 4059 BLF, 4059 MGT, 4058 update

Note that 4000/4755 Directory Client App remains compatible

For 4058 in existing 4400 releases (R1.4, R1.5 & R2) update to 4400 R3 is possible :
 * with/without multimedia keyboard & Terminal Adapter
 * with minimum : PC Pentium 166, 32 M, running Win95/Win NT4

13.3.7 Ordering

Items which can be ordered are as follows :
- 4059 MAC :
  * 4059 software & keyboard without PC for all countries
  * 4059 software & keyboard with PC for : French, English, Spanish, German

- Software Options :
  * 4059 BLF (Busy Lamp Field)
  * 4059 BLF in network :
  * additionally a license per node for remote supervision

Note that 4059 MGT (Management) is free of charge.

Two packages are available for 4058 > 4059 update :

- 4058 > 4059 Software update on one CD-Rom :
  * 4058 software > 4058 update software
  * 4812 BLF > 4059 BLF (optional : only if used before)
  * 4059 MGT (optional : only if used before)

- 4058 > 4059 Product update package :
  * multimedia keyboard and MMK Terminal Adapter
  * 4058 software > 4059 MAC software
  * 4812 BLF > 4059 BLF (optional : only if used before)
  * 4059 MGT (optional : only if used before)

  Update of 4058 to 4059 Software is delivered on one CD-Rom :
  ÷ 4058 software > 4058 update software
  ÷ 4812 BLF > 4059 BLF (optional : only if used before)
  ÷ 4059 MGT (optional : only if used before)

  Update of 4058 to 4059 Product package :
  ÷ 4058 software > 4059 MAC software
  ÷ multimedia keyboard and MMK Terminal Adapter
  ÷ 4812 BLF > 4059 BLF (optional : only if used before)
  ÷ 4059 MGT (optional : only if used before)

13.3.8 Software Licenses

The 4400 software licenses are as follows :
- number of 4059 consoles
- number of Pop-up
  * for Alcatel Directory or TAPI compliant Directory
- "automatic Directory dialing" per node
- "InfoCenter link" per node
- number of 4059 BLF
- "4059 BLF in network" per node

No 4400 software license is required for "4059 Management

13.3.8.1 High performance and reliability

High performance and reliability is provided by :
Win NT platform
new performance of 4400 <-> 4059 link

13.4 NEW SERVICES FOR ATTENDANTS

- ∗ Attendant to Attendant conversation (for direct call or before transfer)
- ∗ Manual connection with automatic distribution mode
  - ÷ automatic distribution is based on longest idle time of the Attendants

14. CALL HANDLING

14.1 HUNTING GROUP IN ABC HOMOGENEOUS NETWORK

Need

- have hunting groups in network for call distribution or security reasons

Offer

- Sequential, cyclic or parallel hunting groups are now available homogeneous private network wide (ABC F)

Availability :

- in Alcatel 4400 R3
- for all groups except Attendant, ACD, VPS and S0 groups

The maximum number of network groups is : 200
The maximum number of groups (network groups and local groups) in a node is : 200

14.1.1 Hunting Groups in Network : Mechanisms

Same service as for a group local to a node

A new management data is created to design :

- local group
- network group

The group is created in the ***reference node :***

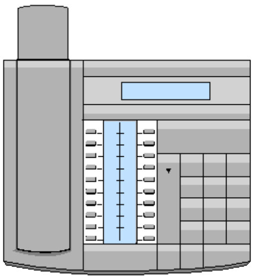
- broadcast of group states is done to the reference node
- choice of reference node is important to optimize the call distribution :
  - ÷ node with external incoming calls
  - ÷ node with the most important number of sets

Standby mode :

- in case of reference node failure, each local group works normally (master node or not)
- if reference node comes out of failure, the network group comes back automatically call by call for the new calls

14.2 ISDN CALLS LOGGING ON ALL SETS WITH DISPLAY

"Logging of non answered ISDN calls" is now available on all sets with display (4011, 4012, A4023, A4034, A4010, 4020, A4035)



Alcatel 4012

Alcatel 4010

Need :

* offer the useful "Logging of non answered ISDN calls" service on all UA sets with display (i.e. : not only on phone sets with softkeys)

Offer :

* "Logging of non answered ISDN calls" service is now available on UA sets with a small one line display of 20 characters

Availability :

* in Alcatel 4400 R3
* on A4011, A4012, A4010, A4020 phone sets (additionally to the phone sets with softkeys)

14.2.1 ISDN Calls Logging : Service Description

Notification

* when "ISDN" key icon is on

Consultation

* the user hits ISDN key
  ÷ the number of new ISDN calls is displayed
* the user dials "2"
* the user hits guide key "I" to scroll one ISDN call :
  ÷ Caller, date & hour, "Save" softkey
* the user hits left arrow to read previously saved ISDN calls
* the user hits right arrow to read next ISDN call
* the user hits the validation key "Loudspeaker" to initiate or to save the ISDN call

14.3 INTERNAL CALLS LOGGING

Need :

* offer the "Logging of non answered Internal calls" like for ISDN calls

Offer :

* "Logging of non answered Internal calls" service is now available on all UA sets with display

Availability :

* in Alcatel 4400 R3
* on UA 2G : 4011, 4012, 4023, 4034, 4040
* on UA 3G : 4010, 4020, 4035

14.3.1 Internal Calls Logging : Service Description

Notification

* when "Mail" key icon is on

Consultation

* like for "Non Answered ISDN calls logging" but with "Mail" key in place of "ISDN" key

14.4 NEW SERVICES FOR SECURITY

* New categories for trunk to trunk transfer
* Secured DISA

* Barring category for external forwarding

For those services, please see description in the Security part of this document.

14.5 ENHANCEMENTS FOR EXTERNAL DIRECTORIES CONNECTION

For automatic dialing from 4000/4755 or third party Directories, the "STAP" (Simple Telephony Application Protocol) protocol was only available through V24 connection between 4400 and the Directory server.

Now "STAP" protocol is also available on IP LAN connection.

14.6 CALL HANDLING ENHANCEMENTS IN R3.1

- 50 attendants per node and per attendant group
- Automatic DISA for non ISDN signaling (e.g. PCM)
- DISA allowed with GPA board

15. SECURITY SOLUTIONS

15.1 INTRODUCTION

One of the main social trends today is remote working. Customer requirements for mobility and tele-commuting, have led to the provision of remote access telecommunication facilities (DISA, voice processing applications), which opened the access of the corporation telecommunication resources to the external world.
Another major advance in computing and telecommunications of the two last decades, was the move from proprietary, closed systems, to standard and open architectures, enabling a worldwide interconnection and information share, as with the Internet today.
The advantages of system openness are widely recognised : software reusability (JAVA applets), standard communication protocols (TCP/IP), interoperability, wide access to information (Internet/Intranet booming), distributed computing are among the most relevant.

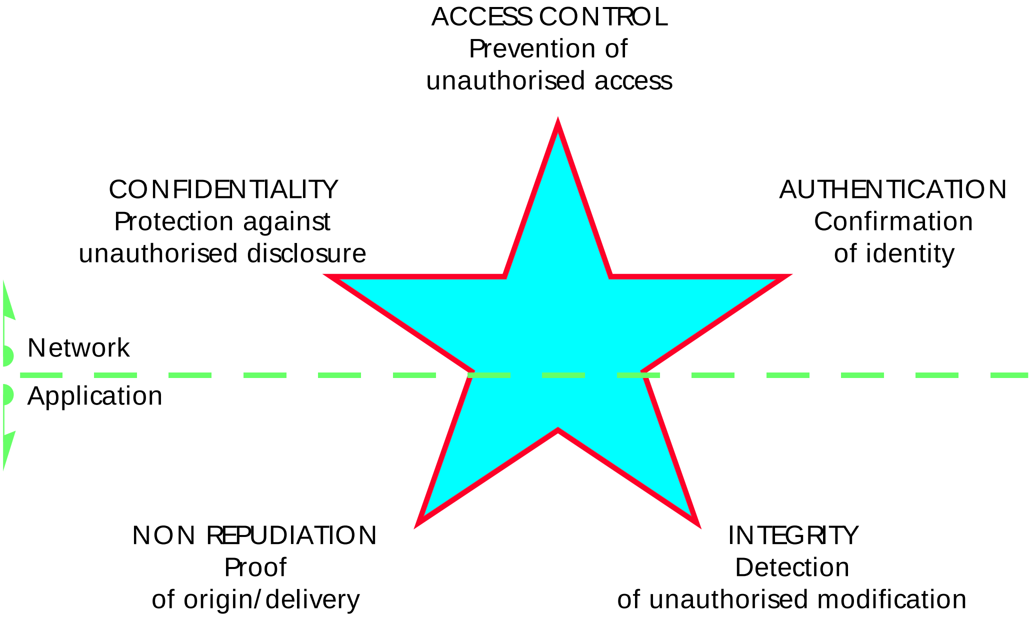
Telecommunication systems and PBX in particular, have also moved to open architectures, by adopting standard operating system (Chorus for A4400) and by offering standard interfaces for value added applications development.

The drawback of this evolution is that openness also means vulnerability of the system itself against fraudulent use or damage of its resources. This vulnerability has a cost. Telecommunication fraud, also called toll fraud, was estimated to more than $4 billion per year, in the US (end 94 figures from Pacific Bell), affecting nearly 25% of US companies.

This document focuses on A4400 security against toll fraud or fraudulent A4400 data access.

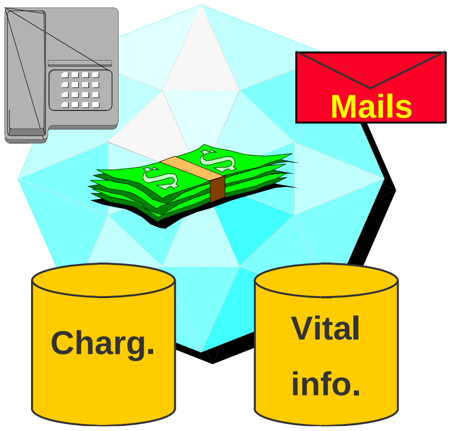
15.1.1 <u>The 5 + features of modern security</u>

The 5 key points of modern security are the following :



As far as the A4400 is concerned, three of these key points are relevant : access control, authentication and confidentiality

15.1.2 PBX at a glance



If we take a look at a 4400 installation, we have critical data (or vital information) such as system information, customer configuration data and charging information.
We also find a lot of telephony services which can cost the company : external outgoing calls (direct calls or through DISA services), and forwarding/transfer on external numbers (national/international and GSM).
Then, we have confidential information such as charging information, voice mails in voice mail server.

All types of data need to be protected against fraudulent access.

The presentation of the A4400 security solutions is divided in two parts :

÷ the first describing the protection against excessive telephony calls which can cost the company a lot of money. It is called toll fraud
÷ the second describing the access control to the A4400 and its critical and confidential data

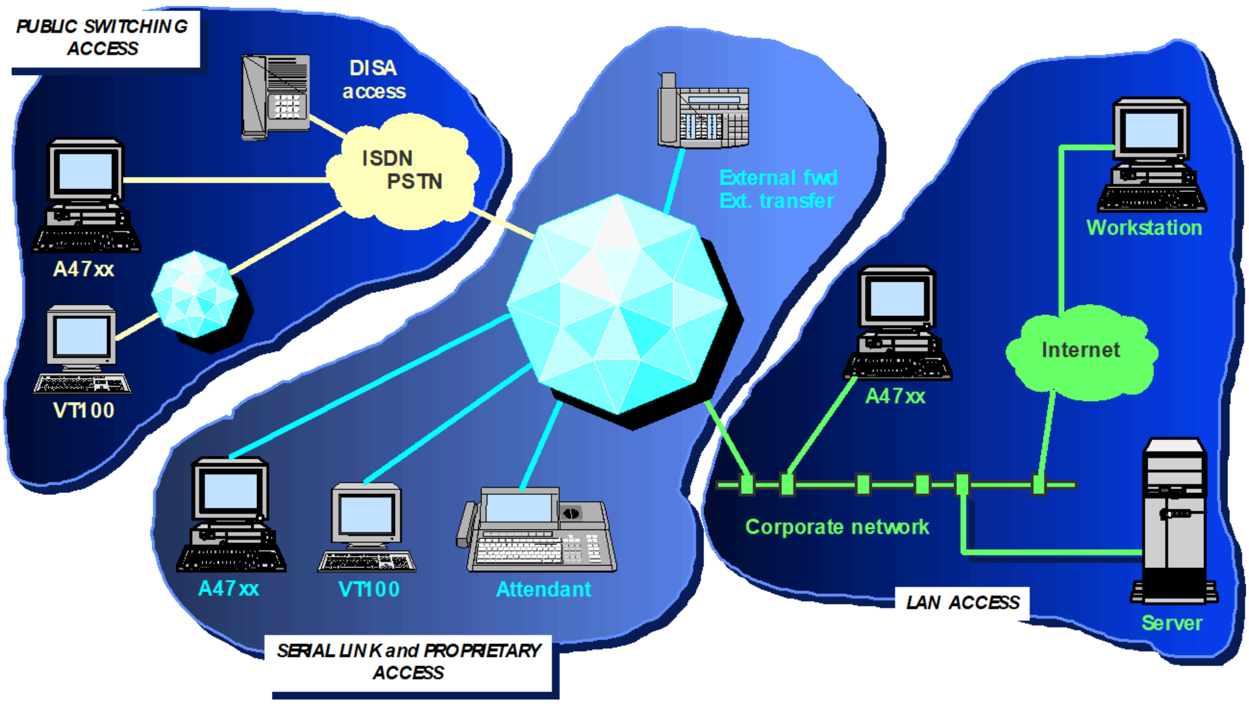
Before this, let's examine the different A4400 access points when connected to the external world.

15.1.3 A4400 access points

An A4400 as it is (if it is not connected to an external network) cannot be hacked. If the server needs to be connected on a corporate data network, security problems arise. If customer needs DISA, he decides to open the A4400 features to the external world ; then security problems need to be taken into account.

The different access to A4400 can be divided in three parts :

÷ access from the public switch exchange
÷ access from Internet/Intranet network
÷ access from the local terminals



15.2 PROTECTION AGAINST TOLL FRAUD ON TELEPHONY SERVICES

This part describes the protection against toll fraud on telephony services. Toll fraud is the usage of the A4400 features which can mean a company loses a lot of money because there are too many calls or excessive long distance calls.

Four internal features are concerned by toll fraud :

* Direct external calls from an internal set
* Transfer between two external calls
* External forwarding of an extension
* DISA (Direst Inward System Access)

15.2.1 Toll fraud protection from an internal set

For a company, excessive external calls can cost a lot of money. From every extension not barred (in a meeting room, in case of absence, in a public area), it is possible to make a long distance call.
For security, the recommendations are :

÷ barring must be adapted to the usage of the extension
÷ locking must be used in case of absence
÷ PIN (business or private) must be used for the external calls

15.2.2 Toll fraud protection in case of transfer between two external calls

A4400 offers the possibility of making a transfer between two external calls (basically, A4400 does not allow a transfer between two external calls). The external calls can be incoming or outgoing. The worst case is when the internal user has established two outgoing calls, then he makes a transfer. The two external correspondents are linked and the company continues to pay the two calls.

In R3, two new facility categories have been implemented in A4400 to improve the control of external call transfer. These two categories will be used by the system to authorise or not the transfer between two external calls (incoming calls or out-going calls). These two new categories are :

÷ transfer out-out
÷ transfer out-in

There are four situations :

- situation 1 : an internal subscriber has dialled two public calls. If the subscriber wants to make a transfer, category « transfer out-out » must be validated. If not, the subscriber will not be able to perform the transfer between the two external calls.
- situation 2 : an internal subscriber has dialled one public call and has received a second one. If the subscriber wants to make a transfer, category « transfer out-in » must be validated. If not, the subscriber will not be able to perform the transfer between the two external calls.
- situation 3 : same as situation 2, but the subscriber has to receive an external call fist of all.
- situation 4 : an internal subscriber has received two public calls. He is always able to make a transfer between the two calls because the company is not charged in this case.

Summary :

Type of communication	Type of communication	Parameter concerned
Outgoing call	Outgoing call	Transfer out-out
Outgoing call	Incoming call	Transfer out-in
Incoming call	Outgoing call	Transfer out-in
Incoming call	Incoming call	Transfert always authorised

In an homogeneous network, the different possibilities of transfer are managed as in a stand alone system even if the public trunks are connected on one node and the internal subscriber on another node.

15.2.3 Toll fraud protection when an extension is forwarded to an external number

External forwarding can also cost the company a lot of money. For example, if an extension is forwarded to an expensive number (e.g. : in the US), it is possible to reach this extension with a short distance call then the call is re-routed to the expensive destination (basically, this feature is not authorised but it is configurable). The company will pay the long distance call, the user will pay the short one. And these kinds of call can cost the company a lot money if they are excessive.
Another example, which is more and more used by nomad people, is the extension forwarded to a GSM set. The cost is very variable and depends on the location of the GSM set.

In R3, we have implemented a dedicated barring category for forwarded calls. As for direct calls, the re-routed call is subjected to a specific barring table. An example is, from his set, a user is able to make an international call but he is not able to forward his set to an international number even an national call if it is well configured.

15.2.4 Toll fraud protection when remote workers reach the A4400 through DISA

DISA brings a lot a benefits to remote workers and to the companies.
The remote working is a present major social trend and will be used with the PBX more and more.
It is a major social trend because there are a lot of situations , where the worker is out of the company. He can be at home if he is a home worker, or can be nomad (with a client, in a car, in a hotel ...). DISA offers him a lot of advantages. Whatever the situation, he can benefit from all the internal telephony features including all those which can save money as least cost routing or break-out.

But DISA must be well protected against hackers

DISA can be a danger if it's not sufficiently protected. A hacker can use DISA to substitute an internal subscriber and then dials as many long distance calls as he wants. The company will be fully charged.

DISA is a very sensitive entry point. This is the reason why we recommend our customers to apply one of the three possible protection levels noticed hereafter :

a) Password control
b) Caller line identification in ISDN environment or other environment providing such as R2, R1.5, R1 signalling followed by an automatic substitution
c) Strong authentication of the remote access → DISA becomes a personal access

a) Basically, A4400 offers the password control. It is standard in A4400. In R3, this first protection level has been enhanced. Up to R3, when a remote user sends a wrong password, the A4400 immediately cuts the line. The remote user had to re-dial the DISA number. In R3, the A4400 does not cut the line any more. He has the possibility to re-dial the password (password of DISA and/or virtual set password) during the same call (it is more comfortable for the remote user). But in terms of security, A4400 will count the number of attempts the remote user dials. When the number of attempts reaches a certain value (value is configurable, the value is initialised at 5), the A4400 will cut the bundle and lock the access. The access is locked for a temporary time duration (duration is a programmable parameter minimum 5 minutes). After this period when DISA is locked, the access is unlocked so as not to penalise other remote users. After 3 additional wrong attempts, DISA is locked again for a time duration twice as long as the previous one. This is a repetitive mechanism (no limit). At each phase (first wrong attempt, temporary lock...) A4400 generates an alarm.
DISA can be unlocked in two ways :

- after a temporary period
- by dialling a prefix number from a set (having the right category to do this). In network configuration, the set must be located on the same node as DISA.

The system manager can manually block the access by a configuration command.

b) Caller line identification followed by an automatic substitution is available since A4400 R2. This protection level is well adapted in ISDN environment from A4400 R3 or other environment providing such as R2, R1.5, R1 signalling from A4400 R3.1 followed by an automatic substitution and for ubiquity services. However, the line is identified but not the caller.

c) When security is considered as important for the client and when he wants or needs to secure the PBX access, A4400 offers a high-level solution to protect DISA based on strong authentication. There are different reasons to adopt this third protection level :

- Remote user must be personally identified (not only the trunk line)
- Password control is not safe enough and is restricting
  - ⇒ reusable passwords are not safe enough
  - ⇒ passwords are easy to guess
  - ⇒ password are difficult to remember
  - ⇒ password have to be changed frequently

- Passwords are cheap to implement, but expensive to operate high level of security

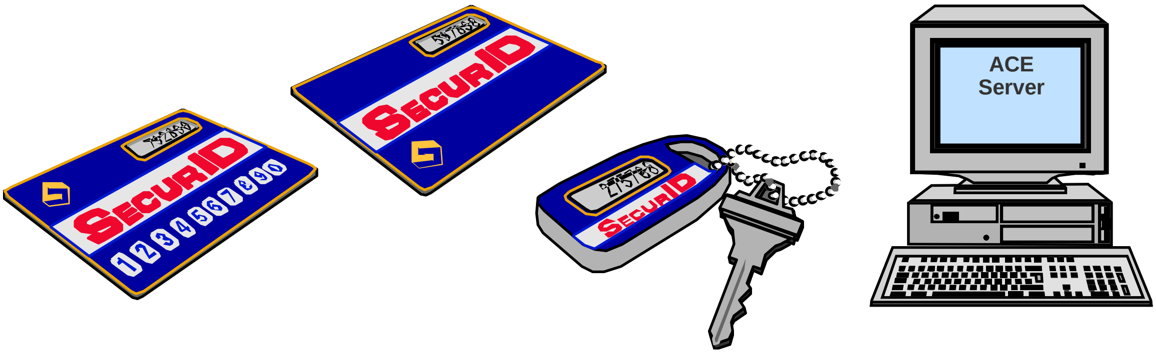
Presentation of the strong authentication solution for DISA :

A) PRESENTATION OF THE SOLUTION COMPONENTS

Strong authentication solution offered by A4400 is based on « Security Dynamics » solution called SecurID (SDI). It contains different external components :

* Token cards (credit card format with / without PINPAD)
* Token cards (key ring format without PINPAD)

* Authentication server (called ACE server)

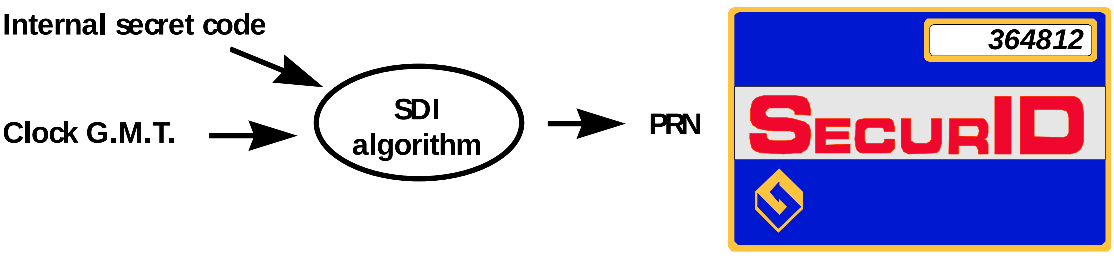


The token card is an access control security measure which is used to positively identify users of computer systems and networks. The SDI token, whatever its form, automatically generates a unique and unpredictable access code every 30 or 60 seconds (there are two kinds of card). The token card does not need a specific reader to read the changing code. It is displayed on the token card.
To properly identify and authenticate an authorised user, two factors are necessary. The first is something secret that the user knows : a memorised personal identification number (PIN).
The second factor is something unique that the user possesses : the SDI token.
The changing access code, displayed on the SDI token, guarantees the user must have the token in his or her possession at the time it is used.

B) ***<u>HOW DOES IT WORK ?</u>***

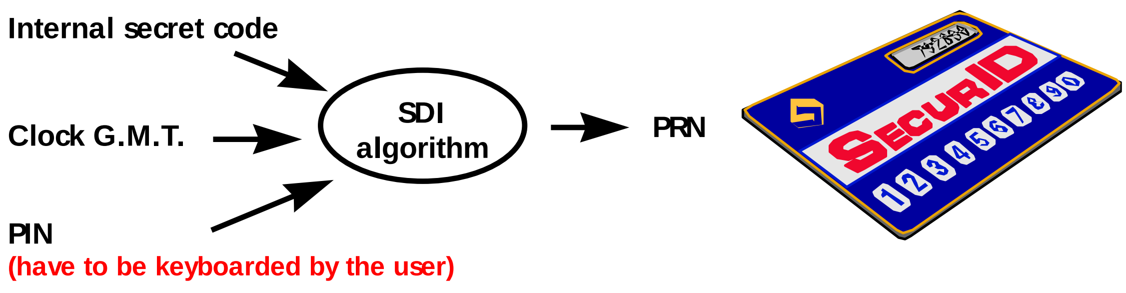
<u>Glossary :</u>

* PIN : Personal Identification Number
* PINPAD : keyboard on the token card
* PRN : token's unique code. It is the result of « Security Dynamics » proprietary algorithm
* PASSCODE : sequence of digit sent to the protected resource

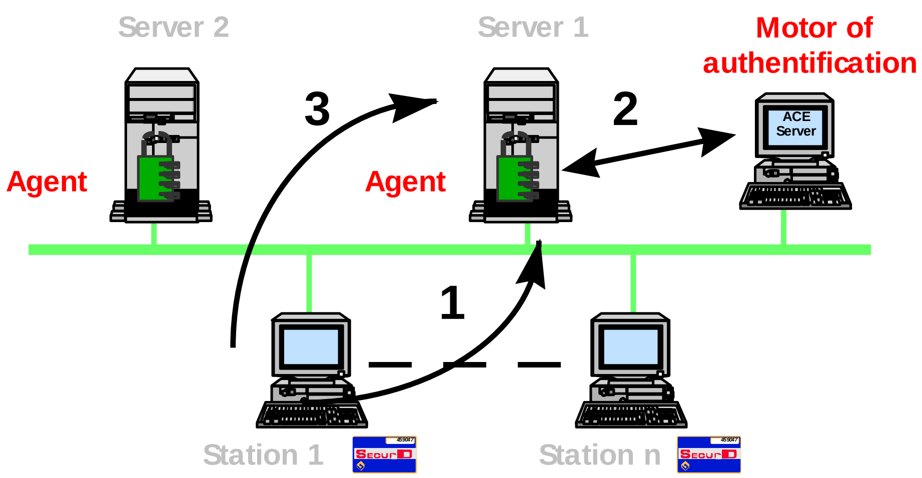


Each SDI token is programmed with a « Security Dynamics » proprietary algorithm, which, combined with unique parameters (internal secret code and the clock G.M.T .), assures that every number displayed is only valid for that user at that time. The result of the algorithm is called PRN.
To gain access to a protected resource, a user simply types his secret PIN, followed by the current access code PRN displayed on the token : PASSCODE = PIN + PRN
On the other hand, the ACE server is synchronised with each token thanks to the GMT clock.
Authentication is assured when the ACE server recognises the token's unique code combined with the user's secret PIN.

If the token has a keyboard (see the figure hereafter), the PIN is entered by the user (This is the reason why the keyboard is called PINPAD). This PIN is combined with the unique parameters. Then the PRN is the result of the three inputs.
To gain access to a protected resource, a user simply types the code displayed by the token : PASSCODE = PRN



Now, let's see how it works when a call is set-up in a datacom network configuration. The ACE server is connected on the datacom network and the user has his token.

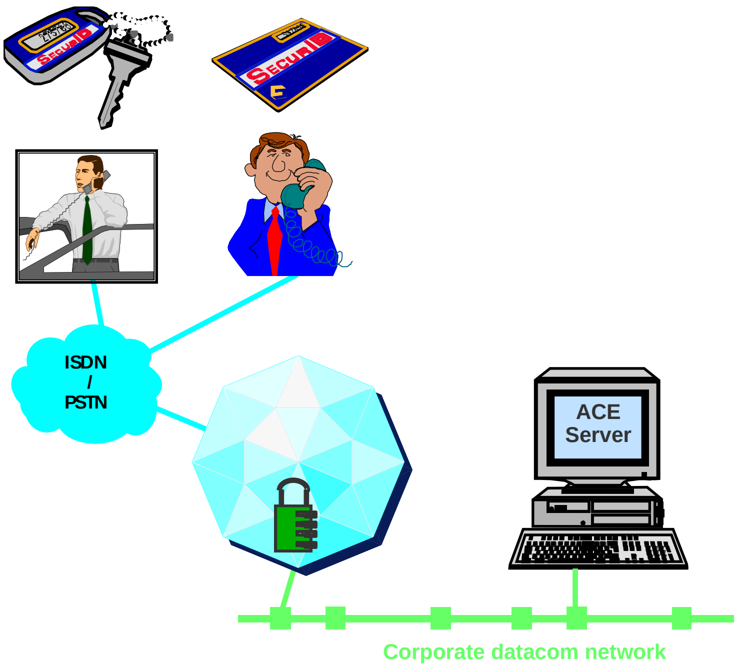


The ACE server (also called motor of authentication) needs an agent in the secured server. The rule of the agent is to lock or unlock the access after the user identification and authentication validation.
The call set-up protocol is divided into three steps :

* the first step is the demand. The user sets up a call from his station 1 to the server 1. He will identify and authenticate himself with the PASSCODE displayed by the token.
* then the server 1 and the ACE server will exchange information in order to validate the PASSCODE. For security reasons, during the validation phase, the messages exchanged between the agent and the ACE server are IP crypt.
* if the user is recognised, the agent will unlock the access for this user and the station 1 will be automatically connected to the server 1.

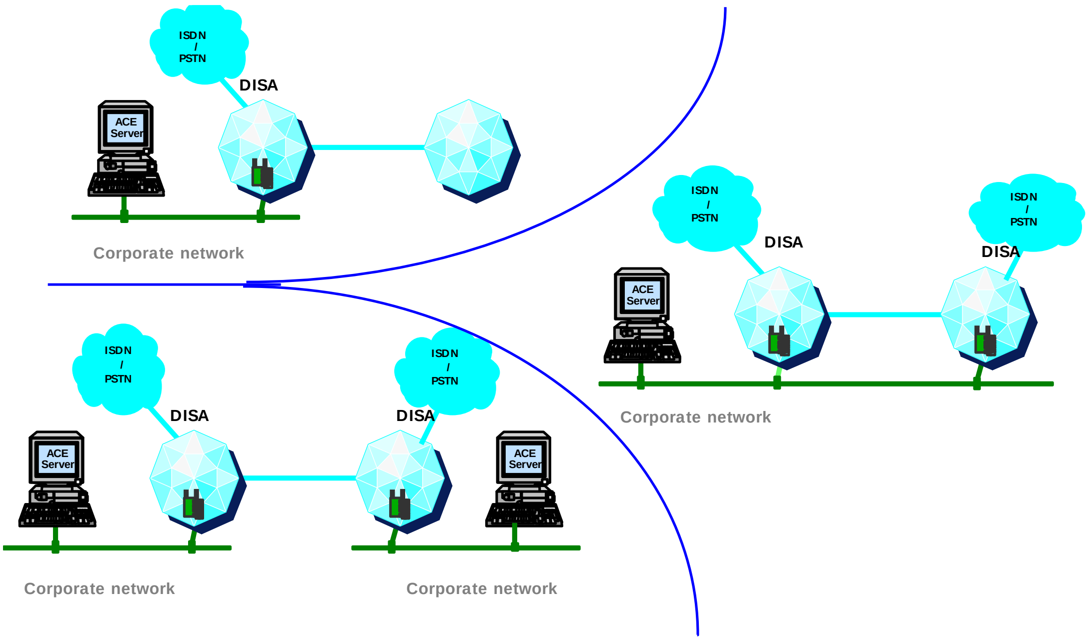
C) THE SOLUTION HAS BEEN ADAPTED FOR A4400 DISA

In R3, the « Security Dynamics » solution has been adapted to A4400 environment because the SDI agent has been ported into the PBX. A4400 now has the strategic partnership label from « Security Dynamics ».



And in R3, DISA can benefit from the strong authentication solution. The DISA users will get a token card or a key ring and the ACE server can be easily connected to the A4400 by the corporate datacom network.
A4400 now offers a high level solution to secure DISA. The customer can easily decide to open the A4400 to its remote workers. In addition to the security aspects, the sequence of digits sent by the remote user is optimised. He has to dial the following sequence <DISA number>+<Virtual set for substitution>+<PASSCODE>+<#>.
When the remote user has dialled the sequence, the ACE server will identify and authenticate him. The validation phase described previously is performed between the A4400 and the ACE server. At the end, the remote worker is automatically connected to the A4400. Several voice guides have been added to help the remote worker during the different phases.
The remote worker has the possibility of changing his PIN from its remote site. Voice guides will help with the change.

In A4400 network topology, DISA can be also secured. The above figure gives the different possible architecture.



D) ORDERING ASPECTS

This solution offers a lot of sales leaflets to the salesman because the strong authentication increases A4400 security solutions. And « Security Dynamics » solution is the leader in this domain. A4400 really has a high security solution to secure DISA used by remote workers . Lots of markets are concerned by security : military, banks, traders but also all companies developing the remote working.
Another sales leaflet is that kind of solution can be used not only for DISA security but also for the datacom world. The ACE server and the token are the same. A company with the ACE server for datacom security can re-use it to secure DISA. The users have the same token for datacom and voice connection.

15.3 ACCESS CONTROL TO THE A4400 AND ITS CRITICAL AND CONFIDENTIAL DATA

There are a lot of reasons to explain why a hacker wants access to a PBX. Some of the reasons are :

- ∗ he wants to destroy the PABX in order to stop the activity of the company
- ∗ he wants to read or destroy critical or confidential information in the A4400
- ∗ he wants to change configuration parameters to make toll fraud
  - ÷ class of service changing
  - ÷ barring category removing
  - ÷ extension creation
- ∗ he wants to delete or read charging information
- ∗ he wants to update ARS programming to re-rout external calls on a dedicated carrier
- ∗ he wants to use the A4400 as a gateway to the corporate datacom network

The first recommendation we have in terms of security is the physical access protection of the PABX system. A dedicated technical area locked must be able to control the access to the installation and to the main distribution frame.

Then A4400 has a lot of security solutions (embedded or externals). This part of the document presents the solutions. It is organised as follows :

- ∗ description of the basic protection by password
- ∗ description of the secured access protocol for A47xx management platform
- ∗ presentation of the solutions to protect the local access through a serial or proprietary link
- ∗ presentation of the solutions to protect the remote access through the public switch exchange
- ∗ presentation of the solutions to protect the remote access through Internet or Intranet

15.3.1 Description of the basic protection by password

A4400 is an UNIX system. It is based on standard Operation System and application software.
Basic A4400 security solution consists of protecting the access to the system by a password. This solution is fully independent of the physical access (V24, LAN), or the protocol used (V110/120, IP, PPP, Telnet). As a result, every time a user connects a A4400 irrespective of his location, terminal and protocol, a password will be required.

Protection by password is our first answer to the security problem.

Basically, access to A4400 software platform (including OS and file system), to management services (configuration, accounting) and to maintenance functions is protected by a password.

We recommend managing the password responsibly. A password handling policy must be put in place by the company. Don’t use passwords :

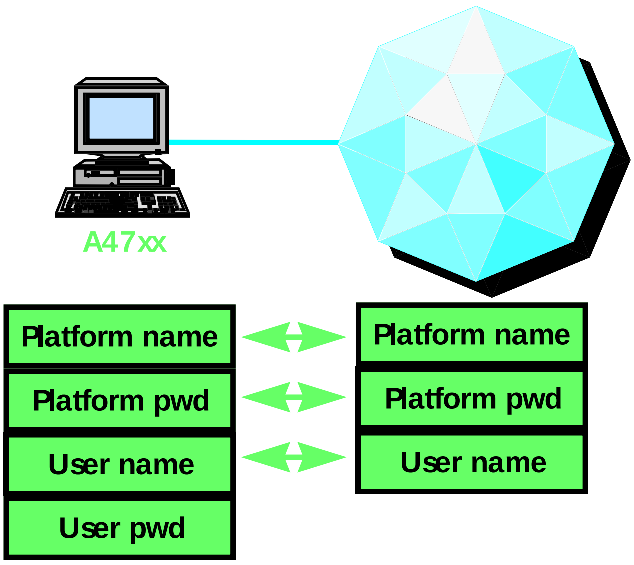
- ÷ based on the account’s name, in any form : as is, reversed, capitalised, doubled

- based on the user's name, or first name, in any form
- based on the user's family first name, in any form
- that match a word, or a reverse word, contained in a dictionary, spelling list or other lists of word
- that match a word in a dictionary with an arbitrary letter changed by a control character
- that match a word in a dictionary with some or letters capitalised or replaced by digits
- that are grammatically derived from a dictionary word (ex : plurals, conjugations)
- that are keyboard patterns (ex : qwerty)
- that consist only of digits or of the same letter
- shorter than six characters

and never use an UNIX account without password.

15.3.2 Description of the secured access protocol for A47xx management platform

To enhance security, A4400 controls the identity of the A47xx platform (A4715/30/40 A4755) and the user. The A47xx platform is defined by two items : a name and a password. The user is also defined by a name and a password.



Principle :
Each authorised A47xx platform and each authorised user are defined twice : in the A47xx platform and in the A4400. As described by the figure, the name of the platform and its password are configured on both sides. For the user, only his name is known by the A4400 and the A47xx. The user password is local to the management platform.

The access control is carried out during the connection phase. When a A47xx sets up a call to the A4400, the PABX will check the consistency of the three parameters (platform name, platform password, user name) and will authorise or prevent the connection.

This A47xx connection protocol prevents unauthorised PC or Sun stations from reaching a A4400. By default the table configuration offers the connection to all stations. For security reasons, the tables must be filled in after the cut-over.

This solution is fully independent of the physical access (V24, LAN), or the protocol used: (V110/120, IP, PPP, Telnet). As a result, every time a user connects a A4400 with a 47xx platform, wherever the location and the protocol, the control will check the identity of the platform and the user.

In network the data are broadcast into the different nodes.

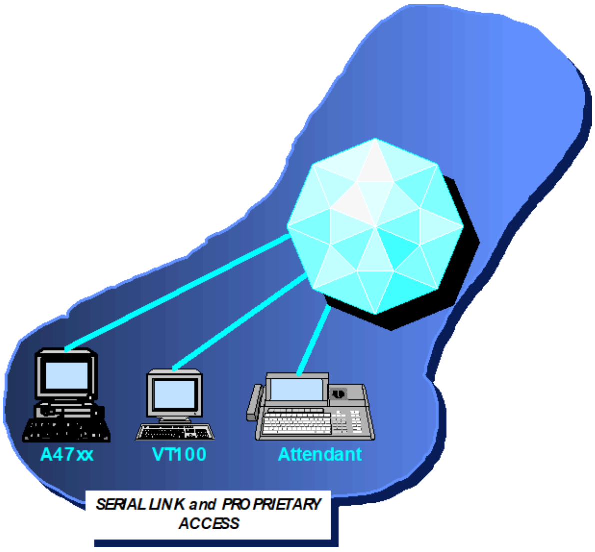
Limits :
Maximum number of platform authorised : 32

Maximum number of user authorised : 32

Compatibility between the software releases :

	A4400 <R3	A4400 >=R3
A47xx < R4.4	connection is always accepted	an option offers the possibility to reject (by default) all A47xx connection or to accept all A47xx connection
A47xx >= R4.4	connection is always accepted	connection is secured as described

15.3.3 Solutions to protect the local access through a serial or proprietary link



There are three possible ways to access the A4400 through a serial or a proprietary link :
- by the attendant console
- by a VT100 terminal
- by a A47xx management platform

Each one is protected by a password.
A password secures the access to configuration application from the attendant console.
Other passwords protect the access to maintenance services and configuration application from the VT100 terminal.
Then when the A47xx platform wants to reach the A4400, the secured access protocol described before is applied.

Every time access to configuration data is gained, A4400 stores a trace into an history file. Before the release 3, A4400 stored who made the access : the A4730, the attendant, the broadcast in network mechanism.... Now in R3, the A4400 is able to give in detail exactly what MAO commands where made for each access : creation, setting, deleting, on boards, sets, on repertory keys.... Hereafter, we show several examples of information given by this feature.

- Example of information given by configuration history file : (Last 2000 event are stored into the file)

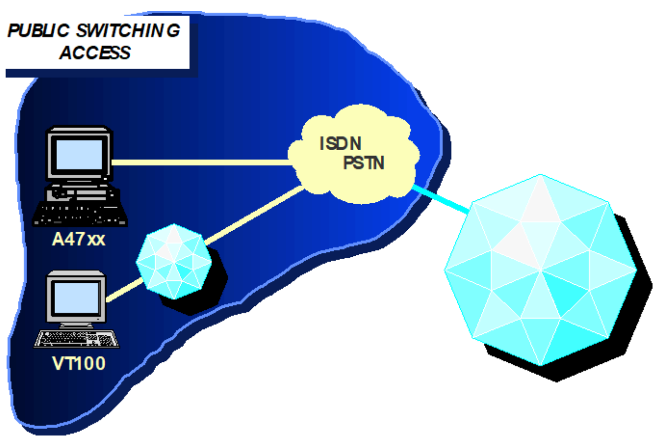
  1- 1997/11/12 15:10:05 Login (MANAGER) accepted
  2- 1997/11/12 15:10:44 Update Users 1 31005 (MANAGER)
  3- 1997/11/13 15:51:34 Login (MANAGER) accepted
  4- 1997/11/13 15:52:12 Create board 1 0 0 (MANAGER)
  5- 1997/11/14 17:17:22 Login (MANAGER) accepted
  6- 1997/11/14 17:18:57 Delete users 1 31030 (MANAGER)
- Example of information given by configuration history detailed file :

  Creating Object [890023, Board]

Node number (reserved) : 1
Shelf Address : 0
Board Address : 0
Interface Type : UA_16
Usage State : Idle
Operational State : Disabled
Main/Standby State : Main (Master)
Number of Sets Being Connect : 0
Country Protocol Type : Default
Send Init Dynamic Msg : FALSE
Parameter By Default : TRUE

- Example of information given by call handling history file :
  1998/03/27 10:08:34 Update Users 6320 (CH)
  1998/03/27 10:09:47 Update Repertory Keys 6320 003 (CH)
  1998/03/27 10:09:47 Update Repertory Keys 6320 005 (CH)
  1998/03/27 10:10:03 Update Prog.Keys 6320 002 (CH)
  1998/03/30 12:05:42 Update Attendants Group 2 (CH)
- Example of information given login history file :
  1- 1997/12/11 15:10:05 Login (MANAGER) accepted
  3- 1997/11/13 15:51:34 Login (MANAGER) accepted
  5- 1997/11/14 17:17:22 Login (MANAGER) accepted
  9- 1997/12/05 09:20:43 Login (broadcast) accepted
  11- 1997/12/06 16:46:56 Login (afe) accepted
- Possibility to search match string in configuration history file :
  String to find [List of objects = ?] : Users
  2- 1997/11/12 15:10:44 Update Users 1 31005 (MANAGER)
  6- 1997/11/14 17:18:57 Delete Users 1 31030 (MANAGER)

15.3.4 Solutions to protect the remote access through the public switch exchange



A4400 offers three security solutions to protect the remote management access through the public switch exchange :

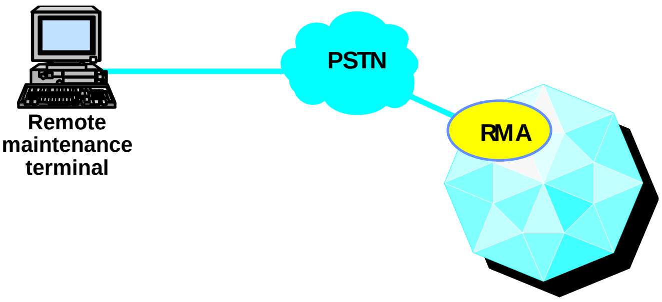
a) The first solution is used in PSTN environment. RMA equipment ensures the control of the access from maintenance and/or management terminals located on a remote site.
b) The second solution is used in ISDN environment. In R3, A4400 is able to identify and authenticate the remote terminals by caller line identification sent by the public switch exchange.
c) In both cases (PSTN or ISDN), A4400 R3 will identify and authenticate the remote A47xx platform and the associated user with the protocol described above.

a) In PSTN environment, this solution provides a high security level for remote management terminal at a predetermined location.

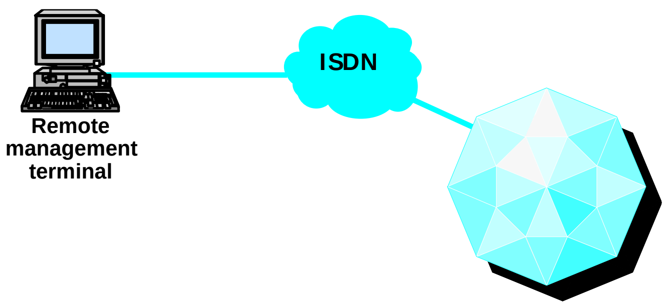
Remote access security through public switch exchange is provided by RMA device. RMA device is available in all A4400 releases. Its main security services are :

- the identity control of the remote engineer by a login/password
- a call back mechanism

When the remote worker is logged into the RMA, a second login/password will be required when he will entry into the A4400.
In this case, security is ensured by a double login/password.



b) In ISDN environment, A4400 also offers a feature providing a high security level for remote management terminal at a predetermined location.
A4400 R3 controls the access with the caller line identification sent by the public switch exchange. Access is mainly achieved through the A47xx platform for remote management (file transfer of the accounting tickets, configuration, ...). The A47xx sends its caller line identification when it establishes the call. If the caller line identification has been configured into the A4400, the call is accepted. If the caller line identification is unknown by the A4400, the call is rejected. In this last case, A4400 notices the rejection by an alarm generation.

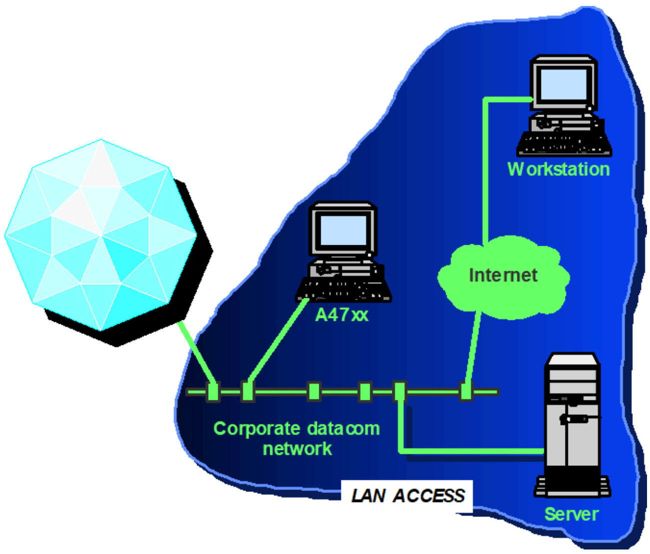


c) The remote A47xx identification and authentication is performed in addition and after the previous ones. The identification and authentication is realised with the secured access protocol described before.

As for local access, each access from a remote site is traced and stored into the history file described above.

15.3.5 <u>Solutions to protect the remote access through Internet or Intranet</u>

The number of Internet users is growing day by day. Consequently, hackers increase with the growth of the users. The companies need to take the security problem seriously into account. A mandatory need is firewall protection (or firewall-like protection) if the corporate datacom network is linked to the Internet or to any remote network. Firewall are the only mechanisms that can adequately protect the computing and communication resources against thieves and vandals.
<u>What is a firewall ?</u> To caricature, think of a firewall as a giant door (or a large wall) blocking the only entrance into a house. Friend and foe, alike, must knock on this door to request admittance. When foes knock, the door recognises them as intruders and repels them immediately before they ever set foot inside the house. But when friends request entry, the door opens and admits them at once. To friends, the door into the house is invisible.



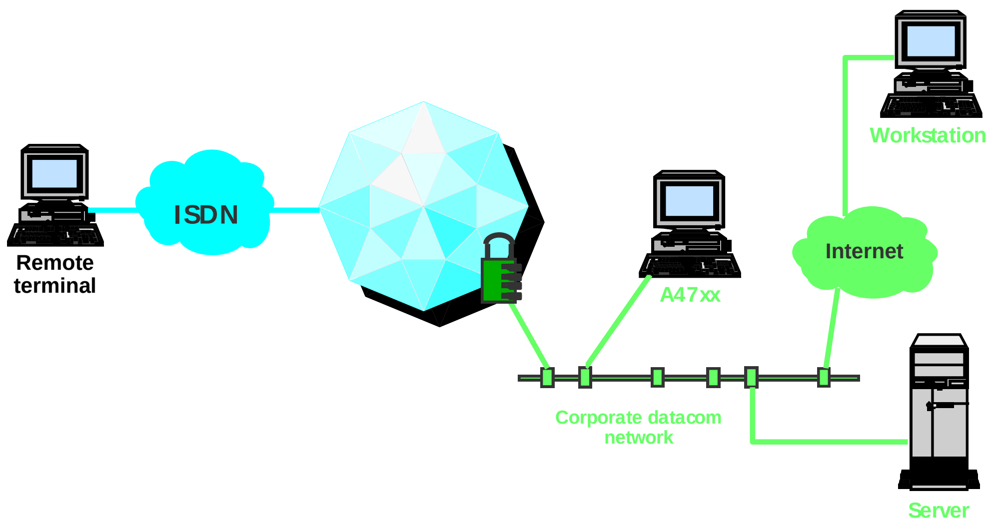
A4400 server offers four security solutions to protect and control remote access through IP networks.
Two of them are integrated into the A4400 server and provide a real embedded « firewall-like » protection :

- Trusted host (available since R1.4) : A4400 server controls the IP address of the workstation which wants to be connected
- TCP wrapper (new in R3) : A4400 server controls the service exchange with the external IP workstation.

The combination of these two first solutions brings to A4400 an embedded « firewall-like » protection.
The third security solution is, of course, an external firewall which can be purchased by the customer. Nevertheless, we have signed a distribution contract with the company « Check-Point » which is the leading company on this market segment.
The last solution can be added to the previous ones. It is the secured access protocol reserved for remote A47xx platform (see description above).

a) Description of the A4400 integrated feature : **trusted host** (available since R1.4)

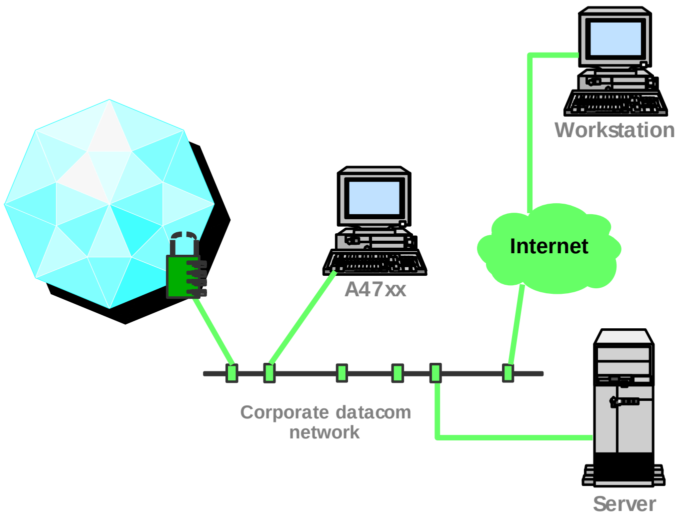
The feature called trusted host in the data world, consists in controlling the IP address of the workstation which wants to connect the A4400. The control is done at the network level (refer to OSI model). Basically, A4400 server offers this security solution.



The control is done by the A4400 for connections in both directions. The first direction is a workstation, connected on the corporate datacom network or on the Internet network, which wants to reach the A4400 server. In this case, IP address is control by the A4400 server which is able to authorise or not the connection.

The second direction is a remote terminal, connected behind the public switch exchange, which wants to reach the A4400. When it is connected, the remote terminal has the possibility of recalling an IP system on the corporate datacom network (A47xx, server, workstation....). In this case, A4400 server is also to control and filter the IP address.
This security service is the minimum service which must be installed when A4400 becomes a server.

b) Description of the A4400 integrated feature : **TCP Wrapper** (new in R3)
The feature called TCP Wrapper in the data world, consists of controlling the service exchange between the A4400 server and the workstation. This feature is complementary to the previous one : trusted host. It improves the A4400 security. The combination of both, trusted host and TCP wrapper brings to A4400 server a real embedded « firewall-like » protection.



The control is carried out by the A4400 server for connections from an IP stations connected on the corporate datacom network. The control is carried out at level 5 of the OSI model. The services monitored and filtered on the A4400 server are : FTP, TELNET, SHELL, LOGIN...

Example :
The following example will illustrate the two previous descriptions, trusted host and TCP wrapper. As an example, we have a corporate datacom network where servers, workstations and A47xx platform are connected. The corporate network is linked to Internet world.
The security policy put in place by the company is as follows :
÷ The accounting manager should be able to transfer charging ticket from the A4400 server to the A4715 platform and nothing else (no maintenance , no configuration...)
÷ Any external users from Internet network should be able to connect the A4400 server

A4400 proposes an administration network menu where trusted host and TCP wrapper features can be configured.
The following questions will be asked to the A4400 network administrator (the menu is only available on a specific account protected by a specific password).
÷ Trusted host IP name ?...................................... A4715
÷ Trusted host IP address ? 10.253.1.10
÷ Gateway IP name ?.. router1
÷ Do you want to accept all services ?................ n
÷ Do you want to denied all services ?................ n
÷ *(These following questions are optional. They are asked if the two previous answers are « n »)*
÷ Do you want to allow FTP services ?................ y
÷ Do you want to allow TELNET services ?.........n
÷ Do you want to allow SHELL services ?...........n

- Do you want to allow LOGIN services ?........... n
- Do you want to allow NETACCESS services ?.. n
- .....

c) Description of the security solution based on an external **firewall** (independent of A4400 release)

For high security, A4400 proposes external firewall. External firewall brings additional security services to the embedded ones. As an example, timed session, encryption, address translation and notification on SNMP traps can complete A4400 security solutions. Hereafter, a description of the additional services brought by an external firewall will be given.

There are two types of firewall technologies available today : static packet filtering and dynamics firewall. Static packet filtering and dynamics firewall can be implemented as either stand-alone devices or integrated into routers or remote access servers.

Static Packet filtering :

Static packet filters examine every packet passing through the network interface to see if it meets pre-established requirements about source and destination IP addresses. However, static packet filters do not monitor the « session state » ; the real-time events involved in sending and receiving data during a TCP session.

Because they do not perform session state monitoring, static packet filtering limits your control and potentiality places your network at risk. During an FTP session, since static packet filters do not know which port number a remote caller and a FTP server have negotiated for a file transfer, they keep open all the high-numbered ports. This makes thousands of ports vulnerable to probing attacks from unauthorised users for the duration of the FTP session. If all the ports are open, intruders can break into your network. If all the ports are closed, even authorised users are prevented from entering the network.

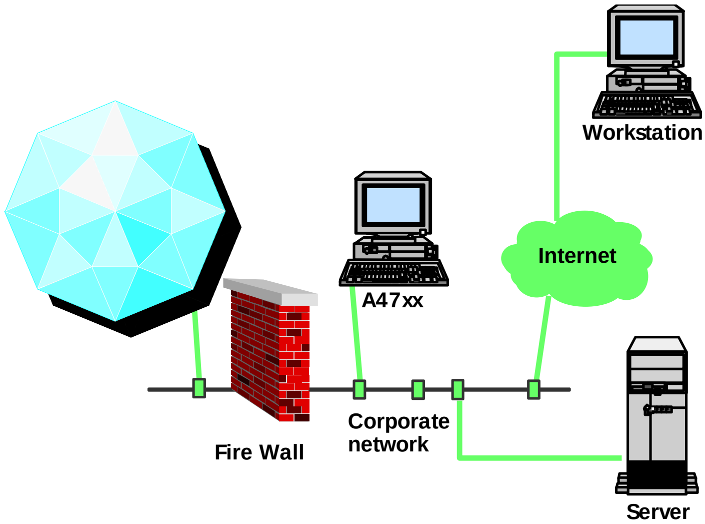
Dynamic firewall :

Dynamic firewall is an intelligent, next-generation firewall technology that provides a more secure solution than static filters.

Dynamic firewall gives you more granular control over users entering the network because they use state-of-the-art technology to create dynamic rules and adapt them to changing network traffic in real time. These rules can be modified to accept or reject conditions depending on specifications such as applications, protocols, network address, session state or direction. Once a session has been initiated, dynamic firewall monitor requests to open ports between terminating points. They open only designed ports and keep all other ports closed. When the session has ended, the ports are immediately closed, eliminating the potential for hackers to infiltrate the network and your company's sensitive data.

FIRE1 description :

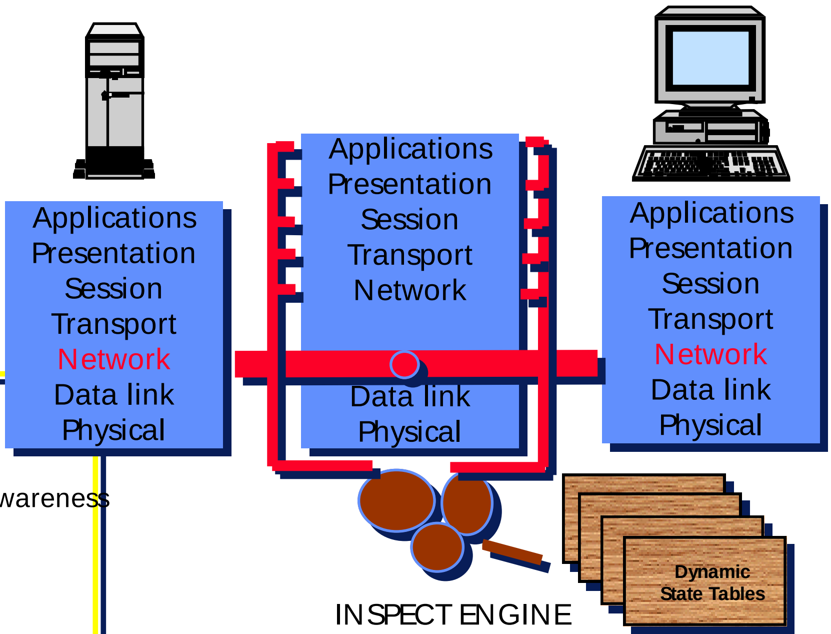
We have signed a distribution contract with the company « Check Point ». Check Point is the leading company on this market segment. We have kept the product FIRE1 (FW1) in our security offer. FIRE1 is a dynamic firewall which completes the embedded « firewall-like » protection in the A4400 server.



FIRE1 is installed on the corporate datacom network and provides a high security level to A4400 server. It offers security features such as :

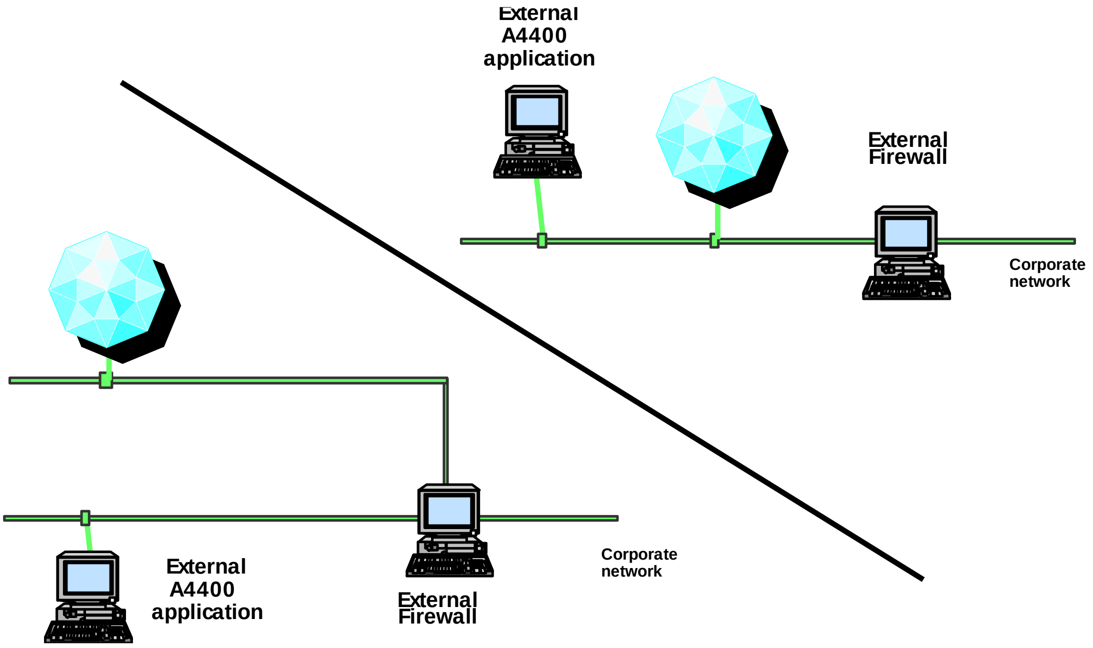
- notification on paging, e-mail or SNMP traps in case of non authorised access
- network address translation
- timed sessions on port
- timed security policies
- virtual encrypted networks
- encryption
- timed based user access
- strong authentication (with securID)

FIRE1 is an application software including a stateful inspection technology. It provides a full application layer awareness without breaking the client / server model. The packet is intercepted at the network layer, but then the INSPECT engine takes over.
Stateful inspection extracts state-related information required for the security decision from all application layers and maintains this information in dynamics state tables for evaluating subsequent connection attempts.
This provides a solution which is highly secure and offers maximum performance, scalability and extensibility.



PRO :
* Good Security
* Full Application-Layer Awareness
* High Performance
* Extensibility
* Scalability
* Transparency

To protect A4400 server and its external applications, the firewall can be inserted into the corporate datacom network by two different ways :



FW1 requires a hardware platform to run the application. The hardware platform can be a PC or a SUN station. The ordering is based on licenses. There are licenses for the number of IP address to protect and for services (encryption, @ translation ...).

These previous orderable items are available in the DATA BUSINESS UNIT catalogue in order to simplify the distribution contract with « Check Point ». People from DATA BUSINESS UNIT are certified and to avoid a double organisation in DSU (sales, experts, logistic...), the components are not available in the A4400 catalogue.

16. HOTEL/HOSPITAL

16.1 HOTEL APPLICATION ENHANCEMENTS

The following enhancements are now available for hotel business :

Enhancements on guest services :

From R2.1 :
- Dual line on digital sets

In R2.1, R3.0 support, and R3.1 :
- Password = extension number (last 4 digits)

From R3.0 :
- Privacy feature
- Multi-language : up to 8 languages

From R3.1 :
- A new hospitality Voice Mail (4635 R3.0)
- Japanese language on 4635 R3.0
- Multiple wake-up and snooze
- Wake-up in network
- set (guest) locked after n password entry errors

Enhancements on accounting services :

From R2.1 :
- Itemized billing for a given period of time
- Double currency billing

In R2.1, R3.0 support, and R3.1 :
- Call accounting by duration with pulse simulation
- Call accounting by duration parameters : up to 10 000 directions and 200 charge scales

From R3.0 :
- Call accounting extended ticket

From R3.1 :

- Call accounting in network

Enhancements on A4400 interface to external applications :

From R2.1 :

- Manageable buffer size : from 500 to 2500 tickets

From R3.1 :

- AHL on TCP/IP
- AHL centralization
- Certified Partners Program

Enhancements on sets :

- Op console : conference facility
- Guest Voice mail access from 4010 and 4020
- Wake-up programming tools
- Analogue hotel sets : a new range of sets
- The smart card write and read system (badge sets)

Other R3.1 enhancements :

- Room number : from 5 to 8 digits
- Static association guest / room
- Guest information on room service sets in network
- GPIN or room number data displayed on guest sets : option
- Time zone change transmission in network

16.2 INTRODUCTION

Hospitality recovers two sectors: the hotel industry and the healthcare industry.

The two of them have a basic common need: providing telephone services to their pensioners and re-invoicing them these services. Due to this common need, the A4400 Hospitality version includes specific software.

But as we look deeper at these sectors needs, it appears they become different. Thus some new services are more designed for the hotel industry and others for the healthcare industry. This orientation will be specified in the following document.

An hotel is a sort of carrier for its guests: it offers them basically the possibility to receive and give phone calls, and this represents a real source of revenue due to the price additional charge fixed by each hotel.

But, to face the international GSM expansion, the hotel industry has to enhance its services offer, as well as it has to manage costs reduction.

Therefore Alcatel gives the hotel industry some specifically new designed tools and services which are going to be presented hereby.

A hospital is also a sort of carrier for its patients, who do not have automatically the same use of telephone, as guests do (no modem connection, less voice messaging…). At least the patients given services do not need to change so much.

A hospital population is constituted of a large part of administrative and auxiliary nursing staff, who has definitely different needs such as mobility, performing alarm systems...

Of course, costs reduction and management is also a great issue in the healthcare industry.

Thus Alcatel systems for the healthcare industry is less patients oriented than new technology oriented (AHL on TCP/IP, DECT, ARS…).

At the end of this document, you will find a summary table indicating which features are more hotel industry oriented or healthcare industry oriented.

16.3 GLOSSARY: REMINDER

- **HOTEL / HOSPITAL APPLICATION PACKAGE**
  From one general software, we generate different package :
  - one basic Hotel Package
  - different options for Hotel

  Multilanguage Voice Guides
  Alcatel Hotel Link (AHL)

- **ROOM BASED CONFIGURATION**
  It is what was used to be named "mono-occupation mode". The internal A4400 hospitality software considers the room number as the identifier.

- **GUEST BASED CONFIGURATION**
  It is what was used to be named "multi-occupation mode". The internal A4400 hospitality software considers the guest number (GPIN) as the identifier.

- **GUEST & ROOM BASED CONFIGURATIONS DELTAS**
  These two sorts of configurations imply differences in terms of available services. Let's have a look to the main ones.
  Guest based configuration allows:
  - multi-occupation (more than one person per room with separate billing)
  - flexible suites
  - rotating DDI
  - club DDI (pre-check-in)
  - room allocation
  - room change

- **GPIN : GUEST PERSONAL IDENTIFICATION NUMBER**

- **PMS : PROPERTY MANAGEMENT SYSTEM**
  A PMS is a server used in hotels at the front office. All in all, its functions are: reservation, check-in, checkout, billing, guest database…
  Examples: Fidelio, Sulcus, HIS, Prologic…

- **CMS : CALL METERING SYSTEM**
  A CMS is a server used in hotels at the back office. All in all, it is a call accounting tool, and most of the time acts as a gateway between the PABX and the PMS. CMS providers also often have some Voice mail systems and Wake-up systems.
  Examples: TMS, FCS, ComTelco…

- **AHL : ALCATEL HOSPITALITY LINK**
  AHL is a CTI link between the PABX (A4400) and a hospitality server. It is used both in hospital and hotel environment. Its protocol is secured because of its acknowledgement structure. Physical connection can be done through a V24 or Ethernet.
  For more information, see aww.edd.alcatel.com/business/css/cataport/index.htm .

- **BASIC TELEPHONY FEATURES**
  - administrative people : they have access to every features offered by the A4400
  - guest : they need the basic telephony features plus some specific features described hereafter
  - housekeeping and so on : if they don't have an attributed set, they can benefit of a password or a smart card for controlled access to specific services

- **FLEXIBLE NUMBERING PLAN**
  - In hotel the "Set number" can be the same as the existing "Room number".
  Different set numbers may have simultaneously different lengths.

- **DIFFERENT OUTGOING TRUNK GROUPS FOR GUEST AND STAFF USE**
  The staff trunk group can overflow on guest trunk group.
  The guest trunk group cannot overflow on staff trunk group

- **GENERIC VOICE GUIDES**
  The generic voice guides are the 6 registered system voice guides containing 3 numbering plans (4x ; 7x ; *x).
  The 6 languages are : English (E), French (F), German (G), Italian (I), Portuguese (P), Spanish (SP).

- **STANDARD VOICE GUIDES**
  The standard voice guides are the locally registered system voice guides containing the local numbering plan.

MARKET	STANDARD VOICE GUIDES
Austria	• L1 : Austrian
Belgium	• L1 : Flemish, French, English
Brazil	• L1 : Brazilian • L2 : English • L3 : French • L4 : Spanish
Denmark	• L1 : Danish
Export 1	• L1 : Arabic male voice, Arabic female voice, Egyptian male voice, Egyptian female voice, Turkish
Finland	• L1 : Finnish
Greece	• L1 : Greek
Ireland	• L1 : Irish
Norway	• L1 : Norwegian
Spain	• L1 : Spanish • L2 : Catalan
Sweden	• L1 : Swedish business, Swedish Infocenter, Swedish Infocenter/VM
UK	• L1 : English

- **CUSTOMISED VOICE GUIDES**
  The customised voice guides are not defined. Voice guides are customised according to the customer's needs. They are registered locally by the DSU.

16.4 GUESTS & PATIENTS SERVICES

What can guests and patients expect from the set situated in their room? They may expect receiving and giving phone calls, not being disturbed or being waken-up, accessing directly to some services (laundry, room service, restaurant…), hearing voice guides in their native language, listening to voice messages … etc.

This chapter is devoted to the services delivered by the A4400 to guests and patients.

The first part presents a resume of all these services (including "dual line", "MCDU based password", and "privacy feature").
The second part treats the voice mail issue and Japanese.
The third one deals with multi-language.

The fourth one presents multiple wake-up & snooze.

16.4.1 Guests services : reminder

16.4.1.1 Direct access to telephony services

- DOD : DIRECT OUTPUT DIALLING
  Direct access to PTT: no need to go through the attendant to give external phone calls.

- DID : DIRECT INPUT DIALLING
  Direct access to a room: no need for an external caller to go through the attendant to call a room.

- CYCLING DID
  Automatic DID allocation: the oldest used DID number is automatically attributed. Thus a guest / patient would have more chance not be disturbed by an external phone call addressed to a previous guest / patient.
  Cycling DID can be managed:
  -By A4400 hotel terminal
  -Through AHL by an external application (CMS or PMS)

- ONE OR TWO DIGITS DIALLING FOR COMMONLY USED SERVICES
  Direct access to commonly used services such as room service, front desk, bar, restaurant, laundry … etc.

  Rm : A settable time delay is used if a digit is also used as the first digit for longer numbers. For example, room "2602" and service "2".

- DND : DO NOT DISTURB
  If a guest / patient wish not to be disturbed, he can dial a prefix or press a pre-programmed key to activate the DND function. Then no incoming call (both internal and external) will make the set ringing.
  For security reasons, both attendant calls and fire alarms can pass over it.

  Rm : A voice guide "your operation is recorded…" indicates that the operation has been successful, and the A4400 lights the DND lamp (or pictogram) on the UA set if any.

  Rm : DND is automatically cancelled at check-out if the guest did not cancel it by himself using the "cancel DND prefix" or the corresponding key if any.

  Rm : In case of Attendant routing of a call to a set in DND status, display of attendant indicates:
  - the call source (calling party)
  - the number of the trunk group
  - the ISDN calling number for external calls (if available)
  - an Icon meaning "DND"
  - the number of the set in DND (called party)

  Rm : When the set in DND mode off-hooks, a voice guide, in guest language, is heard indicating that the set is in DND but that the set can be used normally.

  Rm : The list of sets in DND mode can be printed or displayed on the hotel terminal.

  Rm : Wake up and message information are still sent to the set in DND status.

  Rm : If the set has a lamp associated to a DND key, the associated lamp is lighted when the set is in DND mode.

- WAKE-UP
  See § 4.

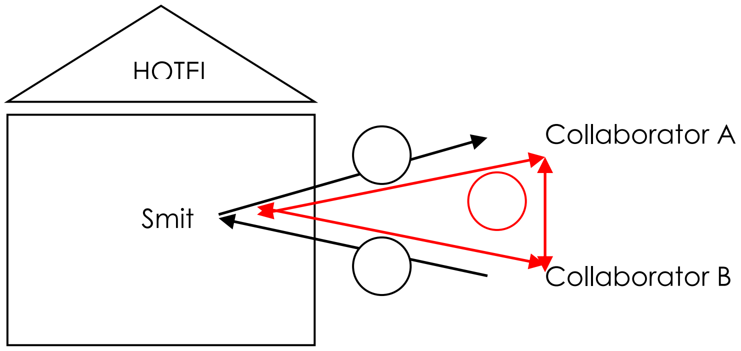
- VOICE MAIL
  See § 2.

- DUAL LINE & CONFERENCE
  The dual line feature or how to benefit of two lines by using one key.
  In case a guest receives a second call as he is in conversation, he can directly access to this second call by pressing a pre-programmed key and go back to the first caller by pressing the same key …etc.
  To do a conference with the two callers, a second key has to be programmed.

  Attention: accessible only on UA sets.

  Let's have a look at some scenarios (Mr Smith is a hotel guest):

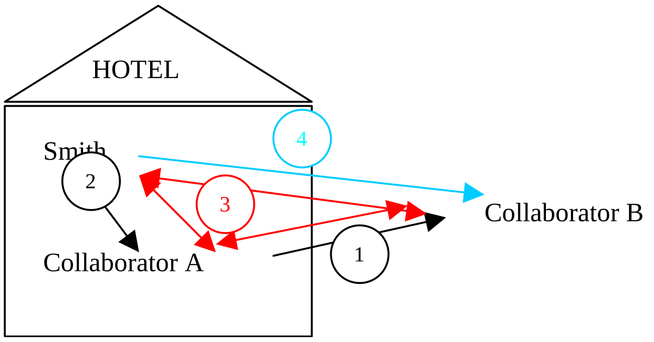
  Scenario 1



Smith can alternatively talk to his collaborators A and B by pressing one key, the dual line key (arrows 1 & 2).
Smith can also make a conference call with his two collaborators by pressing the conference programmed-key (arrow 3).

In that case, Smith's bill will indicate only his call to collaborator A.

Scenario 2



Smith calls collaborator A as this one is in conversation with collaborator B (arrows 1 & 2).
They do a conference call (arrow 3).
Collaborator A quits the conference. Thus Smith is still in conversation with collaborator B (means an external call) (arrow 4).

In that case

- Collaborator A's bill will indicate his call to collaborator B. The counter ends as collaborator A quits the conference.
- Smith's bill will indicate his call to collaborator B. The counter begins as collaborator A quits the conference.

16.4.1.2 Call barring functions

- AUTOMATIC CALL OF ATTENDANT AFTER TIME-OUT

  If the hotel manager / hospital director wants his guests / patients not to be disturbed by external phone calls within a daily fixed period of time, this feature allows to route directly external incoming calls to the attendant console.

- TRANSFER TO OTHER ATTENDANTS WITHOUT CONVERSATION

- INTER-ROOM BARRING FUNCTION

  This feature completes the previous one concerning inter-room calls: inter-room calls can be forbidden within a daily fixed period of time; then inter-room calls can be re-routed to the attendant console.

- SET LOCK / UNLOCK

  A guest / patient can decide to forbid the use of his set by dialling a prefix (or pressing a programmed function key).

  To unlock the set, the use of a password is necessary.

  Rm : the user is helped by voice guides.

- ROOM SET LOCK AFTER N ERRONEOUS PASSWORD ATTEMPTS

  The room set can be automatically locked with the use of that function. This is a security feature. "n" is manageable.

16.4.1.3 Booth sets calls

Any guest / patient is able to call from a booth set. Two options are available:

- Using personal password

The guest / patient enters his room number and his personal password (given at check-in). His bill is automatically updated.

- By attendant console

The guest / patient indicates to the attendant his room number and the number he wants to reach. The attendant has to update his bill.

The first solution is more secured: there is no way to cheat and indicate the attendant a wrong room number.

16.4.1.4 Room move

Auto allocation (room moving) from set.
A room move can be done without doing a checkout by using the "auto-allocation feature".
The number of the new room has to be indicated from a set; then every service is automatically transferred to the right room.

16.4.1.5 Message on free or busy set

16.4.1.5.1 Message drop

Three possibilities :

- From the message service or attendant console by :
  - dialling the set number or the GPIN
  - dialling the message code
  - pressing the "message drop" pre-programmed key
- from the external application through AHL
- from the Alcatel hotel terminal

16.4.1.5.2 Message information on set

The message lamp, if any, starts to flash after message drop

16.4.1.5.3 Message collection

Two solutions depending on hotel choice :

- Fully automated : On off-hooking, the guest receives a routing tone and is automatically connected to the message service or to the message originator (Voice Mail...) after a few seconds (programmable).
- Not fully automated : While this tone is heard, the set can be used normally and a number dialled.

Rm : the Alcatel hotel terminal allows the hotel to consult on the screen or on a hard copy the list of waiting messages.

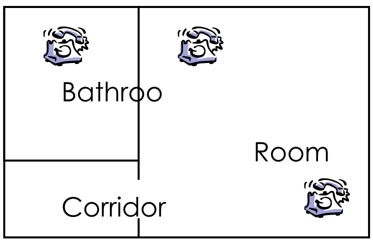
Rm : DND does not affect the reception of messages (the message lamp is activated)

16.4.1.6 Hotel suites

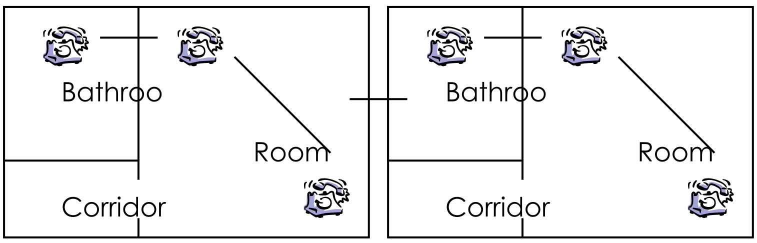
Attention: A suite is considered by the hospitality software of the A4400 as an association of sets. It is not necessarily the definition that can be given by a hotel manager: a suite is not necessarily constituted of different rooms.

Attention: Creating suites requires a guest-based configuration.

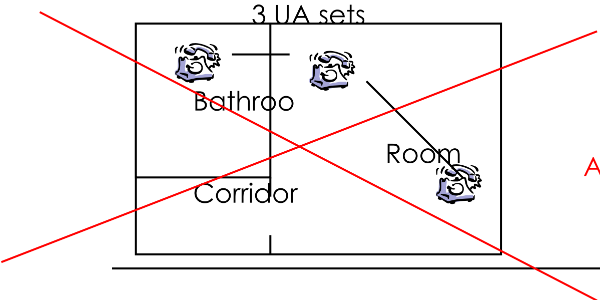
What may be a suite?



Suite 1
3 UA sets



Suite 2
2 rooms



IT IS NOT A SUITE

Analogue sets in parallel

16.4.1.6.1 *Suites sets*

A suite may is an association of sets that may be:

- S0 interface
- UA set
- DECT set
- Analogue set

16.4.1.6.2 *Suite management*

The constitution of a suite is not necessarily fixed: the hotel manager can modify the suites according to his day to day needs.

For instance, he may need a suite for a 5 persons family constituted of 5 rooms; and a week later, he may need a suite for 2 persons, plus 3 rooms. He may associate first 5 rooms to constitute a suite for the family. A week later, he may associate 2 of these 5 rooms to constitute a suite for the couple, and let the 3 other rooms on their own to keep 3 rooms for the other guests.

 Suite management is flexible.
There are two ways to manage suite constitution:

- From the A4400 hotel terminal
- Through AHL from an external application

16.4.1.6.3 *Making calls*

Attention: calls between suites' sets are always available (the inter-room barring function has no influence on that).

There are two possibilities:

- If prepayment is required, then only one call at a time per suite is available
- If no prepayment, then the guests can call from any set at any time

16.4.1.6.4 *Receiving calls*

An incoming call rings all the free sets of the suites.

If all the suite sets are busy, then the incoming call is waiting. The suites guests may hear then a bip, alarming them there is a waiting incoming call, they may use then the dual line function.

16.4.1.6.5 *DND: Do Not Disturb*

Programming a DND in a suite means that DND applies for all the suite sets.

Attention: it does not concern S0 interface.

16.4.1.6.6 *Wake-up*

Programming a wake-up in a suite means that the wake-up will ring all the suite sets.

Attention: individual room wake-up is still available.

For more information about wake-up see § 4.

16.4.1.6.7 *Dual line & Conference*

Conference calls can either be internal or external:

- between rooms of the suite
- external room / sets / trunks

Dual line presents the same functionality in a suite as in a room.

16.4.1.6.8 *Voice mail*

- Voice mail attribution
  It remains on one guest.

- Message led activation
  The message led is activated in case of a new message presence. This is true on all the suite sets.

- Voice mail function
  See § 2.

16.4.2 Other guests services

16.4.2.1 Fire alarm : emergency call on room set (R3)

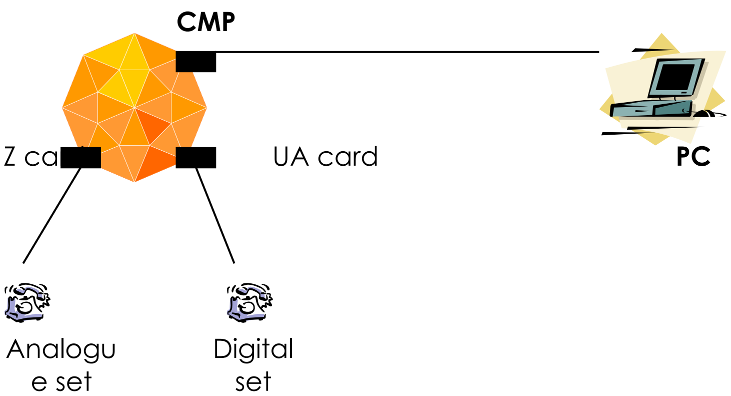
Characteristics

The fire alarm is emergency voice guide diffusion send via the A4400 to determined sets (which can be administrative or guest sets).

The generated alarm consists in:

- Calling these sets (even if they are busy)
- Diffusing a voice guide in the guest language
- Asking the called person for an acknowledgement

Installation



A PC is linked to the A4400 through a CMP card. It is used to create the list of persons to alarm. Its database is renewed by the A4400 thanks to the check-in & out information. The A4400 at each affectation (check-in) and de-affectation (checkout) generates an incident. Thus no alarm is generated in non-occupied rooms.

The incident message contains:
- The room MCDU
- The operation type (change of the parameter value)
- Concerned object type (subscriber dynamic data)
- Action type (check-in or checkout)
- Guest language (information send for the first checked in guest or the last checked out guest)

The CMP card is considered as a PRA card linked to the A4400 by an ABC-F link. Thus alarm calls are network calls.

Fire alarm calls

The fire alarm call concerns guest sets, suites sets, administrative sets, but no OP sets. The fire alarm call directly calls the set itself, thus it passes through DND, call forward, filtering, pre-off-hook voice guides.

Set ringing

The set ringing for fire alarm can be defined as a specific ringing:
- Manageable cadence (like the priority ringing)
- Maximum volume

On free sets: set ringing (all sets ringing in case of a suite occupied by a single person).

On busy or called set:
- Violent set liberation (no warning)
- Then classical fire alarm call

On busy or called set(s) in a suite:
- Violent set(s) liberation (no warning)
- Classical call of the free set(s)
- Then classical fire alarm call on liberate set(s)

  (the first called set is the so defined "main" one; it is in charge of liberating the "slave" sets of the suite)

In multi-occupied suites: only physical sets are called.

Beam from the CMP card to A4400

Three choices:
- T0/T2
- ABC-F
- Urgent diffusion beam

Restrictions

Please take care of these three restrictions:

- Analogue sets: do not call more than 4 analogue sets at the same time because of the cadence.
- In a mono-occupied suite, all the sets are called (like a normal call).
- As soon as a suite set is hooked off, the other suite sets stop ringing and are able to receive or give phone calls.

16.4.2.2 Guest Personal Code

The guest personal code is the private code of a single guest or patient.

It is given to the guest/patient at check-in or pre-check-in time.

- At "check-in" or at set "allocation" a "personal code" can be given to guest. These **1 to 4 digits code** (generally 4) can be chosen by the guest to help him to remember it.

- The association "Room/set number" and "Personal Code" is unique and is used for protected access.
  Maids, staff people, agents, etc…, may also used "personal code" for protected access to some Alcatel 4400 features.
  The guest set will be locked after “n” (manageable) password entry errors.

- If the computer need also a "personal code" it is convenient to choose the same one on the Alcatel 4400.

- When a badge is used, the data read on the badge correspond generally to the "user number" plus the "personal code".

- If a problem occurs (stolen badge...) the personal code can be changed.

The GPC uses are:

- In room based configuration:
  - booth call
  - internal voice mail access (option)
  - external voice mail access (mandatory)
  - room call (option)
  - DND
- In guest based configuration:
  - booth call
  - room call
  - internal voice mail access
  - external voice mail access
  - set use
  - DND

The GPC is determined by the A4400 and given through:

- The hotel terminal
- An external application

MCDU based password (R2.1 & R3.0 support & R3.1): the guest personal code can be based on the guest MCDU.
This allows the guest / patient to remember more easily his personal code.

16.4.2.3 Privacy

The privacy feature has been developed to protect the privacy of guests / patients in their room.

As you must know, some information is displayed on the set: his name and phone number identifies the caller. In the hospitality industry it can be disturbing for guests who want to be "anonymous" towards the other guests.

That is the reason why an option has been designed: the hotel / hospital (IT) manager can decide to protect the privacy of its guests / patients, thus not to let any guest / patient information displayed.

Attention: if the privacy option is chosen, then administrative sets will still display guests / patients information (name, called service, language …and so on.). There is no information loss for the hotel management, but a privacy enhancement.

16.4.3 Voice mail 4635 (R 1.3.5 & R2.2 of the 4635)

In the healthcare industry, most of the persons who need a voice mailbox belong to either the auxiliary nursing or administrative staff. In that case they need a 4635 voice mailbox. Please refer to the 4635 Product description.

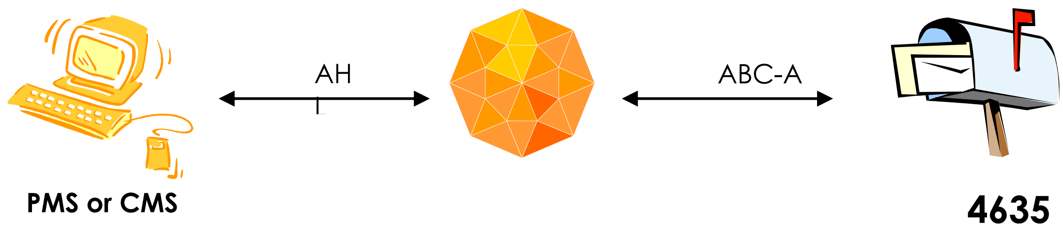
In the hotel industry, most of the persons who need a voice mailbox are hotel guests. In that case they need a 4635 hotel voice mailbox. Please find hereby a description reminder of the main 4635 hotel voice mailbox *(1.1. 4635 Hotel features: reminder*), and a description of the new hotel voice mailbox *(1.2. A new hotel voice mail box type).*

16.4.3.1 4635 Hotel features : reminder

From pre-check-in to checkout, the voicemail is taken into account. Hereby is a small reminder of the main functions linked to the 4635 (J & H).

16.4.3.1.1 Integrated check-in / checkout support

At check-in/check-out time, mailbox is created/deleted automatically on A4635. A4400 interprets PMS or CMS commands carried over AHL link, and translates them into ABC-A commands. A4635 performs create/delete actions according to ABC-A commands.



Note: These features are actually available according to the Hotel Application itself.

16.4.3.1.2 Pre-check-in support

The mailbox creation does not need a room allocation. In guest based configuration, pre-check-in allows to enter the guest into the database without allocating him a room.
Thus, a mailbox can be created at pre-check-in, and A4635 is ready to record message, customise greetings (greeting message or name), and set personal password (which is the same as the guest given password at check-in time).

16.4.3.1.3 Room move support

A guest can move to another room and keep his mailbox automatically (no change to be done neither by the hotel staff nor by the guest).
When guest changes his extension number, the mailbox number can change accordingly.
This avoids deleting the previous mailbox and creating a new one.

16.4.3.1.4 Deferred checkout = deferred mailbox deletion

Guest can be offered the possibility to review his own messages after checkout time.
This feature is active only when mailbox contains new messages at checkout time.
Additional lifetime of a mailbox is a system parameter.
In any case, at next check-in on this mailbox, previous mailbox will be deleted and a new one will be created.

16.4.3.1.5 Wake-up calls

- Wake-Up Programming

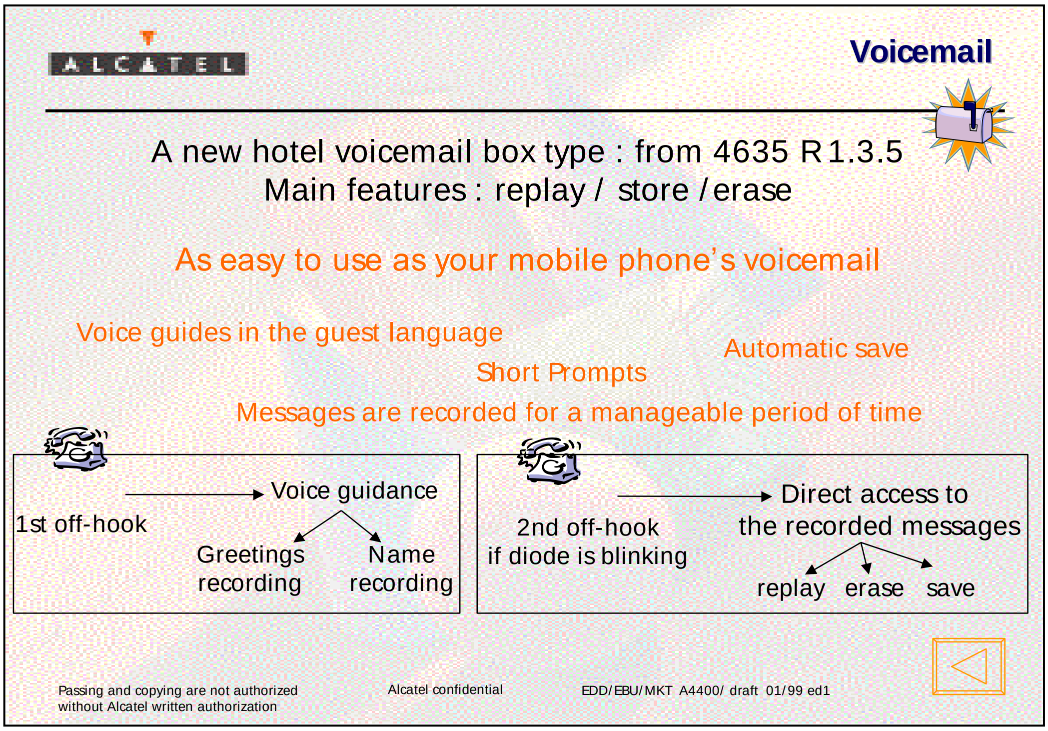
- The guest dials the wake-up programming prefix (can be pre-programmed on a UA set function key) from his room set.
- Or a hotel employee does it from the room service set, the attendant console, the hotel console.
- The guest is prompted by the PBX for wake-up time and destination phone number.

• Wake-up Call
  - A4400 establishes a connection to A4635
  - The guest is connected to A4635, which plays until the guest hangs up : <Time> < This is the time of your alarm call>
  - If the guest does not answer to the first wake-up call, the system produces two more wake-up calls.
  - In case of both no answer at all and engaged number, A4400 generates an alarm on a dedicated printer (that avoids the hotel having problems with guests who could complain against the hotel of not having been waken-up).

• Wake-up call's prompts language
  - Language is indicated by the A4400 to the A4635. It corresponds to the guest language.
  - It concerns both programming voice guidance and wake-up voice alarm.

• A4635 port usage
  - On wake-up programming, one A4635 port is used per session.
  - At wake-up time, all the simultaneous wake-up calls with the same language are connected unidirectional to one A4635 port.

16.4.3.2 A new hotel voice mailbox type



4635 R1.3.5 offers a new & enriched hotel voice mail box type. It is no more the mechanism of an answering machine with which the guest could only listen one time to his messages. It is closer to the services given by a mobile phone: user-friendly and customisable. Let's see how it works.

Main added features:

- replaying a message
- storing a message
- and erasing a message.

To make it guest user-friendly and secured, some services are offered:

- voice guides in the guest language (known by the PABX at guest check-in)
- short prompts (so that the guest won't be board or lost)
- automatic save of the listened messages (more secure : even if the guest hang up the phone before the end of a message, this message will be automatically stored so that the guest would be able to re-listen his message)
- messages are recorded for a manageable period of time (manageable time of the message length, manageable time of the message storage time)

1st off-hook: **customisation** →

Voice guidance in guest language

Customisation proposals

Greetings recording | Name recording

2nd off-hook: **access to messages**

→ In case of recorded messages:

- New messages : the diode is blinking
- Stored messages : the diode is lightened

Direct access to messages

Message envelope in guest language

Message

3 proposals

Replay the | Store the | Erase the

Next message envelope in guest language

...

16.4.3.3 Japanese

Japanese are travelling a lot in the countries Alcatel is acting, and hotels want to serve them in a better way. One enhancement has been done for that reason: one of the multiple languages available for voice mail prompts is now Japanese.

Attention: it is true only for voice mail prompts (voice mail prompts including wake-up prompts).

16.4.4 Multilanguage

16.4.4.1 Multi-language expansion (R3.0 & R3.1)

The more hotels have international customers, the more they need different languages to give their guests benefit of understandable mother tongue based voice prompts.

For that reason, the number of system languages has risen from 4 to 8.

16.4.4.2 Hotel Multi-language procedures (R3.1)

Rm : please refer to the glossary to get a definition of "generic voice guides", "standard voice guides", and "customised voice guides".

For the R3.1, a list of hotel voice prompts has been defined (selected upon all the system existing voice prompts). From now on, hotel generic voice guides exist. There are six hotel generic languages : English, French, German, Italian, Portuguese, Spanish.

Basically, it will be possible to register two hotel voice languages on one flashcard (if the defined hotel voice prompts list is respected). Thus to reduce the cost of multi-language !

Let's have a look at some virtual cases :

- The hotel needs 3 languages.
  - L1 : system voice guides on one flashcard (including hotel voice prompts)
  - L2 & L3 : hotel voice guides on one flashcard (only hotel voice prompts)

  □ 2 flashcards instead of 3

- The hotel needs 5 languages.
  - L1 : system voice guides on one flashcard (including hotel voice prompts)
  - L2 & L3 : hotel voice guides on one flashcard (only hotel voice prompts)
  - L4 & L5 : hotel voice guides on one flashcard (only hotel voice prompts)

  □ 3 flashcards instead of 5

- The hotel needs 7 languages.
  - L1 : system voice guides on one flashcard (including hotel voice prompts)
  - L2 & L3 : hotel voice guides on one flashcard (only hotel voice prompts)
  - L4 & L5 : hotel voice guides on one flashcard (only hotel voice prompts)
  - L6 & L7 : hotel voice guides on one flashcard (only hotel voice prompts)

  □ 4 flashcards instead of 7

16.4.4.2.1 System voice guides list

16.4.4.2.2 Defined hotel voice prompts

The following list of voice prompts is a selection of the voice guides that are useful for guests or patients to access A4400 guests services. This is the composition of the Voice Guides Generic Hotel.

Warning : this is a shorter list than the system voice guides list. These prompts have just been selected for hotel needs.

Index	Message	Comment
3	Your request has been recorded, you may hang-up / *Votre manoeuvre est enregistrée, vous pouvez raccrocher*	
4	Please dial your personal code / *Veuillez composer votre code personnel*	
5	The number dialed is not authorized, please make inquiries / *Cette manoeuvre n'est pas autorisée, veuillez vous renseigner*	
6	Please dial the destination set number / *Composez le numéro du poste destinataire*	
7	Your calls are forwarded, you can still make a call. To cancel forwarding, dial 64 <Forward cancellation Pfx> / *Votre poste est renvoyé, pour annuler ce renvoi composez le 64 <Pfx d'annulation de renvoi>*	
7	Same content as above, alternate numbering plan number 1	
7	Same content as above, alternate numbering plan number 2	
8	You can request to be called back by dialing 5 <Booking On Free Or Busy Set Sfx.> / *Vous pouvez demander le rappel automatique en composant le 5 <Sfx. de Rappel sur non reponse/occup.>*	
15	Please enter the alarm time / *Veuillez composer l'heure à laquelle vous souhaitez être rappelé*	
18	Your extension is in Do Not Disturb mode, you can still make a call. To cancel, dial 78 <DND Pfx> / Vous avez demandé à ne pas être dérangé. Pour annuler, composez le 78 <Pfx. Ne Pas Déranger>	
18	Same content as above, alternate numbering plan number 1	
18	Same content as above, alternate numbering plan number 2	
65	This is the moment you want to be rang / *Il est l'heure à laquelle vous avez souhaité être rappelé*	
102	You have voice messages, to consult your mailbox dial 77 <Voice Mail Consultation Pfx> / *Vous avez des messages vocaux. Pour consulter votre messagerie vocale, composez le 77 <Pfx Consultation mesagerie .vocale>*	
102	Same content as above, alternate numbering plan number 1	
102	Same content as above, alternate numbering plan number 2	
103	Do not hang up, we are paging the called party / *Nous recherchons votre correspondant, merci de patienter quelques*	

	instants	
107	You can cancel the automatic call back by dialing 45 <Canc.auto.call back on busy Pfx> / *Vous pouvez annuler le rappel automatique en composant le 45 <Pfx Annul.rappel poste occupe>*	
107	Same content as above, alternate numbering plan number 1	
107	Same content as above, alternate numbering plan number 2	
108	Your prepayment is used up, would you be so kind as to refill your account / *Vous avez consommé votre prépaiement ; pour pouvoir téléphoner, veuillez recharger votre compte.*	
110	Would you be so kind as to wait, the operator will go to answer you / *Veuillez patienter, l'opératrice va vous répondre*	
139	You can overflow onto the called party's voice mail by dialing 8 <Voice Mail Message Deposit Sfx> / *Vous pouvez demander à être dirigé vers la messagerie vocale de votre correspondant en composant le 8 <Sfx Depot de messages vocaux>*	
252	This extension is padlocked ; to cancel, dial 80 <Padlock Pfx> / *Votre poste est verrouillé, pour annuler, composer le 80 <Pfx cadenas>*	
252	Same content as above, alternate numbering plan number 1	
252	Same content as above, alternate numbering plan number 2	
253	You have call-back request, you can still make a call. To cancel, dial 67 <Consult Call back list Pfx> / *Vous êtes en rappel automatique, vous pouvez effectuer une communication. pour annuler composer le 67 <Pfx de Consult.personnes a rappeler>*	
253	Same content as above, alternate numbering plan number 1	
253	Same content as above, alternate numbering plan number 2	
539	Incorrect code, please reenter your personal code. / *Numéro incorrect, veuillez recomposer votre code personnel.*	**Secured DISA access** : from R3 Multilanguage

16.4.4.2.3 How to order a mono-language hotel

- **GENERIC GUIDES**

L1 choice : one choice upon 6 (E, F, G, I, P, SP)

▯ are generated :
- 1 card (VG or GPA)
- 1 flash card "VG generic L1"

- **STANDARD AND CUSTOMISED GUIDES**

L1 choice : one upon n

▯ are generated :
- 1 card (VG or GPA)
- 1 virgin flash card

It is possible to order the recording to the flashcard to the manufacturer. Therefor, the following procedure has to be respected.

1. check the list of voice guides to record :
   - create a WORD document in copying the list of guides (see § "1.4.4.2.1. system voice guides list")
   - give the name of your country to the WORD document (STDVG**country**.doc)
   - replace the name <prefixe > by your prefixe (ex: <45>) in the English language
   - add under each guide the translation of the guides in your language.

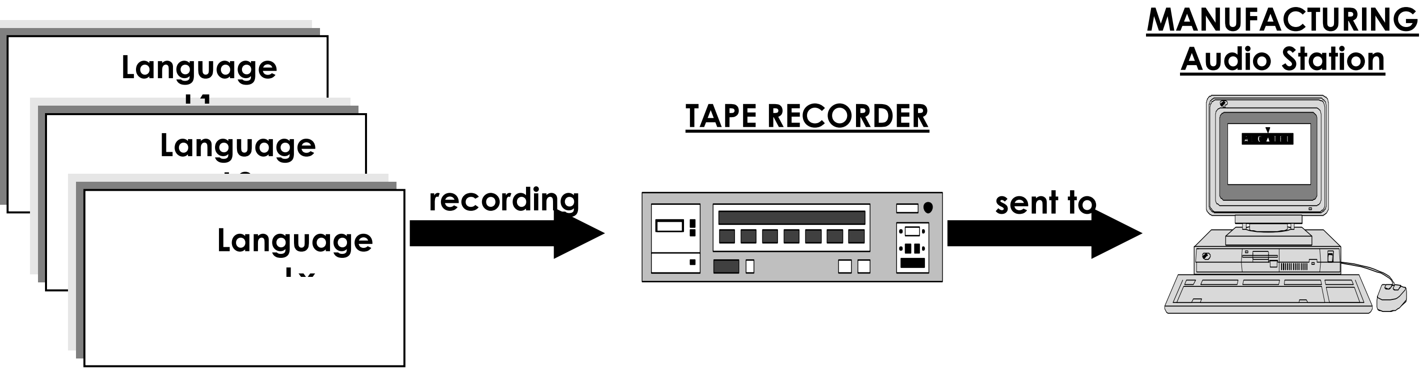
2. 2-2-Recording on :
   - a tape (see next page "Recording on tape")
   - a CD-ROM (see next page "Recording on CD-ROM")

3. to re-send to your Program Manager:
   - The tape + the list completed (WORD document STDYYLxcountry.doc)
   - Or the CD-ROM + the list completed (WORD document STDYYLxcountry.doc)
   - In case of Multilanguage, do not forget to define the 1st language (L1), 2nd Language (L2)…. **In your documents / tapes /CD-ROM**

Explanation ▯ STDYYLx**country**.doc.
YY => letters of the language recorded (example : FR,EN,SP,…)
Lx => priorities of the languages (L1, L2,….)
Country => name of your country (France, Spain, ….)

Example ▯ If ALCATEL France need 3 languages they must create :
A WORD document called STDFRL1**FRANCE**.doc (in French)
A WORD document called STDSPL2**FRANCE**.doc (in Spanish)
A WORD document called STDENL3**FRANCE**.doc (in English)

RECORDING on TAPE



Criteria for the recording

- The recording must be made at level –20db
  (The range must be between –30db and –10db)
- Every message must be preceded by its guide number (***in English***) (**EXAMPLE 1)**
- A silence of 5 seconds must be observed between the guide number and the message
- The audio tape must be accompanied by its documentation (word document on floppy-disk or paper), with the corresponding guide number opposite each text . (**EXAMPLE 2)**
- In the case of **multi-language**, it is necessary to send **one flash card per language.**

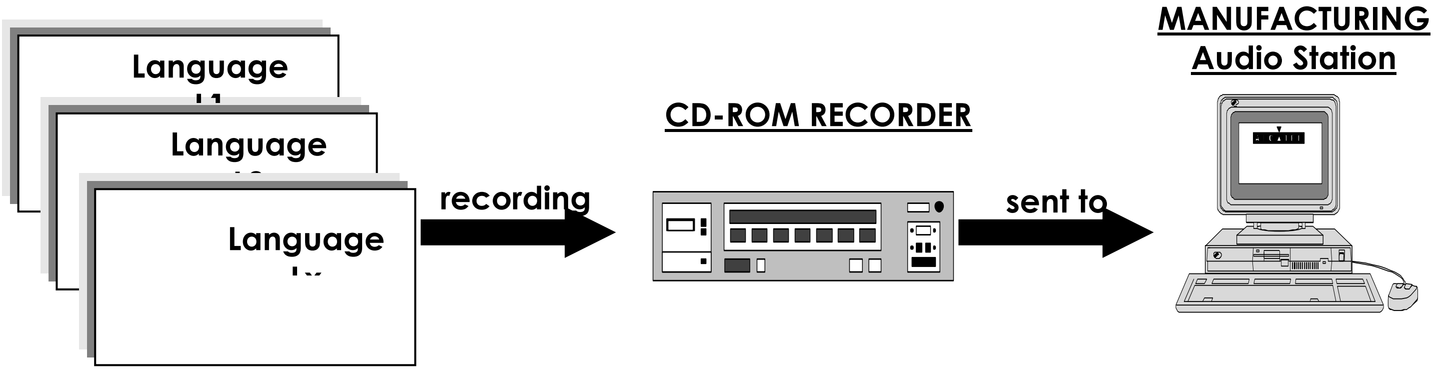
EXAMPLE 1 : **English language to record**

File	**Silence**	**Content**	**Silence**	
VG0003	5 seconds	**Your request has been recorded, you may now hang-up**	5 seconds…	Next message
VG0004	5 seconds	**Please dial your personal code**	5 seconds…	Next message
VG0005	………	…………………………………………………………………………	……………	……………….

EXAMPLE 2 (Word document on Floppy-disk or paper) : **English language to record**

File	**Content**
VG0003	Your request has been recorded, you may now hang-up
VG0004	Please dial your personal code
VG0005	The number dialed is not authorized, please make inquiries

RECORDING on CD-ROM

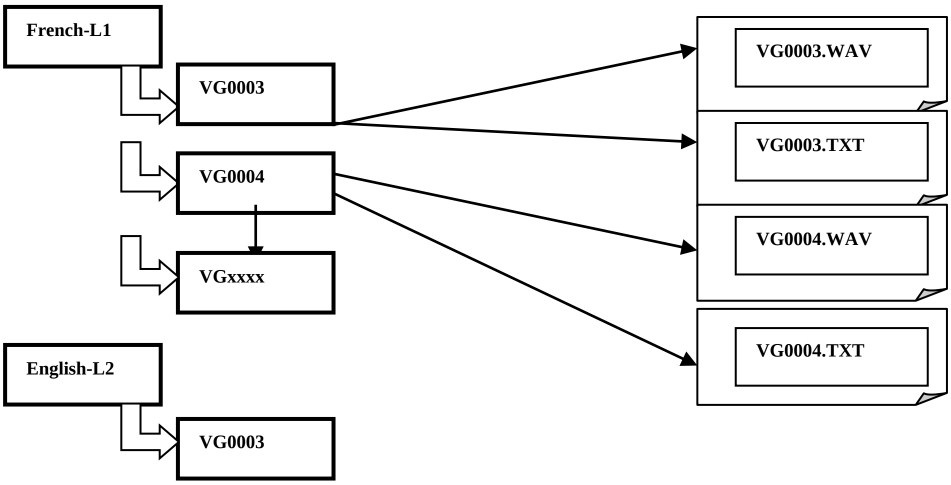


Criteria for the recording

Recording format for the .WAV files => 16 bits PCM non compressed
Sample frequency => 8 KHz
Recording level => -2db (absolute)

Structure

Structure to build on your CD-ROM

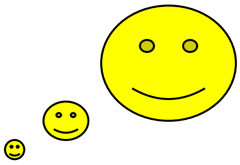


- **directory :** give the name of the " language + Lx " to indicate the row of the Language.
- **Sub-directories :** give the name of the " voice guide".
- **file.WAV :** the file name is the "voice guide number" (to record it see criteria required)
- **file.TXT:** the file name is the "voice guide number" . This file content the written message of the file.WAV recorded.

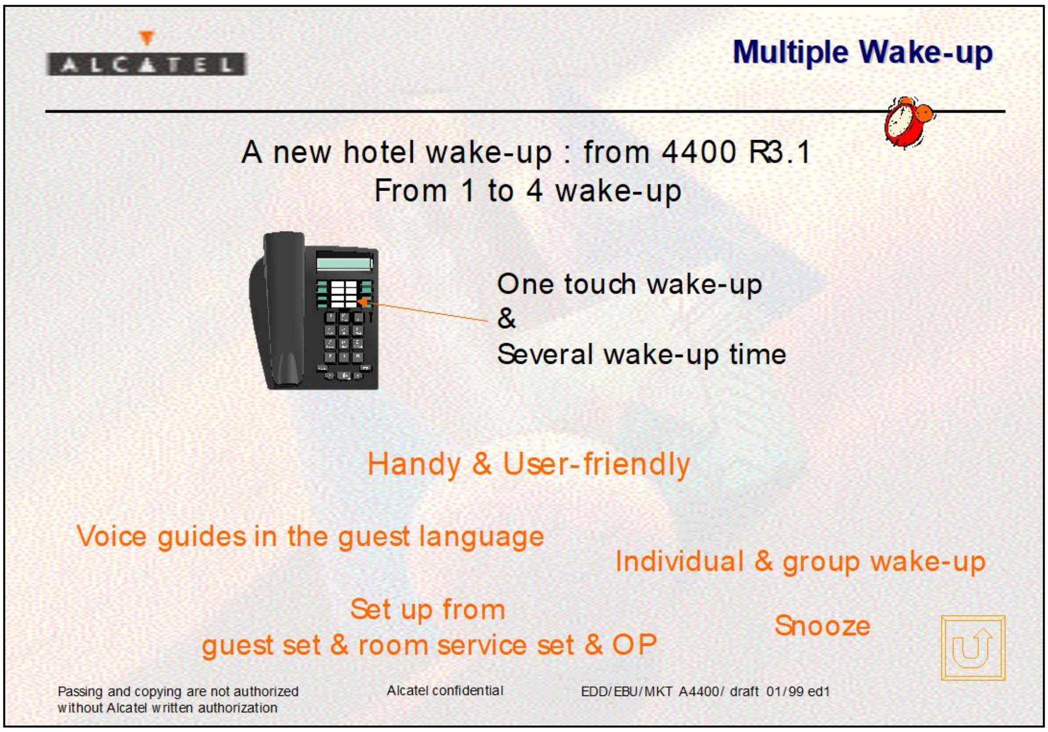
16.4.4.2.4 How to order up to 3 languages for a multi-language hotel

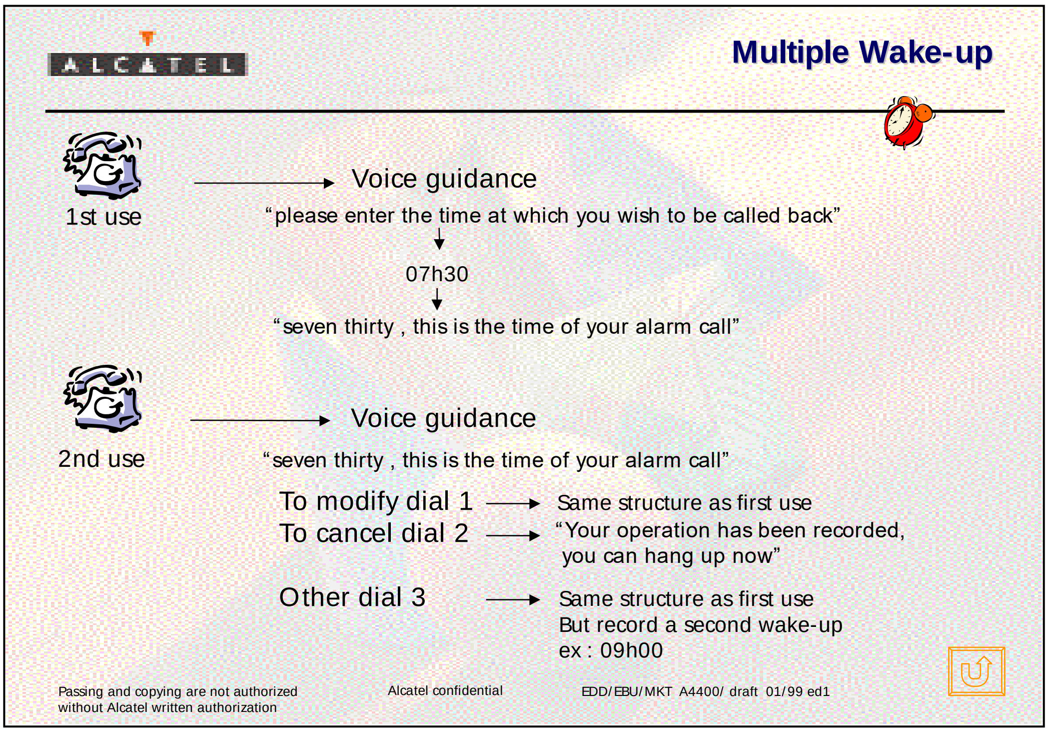
16.4.4.2.5 How to order more than 3 languages for a multi-language hotel

16.4.5 Wake-up (R3.1)



16.4.5.1 Multiple wake-up & Snooze





16.4.5.2 Case study

Let's imagine a wife and her husband sharing a hotel room and needing two different wake-up calls, let's say 7AM for the man and 9AM for the woman. It would be easier either for them or for the hotel staff to program the two wake-up calls at the same time, rather than to program the 7AM one at one time (the day before for instance), and the 9AM one after having received the first one.

The reason is simple: if the husband has to program a wake-up call for his wife as he gets up at 7AM, it's virtually certain that he would forget it. The reason is approximately the same one for the hotel staff.

Everybody would benefit of a multiple wake-up program function.

Now let's imagine you as a hotel guest. You want to have a rest, but not to sleep a long time. It would be reassuring to know you simply have to press any set key to be alarmed few minutes later by a phone call.

This is the beauty of the snooze feature.

16.4.5.3 Description

In order to guide the guest / hotel employee while he/she is programming one wake-up call among 4, a voice guidance is needed (or 5 softkeys on the phone set).

Attention: Hence if the installation has no voice guide or the set is not an MR2, the multiple wake-up call feature is not available on this set.

- Voice guides

Here voice guides means a voice-mail able to provide the multiple wake-up service: A4635 R3 and upwards. Plus the correct system voice guides.

System guides:

- N°512: "To modify this time, press 1. To cancel it, press 2. For next programming, press 3."
- N°513: "To modify this time, press 1. To cancel it, press 2. Else hang up."

- Programming modes

The guest can program and cancel a wake-up call either by dialling the prefix number or by pressing the key dedicated to the wake-up feature (programmed during the installation).

- Number of wake-up calls

Some figures:
- The actual figure is manageable between 1 and 4 for a given installation.
- The default value is 1.
- The absolute maximum number of wake-up calls is 4.

Attention: if the maximum number of wake-up is fixed at 1, there is no change in the customer interface compared to the previous single wake-up feature.

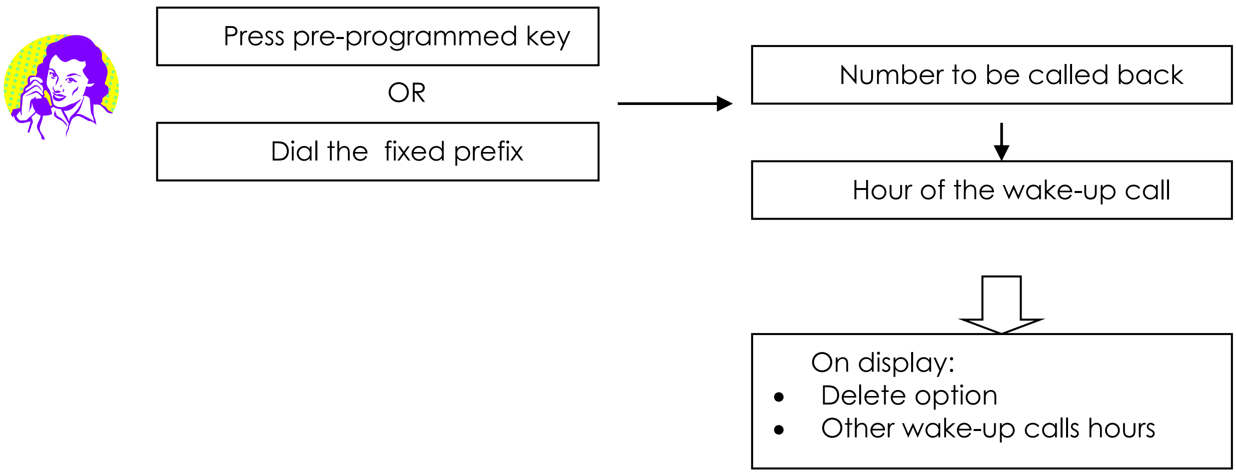
16.4.5.4 Wake-up programming

- If the set has softkeys (UA set as 4035)

Attention: a set with softkeys can be a room service set or a guestroom set. To be a room service set, it has to be able to program a wake-up call for an other extension number. That means a room service set with softkeys belongs to a Phone Facilities Category where the Remote wake-up/appointment option is set to 1.

How to program a wake-up call:

- From a room service set with softkeys
- From a guestroom set with softkeys



- If the set has no softkey

The following behaviour is available for any type of phone like analogue, UA sets without softkeys …and so on. Make sure that the correct voice guides are available on the system, and that there is a A4635 voice mail with R3 software.

Programming the first wake-up

Press pre-programmed key

OR

Dial the fixed prefix

→

"please dial the time required"

Dial the wake-up time (ex: 0730)

"your request has been recorded, you may hang up"

Programming another wake-up

Press pre-programmed key

OR

Dial the fixed prefix

→

"seven hours thirty minutes, this is the moment you want to be rang"

"to modify, press 1"

"to cancel it, press 2"

"for next programming, press 3"

Dial 3

"please dial the time required"

Dial the wake-up time (ex: 0800)

"your request has been recorded, you may hang up"

Modifying a wake-up time

Press pre-programmed key

OR

Dial the fixed prefix

→

"seven hours thirty minutes, this is the moment you want to be rang"

"to modify, press 1"

"to cancel it, press 2"

"for next programming, press 3"

Dial 1

"please dial the time required"

Dial the wake-up time (ex:

"your request has been recorded, you may hang up"

Checking the wake-up times

Press pre-programmed key

OR

Dial the fixed prefix

→

"seven hours forty five minutes, this is the moment you want to be rang"

"to modify, press 1"

"to cancel it, press 2"

"for next programming, press 3"

Dial 3

"eight o'clock, this is the moment you want to be rang"

"to modify, press 1"

"to cancel it, press 2"

"for next programming, press 3"

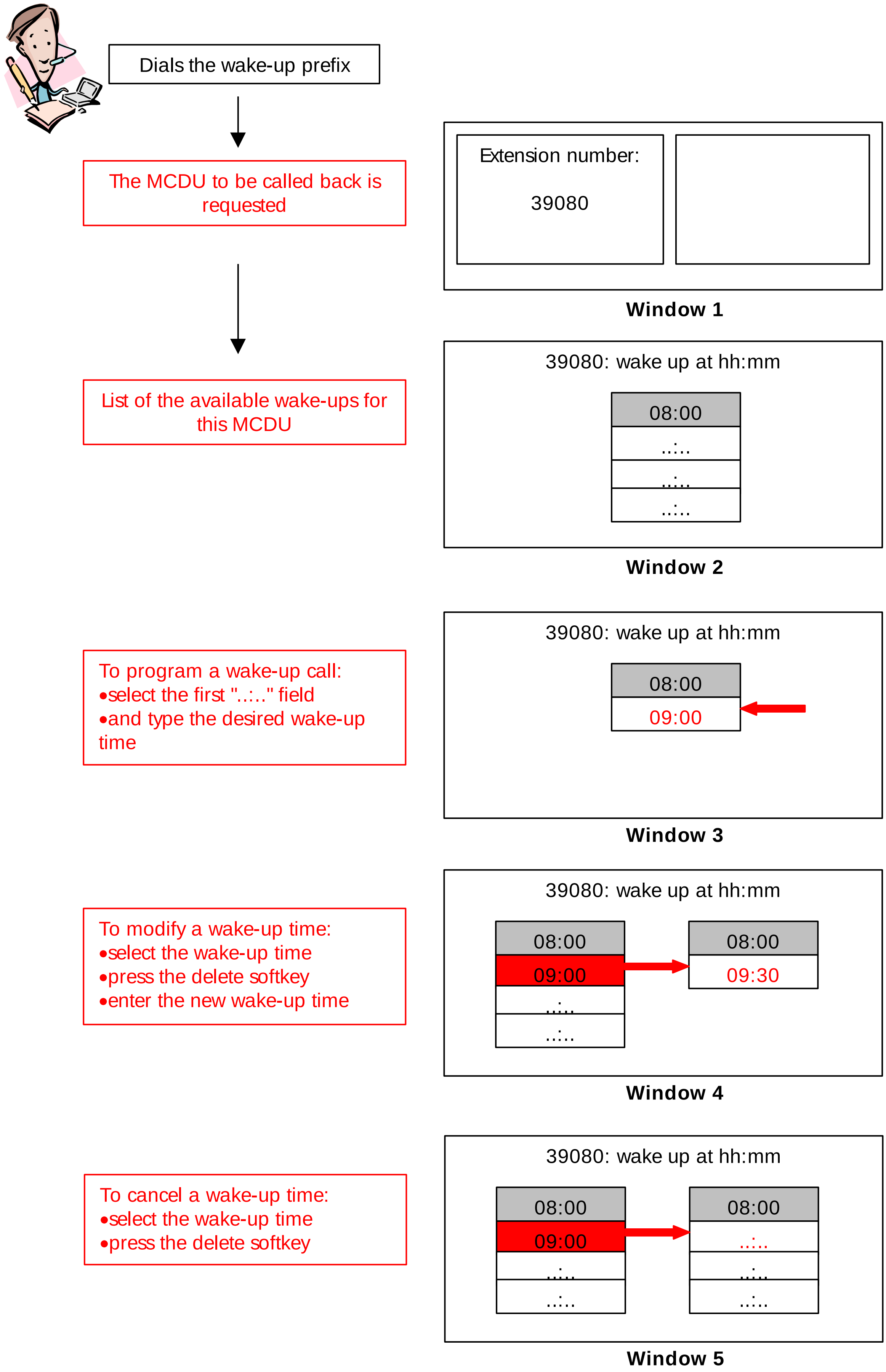
Hang up

Cancelling a wake-up call

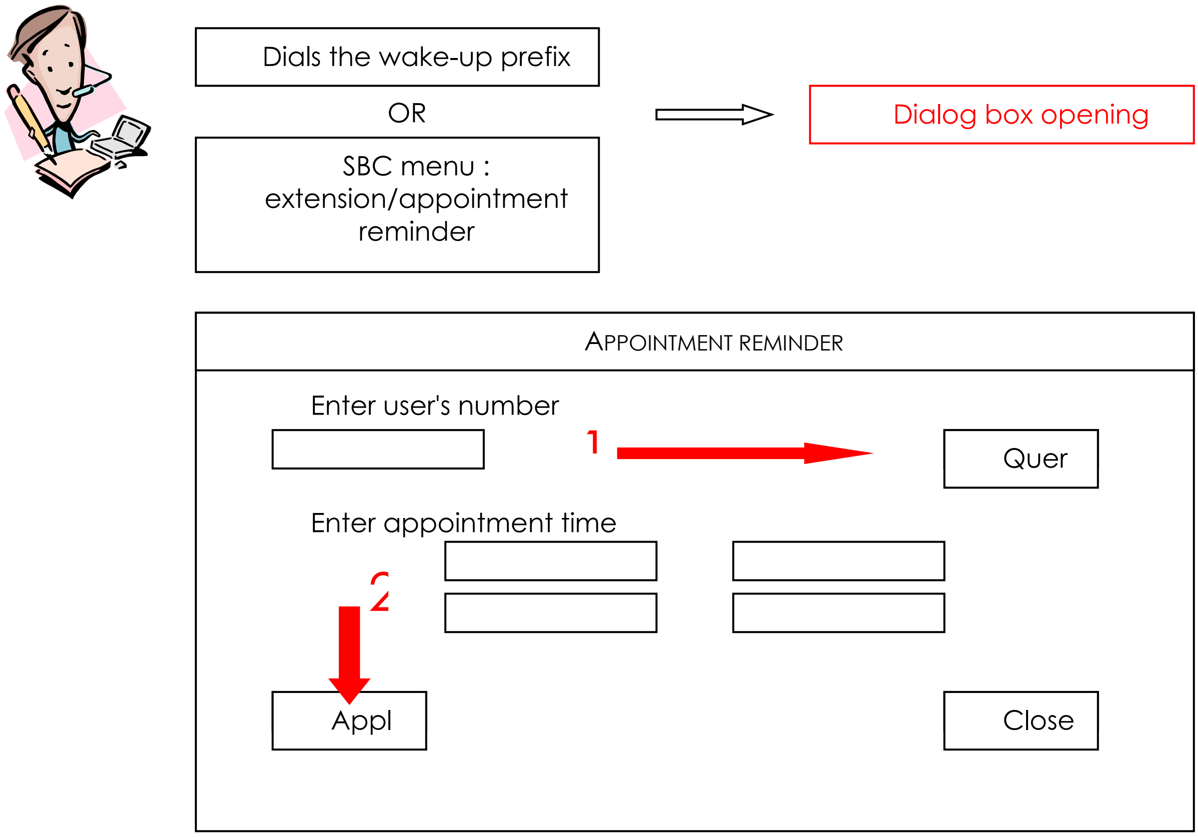
Press pre-programmed key		→	"seven hours forty five minutes, this is the moment you want to be rang"
OR			"to modify, press 1"
Dial the fixed prefix			"to cancel it, press 2"
			"for next programming, press 3"
			Dial 2
			"your request has been recorded, you may hang up"
			Hang up

- From the OP console

From a FBC (Flat Based Console)


Dials the wake-up prefix
The MCDU to be called back is requested
Extension number:
39080
Window 1
List of the available wake-ups for this MCDU
39080: wake up at hh:mm
08:00
..:..
..:..
..:..
Window 2
To program a wake-up call:
•select the first "..:.." field
•and type the desired wake-up time
39080: wake up at hh:mm
08:00
09:00
Window 3
To modify a wake-up time:
•select the wake-up time
•press the delete softkey
•enter the new wake-up time
39080: wake up at hh:mm
08:00
09:00
..:..
..:..
08:00
09:30
Window 4
To cancel a wake-up time:
•select the wake-up time
•press the delete softkey
39080: wake up at hh:mm
08:00
09:00
..:..
..:..
08:00
..:..
..:..
..:..
Window 5

From a SBC (Screen Based Console)



16.4.5.5 Suites

16.4.5.6 Snooze

- Description
- Management
- Interactions

16.4.5.7 Hotel terminal

16.4.5.8 AHL

16.4.5.9 Security: system protection against failure

16.4.6 Wake-up in network

16.5 STAFF SERVICES

16.5.1 ROOM SERVICE

- Here below we extend "Room Service" term to any "hotel service" which can be reached by guests.

16.5.1.1 General features

(**a**) Unlimited number of room service

- Room service can be reached by dialling any numbers of digit **(from 1 to 8).**

(**b**) Multidispatching room service

- A call for one specific room service can be dispatched on several room service sets depending, for example, of the floor in hotel, or the service in hospital.

For example :

* if a guest in floor seven dials "2" for "Bar" room service, he gets the "Bar room service" of floor 8
* a guest of floor two, that dials the same number, gets the "Bar room service" of floor 3.

(**c**) Multiservice room service

- Very often hotels want to give the impression that they have more Room Services that they can afford.
So, calls for different room services must be sent to the same room service set with indication of the room service requested.

- This feature is necessary also when room service people are momentary busy and must forward their room service set to an other one.

For example : if a guest dials "2" for "Bar", he gets Room service "2" and hears "Good morning Mister X this is the Bar...
If later he dials "3" for "Restaurant" he gets the same Room Service "2" but hears "this is the Restaurant..."

(**d**) External call baring

- Depending of their own category, Room Service Station may not have the possibility to call outside the Hotel / Hospital.

16.5.1.2 Set type

(**a**) General

- The set **has to be any multiline set** with display :

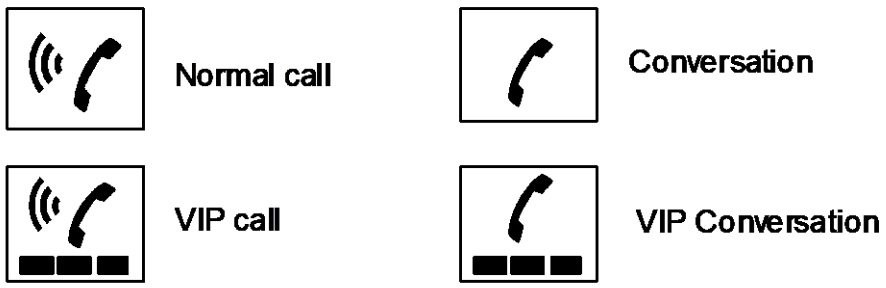
(**b**) V.I.P. Calls

- VIP calls are differentiated from normal call to allow to take care first of VIP calls.

- The VIP attribute is given by the MAO PC for a room and by the check-in for the guest.
  The "V" displayed will be in order to guest 's data or room's data

*	Room service calls a guest	guest VIP
*	Room service calls a room	room VIP
*	Room calls a Room-service	room VIP.

- When a guest call arrives on a UA set, the icon is as follows :



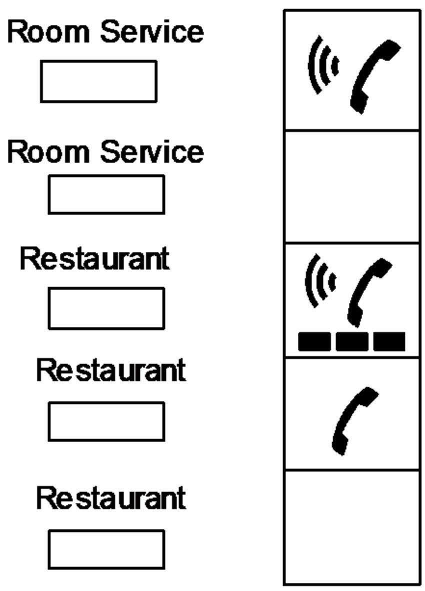
- The calls are selected by the UA user in the wanted order (in principle VIP first).

(**c**) <u>Called service identification</u>

- The Room Service set is a multiline set. So, it can be called through several numbers.

- For each of these numbers several lines can be given.

- To each line a key is attributed.



(**d**) Guest language

- When guest calls the Room service or when the Room service calls the room, the language is displayed:

* if room empty : language of the room
* if room with one guest : language of the guest
* if room with some guest : language of the last guest

(**e**) Room multi-occupation

- The Hotel terminal manages multi-occupation.

- The "*" is displayed if :
* in guest based configuration : if more than one guest are in the room
* in room based configuration : if at the check-in, the room is in multioccupation mode

(**f**) Display fields

÷ line number 3 characters
÷ first name - last name 16 characters
÷ guest number (extension / room) 8 characters
÷ VIP code 1 character ("V" for VIP)
÷ multi-occupation code 1 character ("*")
÷ language 2 characters corresponding
to the translation of the 1digit language code

(**g**) UA sets displays

Room service set display in call return

* A Room service set calls a ROOM NUMBER

Room empty

Room name	Room numberV*FR	

Room with one guest

Guest name	Room numberV*FR	

Room with some guests

Room name	Room number	V*FR

Example : The Room service set calls the room number 5911 with one guest

A4034 & A4035 sets display

LAMBERT Anne 5911 free	V*FR

A4023 set display

LAMBERT Anne V*FR

A4010, A4011, A4020, A4021 sets display

LAMBERT . V*FR

* A Room service set calls a GUEST NUMBER

Room empty, with one guest, with some guests

Guest name	Guest number	V*FR

Room service set display in conversation

* A Room service set calls a ROOM NUMBER
* A Room service set calls a GUEST NUMBER
* A Room calls a Room service

Room empty

Room name	Room number	V*FR

Room with one guest

Guest name	Room number	V*FR

Room with some guests

Room name	Room number	V*FR

Example : The Room service set calls the room number 5911 with one guest

A4034 & A4035 sets display

LAMBERT Anne 5911	V*FR

A4034 & A4035 sets display in Enquiry mode

LAMBERT Anne 5911	V*FR//BOUSQUET Alain

A4023 set display

LAMBERT Anne 5911

A4010, A4011, A4012, A4020, A4021 sets display

LAMBERT. 5911

Room service set display in ringing

* <u>A Room calls the Room service set</u>

<u>Example</u> : A Room set 5927, with one guest, calls the Room service

A4034 & A4035 sets display

BOUSQUET Alain 5927	calls	V*FR

A4023 set display

BOUSQUET Alain

A4010, A4011, A4012, A4020, A4021 sets display

BOUSQUET A. V*FR

(**h**) <u>Attendant console display</u>

Attendant console display in call return and conversation

* <u>An Attendant console calls a ROOM NUMBER</u>

Room empty

Room name	Room number	V*FR	(
)			

Room with one guest

Guest name	Room number	V*FR
( )		

Room with some guests

Room name	Room number	V*FR	(
)			

<u>Example</u> : The Attendant console calls the room number 5911 with one guest

A4048 display

Free LAMBERT Anne 5911	V*FR
(2)	

* An Attendant console calls a GUEST NUMBER

Room empty, with one guest, with some guests

Guest name	Guest number	V*FR	( )

The guest number is displayed for the metering of the guest's calls

Attendant console display in ringing

* A Room calls the Attendant console

Room empty

Room name	Room number	V*FR	( )

Room with one guest

Guest name	Room number	V*FR	( )

Room with some guests

Room name	Room number	V*FR	( )

16.5.2 ROOM STATUS

- For each room there is 3 independent parameters.

16.5.2.1 Room Type: On 2 characters.

16.5.2.2 Occupancy: On 1 character. (Vacant or occupied)

- This is automatically managed by guest "check-in", "check-out", and "allocation" commands
  "1" indicates "occupied room"
  "0" indicates "vacant room"

16.5.2.3 Status: On 1 character.

(**a**) Parameter value :

- Some values for this parameter are reserved and have the following meaning :
* "0" room being cleaned
* "1" room cleaned
* "2" room to clean
* "3" sheets to change

- The other values are chosen and used by the hotelier in order to request room inspection, indicate the type of repair needed, etc...

(**b**) Automatic change over

At defined time
At 3 A.M. all "occupied" and "cleaned" rooms are automatically changed to "to be cleaned again" status (status "3" : sheets to change).
The hotel management can program another time for this change over.

At check-out
The room is move to "to clean" status (status "2")

(**c**) Manual change over

- Can be done from room or from attendant

Without identification

- From the room, the maid (or maintenance service) dials :

* the "change room status" prefix
* the new status code

For example, "22 1" means that the room where the maid is has just been cleaned and that there are no problem.

A voice guide in system language confirms that the operation has been successful.

With identification

- This is mandatory if every room is equiped with badge set, and each maid (or maintenance service) has her own badge.
If there is no badge set, a "Code" is given to each maid (as well as maintenance service personnel).

- This code has between **0** (no code) **and 4 characters** that are set by management on installation or later.

- In this case the maid (or maintenance service) dials :
on arrival in the room (if required by hotel management)

* the "room status change" prefix
  ÷ her own code
  ÷ the "0" status code, indicating arrival in the room

* on leaving the room
  ÷ the "room status change" prefix
  ÷ her own "room/set number"
  ÷ the new status code

for example :

"22 38 0" means arrival in room by maid "38"
"22 38 1" means that the room has been cleaned by maid "38" and that there are no problems.

A voice guide in system language confirms that the operation has been successful.

d) Printing

- All status changes can be printed on a designated printer in real time, with indication of room number and time/date.

16.5.2.4 Hotel terminal change over

- The house keeper or any authorized staff member can modify, from an Alcatel 4400 hotel terminal or Minitel, the room status or the type.
  The access is protected by "personal password" control.

- The house keeper can ask for research (by type or by status) and gets hard copy on the associated printer.

16.5.2.5 Print out

- Any research and corresponding print out is possible on following criteria (crossed research)

* status by room type
* room by status
* free or occupied rooms
* list of room with set number and number of bed per room

- A general summary table can be displayed or printed. It show for each room : "Number", "type", "occupation", "status".

16.6 OP CONVERSATION

16.7 DECT

16.8 NOTIFICATION SERVER

16.9 BUSINESS GENERATING SERVICES

16.9.1 Call accounting

16.9.1.1 Reminder on charging

- For staff sets the Alcatel 4400 business charging applies. The facilities described below (apart from the Printing Staff Station Tickets section) only concern room, booth, shop and meeting room sets put together in the term "guest sets".

- A4400 system offers two charging mode possibilities:
* Accounting with pulses
* Accounting on duration: for countries without tax pulses.

16.9.1.2 Accounting with pulses

- A4400 system works with accounting on pulses mode including three rates.

- The price is calculated according to the following formula : **P = N1P1 + N2P2 + N3P3**

- Bearing in mind :
* price P1 of the first N1 charge units
* price P2 of the following N2 charge units
* price P3 of the remaining charge units (N3)

Prices P1, P2 and P3 have 3 digits and a floating point.

16.9.1.3 Accounting on duration

a) Preliminary

- Due to deregulation the environment of Hotels is changing :
* some operator do not provide the advice of charge indication ;
* some hotels are interested by using several operator (to reduce the telecom cost). In this case the hotel owner will use the services of ARS.

Hotel owners are requesting flexible time based metering.

- In this release, the hotel manager can define in A4400 how he wants to charge the outgoing calls. He can define the cost per direction , the cost per call duration (e.g. tariff change after a given duration) and even a fixed cost. The hotel manager is free to introduce the tariff he wants ; it can be different from the tariff of the operator.

With these parameters A4400 will calculate the cost of the calls. With this new service :
* the integrated hotel package can be used even in countries where the operator does not provide advice of charge indication
* the accounting tickets produced by 4400 on the AHL will contain the call duration and the cost; that means that the external hotel application is not obliged to do the job
* the hotel owner can decide how he will charge the calls in case of several operators without advice of charge

b) Detailed description

- There is an integrated time based metering service that can be used in two different cases :
* in hotels were the national operator does not provide advice of charges
* in hotels connected to several operators (some of them with advice of charge ,some without)

the time based metering

- The cost calculation is based on **unique** table in 4400 where the hotel manager indicates :
* the cost per direction ; up to 1000 area can be defined
* the tariff for a given time of day period
* the tariff depending on call duration
* if he wants to apply a fixed cost.

The cost calculation is applied on all call placed on a bundle declared for that purpose

Hotels in countries having PSTN without advice of charge

- In this case , for each call placed on PSTN , A4400 will calculate a cost based on the parameters that the hotel owner has introduces in its table. The bundle used to access PSTN must be declared « with time based metering ».

At check out it will be possible to charge the guest and if necessary to show him all the calls he placed on PSTN.

In case of external hotel application , A4400 will deliver on AHL , tickets with:

* call duration and cost
* the number dialled by the guest
* the trunk group number

Hotels connected to several operators

- More and more , to reduce cost of PSTN calls, hotels are connected to several operators ; for example one operator for national calls , another for international calls . Generally , new operators will not send advice of charge.

Due to that the hotel will be in following configuration :

* several operators are used by the hotel
* the ARS function of 4400 will be used to select automatically the cheapest operator
* some operator will not provide advice of charge
* the digits sent to the operator might be different from the digit sent by the guest to place his call

- In this configuration , the hotel manager has two possibilities to configure (it is as system option) the A4400 time based metering function:

* either he decides that the cost will be calculated with the digits sent by the user (before ARS transformation). The Pros and Cons of this way are:
  - ÷ pro : the guest will see at check out the calls with the digits he sent
  - ÷ pro : the hotel manager will win money (different tariff between operators)
  - ÷ con : the guest cannot have cheap call by accessing a given operator (e.g. by dialling its access code)
* either he decides that the cost will be calculated with the digits sent on the line (after ARS transformation). The Pros and cons of this way are :
  - ÷ pro : it is possible to apply the same tariff as the selected operator
  - ÷ con : the digit that are in the final ticket are those sent on the line and not those sent by the guest

- At check out it will be possible to charge the guest and if necessary to show him all the calls he placed on PSTN.

In case of external hotel application , 4400 will deliver on AHL , tickets with:

* call duration and cost
* either the number dialled by the guest or the number sent to the line, depending on the hotel manager choice
* the trunk group number

limits

- Following limits have to be taken into account :

* the time based metering does not work in case of operators connected behind PSTN with an indirect access (in DTMF for example)
* the deposit will not work in case of time based metering
* the calls placed by the administration staff of the hotel will use the same time based metering
* in case of multiple operators only following configurations are possible:
  - ÷ 1 operator with AOC + 1 operator without
  - ÷ no limit when several operators without AOC are selected via different trunk groups

<u>Counters</u>

- For each room or user there is :

* <u>A "total cost" counter</u> which counts the cost of calls made by the guest. **It has 8 digits and a floating point**. (*1)

* <u>A communication counter</u> which counts the number of calls for which a ticket is issued. **It has 4 digits**. (*1)

∗ A "daily cost" counter (*1, 2)

∗ A "total deposit" counter which totalises any pre-payments at the check-in. **It has 8 digits and a floating point**.

∗ A "remaining deposit" counter. **It has 8 digits, floating point and sign** (see deposit section).

∗ A daily "total deposit" counter. (*2, 3)

∗ A "total deposit" counter for "global deposit" (*3)

Reset

(*1) reseted automatically at the guest check-out.
(*2) reseted after night-audit.
(*3) common reset of the deposit counters or by reset of all accounting counters.

Furthermore, the booth set counters can be reseted by the attendant console with automatic printout of the corresponding information on the real time printer (see "Booth set").

- Counter printout on guest check-out

∗ On guest check-out or on request, the price (see § 8-2-21-1 "fixing") is automatically calculated and printed with no indication of the number of charge units.

16.9.1.4 Billing

- For each successful outgoing call, a ticket is issued :
- Ticket is available with 2 currencies: EURO + local

Content

- This contains the data described in "information sent to the external computer" (minimum ticket).

Storage

- The tickets are stored on disk (on request).

- On check-in by the following guest, the data stored on the allocated sector are deleted and any additional sectors returned to the "common pool".

Printing guest tickets

- The following information can be printed in real time or on request on **a 80 characters line** for each ticket. It shows :
  ÷ set number
  ÷ cost centre (service)
  ÷ how obtained
  ÷ time and date
  ÷ duration
  ÷ cost calculated from the number of charge units
  ÷ project number
  ÷ number dialled

Global or detailed guest ticket

Detailed ticket for specific period of time

Printing staff set tickets

- Using a system management command, the staff set tickets can be printed using the same format.

- However, the cost is calculated by multiplying the number of charge units by a "Staff set basic charge unit" price.

Dialled number mask

- For screen-outs, printouts or external transmissions, it is possible to mask **the last 1 to 4 digits** of the number dialled (programmed on installation). Masked digits are replaced by periods.

- For numbers with fewer than 5 digits, the first two digits are kept.

Specific on line printing

- On request, it is possible to print tickets

 ÷ for one or for a serial of rooms or users
 ÷ for specific number dialled or specific direction (for example, 3615)
 ÷ for cost overflowing a threshold
 ÷ for duration overflowing a threshold
 ÷ for specific obtaining mode (transfer...)

Transfer of communication (outgoing call) to a guest

- If a call is made by an attendant console or an hotel staff and transferred to a guest, the guest is charged for the totality of the call and only one ticket is issued with a specific obtaining mode.

Research

- Possible on the same criteria as line printing.

Transfer of ticket

- From a hotel terminal it is possible to transfer a ticket from one extension to an other (call of taxi from a room service...)

Ticket preallocation

- From booth (booth or bar...), it is possible to specify to which extension the next call made from that booth must be attached.

Outgoing call SMDR

- The Set Message Dial Recording is an event stored or sent at the end of each outgoing call
The SMDR are issued for any "monitored device" (set, user or group).

- The duration are in minute (**3 characters**) an seconds (**2 characters**) (99999 means overflow).

- The cost is **7 characters plus 1 floating point**, and the number of basic taxes (pulses) is **8 characters without point** ; 99999999 means overflow.

- The last numbers (1 to 4) of the external called device can be replaced by periods for confidentiality.

16.9.1.5 Itemised bills

- The hotel management can request a guest's itemised bill. This can be obtained until check-in of the next guest occupying the same room.
  When this bill is printed, there is no reset or modification. Itemised bills may be requested several times for the same guest at different times.

- Thanks to the chaining system used for storage, there is virtually no limit as for the number of tickets for one guest.

The bill has the following format :

Hotel name (PBX name) and message   Date and time

Arrival date and time

Room no. : "XXXX"   Name : "Guest name"

```
RM     OBTND    D  M  Y  H  MN   DURATN     COST          NO. DIALLED
XXXX   XXXXXX   XX/XX/XX  XX/XX   X XXXMXXS   XXXXXXXX   XXXXXXXXXXXXXXXXXXXX

XXXX  XXXXXX    XX/XX/XX  XX/XX   X XXXMXXS   XXXXXXXX   XXXXXXXXXXXXXXXXXXXX
```

Number of calls : XXXX

Total cost* : $ XXX XXX.XX

Deposit : $ XXX XXX.XX

Debit ** *** : $ XXX XXX.XX

Personalised hotel message (40 characters m)

---

* See "Guest check-out"
** or credit
*** does not appear if pre-payment not used

-The column headers are printed at the top of each new page where appropriate.

16.9.1.6 Printing bill with VAT (Value Added Tax)

- The VAT can be managed. If it is different from 0 the VAT cost appears on the guest bill.

16.9.1.7 Output

- Sorts and limits printing are possible.

16.9.1.8 Fixed Fare for telephone use

- A fixed fare for telephone use can be introduced by MAO management.

When check-in, if a fixed fare is defined at installation level, it will be immediately deducted from the guest account (this feature is mainly used with deposit).

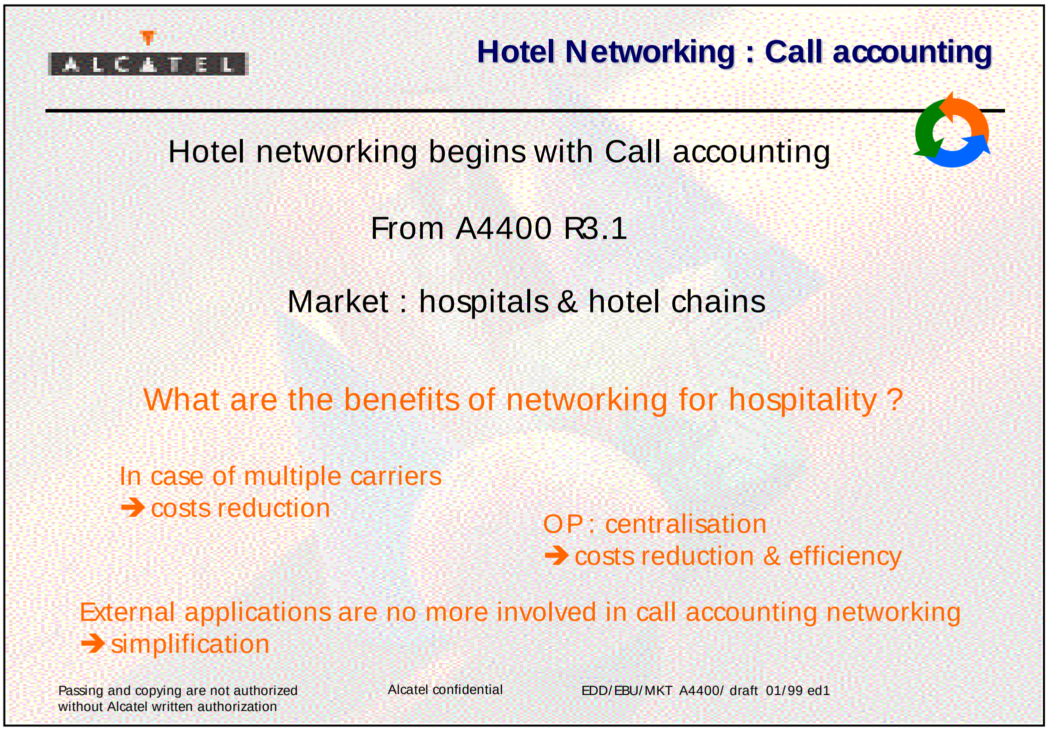
If the deposit is negative or nil the guest cannot make an external call.

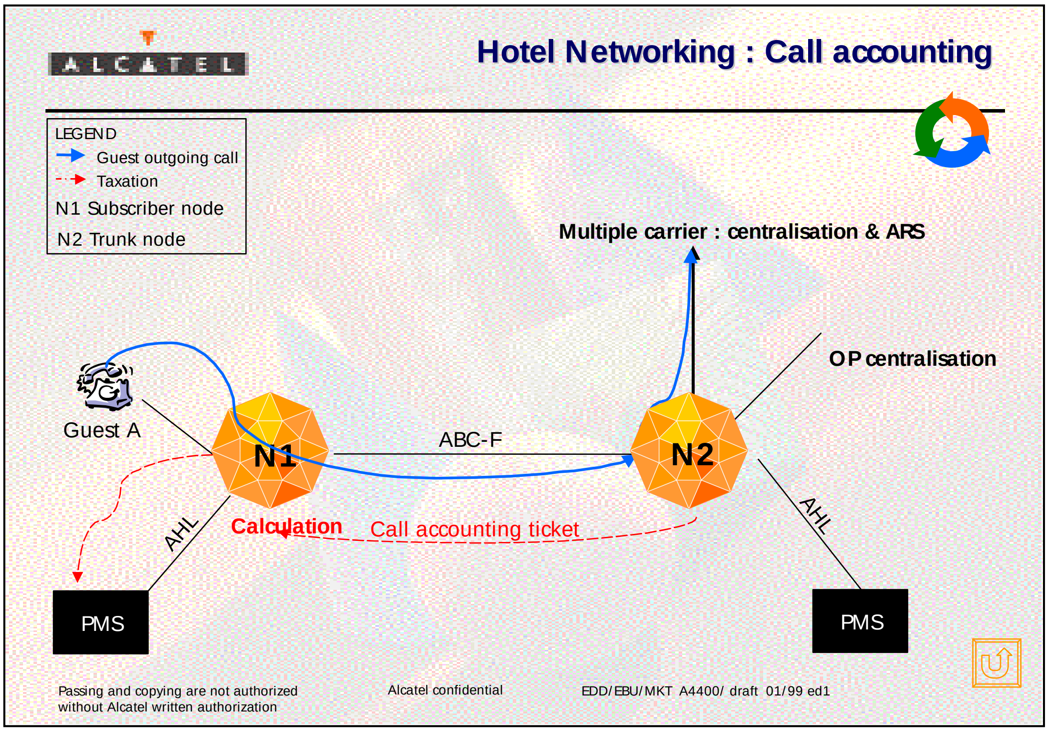
- The fixed fare is deducted, once only, during all the stay of guest.

Fixed fare is defined for:

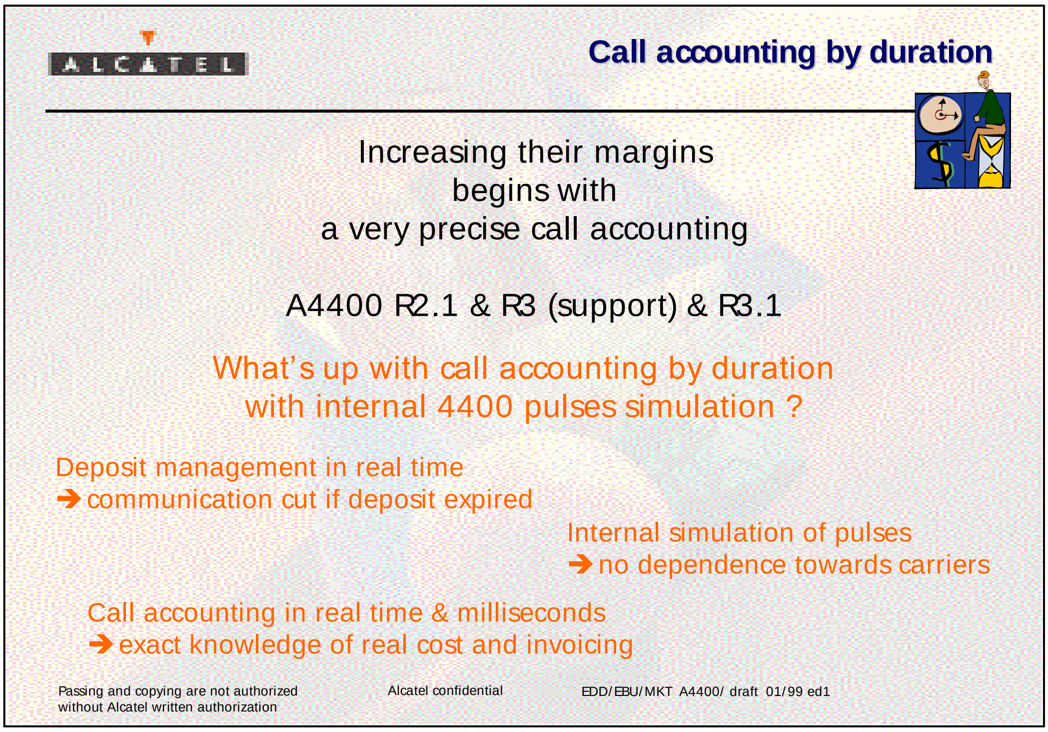
* line allocation
* DDI number allocation
* Voice mailbox allocation
* DECT and S0 sets

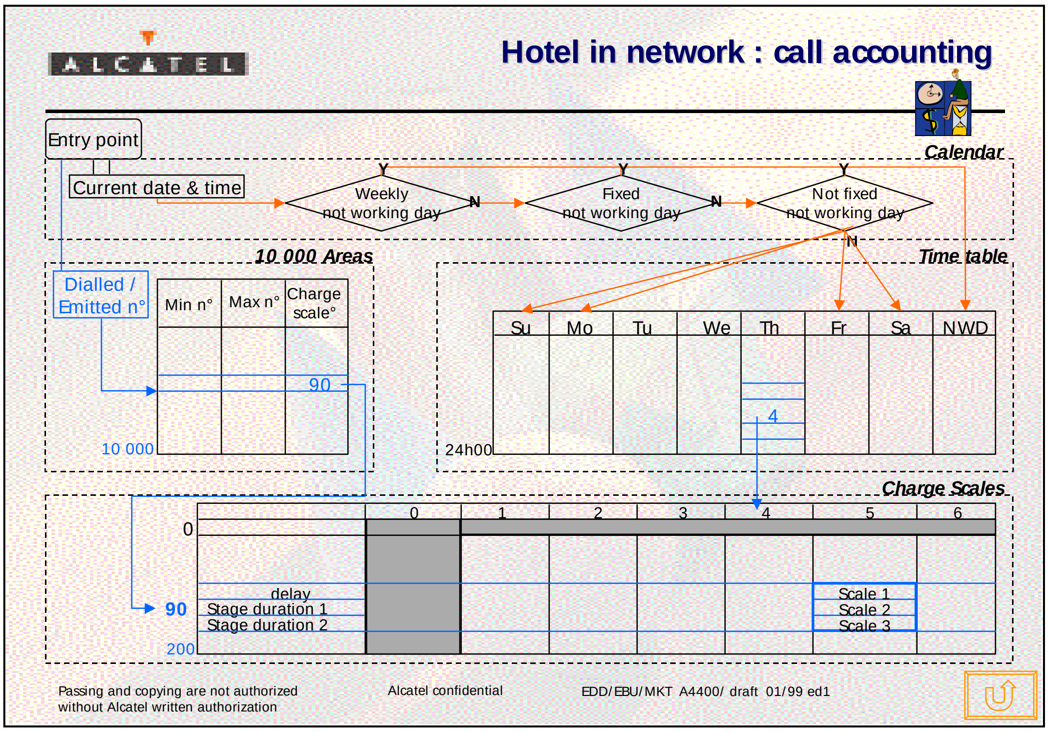
16.9.1.9 Call accounting in network





16.9.1.10 Call accounting by duration with pulse simulation





16.9.1.11 Call accounting parameters expansion

16.9.1.12 Call accounting extended tickets

16.9.1.13 Buffer size expansion

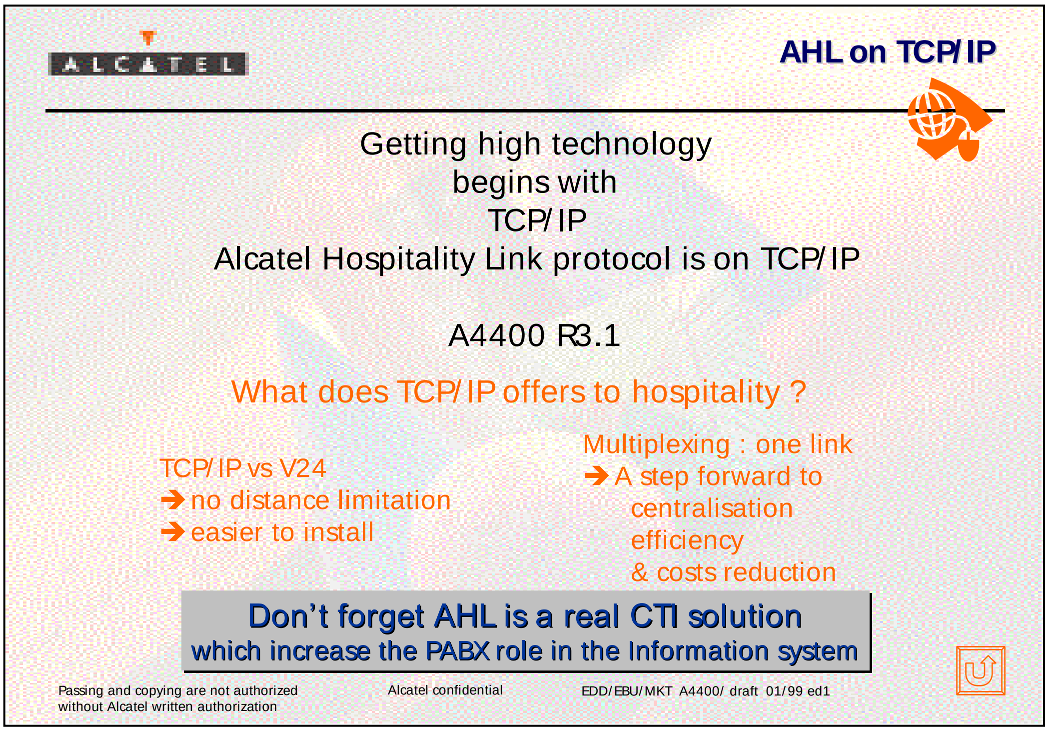
16.9.2 Billing

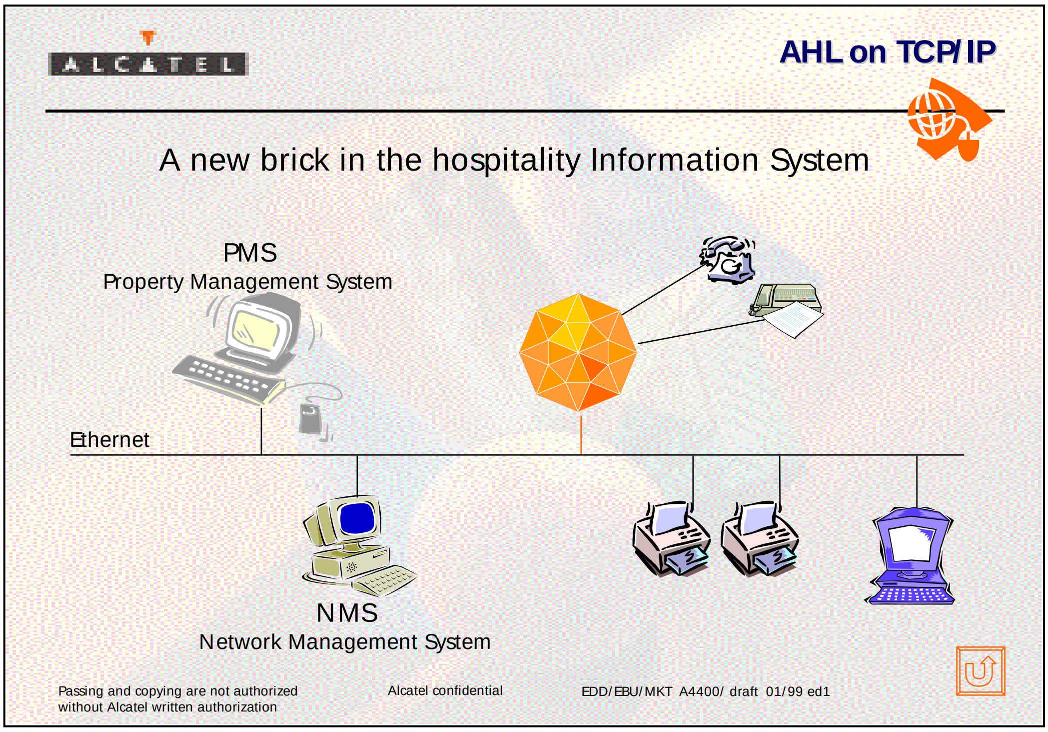
Billing in network
Itemised billing for a given period of time
Double currency billing

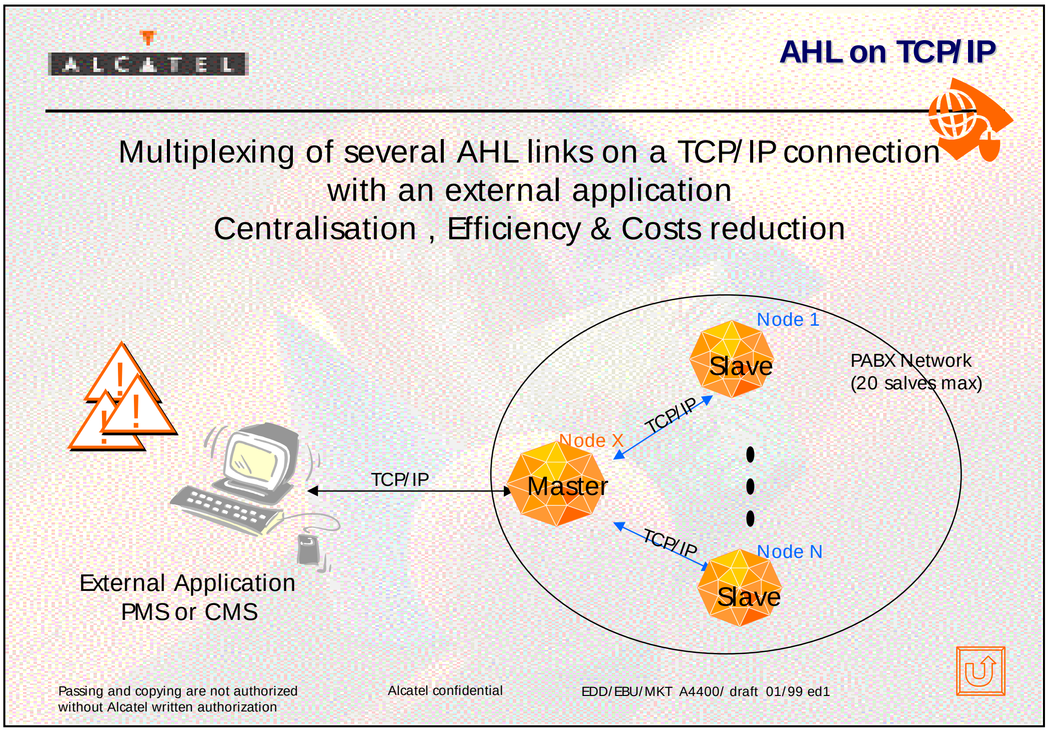
16.10 AUTOMATIC ROUTE SELECTION

16.11 HIGH TECH SOLUTIONS

16.11.1 AHL on TCP/IP




AHL on TCP/IP
A new brick in the hospitality Information System
PMS
Property Management System
Ethernet
NMS
Network Management System
Passing and copying are not authorized
without Alcatel written authorization
Alcatel confidential
EDD/EBU/MKT A4400/ draft 01/99 ed1



16.11.2 AHL centralisation

16.12 OTHER ENHANCEMENTS

16.12.1 Room number expansion

16.12.2 Static association guest / room

16.13 THE SMART CARD WRITE & READ SYSTEM

16.14 SUMMARY TABLE

SERVICES	HOTEL INDUSTRY ORIENTED	HEALTHCARE INDUSTRY ORIENTED
Voicemail 4635 : business mailbox		▯
New hotel voice mail box type	▯	
Centralised voice mail	▯	▯
Japanese	▯	
Multiple wake-up	▯	
Snooze	▯	
Wake-up in network	▯	▯
Multilanguage expansion	▯	
Dual line on digital sets	▯	
MCDU based password	▯	▯
Privacy feature	▯	
OP conversation	▯	▯
DECT	▯	▯
Notification server		▯
Call accounting in network		▯
Call accounting by duration with pulse simulation	▯	▯
Call accounting parameters expansion	▯	
Call accounting extended tickets	▯	▯
Buffer size expansion	▯	▯
Billing in network	▯	▯
Itemised billing for a given period of time	▯	▯
Double currency billing	▯	▯
Automatic Route Selection	▯	▯
AHL on TCP/IP		▯
AHL centralisation		▯
Room number expansion		▯
Static association guest / room	▯	
The Smart card Write & Read System		▯

17. MISCELLANEOUS APPLICATIONS

17.1 CSTA ENHANCEMENTS

17.1.1 <u>CSTA in 4400 R1.4 and R2 : Recap</u>

CSTA is available in :

* in R1.4
* in R2 with additional services

CSTA is used by :

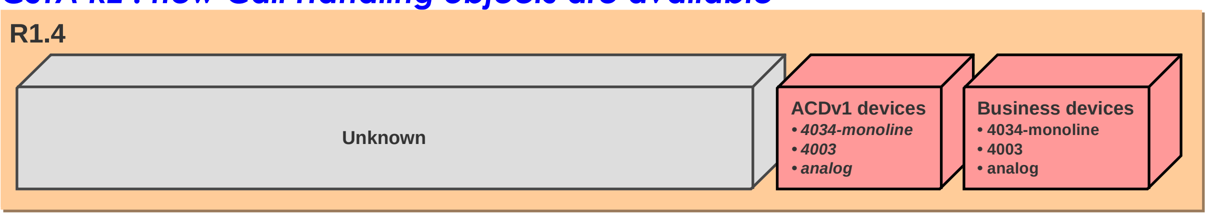
* Alcatel 4400 Business
* Alcatel 4400 ACD V1
* Alcatel 4400-CCD (V2.1)
* Alcatel 4400-CCS
* Alcatel 4400 Notification Server

What is it ?

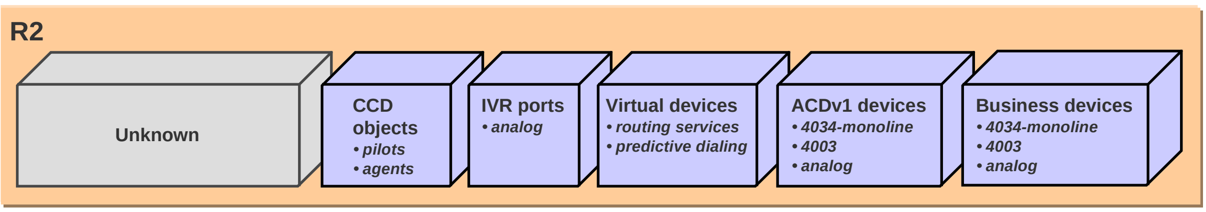
* a way to monitor/control Call Handling objects

17.1.1.1 <u>CSTA in 4400 R1.4 and R2 : Recap</u>

CSTA R2 : new Call Handling objects are available



Note that analog devices can either be real or fictive (aka w/ a coupler but w/o the telephone set)



New CSTA services in 4400 R2 for Alcatel 4400 Business or CCD :

- Virtual devices for external CTI application routing
- New services :
  * route request (routing device = set)
  * route select
  * route end
  * hold call
  * retrieve call
  * A4400 Date and Time setting
  * divert call for all incoming calls
    (was only for external incoming calls in R1.4)

- ECMAv2 compliance

Those new services are offered for an external CTI application on Alcatel 4400 Business or CCD

New services requirements lead to switch to ECMAv2

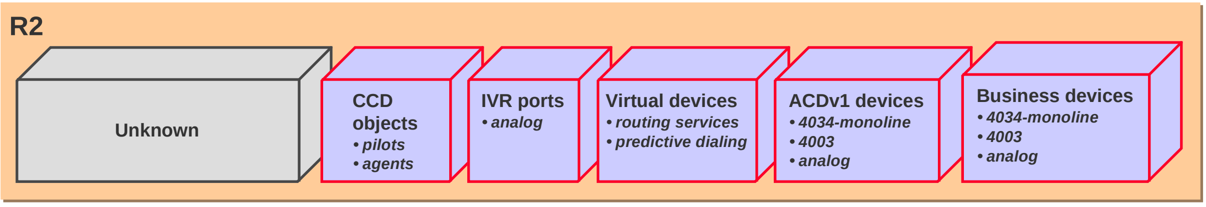
- AppCorrelatorData (was private before), Agent monitoring model (new), ...
- Compatibility tools were designed to ease migration path

In addition, extensions using Escape services & Private Data were designed

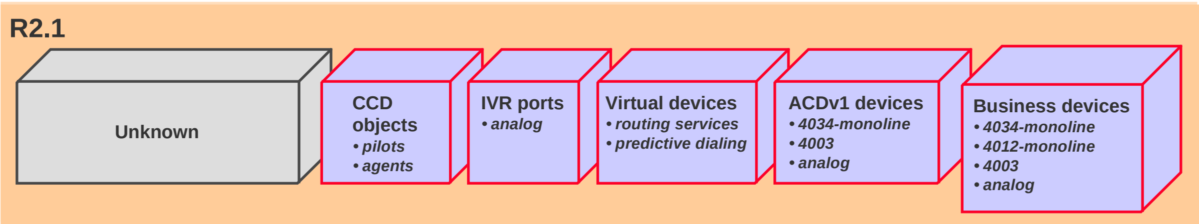
- Whenever required, these extensions are proposed to ECMA

CSTA in 4400 R2.1 : Recap

CSTA R2.1 : 4012 monoline sets are handled by CSTA

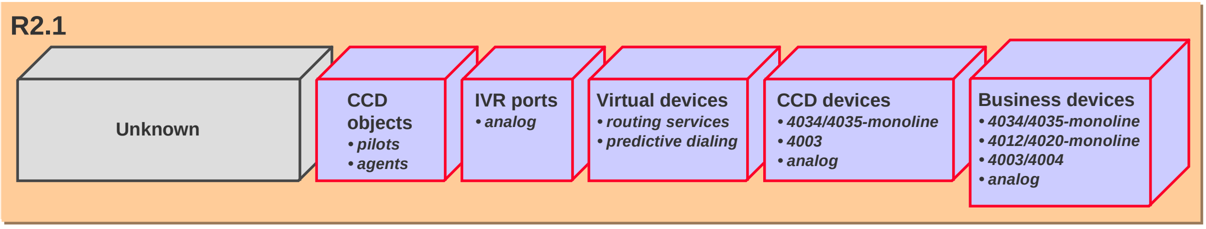


Note that A4012 are only handled for A4400 Business (not for A4400 CCD)

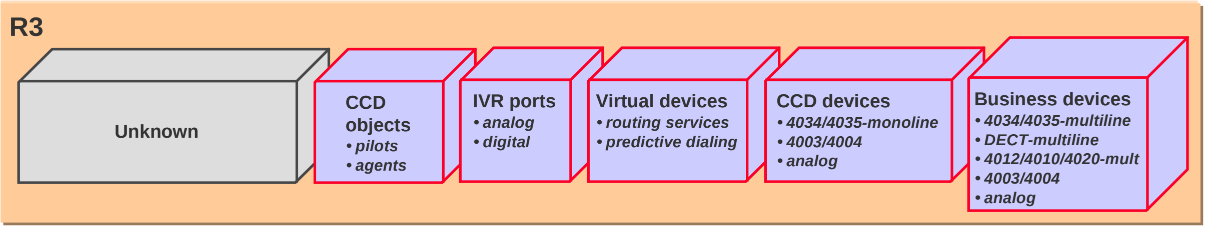


17.1.2 What's new in CSTA of 4400 R3 ?

CSTA R3 : DECT (UA/GAP) & multiline sets are handled by CSTA



Note that DECT & multiline sets are only handled for 4400 Business (not for 4400 CCD)



In 4400 R2.1 the new 3g phone sets are in 2g emulation mode and are **transparently** handled by CSTA (i.e. like 2g sets)

Therefore, 4010 (compatible 4011 for 2g emulation) is not monitored by CSTA ...

Note that 4004 is only monitored in 4400 Business (not in 4400 CCD), this is due to a problem in headset mode

In 4400 R3 the new 3g phone sets are in full 3g mode and all of them including 4010 are handled by CSTA

Now DECT handsets UA/GAP are also handled by CSTA

As shown in the figure above, all the multiline sets are now handled by CSTA in multiline mode.

CSTA monoline/multiline mode monitoring automatically fits the set configuration in 4400 : monoline/multiline

In 4400 R3 IVR ports can be digital or analog

New CSTA services for Alcatel 4400 Business or CCD :

- Multiline mode (only in Business)
- Calling Name on incoming call (only in Business)
- DTMF sending
- Automatic MF sending (bundle depending)
- Immediate Forwarding
- Cancel Conference (for master)
- Correlator data in private/public ISDN network
- Make Call auto originate (automatic handsfree)
- Set device in/out service
- Start/stop record (4630, 4635)

Those new services are offered for an external CTI application on Alcatel 4400 Business or CCD

Notes :

- Multiline mode
  - limited to 5 simultaneous calls
  - no monitoring of supervision lines : not necessary as the incoming calls are taken on the line keys with pick-up indication

* works also on Manager/secretary set
* no monitoring of DSS/BLF : but in case of CSTA monitoring, a DSS/BLF outgoing call will call the line keys on called party (Manager or Secretary)

- DTMF sending is like on 4400
- Automatic MF sending is bundle depending, like 4400 feature
- Immediate Forwarding is as on 4400
- Correlator data in private/public ISDN network
  * for call transfer with associated data
- Make Call auto originate
  * automatic handsfree even if no automatic answering
  * allows no automatic answering for incoming calls
- Set device in/out service
  * request for CTI App for incoming call routing on business sets, useful for virtual Z device