行政院所屬各機關因公出國人員出國報告書

（出國類別：考察）

國際級電信業者大企業客戶之服務及銷售運作模式

服務機關：中華電信總公司　　中華電信北區分公司

出國人　職　稱：　處長　　處長

　　　　姓　名：　陳崑雄　　黃秀谷

出國地點：英國、德國、法國

出國期間：92年11月12日至92年11月21日

系統識別號：C09300206

公務出國報告提要

頁數：114　含附件：是

報告名稱：赴歐洲考察國際級電信業者大企業客戶之服務及銷售運作模式

主辦機關：　中華電信股份有限公司

聯絡人／電話：柯志勇／2344-4094

出國人員：

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出國類別：　考察

出國地區：　英國　德國　法國

出國期間：　民國 92 年 11 月 12 日 — 民國 92 年 11 月 21 日

報告日期：　民國 93 年 2 月 1 日

分類號／目：H6／電信

關鍵詞：　中華電信，英國電信，阿爾卡特，德國電信，西門子，法國電信

內容摘要：中華電信公司在面對民營固網業者強烈競爭及 VoIP 業務與客戶端迴路(Local Loop)開放租用在即之雙重壓力下，如何鞏固既有企業客戶及提供創新服務，實是刻不容緩的課題。本案赴英國電信(British Telecom)、阿爾卡特(Alcatel)、德國電信(Deutsche Telekom)、西門子公司(Siemens)及法國電信(France Telecom)考察國際級電信業者大企業客戶之服務及銷售運作模式。本報告乃將實地考察英、德、法等地先進電信公司企業客戶服務之作法，依序說明各電信公司組織、經營策略及未來方向，以供學習其經營管理企業客戶及提供差異化之套裝服務。最後提出對英國電信之滿意度調查作法等感想與建議，期能提昇本公司企業客戶服務之應變能力。

目　錄

壹、 前言

本公司在85年7月由官署的電信總局，改制為中華電信公司，並同步成立企業客戶單戶(當時稱為專戶處，設立專案經理人提供大型企業客戶單一窗口及整體服務，扮演本公司與專戶的橋樑；91年1月更名為企業客戶處擴大服務範圍及於中小企業客戶)。在電信市場已然全面開放、競爭愈趨激烈之際，提供完善的服務來提升客戶的滿意度，以鞏固客戶提昇企業形象、拓展業務增裕營收。

從另一個角度來看，管理學上80/20法則，80%的營收來自20%的客戶，對本公司而言這20%的客戶就是企業客戶－需求多樣化、希望量身訂做的專屬服務、對電信服務的品質要求高、對電信營收貢獻度較高、任何一家的大型企業客戶移轉都將成為競爭者的宣傳利器。

此次職等奉派前往英國電信(British Telecom)、德國電信(DEUTCH Telecom)、法國電信(FRANCE Telecom)實地考察「國際級電信業者大企業客戶之服務及銷售運作模式」，就是本著他山之石可以攻錯的精神，藉由歐美等知名的電信公司服務企業客戶的方法，檢驗我們自己模式做適度調整，希望提供企業客戶更精緻更細膩的服務。

貳、 行程

天	月/日	工作摘要		地點
1	11/12 (三)	**台北 → 倫敦(行程)**		**Taipei/London**
2	11/13 (四)	British Angel Centre 參訪、簡報		**London**
3	11/14 (五)	Alcatel &British Telecom 會議與意見交流		**London**
4	11/15 (六)	**整理資料**		**London**
5	11/16 (日)	**整理資料** **London → MUNICH(行程)**		**MUNICH (ARRIVAL 16:55)**
6	11/17 (一)	SIEMENS Telecom (**MUNICH**) 參訪、簡報		**MUNICH**
7	11/18 (二)	DEUTCH Telecom 會議與意見交流		**MUNICH**
8	11/19 (三)	**MUNICH→ Paris(行程)**	Alcatel &FRANCE Telecom 會議與意見交流	**Paris**
9-10	11/20-11/21 (四)(五)	**巴黎 → 台北 (行程)**		**Paris/ Taipei**

叁、考察內容

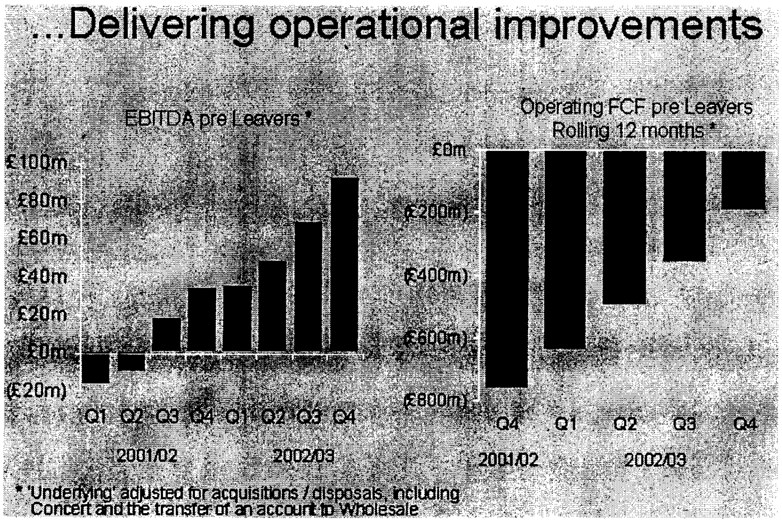
3.1、英國電信 (British Telecom, BT)

11 月 13 日主要行程為英國電信公司，上午 9:00 到達 BT 所屬之簡報中心，由其介紹 BT 整體營運狀況及新 CEO 上任之後的工作重點。目前 BT 擁有十萬零四千七百名員工，2002 年營業額達 130 億英鎊。BT 是一個兼具電信語音 (Voice)骨幹網路 (Carrier)、無線網路(Wireless)、接取網路 (Access) 及企業網路 (Corporate) 的大型電信服務供應商。在歐洲，BT 是服務水準名列前矛的公司，服務範圍包含軟硬體整合、寬頻平台提供、及大客戶 CPE 建置服務。在策略上，BT 目前為政府、商業集團及金融公司主要提供 MPLS 網路服務，此外客服中心 (Call Center) 服務亦相當流行，目前規模約為 10~100 人，BT 並且同時提供以 Web-base 為基礎的海外 (Off-shore) 客服中心服務。為了加強行銷產品給用戶，英國電信與終端產品設備廠商進行策略聯盟，並擬定標準作業程序 (SOP)，以加速達成客戶所需之網路設計及行銷服務。茲將 BT 以 (一) 公司策略構面、(二) 組織架構、(三) 積極固守現有客戶、(四) 未來方向等四方面分別闡述於后：

(一) 公司策略構面

2003 年英國電信公司策略主要重心放在以下幾個構面：

- 提昇客戶滿意度
- 針對不同顧客群提供差異化服務
- ADSL 寬頻上網服務
- 對歐洲網路營運商提供加值服務
- 將英國境內及歐洲部分的網路建立統一管理介面
- 技術純熟的員工組成之工作團隊
- 財務改善計劃



圖一、BT 財務改善情形

(二)組織架構

主要由 BT Retail、BT Wholesale、BT Global Services、BT Openworld 及 BT Exact 五大分公司所組成，分述如下：

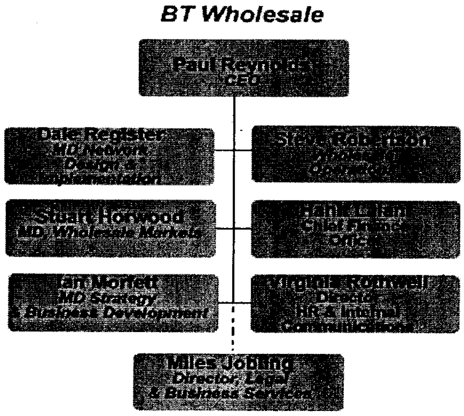
1. BT Retail

BT Retail 定位在市場通路上，提供傳統電話服務、行動通訊、網路存取及通訊整體服務方案，主要客戶群為中型企業、小型企業及一般消費客戶。年度營業貢獻度為一百三十億英鎊，共有二千一百萬位客戶及五萬位員工。今年 6 月 1 日起，將夜間及週末假日之通訊費率，從按分計費之方式調整為單一費率之方式頗獲一般消費客戶好評。根據 2003 會計年度報表顯示，在企業市場中，通訊服務外包之企業成長百分之八，IP 基礎建設服務 (IP Structure) 成長百分之五十八，寬頻服務則大幅成長四倍。

圖二、BT Retail 組織架構

2. BT Wholesale

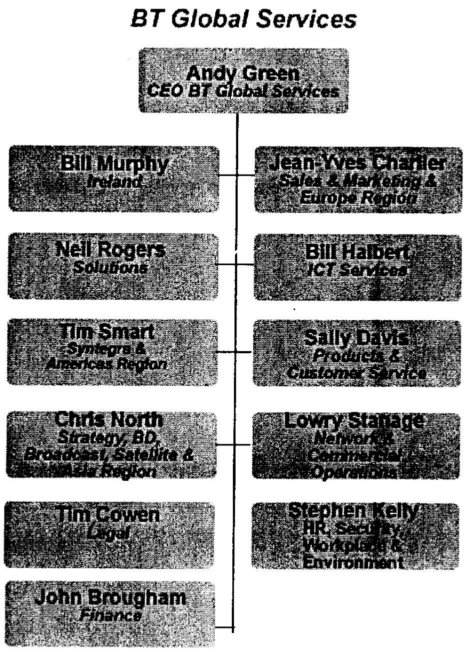
主要業務為提供完整的網路服務給英國國內電信公司、網路與服務業者、以及 BT 事業群其他營運公司，平均每日語音通話量超過三億通，每月網路傳輸連結次數超過三億五千次。2003 會計年度員工人數為二萬七千六百餘人。



圖三、BT Wholesale 組織架構

3. BT Global Services

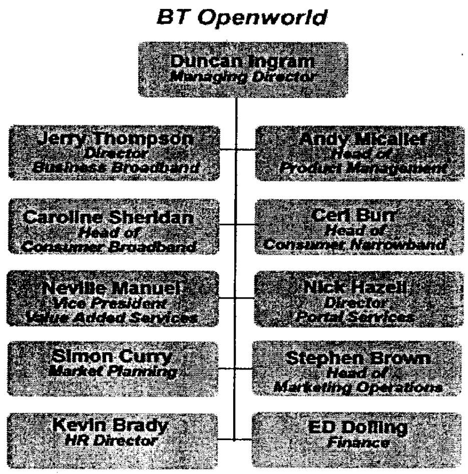
本部門原名 BT Ignite，負責全世界跨國企業網路服務及加值服務，其中以連接全世界六十餘國之 MPLS based IP-VPN 最受歡迎，2003 會計年度之營收達五十二億五千一百萬歐元，比上一年成長百分之十七，員工人數為一萬七千二百餘人。

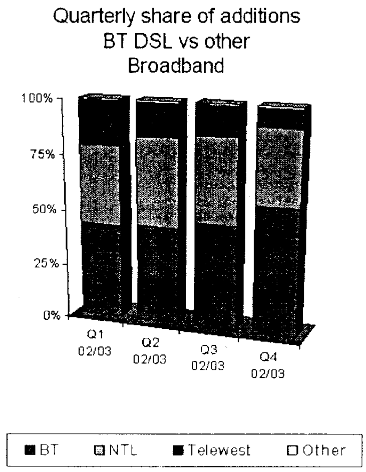


圖四、BT Global Services 組織架構

4. BT Openworld

客戶數達一百七十五萬個，提供一般客戶與中小企業客戶上網及加值服務，旗下有 BTinternet 及 BT Openworld broadband 二大事業群。其中 BTinternet 為英國境內最大的 ISP 業者，而 Openworld broadband 則是提供一般消費客戶及企業客戶之 ADSL 高速上網服務。

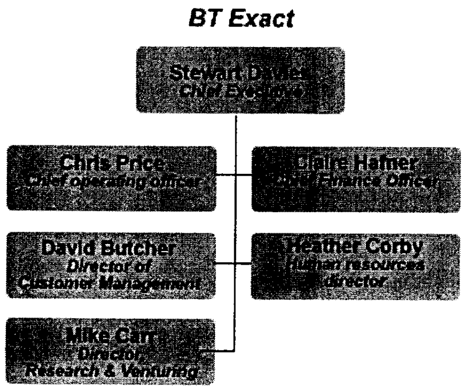




圖五、BT Openworld 組織架構　　　圖六、英國寬頻市佔率

5. BT Exact

為爭取商機，英國電信另外獨資成立顧問部門 (BT Exact)，專職 6 人，技術顧問服務項目內容包括：市場行銷、網路規劃與設計、國際行銷及人力資源管理，並配有 450 餘位自英國電信退休之同仁支援技術顧問工作。在 2003 會計年度投資三億八千萬英鎊，為客戶研究並開發通訊及 IT 方面之整體服務解決方案。



圖七、BT Exact 組織架構

(三)積極固守現有客戶

BT 所服務之客戶群分為大型跨國企業、中小企業、一般住宅用戶三類，敘述如下：

1. 大型跨國企業

國內 500~1,000 之大型跨國企業由 6,000 位專案經理 (SPOC, single point of contact) 為主的 BT Global Services 公司提供單一窗口及專業諮詢服務，於 2002 年創造 40 億英鎊之營收。為了鞏固大客戶忠誠度，行銷多採用綑綁產品服務 (bundle service)，並強調加值服務以爭取客戶訂單。

2. 中、小型企業

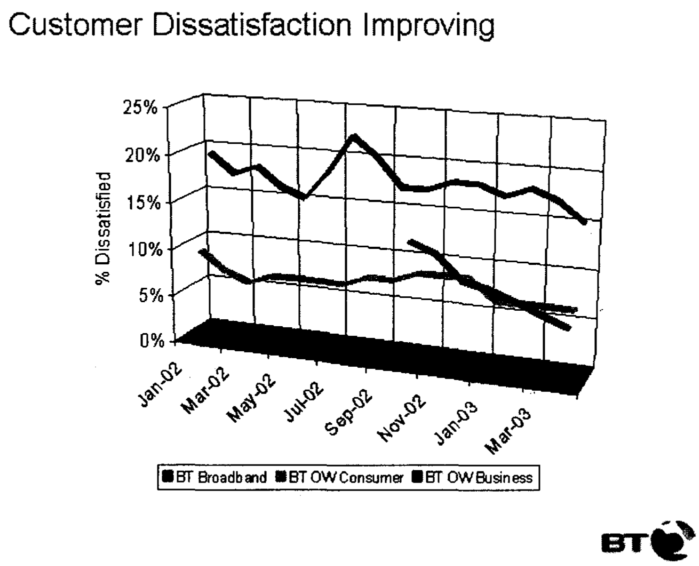
中型企業客戶約 30,000 家，其中的 50%有專案經理服務，其餘 50%仍靠電話行銷 (Telesales) 及客服中心之 Outbound Call 服務。至於 110 萬家小型企業部分，則以直接郵寄 (Direct Mail)、客服中心之 Outbound Call，或經由廣告提供相關服務資訊，整體 113 家萬中、小型企業客戶由 5,000 人服務，從設計至維運、行銷一體，於 2002 年創造 24 億英鎊之營收。

3. 一般消費用戶

一般消費用戶 (Residential Subscriber) 約 2,000 萬戶。包含一般家庭及個人用戶，其服務方式由 BT 在全國所設之 10 個客服中心據點受理服務。

項目	大型跨國企業	中型企業	小型企業	一般消費用戶
區分標準	500～1000 大型跨國企業	多據點企業	員工人數少低使用量	泛指一般家庭及個人用戶
戶數	500～1,000	30,000	1,100,000	20,000,000
服務人力	6,000	5,000		N/A
貢獻度 (英鎊/年)	£666,667 (約台幣 3,933 萬元)	£480,000 (約台幣 2,832 萬元)		N/A
行銷服務方式	派專案經理及專案工程師服務	派專案經理及專案工程師服務	採促銷與保衛戰	客服中心
裝維方式	依照企業類型區分，所有裝機、維運自成體系，自行負責			

為及時掌握客戶對英國電信的服務是否滿意，每個月固定對 1,000 家客戶進行抽樣作為未來改善服務品質之依據。



圖八、BT 客戶滿意度調查 (不滿意部分逐漸下降)

(四) 未來方向

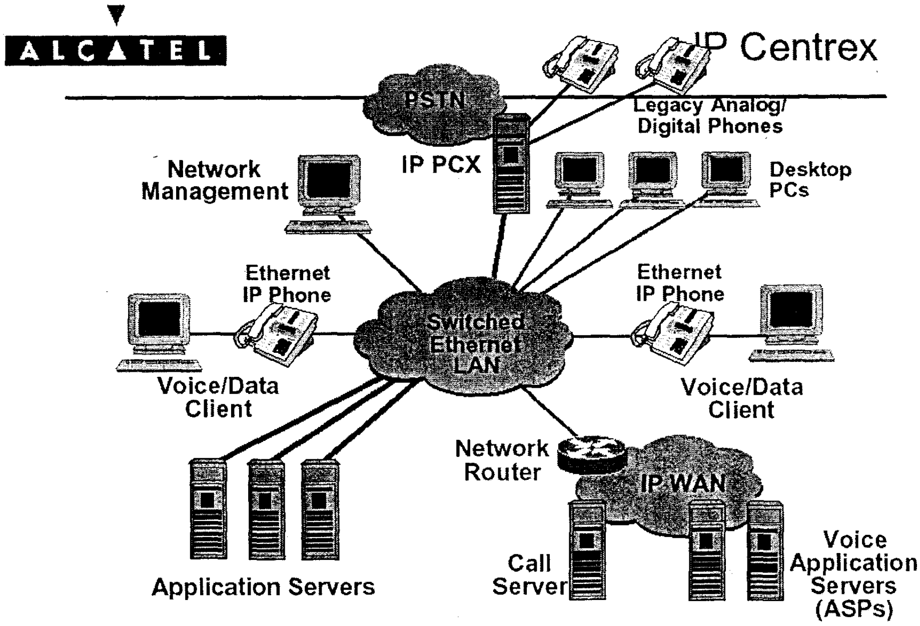
在英國電信市場，只要競爭廠商願意提供較原電信服務提供者低約 10%的價格策略，客戶即會轉換電信服務廠商。

預估未來企業對 IT 消費有逐漸成長之趨勢，未來電信支出應能隨之成長。對於 WLAN 的推動，初期則是設定 500 個熱點 (hot spot) 的目標。

BT 內部對於專案經理有提供獎金 (bonus) 方式的獎勵制度，對於專案經理之銷售具有激勵的效應。BT 對跨國大企業及中小企業採取不同的價格策略，並針對不同客戶群提供差異性的商品與價格包裝策略 (BT Together pricing package)。

3.2、阿爾卡特

阿爾卡特目前在世界上在產品、方案和業務支援等方面，數位用戶線路佔全球市場份額 (market share) 達 18%，全世界每 5 個電話就有 1 個是通過阿爾卡特提供的設備連接的，目前發展係基於現有網路結構進行演進的下一代網路 (NGN) 方案，以提供用戶各種 IP 應用方案。

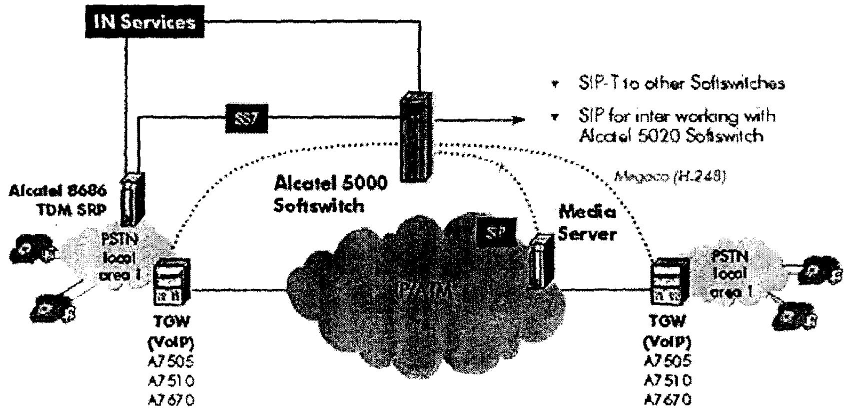


圖九、IP Centrex 網路架構

(一) NGN C4 方案

具備 Class 4 交換機彙接功能的 NGN C4 網路架構可以讓現有的分時多工(TDM) 語音網維持目前的功能，由 NGN 架構中的設備彙接、處理現有的和新增的語音及資料業務。

本方案由 A5000/5020 softswitch 進行呼叫控制，由中繼閘道(TGW)，如 A7505/7510/7670 MG 以及 A1000 S12 IGW，連接 PSTN 與 NGN，並由 1300 CMC 管理。使典型 PSTN 用戶在直接與當地交換局相連下，將產生的話務量透過 TGW 轉移到 IP-Based 的網路中。

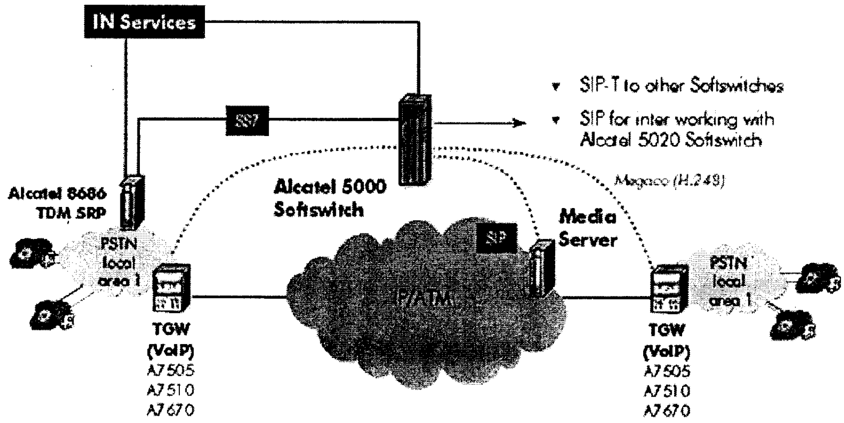


圖十、NGN Class4 方案

對用戶而言，不會感覺與傳統電話線路相比有什麼改變，但是對局端而言，卻可以在不需大幅改變傳統 TDM 網路的前提下提供 VoIP 的功能。目前 NGN C4 系統主要作為 "長途電話的備援"。

(二) NGN C5 方案

具備 Class 5 局端功能的 NGN 網路架構可以讓 TDM 語音網維持目前的功能並處理新增語音和資料業務。提供傳統話音業務的電信業者可以經由 NGN 過渡到單一網路架構。想要將 PSTN 用戶移轉到 NGN 網路，NGN 網路就必須提供與目前 PSTN 品質相近的 Class 5 局端業務。



圖十一、NGN Class5 方案

用戶的接入可以通過在網路中增加如 Litespan-1540 接入閘道，Speed Touch pro 700，以及由 5020 控制的 SIP 電話和 H.323 終端等智慧型終端。搭配使用先進的應用伺服器及媒體伺服器，不僅能提供語音業務，同時也能提供多媒體業務。

3.3、西門子公司 (Siemens)

11 月 17 日上午前往位於慕尼黑的西門子公司訪問，該公司在營運上主要秉持下列五項原則：

- 增強客戶競爭力—我們的成功來自於客戶的成功
- 重視企業創新—將未來概念具體化
- 提昇公司價值—開創新契機
- 加強員工能力—使企業達到世界級的表現
- 落實企業責任—回饋社會

西門子有 40%的客戶需要完整而多元化的解決方案，針對客戶的需求，由專案經理為各種不同產業的獨特性需求提供服務，如同系統整合業者為企業承包量身訂作的整體服務專案。公司並設有專員負責從購買過西門子的產品與服務的客戶調查滿意度，從調查所得的客戶意見中去瞭解改善產品與服務的方向。西門子企業維運由多個客服中心來管理、負責，並經由電話及電子郵件蒐集得之各項問題。

另外介紹 SURPASS VoIP 交換系統設備，敘述如下：

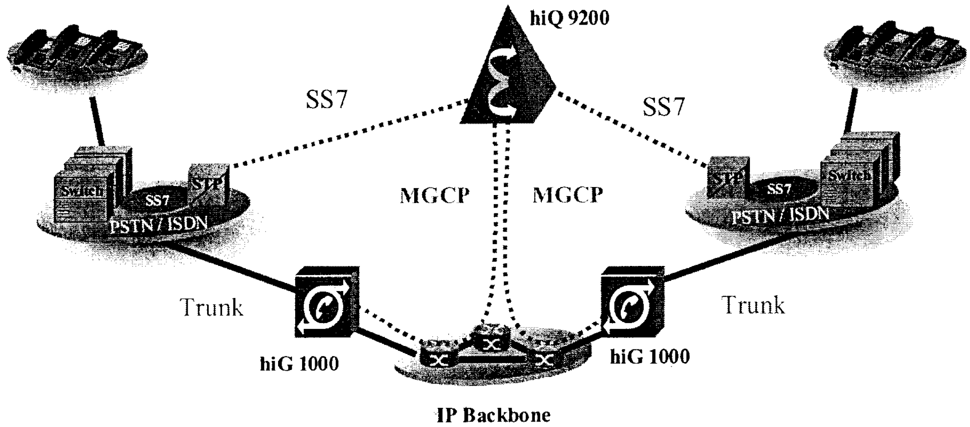
SURPASS VoIP 產品係採用 CoreNet IP 的新軟體協議。CoreNet 基於 H.323 IEEE VoIP 標準，它把更多的 PBX 特色加入到 IP 世界。西門子還將為尚未轉移到 IP 電話的用戶發佈一個新數位電話系列。optiPoint 500 電話支援 TAPI (電話應用程式編程介面)、USB 和普通電話介面。

SURPASS VoIP 交換系統所需使用的相關網路設備：

- SURPASS hiQ 9200 (Softswitch，軟體交換機)
- SURPASS hiG 1000 V3.0T (媒體閘道器)
- SURPASS hiQ 4000 V3 (開放式服務平台)
- SURPASS hiR 200 V1S (資源伺服器)

VoIP 功能

VoIP (Voice over IP) 是在 IP 網路傳送語音、傳真和數據機訊務，即將語音話務在分封網路上載送以取代傳統電路式交換網路。下圖為虛擬中繼 (Virtual Trunking) 典型的 VoIP 應用。



圖十二、SURPASS hiG 1000 的VoIP功能

透過 SURPASS hiQ 9200 Softswitch 控制呼叫建立及釋放，執行信號轉換-與電路式交換網路轉換SS7信號，和利用 MGCP 與媒體閘道器 SURPASS hiG 1000 交換控制信號。一旦建立呼叫，SURPASS hiG 1000 從 PSTN 網路取得音頻信號，依不同音頻信號型式，而使用不同編碼技術，把音頻資訊轉變成數位音頻封包流，這封包流被封裝到 RTP/UDP/IP 和在 IP 網路上被載送到其他閘道器作做進一步處理。

3.4、德國電信 (Deutsche Telekom)

18 日上午前往德國電信與該公司服務部門商談企業客戶服務情形，德國電信由 T-mobile、T-Online (Internet)、T-System (System Integration) 及 T-Com (固網部分) 所組成。

德國境內目前 PSTN 建置量為 6,000 萬路，德國電信的整體客戶約有 4,135 萬位客戶 (市場服務佔有率為 68.92%)、T-mobile 現有 2,500 萬位客戶、T-Online 為 1,267 萬，由於投資 3G (執照費用即需 500 億歐元) 及經營上的其他因素，整體企業負債約 700 億歐元。

對於大企業客戶服務，T-System 在全國設有 25 個服務據點，每據點有 200 位員工，2002 年共創造 113 億歐元。對於中小企業客戶服務，全國設有 100 個銷售據點，而對於一般客戶的銷售部分，則以每超過 5 萬人就設一個銷售據點。

企業用戶的上網方式，中小企業以使用 1.5M ADSL 服務為主，大企業則以使用 2M SHDSL 服務為主。德國電信員工約有 43,500 人，其中 1 萬位員工在國外負責海外市場的經營。

德國電信行銷策略：由於德國電總只對會影響大眾的電信服務有價格管制，對於企業客戶服務的資費均以個案處理，並與企業客戶諮商之後，依雙方協商簽約內涵 (如大客戶 SLA 合約) 而定，此定價機制較本國合理。

在促銷手法上，以不同費率結構或內涵提供月固定價 (Monthly Flat Rate) 的方式促銷 (電話每月 €13 歐元、ISDN 每月 €23 歐元等)。包含住宅客戶在內也可以選擇國內市話、國際市話、行動通信及上網服務整合的套裝服務 (bundle with mobile and internet services)。

約有 90%的客戶選擇自備自裝 (DIY) ATU-R 的方式，並提供客戶免費的試用期，讓客戶熟悉使用後再計費，試用期限依個案而定，例如學生 DSL+WLAN 專案以 Flat Rate 方式提供，試用期以不超過一個月為限。2.5 年之內申請 T-DSL 的客戶急增至 400 萬戶，以上行速率 768K/下行速率 128K 的 ADSL 服務為例，費率計算方式與國內狀況雷同，分為電路月租費 (€31.2 歐元) 及網際網路通信月租費 (€29.95 歐元)，合計每個月月租費為 €61.15 歐元 (台幣約合 2,939 元)。

最令人矚目的促銷活動是創造 3D 模特兒當廣告代言人，頗得當地民眾熱烈迴響。其中，T-Online 的佔有率最高，成功關鍵在於網頁 Portal 獨樹一幟，網頁頗具個人化風格，能夠吸引所有客戶的青睞。另推出在網路上觀看足球即時轉播以提高寬頻的使用率。

德國境內固網業者多達 50-100 家，競爭極為激烈，面臨 IP 服務的開放，也提供 IP Centrex 及 IP PBX 的服務，唯有增加服務種類及新加值服務，才能鞏固固網的營收。

3.5、法國電信 (France Telecom)

19 日上午抵達巴黎，並前往法國電信瞭解阿爾卡特如何支援服務法國電信的企業客戶及提供產品建議方案。法國電信公司組織與中華電信多有雷同，客戶群涵蓋大企業、中小企業及一般住宅用戶。

按照企業客戶的類型提供以 Customer Profile、Package-oriented 及需求分類，xDSL Access、整體服務解決方案 (Total Solution)、語音、技術顧問服務 (Technical Consult) 及寬頻與語音之整合性服務。

法國境內主要固網業者有法國電信、德國電信及 BT，法國電信在面臨德國電信、BT 之強大競爭壓力下，中小企業 (SME) 市場佔有率為 55%～68%，大客戶市場佔有率則為 35%。

阿爾卡特為法國電信最大供應商，與法國電信訂有長期供料合約，為使法國電信符合市場供料需求，並採用依客戶預定下單 (Build to customer order，BTCO) 系統，快速反應客戶需求以即時提供客戶終端設備服務。

固網營收因話務轉向行動網路而節節衰退，為鞏固營收，法國電信當務之急是如何確保 IP 話務在固網網路上發展。

企業常有將電信工作外包 (outsourcing) 的想法，卻又不願意支付太多的費用，在這種趨勢，阿爾卡特推出 IP-Convergence 服務 (整合 IP 語音)，以提供企業價格低廉的語音數據服務。

服務之成功關鍵必須能夠於網路妥善控制語音 IP 訊務量，就目前之技術水準而言，IP 話務難以網管控制，故不易達成 QoS 的目標。法國電信除了提供寬頻接取服務之外，亦提供都會 IP 服務 (Metro-IP Service) 及企業 LAN-to-LAN Service，在市場上同樣受到歡迎。

肆、感想與建議

一、 英國電信每月固定對 1000 家(含住宅)進行滿意度調查，比較能迅速了解客戶對服務品質的反應，本公司每年做兩次客戶滿意度調查(企業客戶與一般消費者客戶)，宜適時增加對各類促銷業務或新服務推出時做產品滿意度的小群化調查。

二、 對於大型企業客戶除以專案經理提供單一窗口服務外，另搭配專案工程師(FE)與系統整合人員(SI)的工作需加速辦理。

三、 由於 VoIP 業務與客戶端迴路(Local Loop)開放租用，對既有經營固網業者的企業客戶營收影響很大，本年度更是關鍵時刻，如何以加值營收彌補因 VoIP 與 LLU 引發的營收減少，將挑戰本公司的應變能力。

四、 服務企業客戶與一般住宅客戶差異很大，在培養專業人力與節省成本的考量下，英德法日本等國電信業者均以獨立垂直體系提供企業客戶服務，本公司只做部分區隔，在民營化後建請擴大獨立營運體系。

五、 近年來因客戶的語音話務已大幅轉向行動話務網路，導致固網營收不斷下降，如何固守大企業客戶市場，在國外的經驗是提供用戶端網路設備(含終端)，由電信業者統一經營維護，以提供具有高品質的語音以及多元化的數據加值服務，目前在歐陸地區電信業者紛紛推出 IP Centrex 服務，已成為市場焦點的新產品服務，本公司應重視此一趨勢，適時推出類似產品以鞏固營收。

六、 為確保企業客戶市場，除價格策略之外，提供令客戶感動的服務及提昇產品品質為刻不容緩的課題，其中提昇品質的方法，在歐洲各大電信業者多採用設計、維護、銷售、服務一元化組織，以確保於競爭市場中快速且有效滿足企業客戶需求，創造營收。

META Group

White Paper

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Voice-Oriented Services

Transforming Commodity Systems Into Strategic Communications Infrastructure

A META Group White Paper

"During 2003-05, we expect IT organizations to become more strategic about developing enterprisewide, voice-oriented services. Technical architecture and infrastructure development skills will be applied to voice technologies to reduce capital and operational costs, leverage existing assets, support faster application development, and streamline current silo-like voice systems into a standard corporate infrastructure service."

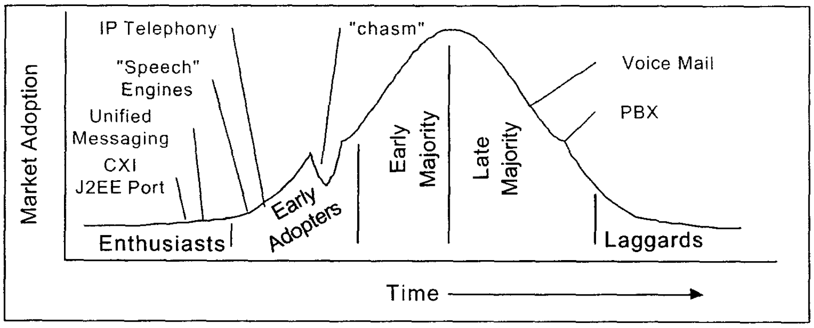
February 2003

Contents

Abstract

Increasingly, the validity of policies used to plan, build, and run core enterprise voice systems (e.g., private branch exchange [PBX], automatic call distributor [ACD]) and value-added voice systems (e.g., computer telephony integration [CTI], interactive voice response [IVR], automated speech recognition [ASR], voice mail [VM]) are coming under question in view of current technology transformations (e.g., IP telephony, speech-enabled voice applications, multiple point-of-interaction [POI] integration platforms [CXI][1], unified communications [UC]). Formerly, obtaining voice system features and functionality were "buy" decisions, and multiple purchases typically were made from the same vendor because it guaranteed a degree of interoperability with the products it offered. Although single-vendor buy decisions can still be good policy, in 2003-05, we expect many IT organizations (ITOs) to think more strategically about developing enterprisewide voice services. We believe that by applying IT architecture and infrastructure development skills to voice technologies, ITOs will reduce capital costs, better leverage existing assets, reduce ongoing operational costs, support faster application development, and streamline current silo-based voice systems to a standard corporate infrastructure service. Moreover, relevant vendors (e.g., Alcatel, Avaya, Nortel, Siemens) are developing multiple products to enter emerging technology markets[2] that promise to be as lucrative as the legacy voice systems market. However, few indicators exist of when and how these markets will mature (see Figure 1). These vendors have historically boosted enterprise system sales by leveraging their roots in supplying equipment to telecommunications operators, but this is rapidly diminishing in the presence of new entrants (e.g., 3Com, Cisco, Alcatel/Genesys, Interactive Intelligence) offering different but competitive products.

Figure 1 — Voice Technology Adoption Life Cycle: A High-Level View



In this environment, ITOs should learn how to spot and exploit emerging market drivers (e.g., lowering total cost of ownership (TCO), increasing infrastructure flexibility) that affect vendor investment decisions (in addition to many vendors' current weak financial performance) to regain control of voice infrastructure development. By defining architecture and infrastructure plans, ITOs will dictate what vendors do to compete effectively for future corporate voice infrastructure services while benefiting from emerging voice technologies.

Voice Technology Transformations

Despite the fact that the current voice technologies (e.g., PBX) are proven, new products are being released with the expectation that current customers will willingly switch platforms. Vendors are hoping to offer customers the following technology transformations and desired results:

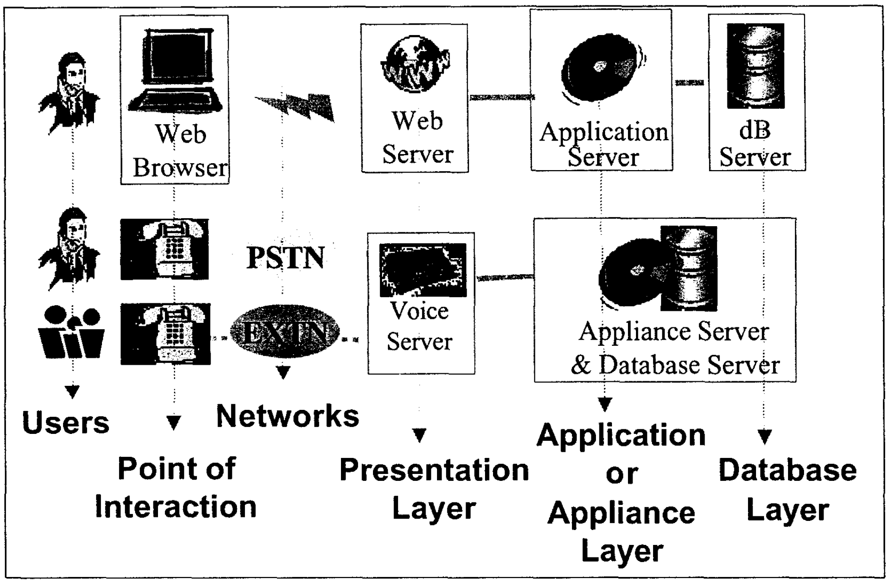
- The entire PBX/ACD industry has invested millions to deconstruct monolithic 1-tier proprietary systems into multiple tiers by porting functionality onto "open" server platforms (e.g., WinNT, Win2000, Unix, Linux), leveraging local- and wide-area networks (LAN and WAN) technology components and advanced intelligent networks (AIN) within the public switched telephone network (PSTN). These investments should ultimately reduce infrastructure TCO of managing voice services (e.g., moves, adds, changes [MAC]) by leveraging open-standard platforms having lower support service costs and IP telephony, permitting the consolidation of LAN and PBX/ACD virtual private networks.
- The CXI industry is investing heavily to port "niche" features and functions (e.g., POI2, universal queuing, skills-based routing) onto the corporate standard J2EE application server (e.g., IBM's WebSphere, BEA's WebLogic), reducing the number of appliance servers (i.e., servers executing dedicated processes in a closed operating environment). CXI will benefit from better support services provided by IT, and front-office processes can get access to back-office systems at a much lower cost, reducing integration costs.
- Automated speech recognition (ASR), text-to-speech (TTS), speech-to-text (STT), and voice verification are increasing in adoption to support self-service applications. Numerous vendors offer "speech" toolkits that systems integrators and traditional IVR vendors use to develop "speech enabled" applications. Other vendors are offering VoiceXML application development environments, and emerging vendors are offering voice application servers (VAS); both promise significantly accelerated development of voice-enabled applications and the flexibility to change them at lower costs. Examples of these applications range from standard IVR, to voice mail and automated attendants.
- Unified communication platforms are being promoted by vendors (e.g., Alcatel, Avaya, Cisco) to initially provide a single "mail box" for all message types (e.g., voice mail, mobile mail, e-mail, fax, SMS), but they ultimately seek to be the infrastructure platform where all future enterprise voice services are developed, managed, and delivered enterprisewide.

Notably, all these trends are affecting the same vendors (e.g., Alcatel, Avaya, Aspect, Nortel, Siemens), and we believe that a large degree of uncertainty in the corresponding research and development (R&D) organizations, especially because none of these companies has been tremendously successful in stimulating market growth from newer products. Moreover, none of these vendors is adequately addressing its problems — that is, the available market to purchase new products is already saturated with functional and reliable voice systems, and creating a better "mousetrap" (i.e., product) is not compelling enough to make users switch. In the current economic circumstances, organizations want to understand how to leverage existing infrastructure and homogenously add new features and functions with the ability to scale out an infrastructure enterprisewide. Therefore, the future "requirement" for enterprise voice services infrastructure must offer high availability and scalability levels, and a core set of communication services that can be further augmented by developing value-added services (e.g., UM, screen pops, speech recognition, voice mail, auto attendant).

We propose that in the typical ITO's e-business "war chest" there be many new processes and skills developed to deploy online systems. These should be leveraged to instantiate the architecture of what future corporate communications should be as well as infrastructure planning to ascertain how far this vision can be accomplished in the short, medium, and long terms. Breaking down voice infrastructure into tiers, it can be shown that there are useful parallels to the

3-tier/"n"-tier e-business infrastructure that can be reused (see Figure 2). Furthermore, we provide our views on what architecture and infrastructure considerations should be taken. From these two perspective, we derive the benefits of reducing capital spending, leveraging assets, reducing total cost of ownership (TCO), and specific actions to take more control of corporate voice services.

Figure 2 — Voice and Web Infrastructure Tiers



Voice-Oriented Services Technical Architecture: The Road Map

Leading ITOs are increasingly raising the importance of strategic (3-5 years) information and technology architectures. META Group research indicates that companies can reduce IT spending by up to 30% with an architecture strategy. The primary goal of architecture is the design of information systems to support fundamental business changes. Architecture is a contributor to the formation of future corporate organizational structures, and it guides the alignment of limited personnel resources for the development and evolution of applications and IT infrastructure for architecture and business realization.

Considering a five-year horizon, technology is promising greater computing power, high speech recognition accuracy, Web services, Session Initiation Protocol (SIP)-based IP telephony, third-generation wireless, higher grades of mobility/devices, and further exploitation of contact centers as the preferred channel to market and service customers. For some organizations, the solution is selecting the right vendor with the best vision and long-term viability (e.g., Alcatel, Cisco). Although this may be adequate for some early adopters, a greater number of companies with embedded and high-performing voice systems are seeking to leverage existing investments. Moreover, in five years we do not expect the Internet to have ubiquitous quality of service (QoS) to enable end-to-end voice communications, thus requiring organizations to continue using existing PSTNs for voice communications. In our opinion, organizations will need to support both networks for the next 10-15 years and should be developing skills to optimize corporate communications with hybrid enterprise voice infrastructure.

Voice-Oriented Services Technical Architecture Domains

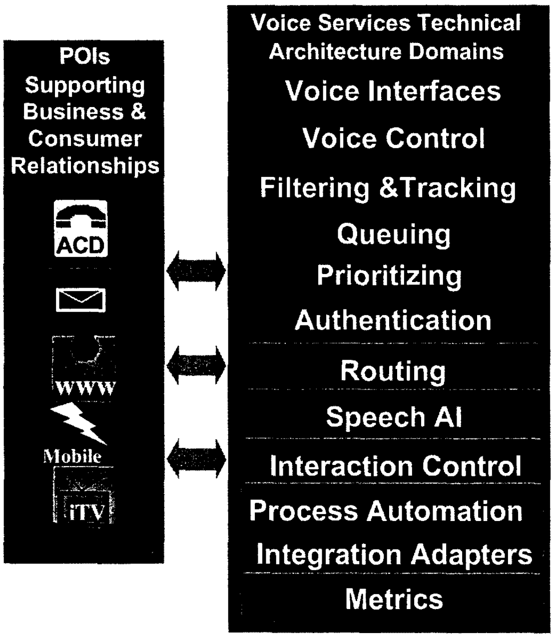
Considering the technology trends and user demands (e.g., mobility, self-service, more contact center models), we propose a high-level domain architecture (see Figure 3) with a functional description of the domains (see Addendum A). Organizations ultimately need to develop their own domain architectures by understanding corporate strategy better and incorporating the most suitable technologies. It should be noted that voice-oriented services are a shared infrastructure service that ITOs will build out over time to meet business demands.

Don't Give Up Control; Define Architecture

The voice services architecture is the best defense of corporate communication policy against vendor "blue sky," "marketecture" pitches that only prop up their offerings. Users must create voice architecture beyond technology component designs and not assume the notion of "everything going IP" will magically solve communication challenges. In our opinion, the communication challenges just begin with IP. Furthermore, networking and voice architects should study current IT architectures and the corresponding shared infrastructures deemed to support the future corporate organization. This will provide meaningful insights to organizational and process changes that influence corporate communications.

One of these organizational changes will be the increasing use of both formal and informal service centers. Formal service centers are typically customer-facing, and informal centers are typically business-to-business (B2B) or internally-focused; determining which model to use is dependent on interaction volumes and quality service levels. ITOs must realize that the future state of the organization will not be driven by customers only, but increasingly by business partners, internal employees, suppliers, and channels to market; providing quality services to these new constituents will drive the use of formal or informal service centers. ITOs should be careful not to replicate technology silos for each service center type to avoid replicating capital and integration expenses. In addition, a common voice-oriented service infrastructure must be adaptive enough to flexibly and quickly meet the demands of corporate service strategies, accenting valued and extended relationships.

Figure 3 — Enterprise Voice-Oriented Services Technical Architecture Domains



Notes.

1) Acronyms — AI: Artificial intelligence; iTV: Interactive television
2) The technical domains are established at an abstraction layer of generic attributes required to establish basic as well as complex voice services. Descriptions of each domain are listed in Appendix A.
3) The domain architecture must ensure that communication services support multiple voice POIs and possibly other electronic interaction points. The domain architecture should also consider the future state of the enterprise and what infrastructure services will and can be shared throughout the company.
4) A simple example of a voice service is a PBX for basic telephone communications. This service then requires voice interfaces for the PSTN and internal telephone extensions; voice control to set up and break down communications between end points; interaction control (typically available at the telephone set to control call arrival, progress, and termination; and metrics to report on the volume of internal calls, external calls, and the cost of communications.
5) A complex example of a voice service is contact center infrastructure with ACD, CTI, and IVRs. This service requires all PBX domains, plus filtering and tracking, to determine whether infrastructure resources (e.g., memory, processing power) should be allocated to the call, queuing to place calls, other POI requests and preserve application state before sending the call to its final destination; prioritization for moving valued customer requests up the queue; authentication to identify and authorize callers, routing that mediates across multiple networks and end points to deliver the call and metrics. Increasingly, more organizations are leveraging speech AI to reduce the complexity in managing the life cycle (i.e., enhancement definition, design, plan, build, test, production) of self-service applications. In

addition, financial services and telecommunication industries are reducing fulfillment activities by integrating contact center processes to back office systems using business process automation (BPA) transactional engines.

Voice-Oriented Services Infrastructure: Recognizing and Using Patterns

Central to IT architecture are shared infrastructure services that render computing value (e.g., transactional integration, networks, Web hosting) to different business units for a negotiated service purchase price and quality (e.g., reliability, availability, response time). This approach is increasingly obtaining executive-level support and funding to create more flexible IT systems in support of corporate productivity initiatives. Further, it enables ITOs to 1) reduce the number of independent systems; 2) increase interoperability among remaining systems; 3) improve the reliability of "handing off" systems from one IT group to another; 4) reduce the effort to implement changes; 5) reduce the number of suppliers to manage; and 6) steadily reduce the TCO. The inability to counter these problems will result in categorizing IT as a high-cost, low-business-value activity. More important, it will ultimately make the business uncompetitive in fast-moving markets.

It should be obvious that voice systems have similar challenges, a few of which are caused by high-performance requirements in contact centers; mergers and acquisitions (M&A) resulting in multiple and incompatible voice systems; and business demand to add new features and functions on fragmented infrastructure (e.g., UM):

?? In the contact center, there can be up to six different product types working together (very few are integrated) to support thousands of voice and data transactions per day for thousands of employees. When product changes need to be made, there is massive uncertainty about how well the system will perform due to weak product interfaces used for integration. Numerous vendors are addressing these challenges with closed platforms that do everything, but with less feature depth (e.g., Interactive Intelligence, Apropos, Altitude, Concerto, eOn Communications, Wicom), and leading CTI infrastructure vendors (e.g., Aspect, Avaya, Alcatel/Genesys, Rockwell) aspire to expand current suite offerings.

?? M&As rarely undergo infrastructure compatibility assessments or calculate how much it will cost to integrate voice systems. Although most challenges are resolved with tie lines between systems, establishing a common numbering plan and voice mail across multiple platforms is not straightforward and can be costly when changes are frequent.

?? Adding new features and functions to existing voice systems is an uphill battle. Voice teams are typically a skeleton crew that manage suppliers; feature/function enhancements require supplier lobbying to get the desired functionality scheduled onto product development cycles. If different voice systems are in place, the upgrading cost of these feature/functions compounds.

Our research indicates that many of these issues are commonplace for the majority of Global 2000 organizations, and it is unlikely that full-scale infrastructure replacement is an option, unless an aggressive return on investment (ROI) is tangible; however, even this option carries high risk. Instead, ITOs must determine how voice infrastructure will be rationalized, leveraged, and developed to realize the voice service technical architecture. An obvious first step is to create an equipment inventory and then standardize on platform types (e.g., preferred IVR, PBX). To maximize the return on this effort, we propose end-to-end, or telephone-to-telephone, voice infrastructure inventories (i.e., telephone, network, and switch) that are then organized by the characteristics they share (e.g., scale, functions, cost of ownership). The next step would be to rationalize these end-to-end inventories to create a smaller set, representing 70%-80% of all the desired characteristics for voice services. We propose to call these voice patterns.

The voice pattern will refine the inventory characteristics, document experiences, costs, management requirements, etc. The pattern also serves as an objective medium to ward off dissent within the organization related to supplier allegiances. The pattern also identifies service levels (e.g., availability, performance, scalability, ease of feature/function enhancements). Finally, a voice pattern will ultimately help ITOs respond to business needs more reliably and quickly while providing options on how value-added voice services are delivered (e.g., voice mail, auto attendant, IVR). For clarity, we propose that the starting point of any future voice services infrastructure development start with a pattern, not a vendor or a PBX/ACD.

Our current research indicates that there are seven different voice infrastructure patterns that ITOs should consider as starter kit in the development of their own set. This should eventually be reduced to a small and manageable set. For the purposes of simplicity, we have assigned only numbers to pattern types to avoid nomenclature debates. In addition, a CXI server and a unified communications (UC) server are shown in the diagrams but are not part of the pattern. The purpose of the CXI and UC server is to demonstrate that multiple voice infrastructure patterns can be supported by a common event-brokering platform (i.e., CXI) and a common communication platform (i.e., UC). More important, the CXI server becomes a building block to realize the voice services' technical architecture domains across multiple patterns; particularly for contact center users. Similarly, the UC server becomes the building block for rest of the enterprise.

Voice Infrastructure Patterns: A Starter Kit

META Group's proposal for the seven voice services infrastructure patterns will be described in high-level detail to establish clarity and the importance of the pattern (see Figures 4 and 5). Our research indicates that many companies have two to four different types in place, and with the migration toward IP telephony, there is a high risk that variants will stem off the current patterns to create more types that need to be managed. A full description of the patterns is necessary (see Figure 6) and should be the primary objective of every organization seeking to standardize voice services infrastructure.

Pattern 1

This pattern describes the classic PBX/ACD. These systems are 1-tier and monolithic, having all functions supporting telephony in one proprietary hardware appliance (e.g., trunk lines, extension lines, main processor unit, time division multiplexed [TDM] and pulse-code modulation [PCM] switching matrix, power supply, audio interfaces, call handling features). Trunk interfaces are connected directly to the PSTN, and extension interfaces are connected to proprietary telephone sets used by employees over a dedicated network (i.e., extension network). With the boom of call centers and advanced PSTN features (e.g., advanced intelligent networks, ISDN), PBX/ACD vendors commenced the march to "open" up appliances with CTI interfaces enabling advanced features to be controlled by an adjunct computer running software developed in a more common language. Many initially embedded these interface management processes with the appliances, but it proved difficult to maintain over time and resulted in creating CTI gateways running on "open" workgroup servers (e.g., Unixware, Windows NT/2K).

The challenges for this pattern escalate with the number of PBX/ACDs that are networked within a company. A major criticism is the high cost of changing the "voice network" and lack of interoperability with other supplier systems. Others have been scalability, cost of scalability, and insufficient processing power to support complex adjunct applications (e.g., predictive dialing).

It should be straightforward to create a laundry list of experiences to describe this pattern. It will also be the foundation of triggering discussions related to the limitations and desired requirements.

Figure 4 — Voice Services Infrastructure Patterns 1-4

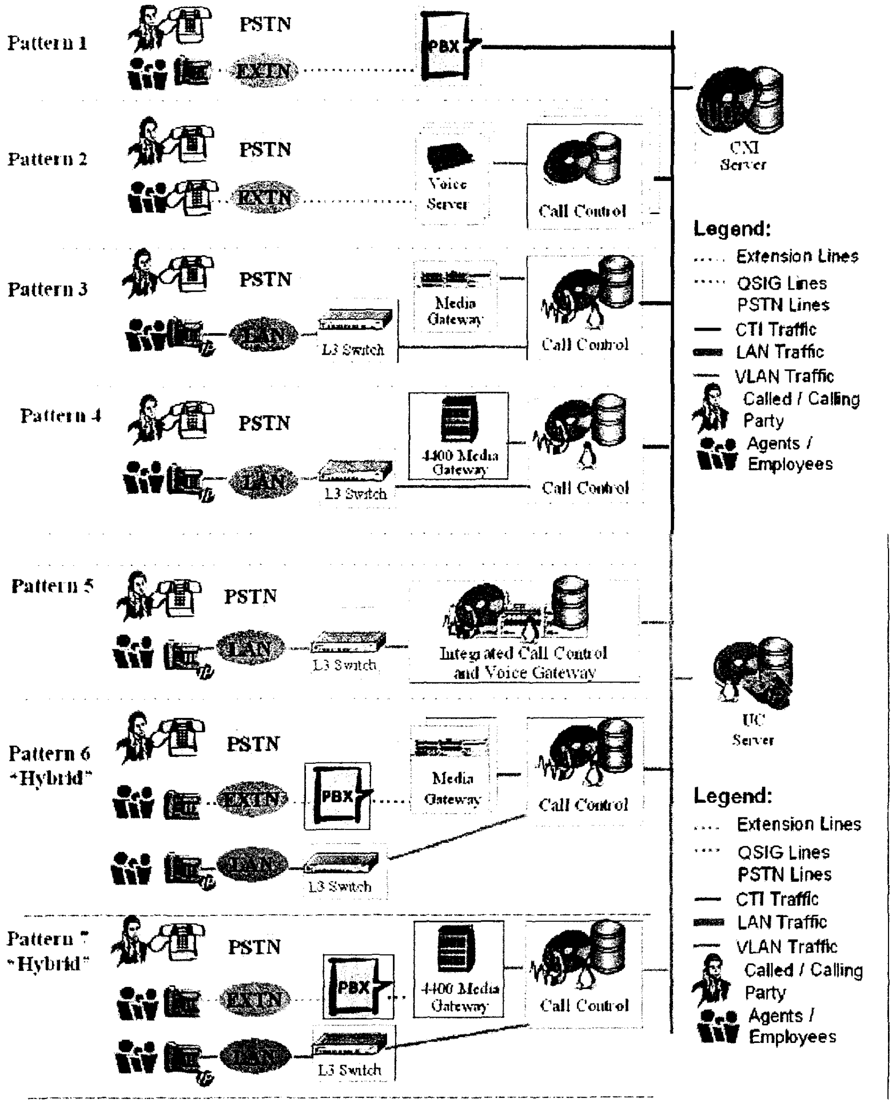
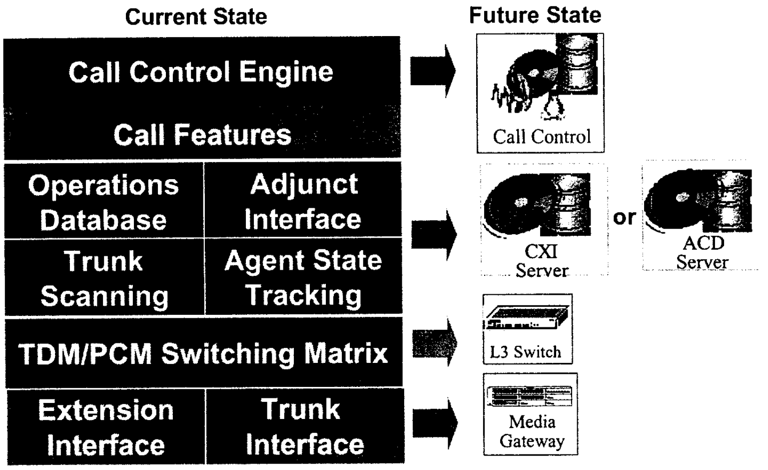


Figure 6 — Fundamental PBX/ACD Appliance Hardware Internal Functions



Pattern 2

This pattern describes the answer to voice services with open servers (e.g., Wintel, Lintel, Unix/SPARC). During the early 1990s, this approach was plagued with significant reliability problems because of the immaturity of the Windows OS and its ability to handle real-time events. This caused intermittent system operation, lack of high-availability features, unhardened ISA/PCI telephony card drivers, and vendors cramming too many processes (e.g., PBX, IVR, voice mail) in one box, leading the industry in general to characterize this approach as a mockery to telephony (e.g., "Unix is the only operating system for telephony"). As the next decade approached, a growing number of customers adopted this approach and reached acceptable production reliability levels. However, the major drawback to this approach is scalability, which decreases as the number of processes executed within the platform .

In this pattern, voice server(s) can be scaled out within a high-availability cluster to support many PSTN interfaces and extension telephones. The telephones are dumb 2500 plain old telephone service used to carry audio only, with call control features displayed on employee workstations for point-and-click navigation. The voice server can be analogously viewed as a Web server, responsible for the presentation of audio and HTML pages, respectively. VoiceXML pundits believe these two servers will work together more closely (but not merge) as Web content develops voice-enabled navigation (based on STT and speech recognition). ITOs wishing to "voice-enable the Web" would do well to develop experiences in deploying and supporting voice servers. Moreover, good experiences would eventually render this pattern a viable approach for low-scale voice service requirements.

The call control server is simply the "brains" of call control. In this pattern, it also has a Web server (not shown) that presents the HTML session (i.e., GUI application) to employee workstations for call control (e.g., call pick-up, hang-up, and transfer; three-party conferencing; call forwarding, call holding). Many vendors in this segment (e.g., Altitude, Interactive Intelligence, Concerto/CellIT) leveraged these servers and added a broad array of voice features (e.g., voice mail, UM, ACD, outbound dialing, quality monitoring, multisite networking, multiple-POI routing) while adding more processing power in high-availability clusters to deliver these services. This pattern is very popular with small and medium businesses because of the many features and functions that can be obtained at a low cost. The limitations to this pattern are that organizations will find many of the features and functions limited in comparison with best-of-breed suppliers, and they often do not work well when they are scaled out. Furthermore, this pattern does not integrate with CXI infrastructure used to support contact center interactions or low-volume, enterprisewide office productivity interactions (e.g., simple automatic outbound dialing). CXI services are actually part of the call control server extended features and functions. They are not as robust as those offered by industry leaders (e.g., Aspect, Avaya, Alcatel/Genesys) and should not be considered for enterprisewide deployments. Therefore, this pattern reveals that call control server-enhanced features/functions should be used with caution and awareness that islands of voice services infrastructure will permeate once these are used. Large organizations seeking to build single instances of voice services and leverage them enterprisewide will not find this pattern useful. Smaller companies (fewer than 200 employees), however, will find this pattern useful because of the many features/functions that are tightly integrated onto a single voice services infrastructure.

Pattern 3

This pattern depicts the "ideal" IP telephony platform, in the view of pure-networking pundits. The voice-enabled gateway has PSTN interfaces and coverts time-division multiplexing/pulse code modulation (TDM/PCM) voice to IP packets (VoIP) based on coding/encoding standards defining VoIP packet sizes (e.g., 5.6 Kbps, 8 Kbps, 64 Kbps). The VoIP packets are placed within a virtual LAN (VLAN) domain supported by Layer 3 or IP address switching and are sent to near-end and far-end parties to establish a session (or conversation) by the call control server. Once the session is in progress, the ongoing control of the session is done with a session protocol H.323 or SIP. Session protocol managers are also located on the call control server.

In this pattern, value-added voice services are delivered by appliance servers supporting H.323 or SIP (e.g., Web phone, UM, call logging, self-service "speech"-enabled applications). However, the products (as well as this pattern in general) are challenging to scale, and early implementations are revealing so at a rapid pace.

This pattern also eliminates the use of a telephone "extension" network within the enterprise, but it does require all telephone sets to be replaced by IP telephones. The pattern is currently most useful to support small remote branch offices and greenfield locations. Other uses of this pattern are based on a business case value proposition (currently, few cases are demonstrating ROI). Users should learn this pattern thoroughly, because it will be leveraged by other patterns to create hybrid scenarios.

Patterns 4 and 7

Pattern 4 is very similar to Pattern 3, with the exception of the voice-enabled gateway, which is a traditional PBX/ACD converted to a gateway. Most leading vendor products already support TDM-to-VoIP conversion and are already connected to PSTN interfaces. In this pattern, the call control functions are in a separate server and call control is no longer handled by the PBX/ACD.

This pattern will appeal to users who want to leverage existing switch infrastructure to reduce capital expenditures. Users should also negotiate with the PBX/ACD vendor for lower maintenance fees.

Pattern 7 is identical to Pattern 4, but it supports both existing telephone sets as well as IP telephones. We expect many companies with fully functioning switches to continue leveraging the existing infrastructure as long as possible and gradually migrate to IP telephony. The challenge of this pattern is that the call control server must be able to control the PBX/ACD's proprietary phones. This often predicates that the call control server be obtained from the same manufacturer that supplies the PBX/ACD, which in certain circumstances may not be very beneficial. An alternative approach is to control the telephone sets via a CTI interface with the CXI server, but this has yet to be proven as a successful approach.

Pattern 5

Pattern 5 is similar to Pattern 3, except that in this pattern, the voice-enabled gateway and the call control server are embedded into one common appliance hardware server (e.g., 3Com, Shoreline, Alcatel's OmniOffice, Avaya). In some vendor products, a small Layer 3 switch is also embedded into the appliance. This pattern is most useful for small branch offices or small companies seeking an all-inclusive packaged product. Although the benefits of a fully enclosed platform are attractive, users should ensure that the specific functions for the voice-enabled gateway, call control, and Layer 3 switching (when applicable) can be configured and managed as unique and independent components to avoid committing to a proprietary hardware appliance.

Pattern 6

Pattern 6 depicts a hybrid configuration supporting IP and proprietary telephones. This pattern also includes a voice server that is configured to be a voice-enabled gateway. This may be a better alternative to Pattern 4 or Pattern 7, because, in some cases, the expense of converting an ACD/PBX to a voice-enabled gateway may be cost-prohibitive or technically impossible, or the architecture policy might state that a particular PBX/ACD will be phased out of production and thus not be worthy of further investments. Alternatively, the technology component that performs the VoIP conversion could be an off-the-shelf voice-enabled gateway. The decision of which technology component to choose will be determined by the experiences of internal skills and value component provides to the pattern.

Pattern Definition Template

The pattern definition template is a recommended tool to be used to document the characteristics of an end-to-end voice infrastructure inventory that will eventually become one of the candidate voice patterns used by the ITO (see Figure 7). A list of characteristics that should be documented is listed below.

To simplify the documentation effort, users should leverage existing voice technology platform standards or policies that dictate which products are used for a particular purpose. For example, Figure 7 has the following tiers: telephone devices (e.g., IP telephones, Web-phones), networks (e.g., PSTN, PBX/ACD telephone extension network, LAN/WAN), voice-enabled gateways, and call control. Any information from standard platforms should be used to develop the end-to-end pattern. However, platform characteristics should not dictate end-to-end patterns rather than simply the direction of infrastructure developments.

?? ***Functionality:*** A measure of the breadth of features and functions, and the ease of which new ones can be added.

?? ***Performance:*** A measure of the consistent operation for all features and functions within expected service levels (e.g., getting an outside line 100% of the time).

?? ***Scalability:*** A measure of how many users can be served with the same level of performance for all features and functions as well as the speed that production worthiness is renewed after expansion.

?? ***Availability:*** A measure of the outages, planned and unplanned, caused by upgrades, failures, alarms, and other service impairing events.

?? ***Cost of ownership:*** A measure of the costs to support and maintain service inclusive of maintenance licenses, external support services (e.g., MAC), upgrade costs, third-party network costs (e.g., telephone monthly rentals, usage charges), internal staffing, internal support processes, and systems.

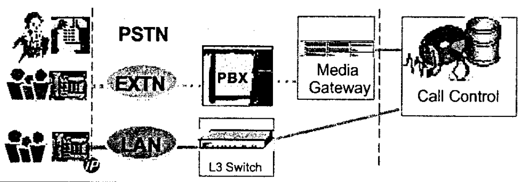
Distributed Enterprise Voice Services Infrastructure Needs a New Language

One of the challenges ITOs will face is identifying the right infrastructure development strategy to take to deliver enterprise voice services. Before arriving at this decision, ITOs need to establish an internal language that enables voice- and data-skilled personnel to clearly communicate the pros and cons of infrastructure design. Making decisions about what designs vendors put forth or labeling designs after vendors will not help IT groups communicate better (and instead will polarize groups into vendor camps, decreasing productivity). Ultimately, this gives too much power to vendors as IT staff members seek vendor-differentiating features to beat the other camp. Moreover, it places ITOs at a negotiating disadvantage because ITOs are busily trying to figure out the details of the vendor design (vs. the vendor understanding user requirements more completely).

We believe ITOs would avoid unproductive internal battles and maximize negotiation power when voice and data teams are able to jointly agree on voice patterns. Debates among staff members could be conducted more constructively, enabling more focus on meeting end-user communication requirements.

Figure 7 — Pattern Definition Template

Pattern 6
Hybrid



Functionality - UM, speech,					
Performance - reliability					
Scalability - integrate-ability					
Availability					
Cost of Ownership - usage, IN services, internal and third-party support, MACs					

Capital Cost Reductions

Establishing voice services architecture and infrastructure enables ITOs to be more selective about buy-vs.-build decisions. Historically, vendors packaged products onto a 1-tier platform and were confined by TDM/PCM architectures, proprietary operating environments, and processing power limitations. This technology generation required multiple products and extensive custom integration to realize the future architecture of enterprise voice services. Moreover, many functional domains would not even be possible to develop. However, current voice technology transformations promise to increase the likelihood of developing a cohesive voice services infrastructure. High-scale performance would still be great challenge, but current technologies are benefiting from higher performance servers and networks, enabling voice service infrastructure to be reliably distributed across multiple appliance servers and technology components, greatly closing on the scalability gap.

Many voice technologies are simultaneously transforming and creating situations where functional overlaps occur between product types, while newer technologies are offering new ways delivering voice services previously found in a standalone system. For example, there is technology overlap between UM and CXI suites and the potential of speech-enabled voice application servers to replicate voice mail system functionality.

CXI and Unified Messaging

Some organizations with CXI suites in contact centers have been able to extend the platform into the enterprise to handle standard telephony for personnel not in the customer interaction center (CIC). These companies integrated CXI with various PBX/ACDs, the Web site, fax, and the email platform. UM integrates to various voice mail systems, fax, and e-mail platforms. The question then becomes whether to buy another platform for UM, to leverage existing CXI infrastructure by integrating missing functions to CXI, or to push the vendor to develop the functionality. The majority of vendors will resist this merger of platforms to preserve their own interests (i.e., they wish to sell two platforms), though some smaller competitive platforms currently offer a single code base for both functions under a single platform. Users should seek opportunities to consolidate as much as possible and establish economies of scale to support contact center and enterprise voice services.

Speech-Enabled Voice Application Servers and Voice Mail Systems

Speech-enabled VASs are offering the ability to create voice scripts with ease as well as record and store audio. Moreover, audio can be stored within existing corporate network-attached storage or storage-area networks as retrieval times improve. Some large organizations are already designing first-generation VAS platforms as replacements to voice mail systems. This provides not only the elimination of purchasing voice mail systems, but also the ability to leverage existing enterprise storage network infrastructure.

Architecture Reveals Infrastructure Savings

Cost savings in IT spending are one of the primary results of developing a technical architecture. The process of developing voice services architecture will reveal many areas where technology overlaps and replacements can result in infrastructure capital and operational expenditure savings. If the effort to accomplish these savings is too great (e.g., integration costs, non-standard interfaces), ITOs should postpone these developments and review this opportunity at a later date or work with viable integrators and vendors that could help complete the integration.

Leveraging Existing Assets

Completing voice pattern definitions will identify which infrastructure design makes sense for a particular user environment and what voice technology will remain in place. The most common reuse of existing voice technology is the repurposing of PBX/ACDs as voice-enabled gateways, enabling support for existing as well as IP telephones. This can save organizations a significant amount of capital expenditure and provide a lower-risk migration path toward an adaptive corporate standard voice service infrastructure.

Reducing Total Cost of Ownership

A single voice service infrastructure, though it will be highly distributed, will yield a lower TCO due to savings in service provisioning, changes, configuration, and problem management. Many vendors are espousing the benefits of IP telephony and are able to show ROI. However, users should take a long-term view of TCO management and not consider this a one-time effort (e.g., buying all new technology) or annual effort (e.g., negotiating lower maintenance and support costs), but an ongoing process. This process seeks to identify how and when investments could be made to get closer to the ideal architecture that promises overall reduced spending. Infrastructure planning and clear communications are key benefits of using voice patterns and the process of defining patterns and developing them. This process provides information and insight on how numerous voice technology components could be streamlined to fewer ones, while ensuring that quality voice services are delivered to the business.

Dictating Enterprise Voice Services Requirements

ITOs must begin to take more control of voice infrastructure and challenge vendors with clear requirements. Significant technology transformations and vendor uncertainty exist in the industry, permitting ITOs to actively participate in molding vendor product developments. Currently, many vendors are building modularity into their product portfolio, enabling ITOs to select the desired components. ITOs must take an end-to-end view on voice services and ensure vendors are well aware of corporate infrastructure designs, the dependencies on different technology components, messaging protocols, performance expectations, and a strict demand for interoperability across vendors. Technical architectures should set the governing guidelines for what future voice technology component purchases and projects should be. They should also ensure that corporate investments are only made in support of technical architecture realization from ongoing infrastructure development initiatives, leveraging voice patterns and internal IT skills.

Addendum A: Voice-Oriented Services Technical Architecture Domain Functional Description

Voice Interfaces

This domain terminates the physical layer connectivity to different communication networks (i.e., PSTN, proprietary PBX/ACD telephone extension network, WAN, LAN, wireless network, and the Internet). Organizations will need to support many of these networks with a few consolidating networks when possible, but only if enterprise infrastructure complexity can be reduced.

Voice Control

This domain is responsible for the call management from the moment the incoming call enters the enterprise space. Management functions include call receipt, transfer, hold, three-party conferencing, forwarding, dialing digits, redial, call termination, and basic call routing to an internal phone. There are many more call control features that manufacturers provide, but essentially the bulk of call control features can be defined in the aforementioned function set. Organizations will ultimately need to support IP and TDM call controllers; therefore, it is essential that both controllers are interoperable.

Filtering and Tracking

This element is the first port of call for all interactions and requires filtering logic to determine what gets tracked. The business rules to determine what gets tracked and how the filtering will get done require cost/benefit analysis to determine which employees will use voice services. For the CIC, the cost/benefit decision is typically made when purchasing an IVR unit, providing caller segmentation and customer identity determination. For employees, the IVR approach is an unlikely solution because customers have direct-dial access. A solution would be to track every call that enters the PBX, but this will not be cost effective for all organizations, requiring business units to determine whether voice services are a viable expense for the value gained.

Technically, filtering and tracking are high consumers of processing power and memory. Essentially, every interaction gets tagged and initiates a state machine to record the lifetime of the interaction, locations visited (see below), capturing relevant information in a container per location, maintain third-party application sessions (e.g., CRM screens) active, and evolve the state of the interaction for the purposes of setting alerts for user and system handling.

Operationally, this element must be monitored by service management "agents" to capture and report events to a service management platform (e.g., from Aprisma, HP, and Tivoli). There will be a need for automated alert resolution by allocating more processing power or memory to tracked interactions. This will be essential as users may place customers on hold and forget about them, or for phone calls that may take several hours to complete, as in the case of a conference call.

Queuing

This element provides a virtual funneling of all interactions for all internal employees to a single point where a business rule can be applied to each interaction before it is routed. The queue is essential to buffer and put interactions temporarily on hold before other communication infrastructure service elements are ready to process the interaction further.

Multiple queues must be set up for different POIs, dependent on lines of business (LOB) to be served. This will enable the ITO to provide different grades of service to the LOBs and allow operational to monitor the performance of business service queues.

Prioritization

This element enables different LOBs to set prioritization rules on how customer segments get treated, potentially bumping higher-priority customers up the queue. This requires integration with CRM and other back-end data stores (see "Process Automation") to extract relevant customer, product, and previous interaction information to determine customer "value" to the business, and correspondingly make the business-rule routing decision.

Authentication

This element leverages the use of keypad entry (e.g., phone pad), caller identification, or voice verification technologies to establish the identity of callers. In an ideal situation, customer identity should be identified before a call is routed, but this is not practical considering the apprehension of many market cultures, and many LOBs do not have the need for this. However, when applied in the CIC or for a specific LOB, the customer can engage in a more meaningful interaction with the business because the identity would be used to initiate the rendition of other communication services to address customer needs quicker and more effectively. An example would be high-value customers of banking services, who have dedicated account management teams and demand superior service.

Routing

This element ensures interactions are delivered to the appropriate resource in the enterprise. Current technologies enable routing to extend to geographically distributed locations (e.g., intelligent network routing, prerouting), enabling interactions to be delivered to the current employee's workplace (e.g., office, home, other office). A key requirement for this type of routing is that it must also maintain interaction state as it gets transferred from one employee to another, while maintaining POI session and CRM application states such that the next employee can continue from the last touch point. Similar to tracking and filtering, this element is a high consumer of processing power and memory.

Speech Artificial Intelligence

This element is based on use of speech engines (i.e., recognition, TTS, STT) and speech application markup languages (i.e., VoiceXML, SALT). "Speech artificial intelligence" (SAI) is a catch-all expression to provide voice-activated self-service navigation for a browser or speech-only application. The CIC examples are relatively obvious, but the employee example requires some explanation. The SAI can, for these employees, engage the calling party with a dialog related the employees' availability (i.e., TTS: "busy for the next few hours"), provide them with options to leave a message (i.e., voice mail), get service from a viable alternate (e.g., "Would you like to speak with ...?"), or request the resource mediator (see below) to locate the employee. The key contributor to the dialog would be the employees' calendar, ultimately requiring a user interface for the employee to establish some basic rules to handle different customer types. Thus, in a sales example, a customer calling with high purchasing power should be treated differently from others.

The challenge facing ITOs will be to ensure that speech engines can meet the scalability and high-availability requirements to service the enterprise from one platform. Furthermore, the centralized platform supporting geographically distributed locations will require QoS and application prioritization over a WAN to minimize latency, which may result in high networking costs.

Interaction Control

This element provides applications (e.g., CRM) with control of the POI sessions that are routed to the user. The POI sessions can be accepted, completed, transferred, shared with colleagues and trigger follow-up interactions, etc. Basic POI session control is currently available within the technology alliances of CTI suites and CRM application vendors. Many CTI suites are offering software development kits to place interaction control on other customer-facing applications, ideally browser front ends to maintain low impact on current IT standards for workstation builds.

Process Automation

This element requires the integration of CTI suites and business process automation (BPA) software[3]. The combined systems will provide opportunities to reduce the number of manual handoffs between front and back offices by automating the workflow between the two. Routing decisions are also enhanced with BPA because it can source relevant customer profile information from back-end data stores, providing dynamic customer prioritization capabilities.

For front- and back-office process automation, it is important to establish that POI session management is a resource- and computing-intensive environment that requires real-time, connection-oriented, and synchronous interaction management. Currently, CTI suites are meeting this requirement and provide organizations with assurance that new self-service POIs can be added with an escalation path to a customer service representative, while maintaining the POI session state during the transfer.

Metrics

This element is the probably the most important, and it is one of the main reasons communication services architecture and infrastructure are relevant. For all tracked interactions, it is important to obtain a measure of interaction handling times correlated to customer interaction case histories. Key metrics for consideration should be first-call resolution, customer "case" resolution costs, and total cost to service customer inquiry caseload.

About META Group

META Group is a leading research and consulting firm, focusing on information technology and business transformation strategies. Delivering objective, consistent, and actionable guidance, META Group enables organizations to innovate more rapidly and effectively. Our unique collaborative models help clients succeed by building speed, agility, and value into their IT and business systems and processes. Connect with metagroup.com for more details.

The ROI of Alcatel IP Communications Solutions

Business Development
Large Enterprise Segment

ARCHITECTS OF AN INTERNET WORLD
ALCATEL

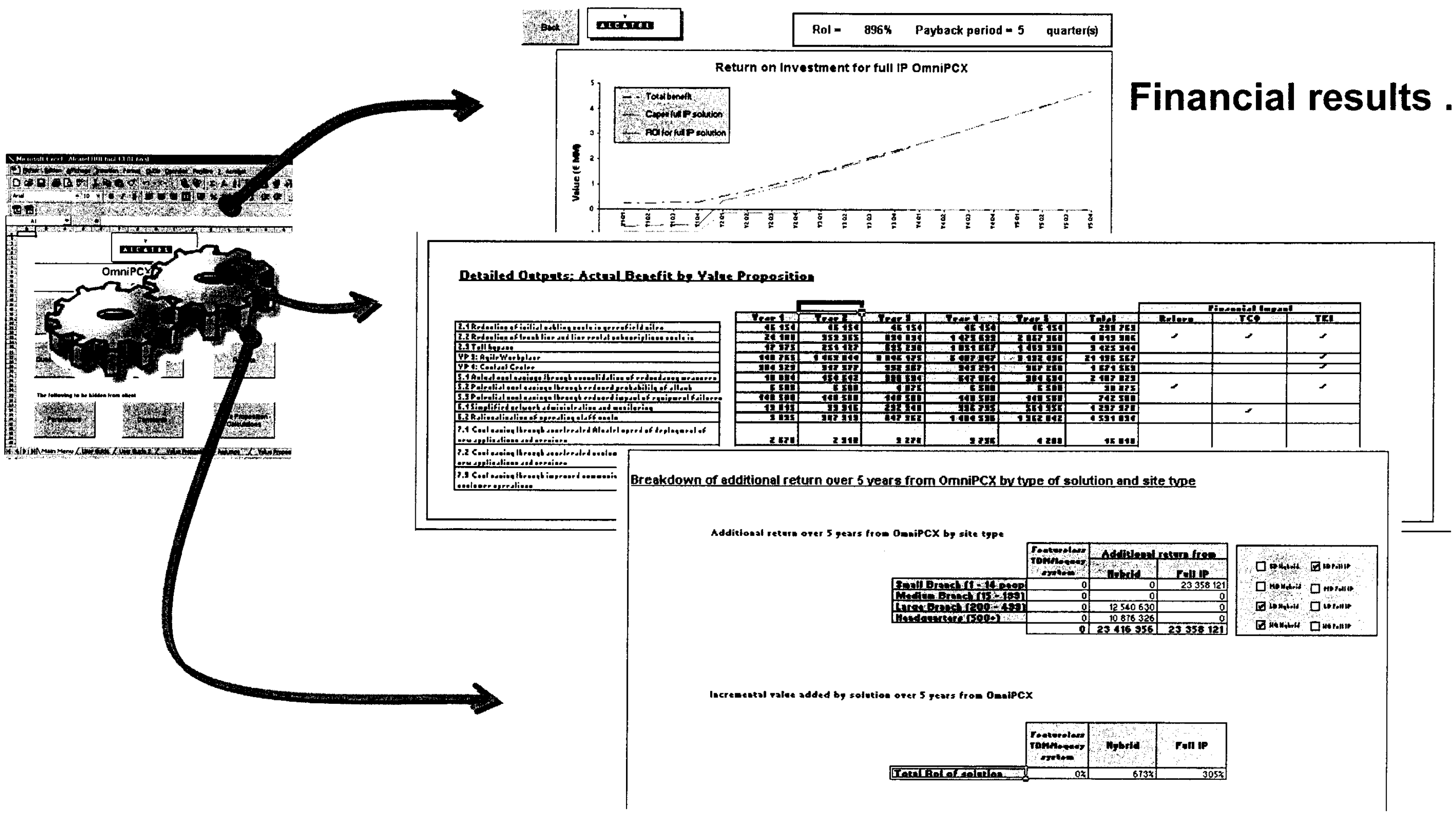
Value Propositions of Alcatel IP Communications solutions…

- Architectural flexibility
- Intelligent networking
- Highest reliability
- Simplified management
- Agile Workspace
- Customer Interaction

The ROI Model =
Measuring the Economic Benefits of 6 Value Propositions

An ROI Calculation Tool Outputs

Financial results ...



Agenda

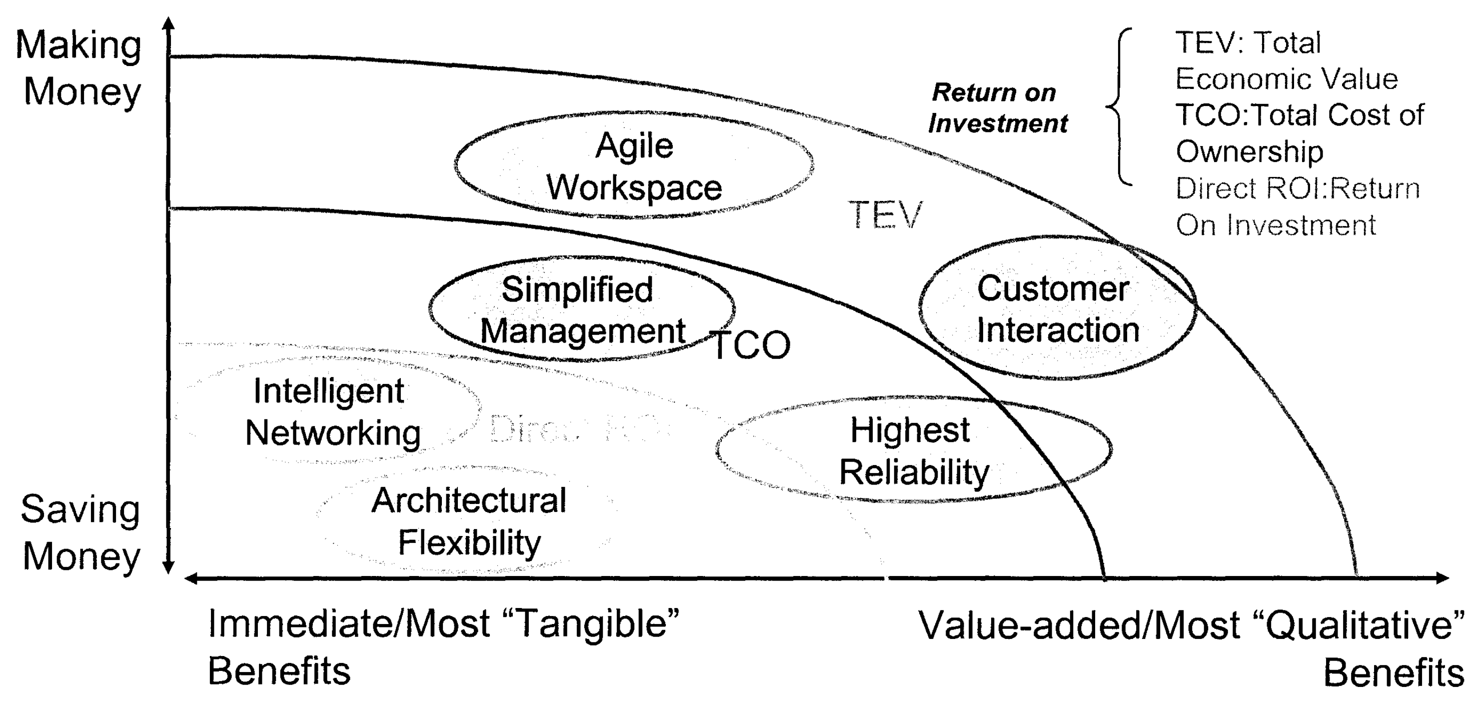
- Defining the ROI Methodology
- The ROI Calculation Tool / Process
- Some key learnings of initial ROI calculations
- Real Case Examples

ALCATEL

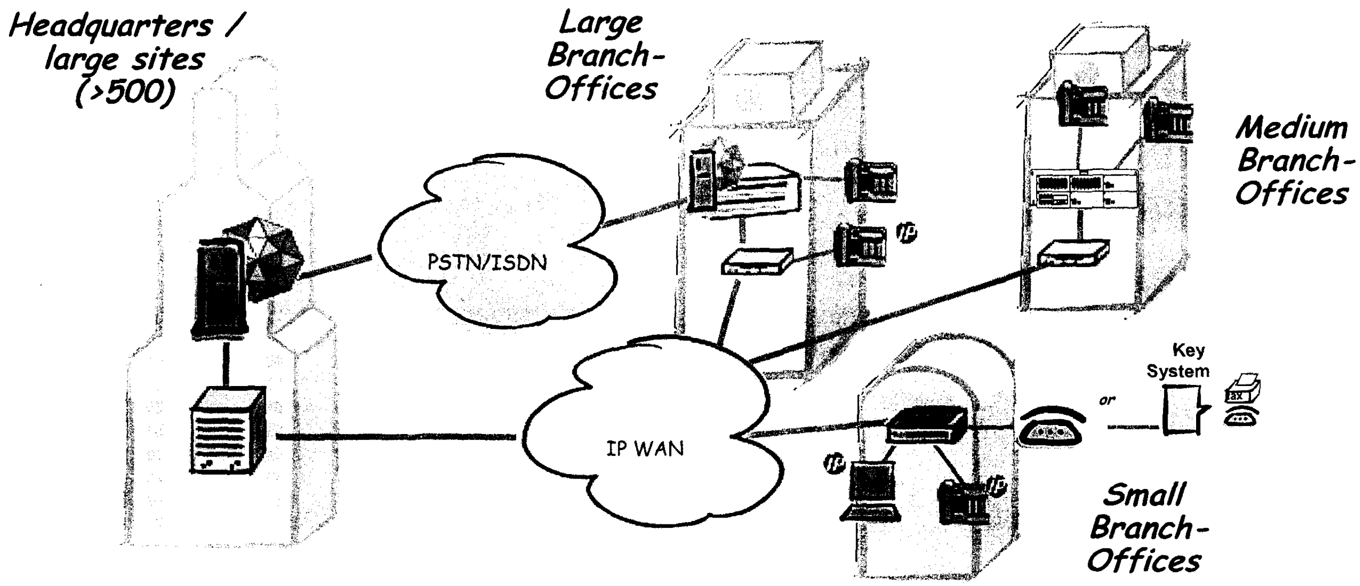
The Three levels of ROI to measure value

- Direct ROI ie Purchase cost savings
- Savings on Total Cost of Ownership
- Savings on soft costs, improvements in productivity and increase in revenues defined as Total Economic Value

Mapping the Value Proposition on the three levels of ROI...



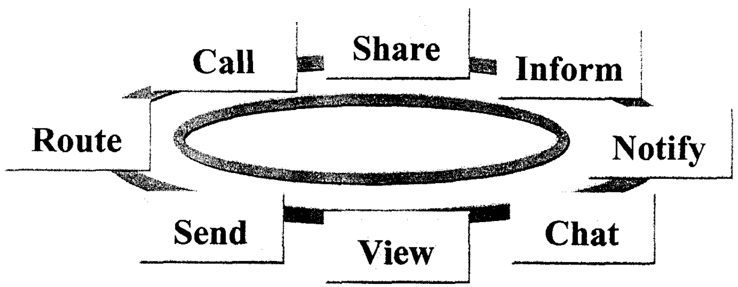
Mapping the Infrastructure VP to the ROI model



A Flexible Model : various scenarios (from pure IP to pure TDM) in multisite networks modelled from large sites to small branch...

Mapping the Application VP to the ROI model

- **A flexible model including end user productivity scenarios depending on :**
  - the type of application used,
  - the user profile,
- **Based on real cases,**
- **From a set to the complete application suite**



Direct ROI – Purchase Cost Savings

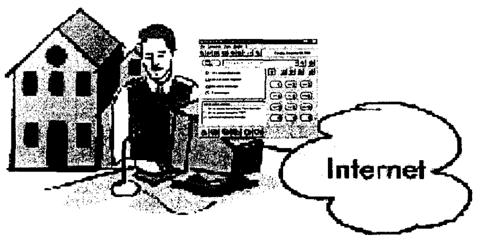
- **Cost savings on traffic through intelligent networking**
  - Cheaper calls between company sites through shared bandwidth
  - Cheaper calls outside company sites : long distance, GSM, Least Cost Routing, even from the smallest branch offices
    - Centralization of subscriptions
    - Access to optimized tariffs
  - Remote and nomadic access to local call number
- **Cost savings on wiring for green field opportunities**
- **Investment protection: through architectural flexibility**
  - Partial, progressive and phased deployment
  - Legacy terminals and applications integration

Direct ROI – Purchase Cost Savings

- Direct costs savings on Telecom bill due to new IP applications for employees
  - VoIP teleworking at home : Save 3,4€ per day
    - internal calls for free with IP Softphone on ADSL connection
    - consult your voice messages for free
  - Reachability for VIP incoming calls when on the move
    - don't recall your voice mail and the caller

Save 0,5€ per day

VoIP remote working

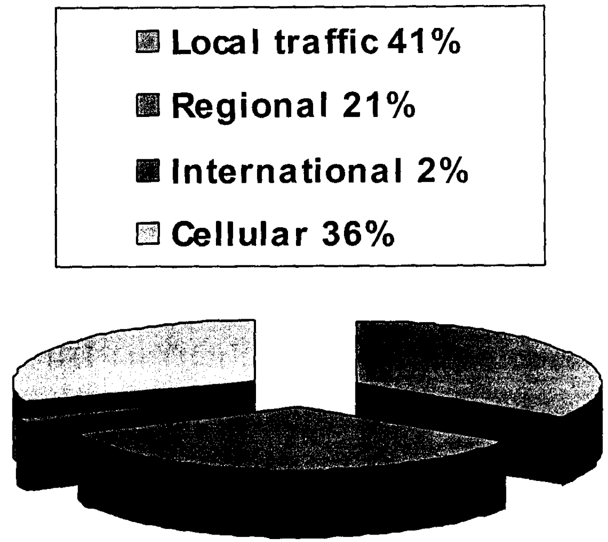


Reachability for VIP calls

ALCATEL

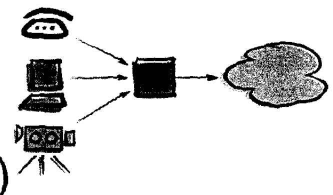
Direct ROI - savings assumptions from real cases

- **Impact on annual branch telecom bill**
  - Centralization of GSM network access enable savings in PSTN to cellular
    - possible cost savings of 10%+
  - Sharing access to the public telephone network across sites:
    - possible cost savings of 15% + on PSTN charges



Total Cost of Ownership : systems and network

- **Costs savings on operating costs**
  - Centralization of maintenance contracts / upgrades
  - Easier moves, add and changes
  - One staff for network administration of all sites
  - Integrated system management (voice / data)
  - Resources optimization (centralized operators)



- **Reduction in Business operations discontinuities**
  - Reduction in failure rate thru security protection and enhanced availability / reliability

Total Cost of Ownership : IP applications

- **Reduced training costs**
  - easy to use IP applications
  - using standard user interface
    - Browser, Voice, Microsoft Outlook, Lotus Notes

- **Minimized support, deployment and maintenance costs**
  - unified log-on
  - thin client. No client software on user desktop
  - single point of message storage

- **Minimizes integration costs**
  - portlets and XML web services technology

Total Cost of Ownership - savings assumptions from real cases

- **Mutualize expenses: equipment upgrade, operation, staff expenses**
    - Estimated minus 5 to 20%
- **Lowering adds, moves and changes costs**
    - Estimated MACs (minus 20 to 40%)
    - Cabling yearly additions (minus 30 to 60%)
- **Integrated management**
  - Reduces software and staff costs by 10 to 40%
    - Due to improved control and traffic analysis, enables team to optimize and reduce telecom bill by 5%

Total Economic Value : the value of IP applications

- **New communication services leading to productivity improvement**
  - Openness for Improving business agility and connecting the knowledge
    - Managing real-time communications
    - Unified messaging
    - Personal routing engine
    - Collaborative working
    - Universal Directory Access

- **Enhancing/optimizing the company Customer service/interaction**
  - Distributed and scalable Contact center
    - Staff optimization
  - Multimedia Contact center
    - Improving customer satisfaction / Increasing revenue opportunities

Agenda

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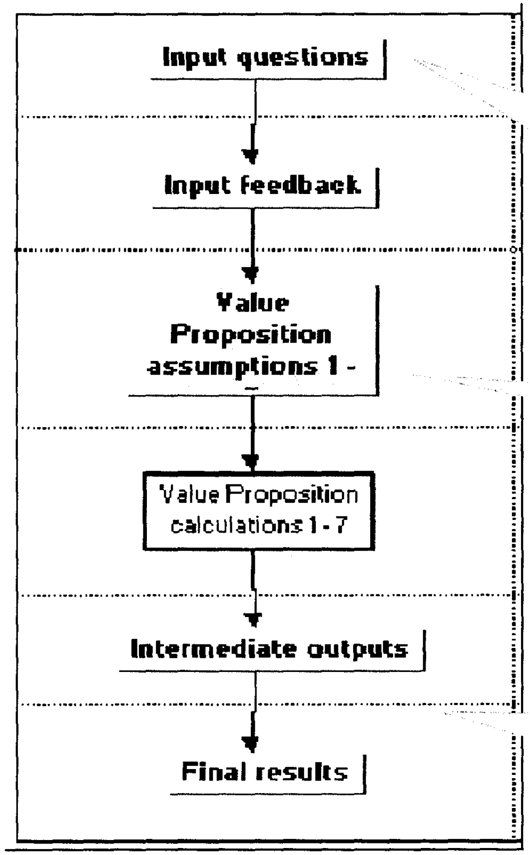
Alcatel OmniPCX ROI Tool
The process for using the tool



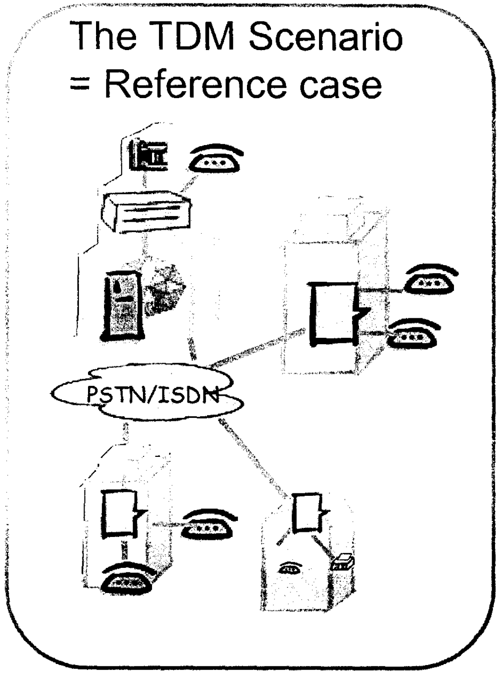
Inputs from the customer, collected in a customer face-to-face meeting :
25 questions
on telecom bill, capex, opex,
company staff typology (default values,…)

Assumptions based on Alcatel experience,

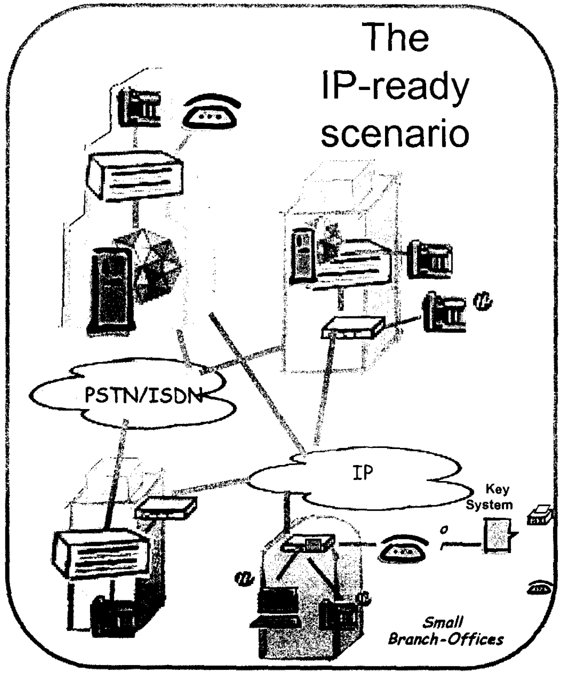
Financial impact of a partial or total IP migration of the customer's network.

Tools to lead of a detailed discussion + try diverse scenarios and assumptions

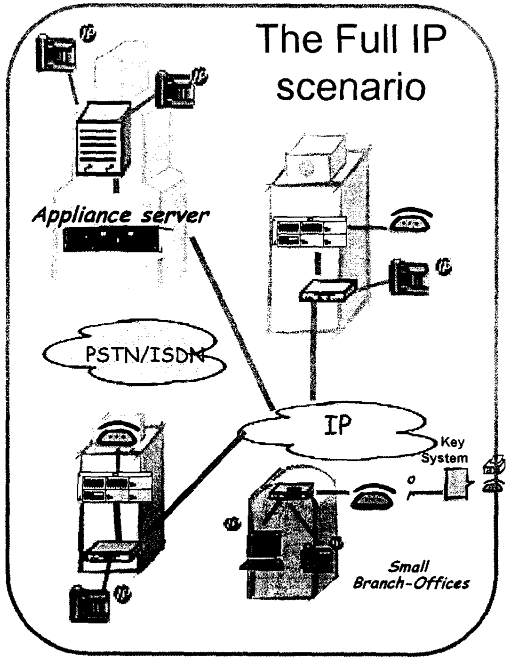
Three scenarios can be tested...



4400 R <4.0
Legacy PBX

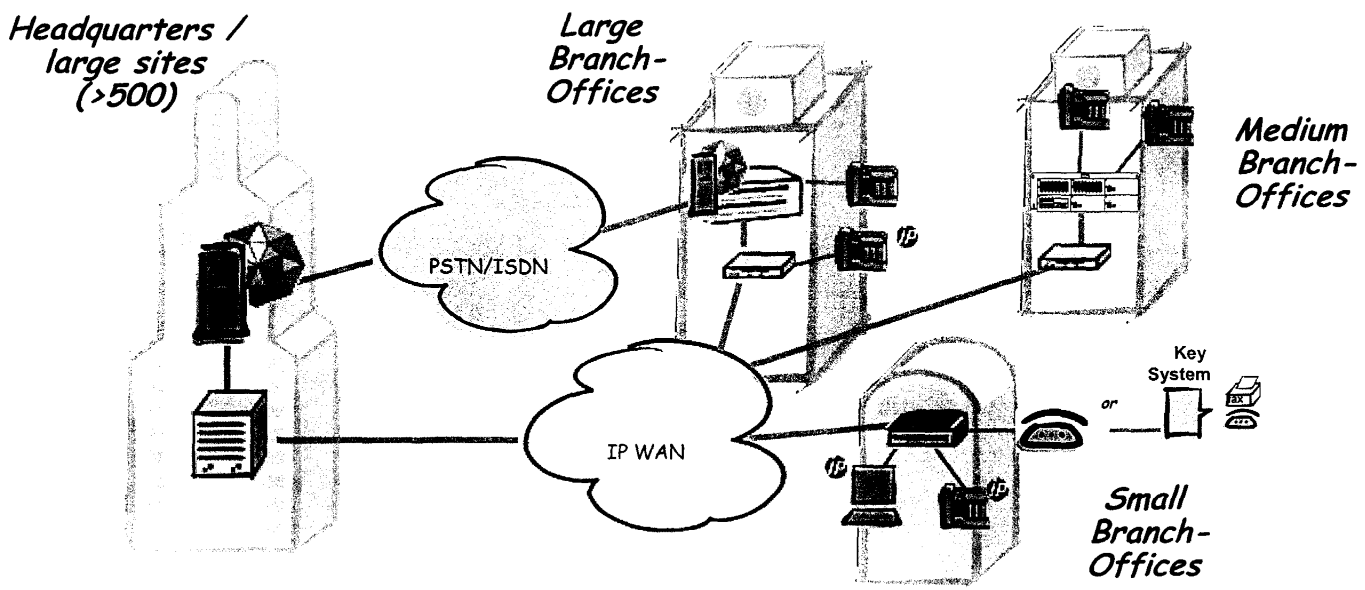


4400 R 4.x up to 5.0 Ux
Advanced PBX



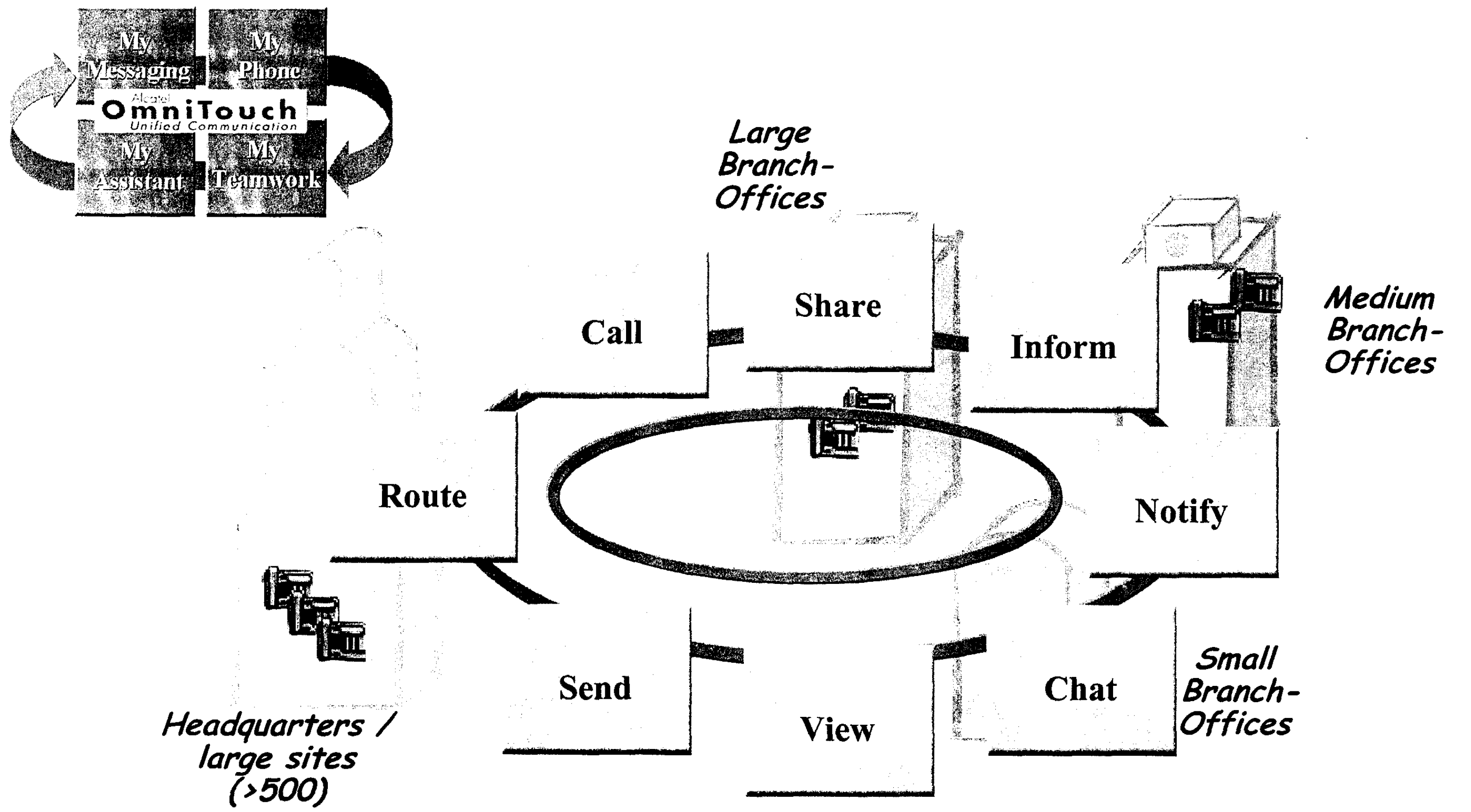
Enterprise

...across various corporate topologies...



A Flexible Model : various scenarios (pure IP to pure TDM) in multisite networks modelled from large sites to small branch

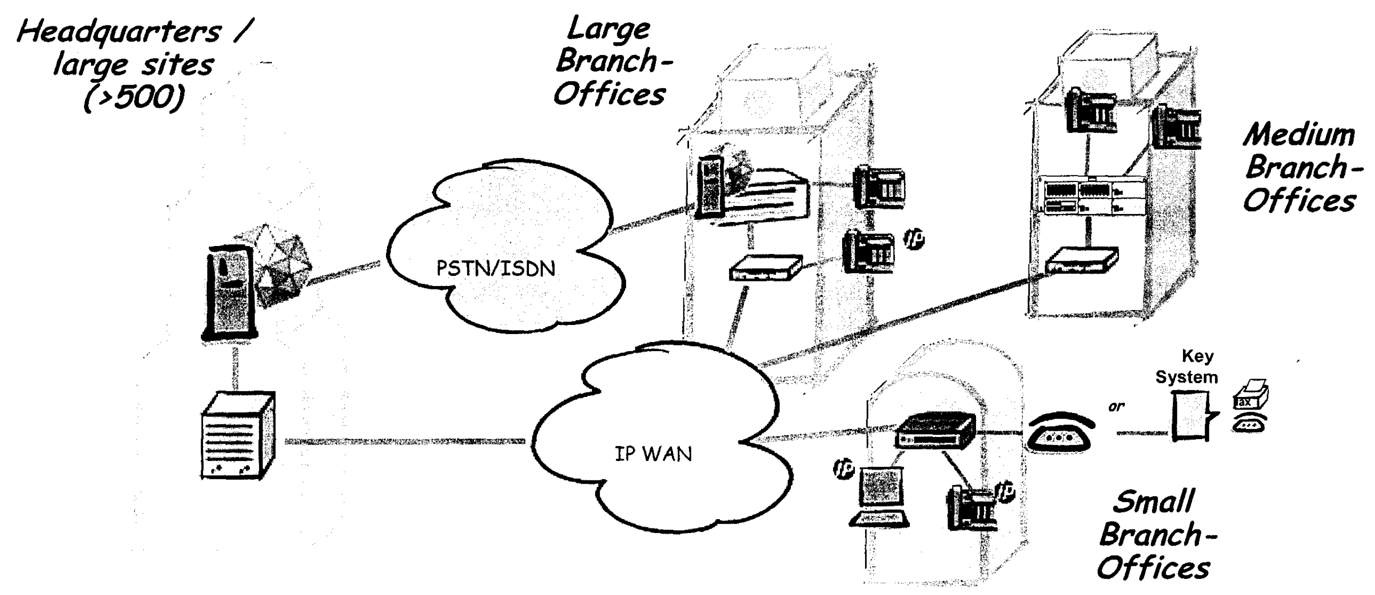
...with different user needs and profile



Agenda

- Defining the ROI Methodology
- Some key learnings of initial ROI calculations
- Real Case Examples

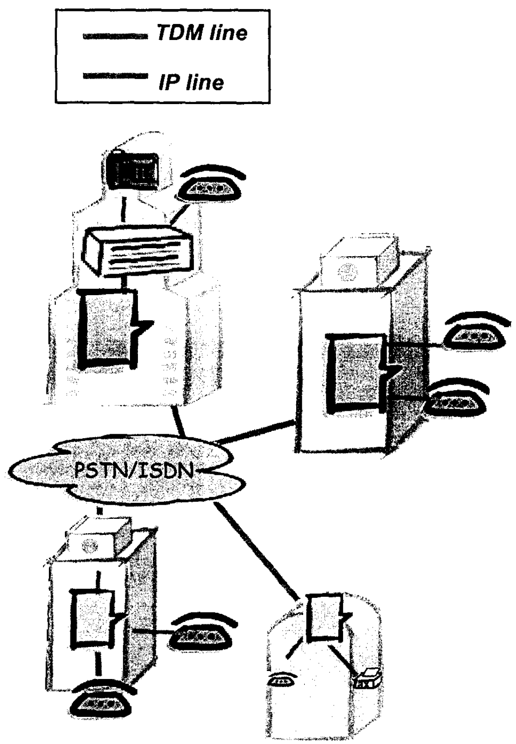
Example of Large Retail Bank



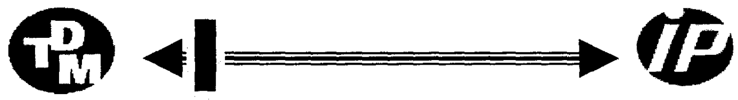
60 000 employees : 50% in headquarters and large sites and 50% spread evenly across branch office

And different application needs

The TDM reference scenario



Set as the Reference Case
All Values at 0 (investment and return)

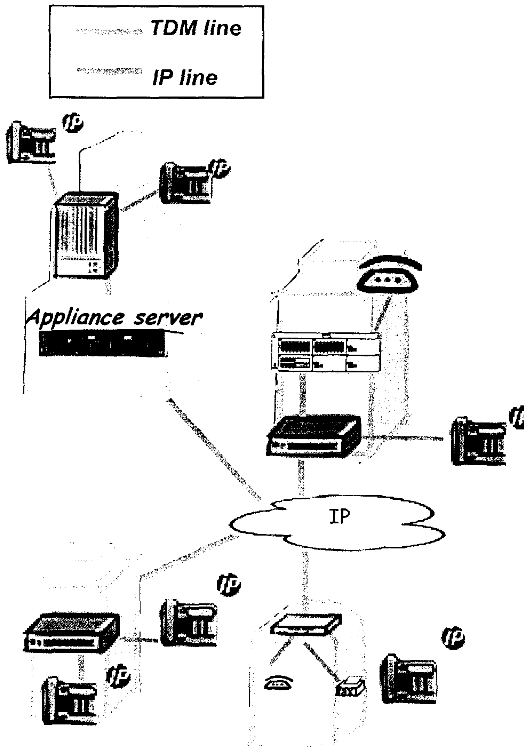


	Featureless TDM/legacy system	Additional return over 5 years	
		IP-Ready	Full IP
Small Branch (1 - 14 people)	0	0	0
Medium Branch (15 - 199)	0	0	0
Large Branch (200 - 499)	0	0	0
Headquarters (500+)	0	0	0
	0	**0**	**0**

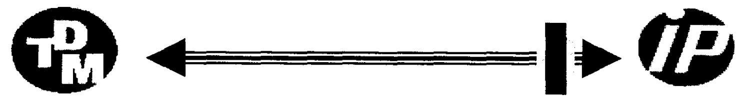
☐ SB IP-Ready ☐ SB Full IP
☐ MB IP-Ready ☐ MB Full IP
☐ LB IP-Ready ☐ LB Full IP
☐ HQ IP-Ready ☐ HQ Full IP

		Full IP Scenario
Achieved return :	0	*34 140 381*
% vs Full IP	*0,00%*	
Required Investment	0	*8 507 447*
% vs Full IP	*0,00%*	

The Full IP scenario



100% of potential benefits for 100% investment



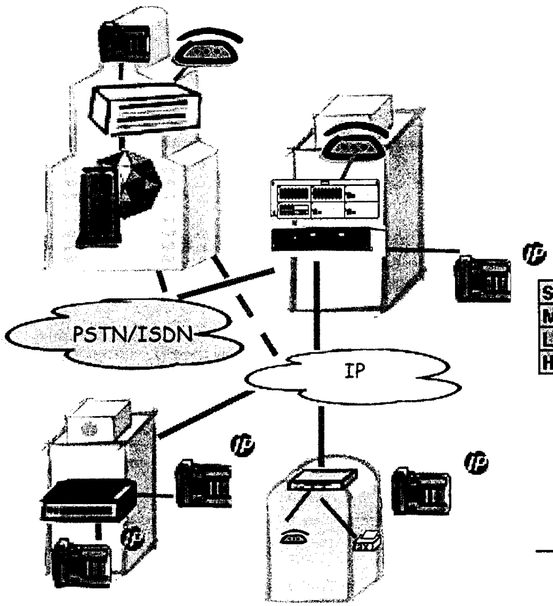
	Featureless TDM/legacy system	Additional return over 5 years	
		IP-Ready	Full IP
Small Branch (1 - 14 people)	0	0	9 532 042
Medium Branch (15 - 199)	0	0	10 846 483
Large Branch (200 - 499)	0	0	10 181 771
Headquarters (500+)	0	0	3 580 084
	0	**0**	**34 140 381**

☐ SB IP-Ready ☑ SB Full IP
☐ MB IP-Ready ☑ MB Full IP
☐ LB IP-Ready ☑ LB Full IP
☐ HQ IP-Ready ☑ HQ Full IP

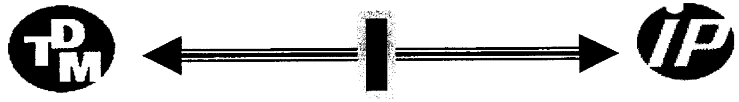
		Full IP Scenario
Achieved return :	34 140 381	*34 140 381*
% vs Full IP	*100,00%*	
Required Investment	8 507 447	*8 507 447*
% vs Full IP	*100,00%*	

Partial 'IP A la carte' Scenario

TDM line
IP line



70 % benefit for 30% investment – Minimize Risk !
Extracting the maximum value from partial IP deployment



Additional return over 5 years from OmniPCX by site type

	Featureless TDM/legacy system	Additional return over 5 years	
		IP-Ready	Full IP
Small Branch (1 - 14 people)	0	0	9 532 042
Medium Branch (15 - 199)	0	0	10 846 483
Large Branch (200 - 499)	0	5 707 353	0
Headquarters (500+)	0	0	0
	0	**5 707 353**	**20 378 525**

- ☐ SB IP-Ready ☑ SB Full IP
- ☐ MB IP-Ready ☑ MB Full IP
- ☑ LB IP-Ready ☐ LB Full IP
- ☐ HQ IP-Ready ☐ HQ Full IP

		Full IP Scenario
Achieved return :	26 085 878	*34 140 381*
% vs Full IP	*76,41%*	
Required Investment	2 933 545	*8 507 447*
% vs Full IP	*34,48%*	

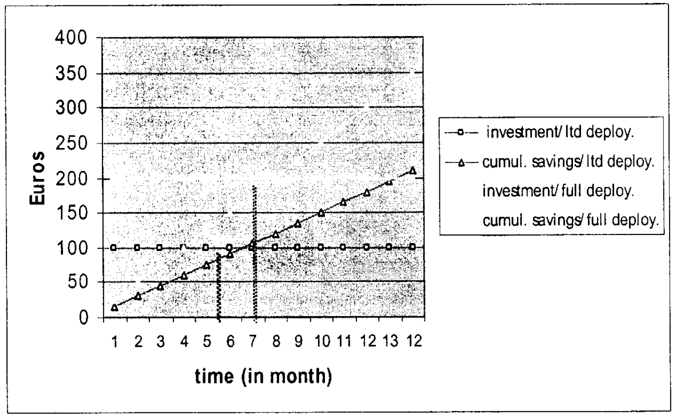
ALCATEL

Employee Productivity : 'a-la-carte'

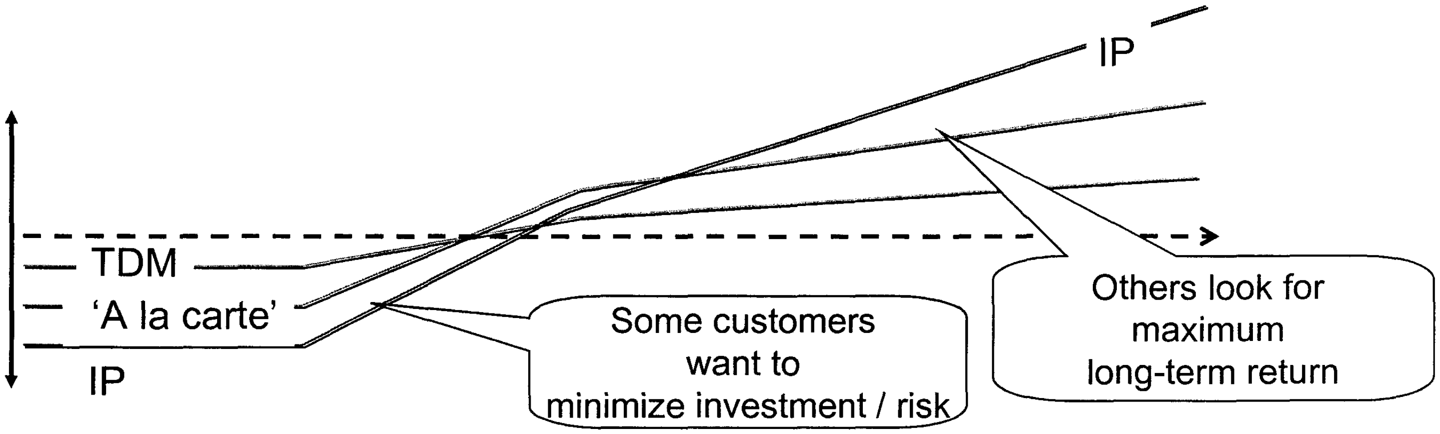
- **Important variables of ROI calculated on individual time savings are**
  - Traffic : average number of phone calls and number of eMails made/received
  - Employees Paterns : average salaries, size of mobile or remote workforce
  - Number of employees to equip : ROI is stronger when initial costs are reparted
- **It will lead to the decision to deploy a part or all applications of the suite to all or a part of the employees**

> ROI example : limited vs full deployment

- In this case, although ROI is quicker with a limited deployment (6.6 months) than with full deployment (8 months), savings of full deployment exceeds those of limited deployment after 10 months

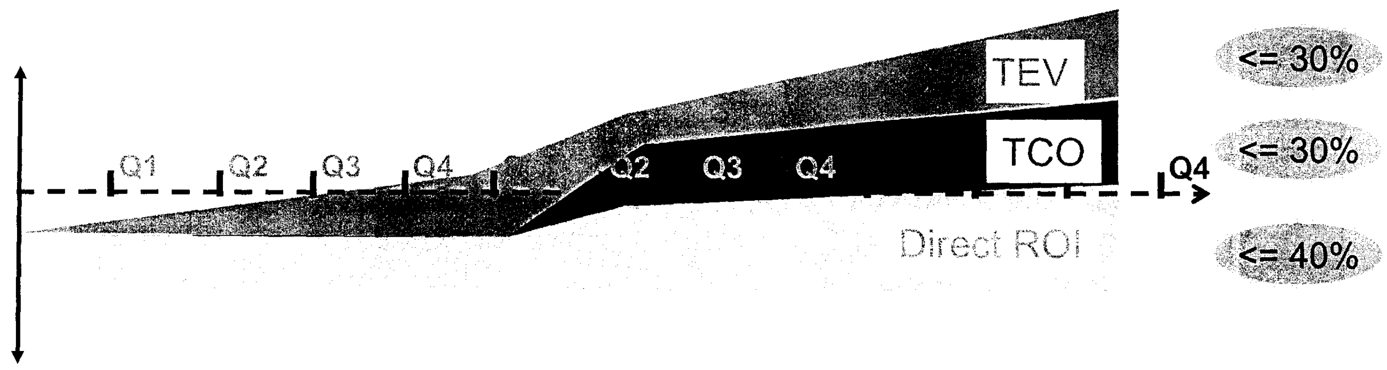


Key learnings : IP where it makes 'cents'



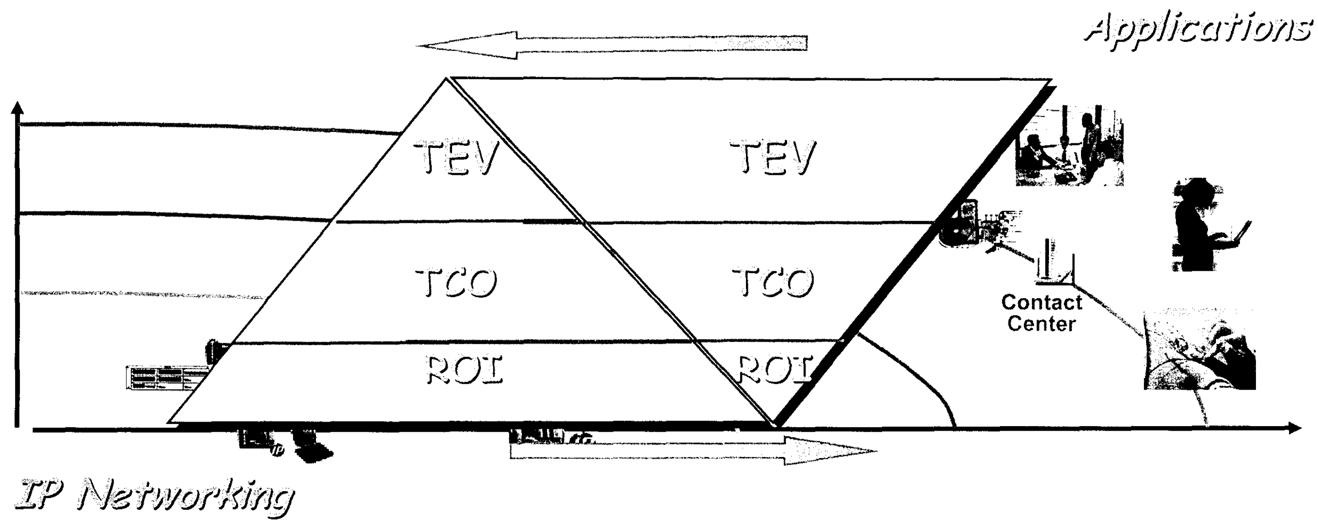
- **Not a single recipe**
  - A Full IP : maximum return but required investment high
  - **The 'A-la-carte' scenario : often the most cost-effective**

Key learnings : TEV makes the difference…



- Any economic analysis needs to integrate TEV ie value contribution of IP applications

Benefit 100% on the value chain



- **Benefit at 100% of the total values Direct ROI, TCO and TEV**

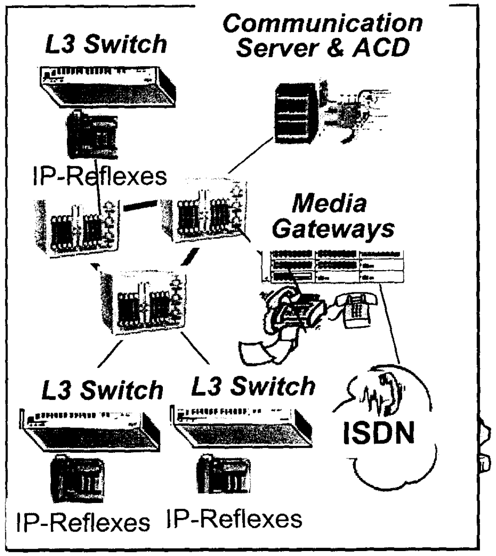
Agenda

- **Defining the ROI Methodology**
- **Some key learnings of initial ROI calculations**
- Real Case Examples

A 'full IP network' across a multisite city network
City of Dothan, Alabama

- **The pain**
  - Management complexity (ATM, Token Ring, Ethernet)
  - Very high communication costs (Centrex)

- **The solution**
  - Full renewal of the voice and data infrastructure
  - Data network based on Alcatel OmniCore and OmniAccess range
  - IP Telephony (LAN, WAN, branch) based on OmniPCX family



A customer case, City of Dothan ROI Evaluation

DOTHAN — EVERYTHING IN A NUTSHELL — ALABAMA

- The city of Dothan : 30 sites
- Total investment 400 K$

- **The gain**
  - **Direct ROI and TCO**
    - 200K$ savings annually ie 50%
  - **Total Economic Value**
    - Advanced services to 30 sites through VoIP and a converged network (ACD, on site mobility with PWT, conferencing, voice mail…)

Payback
8 quarters

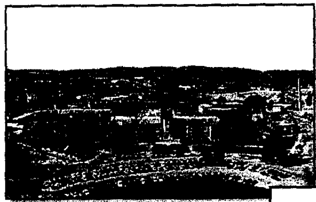
More cases to highlight the value of applications...TEV makes ROI !!

- **Virtual IP contact center to avoid organizing shifts in each location**

DaimlerChrysler Bank

- **Non profit Organization** Traveling Staff : **IP** Softphone **to call with a good** quality, **reasonable** price.

- **Jackson State University** : Sell Voice Mail Services **to Students on a full IP Network**



Payback 6 quarters

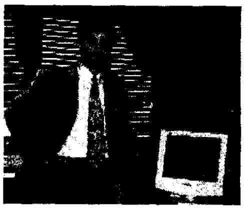
Crédit Agricole

Bank leader chose to converge voice and data networks onto a single platform to greatly improve client care.

«Our choice for Alcatel was based upon an open and distributed communication platform supporting IP, while leveraging our legacy Token Ring infrastructure and avoiding costly cabling upgrades required by

ıd
ke a
t
buyer.»

Jean Pierre CORDEIRO,
Telecom Director,
Cr檢it Agricole Pyr慚業s Gascogne



Challenges

- Replacing separate voice and data networks in a mission critical banking environment to reduce costs, operational complexity and streamlining vendor support issues.
- Giving 150 branch locations the same service level quality and voice service functions available at corporate headquarters.

Solution

- An Alcatel VoIP network integrating separate networks, key cost savings in several areas (e.g., cabling, PSTN line cost, staff reduction), simplifying network management and establishing a future-proof platform for evolving applications and technologies..

Benefits

- 46% GSM and PSTN line Cost savings
- Homogeneous voice features upgrade.
- Increased network manageability.

Crédit Agricole Pyrénées Gascogne

The leading French bank in customer accessibility thanks to its decentralized management structure and high-density branch office network, Credit Agricole is the first bank of France with more than 16 million customers. The group is a decentralized having three organizational structures; the parent holding company, 45 regional banks, and reporting into these are 9,000 branch offices.

Crédit Agricole Pyrénées Gascogne is one of the largest regional banks in the CA group and one of the largest financial institutions headquartered in Pau, France (100 millions annual revenue). Through its branch network and Internet banking division (IT budget: 10 millions – Telecom budget: € 2 millions), CA provides its customers with a full range of commercial and individual banking products and services.

With 152 locations and some 1800 people, Crédit Agricole offers the most diverse banking services found anywhere in the Pyrénées/Gascogne region.

Such responsibilities is one of many reasons Crédit Agricole selected a converged voice and data network to improve reliability, manageability and information access of its communication network.

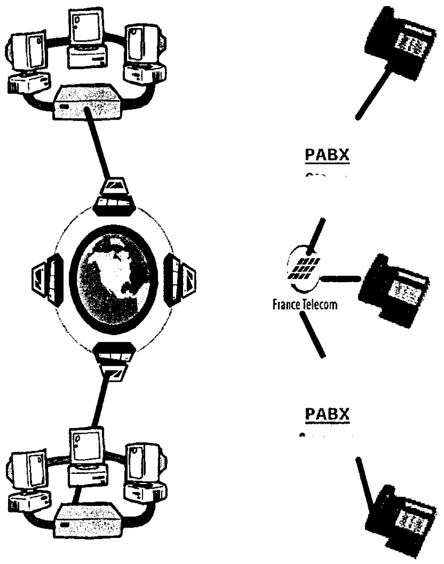
Network aches and pain
Separate voice and data networks are costly and difficult to manage.

Information technology at Crédit Agricole is essential to efficiently delivering banking service. According to Jean Pierre Cordeiro, Telecom director at Crédit Agricole, "It's important that necessary information is made available when and where it's needed to facilitate business decisions. This means providing a variety of communications services among our several sites and agencies, teams as well as to our clients ".

With a communications network playing such a critical role in meeting the needs of clients and staff, it's not difficult to understand why Cordeiro and member IT team found themselves concerned and frustrated with their existing network infrastructure.
Because of its steady growth over the last six years, CA sought to create a unified network capable of providing standardized services to all of its branches. Disparate telephony systems needed to be replaced, and essential features such as voice mail, automated attendants, and four-digit dialing needed to be implemented while maintaining branch independence.
Crédit Agricole's separate voice and data networks covered 152 sites with disparate voice features making the infrastructure costly and cumbersome to manage and change. The headquarters were equipped with 4 Alcatel OmniPCX 4400 while the sites hosted 112 Alcatel 4200 PBXs. The numerous Headquarter-site calls increased the phone subscription fees up to € 305 K and the access line link charges accounted for 11% of the total telecom budget.

"What's more" said Cordeiro, "we found that 30% of our customer calls were abandoned due to discrepancies in voice features levels". These discrepancies were created due to budgetary constraints forcing work around and short cuts that over time resulted in a complex telecom network of PBXs with different software releases, configurations and delays in change implementation".

On the data side, discontinued service support for our existing private IP network was in serious jeopardy. It was absolutely necessary to replace our network end-to-end or be without vendor support".



2 separate networks

"Our goal", Cordeiro continued, "has been to put in place a common infrastructure for all voice, data and video services provided and then take advantage of converging technologies as industry standards became established and adopted. When we took a serious look at Alcatel products line, it was obvious that it was not only a nice fit for our organization but ultimately and most importantly would greatly reduce the cost of providing services to our staff and community".

CA decided to explore Voice over IP (VoIP) as a way of putting excess bandwidth to work, improving operating efficiencies, and reducing operating expenses.

The vendor/Integrator of choice
Alcatel solution delivers more functionality and overall value

Crédit Agricole launched a comprehensive selection process evaluating Vendors and Integrators to determine the best means of meeting their long-term network objectives. "In the final analysis, Alcatel (and NextiraOne) not only delivered more options than we had asked for, but the company did so with a single server configuration, full end-to-end manageability and a very simple-to implement network plan", Cordeiro explained.

The anatomy of a converged voice over IP (VoIP) network
Crédit Agricole and Alcatel partner for an end-to-end network migration.

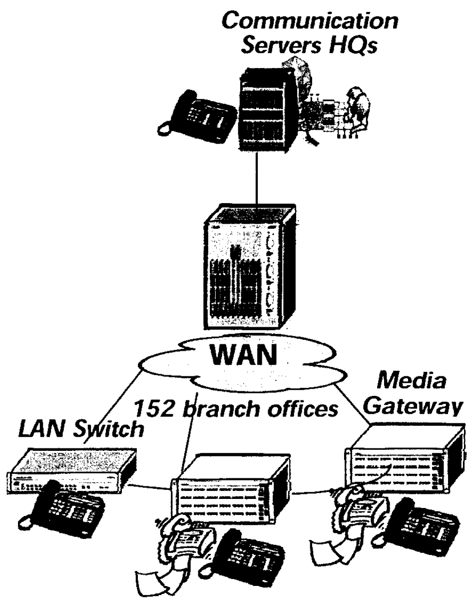
To provide the infrastructure and quality of service to support advanced voice communications, Alcatel simplified Crédit Agricole's network with an open and distributed appliance sever platform versus a monolithic closed PABXs architecture.
The Alcatel OmniPCX Communication server supports a wide spectrum of leading

application and telephony features over IP networks and traditional time division multiplexed (TDM) networks for slower IP transformation projects requiring support of existing PBX platforms.
Crédit Agricole chose a fast IP transformation path resulting in seamless network integration, simplified network management, greater flexibility in feature deployment, and reduced costs for supporting its users across multiple branches and remote access locations.
With more than 1,800 telephone nodes on its network Crédit Agricole is also migrating its voice mail functionality to the latest converged voice and data technology available.

Results

With the deployment currently in progress, CA has begun to realize the following voice and data convergence benefits. Operating expense reductions

- Streamlined, any to any corporate communications.
- Full bandwidth utilization of the data network.
- Standardization of voice mail and Auto Attendant services across the network.
- Independent operation of business-critical telephony systems at each branch.
- Cost-effective centralized management of 152 branches from the central site eliminating the need for third-party managers
- Replacement of costly dial-up connections with cost-effective connections over the corporate WAN
- Reductions in station-to-station long distance charges



Abundant network benefits
Converged voice and IP network puts Crédit Agricole Telecom in control, impress users with advanced functionality, and prepares for the future

During Crédit Agricole one-year deployment process, many benefits of the Alcatel solution emerged. Cordeiro says Crédit Agricole continues to discover new advantages.

"Our biggest benefit was the immediate feeling and realization that we – our IT staff – are now in control of the network" said Cordeiro. The network is easy to implement, configure, manage / administer and recover. "The management of a single converged voice and data network is freeing so much of our time. We used to fight fires and do crisis management. Now we are actually able to be proactive".

The project results were as follows:
The investment payback was 18 months.
The project costs and benefits were as follows (in euros):

	Before	**After**
Access Line Subscriptions	€ 305 K	€ 214 K
Communications HQ/Sites	€ 107 K	€ 0 K *Free !*
Communications Fixed/Mobile	€ 229 K	€ 153 K *Centralized Gateway*
Communications Others	€ 732 K	€ 610 K *Site to site communications*
TOTAL	**€ 1373 K**	**€ 977 K**

Equipment related costs	
Hardware:	€ 350 K
Software:	€ 122 K
Consultancy	
Installation & support:	€ 218 K
Communication charges	€ 0 K
Staff costs, system admin	€ 80 K
Maintenance & MACs:	€ 70 K
Total investment:	**€ 690 K (€ 5 K/ site).**

"Centralize the PSTN access lines for the decentralized outgoing calls " that was the main cost benefit said Cordeiro.

Cordeiro cites other important benefits in addition to substantial savings in costs over the original solution:

Integrated infrastructure. The solution lets Crédit Agricole operate a converged voice and data network and manage it as one 'flat' network. It will make possible the delivery of integrated services, and positions Crédit Agricole to fully leverage next-generation applications such as unified messaging, mobility, centralized directories and other emerging services.

Ease of administration. The network management solutions make monitoring, troubleshooting, managing configurations "a true networking lifesaver" says Cordeiro.

Effective, satisfied users. "I have people lined up and waiting for some of the applications our new network makes possible" said Cordeiro.

VoIP Communications Costs	39 671 €
Communications costs without VoIP	73 521 €
Savings	**46.04%**

November 2002 Telecom costs

Convergence project Pre-requisite

The key success factor cited by Cordeiro was the obligation to succeed data/voice teams convergence before the network convergence. The organization needs to control and be sure of its data network. A network managed by a third party is not trustworthy.
It is also necessary to invest patience and communicate with the users.

For further information see www.alcatel.forum/

At a glance

- Crédit Agricole Pyrénées Gascogne.
- Pau, France.
- www.credit-agricole.fr
- 152 sites.
- 1 800 employees.
- Provides banking services.
- Aims to be France's leading banking services.
- Project ROI: 18 months
- Project investments: € 600,000
- Savings: € 400, 000 annually
- Providers: Alcatel/NextiraOne

January 2003
Pub No: xxxxxx

_Fixed services Innovation and attractive rates make communicating easy and convenient for our fixed-line customers.


1_

49.5
MILLION
CUSTOMERS
WORLDWIDE


2_


3_

1/2/3_Behind the apparent simplicity of the telephone lie non-stop actions to maintain superior network quality and respond to customers' needs.

The phone business

Total competition

Full competition for all telephone traffic was introduced in France as of January 1, 2002. Our share of the long-distance market - domestic and international - nevertheless stabilized at around 64 percent at yearend 2002, and we have more than 80 percent of the local call market. Our strategy is in fact quite simple: do whatever it takes to satisfy and keep our customers!

Competitive rates

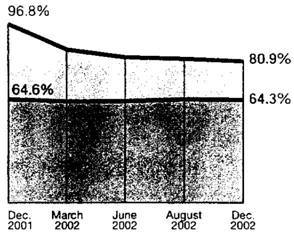
Competition has put strong pressure on rates, and our rates figure among the lowest in the market, with a complete range of attractive service plans and options. The key to services introduced in 2002 was a focus on simplicity and saving money for consumers. For example, "Les Heures France" is a selection of eight plans that let people phone anywhere in metropolitan France for the same flat, per-second rate, with no minimum time. These packages include from two to 20 hours of calls per month. For the first time we also introduced special discounts to reward loyal customers.
The range of local calling packages, known as "Les Heures Locales", was extended with more call time options, accompanied by cuts in rates.
As of April 2002 our customers enjoyed a reduction in the cost of fixed-line calls to wireless phone numbers. A further rate reduction took effect in early 2003.
For small business customers we introduced two simple and very competitive packages, "Forfait Local Pro/PME" and "Forfait France Pro/PME". The first package spans a choice of 11 monthly service options providing between four and 160 hours of local calls. The second package also includes 11 service options for five to 210 hours of calling at a single flat rate to anywhere in France.
Reflecting economic realities, these lower call rates were accompanied by a rise in monthly service subscription fees to match the inflation rate. Despite this, France Telecom's basic service fee remains among the lowest in Europe.
We launched a multitude of special promotions throughout the year as well. As official sponsor of the French national rugby team, we made it possible for customers to sign up for a free hour of calls over the weekend of the France-Ireland match. Customers could also sign up for a free hour of calling during the year-end holiday season.

Living is easy with convenient phone services

Low charges are one thing, but the superior quality of our network is what sets us apart from competitors. Behind the apparent simplicity of telephone service lie a steady stream of innovations, from voice recognition to voice-over-Internet. With over 34.1 million lines installed in France - including 4.9 million Numeris/ISDN lines - our fixed network is one of the most powerful and efficient in the world, backed by best-in-class technologies. And we make sure that sophisticated technology means ever greater ease of use, accompanied by a continually growing selection of new features and services.

France Telecom market share in France

(in %)



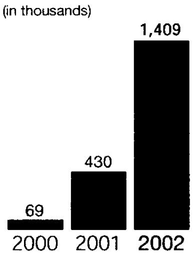
France Telecom's share of the long-distance market was unchanged and our share of the local market stabilized following the introduction of local loop competition at the beginning of the year.

january 2002 > Full competition in fixed-line telephone creates both commercial and technical challenges. Thanks to meticulous preparations, the "big bang" went off flawlessly as specialized France Telecom teams mobilized their talents. **// march 2002 >** Caller ID joins a long list of services that make fixed phones more convenient. In June, we introduced a "Voice Contact" a talking caller ID box. **//**

CONTINUED ON PAGE 52

ADSL connections

(in thousands)



Innovative rate offers from France Telecom have propelled France to No. 2 in Europe in ADSL broadband, behind Germany.

The 14 different convenient "contact" features available to our customers proved as popular as ever. Introduced in late 1999, the "3131" last call return service counted 5.8 billion calls, while the TOP Message voice mail service now manages more than 5 million voice mailboxes. Other services like call waiting and automatic callback make it easier than ever for our customers to be contacted wherever and whenever they want.

Sophisticated voice technologies helped improve different services such as the 3000 interactive customer service platform, our range of toll-free/toll-share numbers ("Numéros Vert, Azur, Indigo") and Audiotel auditex portals (32 20®, 32 40®, 32 60® and 32 23®) which let callers obtain information just by saying the name of a company.

In March, the caller ID service was enhanced with a "Voice Contact" box that both displays and pronounces the name of the caller. This innovation won a prize at the Autonomic trade fair for solutions and services for the disabled.

Staying reachable sometimes means paying for calls received. This is now possible with the "PCV France" collect call service introduced last year.

For websurfers who use a dial-up connection, the "@llo" service notifies them of incoming calls while their line is being used for the Internet. They can either answer with a voice Internet connection, transfer the call to another line, or disconnect from the Internet session and take the call.

Internet energizes fixed services

Broadband for everyone

The Internet has totally rejuvenated fixed phone service, thanks in particular to ADSL broadband, now used by a quarter of Wanadoo customers in France.

Since its launch only three years ago, ADSL has continually boosted the fixed-line market. Sales have accelerated considerably, reaching over 35,000 connections a week, making a total of more than 1.4 million households with ADSL connections at the end of December 2002.

France Telecom is working hard to make sure that everyone can enjoy the benefits of broadband. By the end of last year, ADSL was available for 21 million lines, covering 71 percent of the country. For people in remote areas we are working on dedicated infrastructures in partnership with municipal and regional authorities.

Equally important, we introduced new broadband rate offers in April, with access options offering from 128 kbps to 1,024 kbps of bandwidth. This diversification was accompanied by an overall drop in charges of about 20 percent. In addition, our new **unbundling** fees make France the cheapest country in Europe.

Internet and intranet solutions for small businesses

To sharpen their performance, businesses need to maintain close contacts with their customers, optimizing processes and giving staff access to efficiency-building tools. In the wake of large corporations, smaller enterprises are pursuing their own "e-transformation" thanks

6.8 million service packages sold in 2002

to turnkey solutions that bring them high-speeds and solid security.
To address these needs, we expanded our Oléane range of small business solutions.
Oléane Open is a complete Internet access package, including email, personalized domain names, website hosting and customer support. The range contains a very attractively priced entry-level solution called Oléane Open 3010.
Oléane VPN is a variant of the Equant IP VPN service, previously reserved to large businesses. This lets smaller firms set up a virtual private network to support a full-fledged **intranet**, managed by France Telecom. This solution is perfect for small-to-midsize enterprises with two to ten sites.
The Bizao portal provides an efficient online business platform, creating a personalized working environment tailored to a company's activity. The site provides information, practical services and collaborative working applications, all just two clicks away.

Serving the public

Multifunction payphones

Despite the boom in wireless phones, the 202,000 payphones in service in France continue to provide an essential service for our customers. These payphones are also embracing innovation, joined by 2,000 public access Minitel videotex and Rapidofax terminals. In addition, some 2,300 Wanadoo Internet kiosks enable access to custom-tailored content based on the location or operators of the kiosks, which might be a municipal office or shopping mall, for example.
Today's phones are evolving into "multiservice" kiosks.

1_

2_

Whether in the Paris suburb of Argenteuil 1, or Jordan 2_, the France Telecom group puts innovation to work for consumers.

15.4
MILLION FIXED LINE CUSTOMERS WORLDWIDE OUTSIDE FRANCE

including

10.8
MILLION CUSTOMERS FOR TP GROUP

The location of payphones is determined in liaison with public authorities and is designed to contribute to universal service.
These terminals are ideally suited to credit card payment of services and will increasingly offer Internet-based services, such as email access thanks to an interactive voice interface. New "full-service payphones" are being tested in Paris and Lyon. One promising application is "Aidenoo", which provides information on local services.
Telephone tickets have become a popular and convenient way to pay for calls from any fixed-line phone in France or other countries. The latest entry in the hotly competitive market for prepaid card services, these telephone tickets now account for 50 percent of the market.

Commercial innovation

Our 650 sales offices focus on customer service before, during and after the purchase. A variety of commercial innovations introduced in 2002 helped us deliver on this promise, including payment in three installments, home delivery, personalized diagnostic of needs, a wider selection of phones and "all-in-one" packs. A redesign of sales offices to enable more dynamic customer service was completed at over 280 locations.
In addition to the network of consumer sales offices, 26 sales locations across France cater specifically to small business customers.
Another essential factor that enhances customer service is our information system. The system was further enhanced in 2002 to enable more personalized contacts adapted to the diversity of our customers. This is especially important for phone contacts and helps rationalize our customer service resources. In other areas, ordering and provisioning is now fully automated for 80 percent of new ADSL lines, and the time needed to order a new cellphone was cut from 20 to 7 minutes.

Staying close to customers

To keep close to customers' needs we also established indirect distribution channels in addition to our own sales offices. New distribution outlets via computer, household appliances and other retail chains were set up in 2002 to reach consumers in rural areas.
Wanadoo's 2,500 indirect distribution outlets accounted for over 6 percent of broadband Internet connections in 2002. Thanks to a new website, retailers can instantly find out whether ADSL service is available for a line and sign up a new customer if it is.
"Self service" orders via online or phone services reached 7.6 percent of indirect distribution sales last year, for a total of 2.4 million sales. This rise was stimulated by special gift offers, attracting more than a million customers. The France Telecom Net store (www.agence.francetelecom.com) enjoyed an 80-percent surge in visitors as sales via the site doubled. The "3000" interactive customer platform handled 23 million calls, representing 1.4 million deliveries, over double the year-earlier figure. Orders for home delivered products jumped 180 percent to 370,000.

Covering the world

France Telecom's focus on service continues when customers travel outside France, spanning some 220 countries and territories, including the vast global coverage of our Equant subsidiary.

CONTINUED FROM PAGE 49

// **april 2002 > Sharp cuts** in ADSL rates sparked a surge in France, which became Europe's No. 2 broadband market at the end of the year. The one-million-customer mark was surpassed in November. // **summer 2002 > New local and domestic long-distance call packages** make fixed phones more convenient and economical than ever.

Third leading operator in Spain

In the growing Spanish market our wholly-owned subsidiary Uni2 was awarded the country's third fixed-line telephony license in 1998. It has since pursued growth thanks to a series of technical and marketing innovations. Uni2 was the first **alternative operator** to introduce flat-rate packages for consumer voice services. Thanks to its competitive pricing policy Uni2 boosted revenues by 50 percent in 2002, winning 60 percent of the market growth in lines on which consumers have opted for carrier preselection. The company had 1.6 million customers at the end of the year, representing 2.6 million lines.

In French Polynesia, France Telecom signed an agreement in mid March of last year with Office des Postes et Télécommunications (OPT) to create Tahiti Nui Telecom. TNT has a 20-year license for international telecommunications services. This partnership anchors our long-term presence in the region following the transfer of responsibility for international telecommunications from the French state to territorial authorities.

In El Salvador, our Telecom subsidiary unveiled new branding as it continues to develop universal, high-quality service. The country now benefits from a 100-percent digital fixed network serving 262 communities. Telecom also proposes GSM cellular and Internet access services, along with a full range of telecom services for businesses in Guatemala.

In Jordan, Jordan Telecom was listed on the stock market in September 2002 with the second-largest market capitalization on the Amman exchange. Under the stewardship of France Telecom, which has held 40 percent of the operator since January 2000, the network has been upgraded to international standards, and both a high-speed data network and wireless service have been introduced.

In Africa, our Senegalese subsidiary Sonatel successfully bid for a license to become the second operator in Mali.

In Asia, our subsidiary FCR Vietnam intensified its cooperation with state-run operator VNPT. Plans call for FCR Vietnam to build 540,000 lines in five years. FCR Vietnam also provided technical assistance during the year covering rollout of new services, the information system and network interconnection.

Strategic position in Poland

With a population of 39 million, a vigorous economy and a strategic position in the heart of Central Europe, the Polish market harbors great promise. In conjunction with our Polish partner Kulczyk Holding, we own 47.5 percent of the former state phone company TP (Telekomunikacja Polska). France Telecom's share is nearly 34 percent. Along with its subsidiaries, including cellular operator PTK Centertel, TP forms the TP group, which was consolidated in France Telecom accounts as of April 1, 2002.

In 2002 TP Group pushed its revenues up over 4 percent on a pro forma basis to 3.47 billion euros. The group's EBITDA surged 24.9 percent to 1.45 billion euros on a pro forma basis. TP is profitable in its fixed, wireless and Internet activities and enjoys excellent growth potential.

At yearend 2002 TP Group had over 10.8 million fixed-line customers, up from 10.4 million a year earlier. It also had over 95 percent of the long-distance market.


1_


2_

The France Telecom group is building strong positions in fixed, wireless and the Internet around the world, especially in Poland, Spain 1 and Senegal 2_.

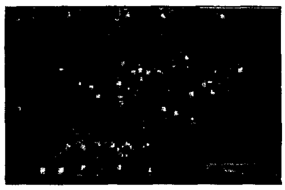
_Business services

Our network services bring enterprises unmatched competitive performance.


1_

3,700
MULTINATIONALS ARE EQUANT CUSTOMERS


2_


3_

Equant is present in 220 countries and territories, including the Netherlands 1 and Germany 2_. Equant's network provides global coverage for businesses 3_.

Innovative products

Proven solutions

In a sluggish economy, businesses must work harder than ever to sharpen their performance. Efficiently putting information technology and Internet-based services to work are thus fundamental imperatives. Our networks and value-added solutions are backed by world-leading customer care to help businesses address today's challenges.

To support the e-transformation of businesses, our primary role is to bring them access to innovative solutions. For the past five years we have used IP-based resources on our own intranet – most of them developed in our labs – before proposing them to our customers. By applying our own solutions we make sure they will fully meet the expectations of demanding customers around the world.

Tools for nomadic communication

The Internet is not our only focus for innovation. For example, we introduced new solutions in 2002 to facilitate nomadic communication in France and worldwide.

Thanks to Orange Enterprise Portal and Orange Bureau solutions, staff enjoy fully secure access to their email, diary and corporate intranet, as well as the Internet. Over 700 businesses have already signed up for these services. Since spring 2002 they have been available via the Orange GPRS network, cutting access time and optimizing costs. Outside of France these solutions are available in Belgium, Denmark, the United Kingdom and Switzerland.

Convergent solutions

The France Telecom group is uniquely positioned to blend voice, data and video solutions, spawning a suite of innovative convergence solutions packaged with value-added. These solutions seamlessly span wireless, data, the Internet and intranets. Trials with large corporate customers will help speed a host of new applications to market in 2003, including video, IP telephony, Web conferencing, e-learning and multimedia contact centers.

Best-in-class outsourcing solutions

As more large corporations refocus on core activities, many of them have chosen France Telecom for complete or partial outsourcing of their telecommunications operations. Club Med, Thomson, Credit Lyonnais, AGF and others benefit from our custom-tailored services as part of multi-year contracts. These services cover France and the rest of the world, with à la carte outsourcing of components – from intranets linking corporate sites to managed LAN – as well as security, hosting and contractual quality of service.

650
EQUANT IP VPN NETWORKS DEPLOYED WORLDWIDE

80,000
GPRS SUBSCRIPTIONS IN 2002

july 2002 > Smartjog, a venture spun off from Globecast, offers the first international platform for international digital programming distribution for broadcast professionals. The service replaces video tape with digital content. **//**
october 2002 > Equant named "Best Global Carrier" by Emap and ECTA for successful integration of Global One, financial stability and superior quality customer care. **//**

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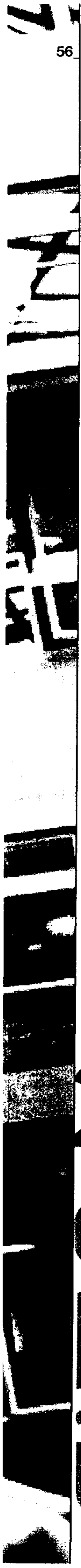

2.6
MILLION KILOMETERS OF OPTICAL FIBERS IN FRANCE

9,000
CUSTOMER SERVICE AGENTS

Networks for every need

Ubiquitous bandwidth without limits

Our unmatched network strength lets us provide businesses of every size with efficient communications solutions, regardless of where they are located or the volume. High-bandwidth services accelerate this process. In France, over 200,000 business sites were equipped with high-speed links at the end of 2002. Most of them use the links for connection to the Internet and data networks, providing bandwidth of up to 1 **Gbps** using optical technology. Some 80 percent of enterprise sites now have access to DSL technologies.

We achieved a significant milestone in 2002 with trials of Gigabit **Ethernet** links, in partnership with Atrica. This should usher in services on multiclient metropolitan networks and even on nationwide infrastructures that tap the power and rich functionality of Gigabit Ethernet **LAN** switches. To interconnect major sites, we offer bandwidth of up to 80 Gbps. Distribution networks will leverage the widespread availability of business-class DSL technology. The Numeris Multisite ISDN solution brings fully secure fiberoptic service to large urban areas for both voice and data transport.

Custom-tailored networks

France Telecom is widely recognized for excellence in providing high-speed networks for the world's largest businesses. Our complete portfolio of solutions addresses the full spectrum of needs, including geographic constraints, flawless reliability, scalability and, of course, security. For insurance firm Allianz, Equant deployed a virtual private network based on the Internet protocol (**IP VPN**). This network links a hundred sites on every continent and supports an intranet with a corporate directory, groupware, and a stock and bond trading application. Equant is providing project and service management, along with high-level security. Transmission speeds at the core of the network are up to 155 Mbps. Equant IP VPN is the industry's leading solution, used by more than 650 multinational businesses.

To shrink design and production cycles and share data, the European automotive industry established the European Network Exchange, or ENX. This **extranet** connects automakers with suppliers and partners. We are one of three carriers selected for deployment of the network. In 2002 we expanded our ENX offer from six to 33 countries, supported by the Equant network.

New horizons for collaborative working

Following the success of Operation Lindbergh (a transatlantic surgical operation) in 2001, France Telecom and IRCAD (Institute for Research into Cancer of the Digestive System) pursued their partnership with the creation of the Argonaute 3D experimental platform that lets medical professionals at remote locations examine patient records together, with three-dimensional elements.

A demonstration of the platform drew on existing solutions from Wanadoo and Oléane, coupled with innovations spearheaded by France Telecom R&D.

200,000 sites equipped with high-speed links in France

Our high-bandwidth networks enabled fluid transmission of 3D images, and the experimental collaborative working platform developed by our researchers harbors great promise for the future.
Argonaute 3D opens up tremendous opportunities to increase productivity for all those who work with complex images, such as urban planners, geologists or architects, as well as for the aerospace, automotive and other industries that rely on computer-aided design.

Satellite broadcasting

France Telecom is also recognized for its expertise in satellite broadcasting through our subsidiary Globecast, the global leader in satellite transmission services for professional broadcast, enterprise multimedia and Internet content delivery. With 16 offices and teleport earth stations on five continents, Globecast offers a complete range of satellite broadcast solutions, including TV and radio distribution, secure Internet content delivery and mobile production for sports and news events.

Value-added solutions

Spearheading e-business

Our value-added services facilitate relations both within businesses and with their customers and partners, covering everything from security and customer contact management, to hosting of servers. These services must deliver 24-by-7 performance, which is why they are an important core activity for the group in France and worldwide, available via Transpac and Equant. Staying on the cusp of the latest

2_

Equant operates one of the world's largest data networks, especially in the United States, including facilities in New York 1 and Atlanta 2_.

technologies requires significant resources, which is why more and more enterprises are opting to focus on their core activities and have specialists host their applications. With more than 20 years of experience, we have the skills and facilities to address their needs, whether basic or highly complex. Over 700 people at France Telecom work in the hosting segment.

Equant, recognized leadership for data beyond borders

Equant is a widely acknowledged world leader in international data and IP network services for multinational businesses, operating a seamless network with unmatched global reach. Equant's network spans major business centers in 220 countries and territories, with local support services in 167 of them. Backed by over 50 years of proven experience, Equant serves thousands of top-tier multinational enterprises thanks to the industry's most comprehensive portfolio of network services.

We currently have 25 to 35 percent of the French market and we expect growth of 35 to 40 percent per year. We aim to bolster our leadership in a market that is seeing consolidation. Our strategy is anchored in high quality of service, security, and scalable hosting solutions for websites and applications.
Our solutions encompass network connectivity, supervision, security and backup, or complete management of applications. Two emblematic examples of our expertise are hosting of the website for outdoor and sporting goods retailer "Le Vieux Campeur" – listing 10,000 items – and a collaborative working application used by aircraft manufacturer Airbus and its suppliers.

Certified security

Businesses are legitimately very concerned about the security of their information system. Our R&D center has let us develop high-level expertise in this area, which is why over 200 large enterprises rely on France Telecom for the security of all or part of their Internet accesses, as well as exchanges with their corporate intranet. In January 2002, the Equant IP VPN solution obtained "Common Criteria" certification from the French government information systems security agency DCSSI, the first time certification had been granted for an entire service. This label guarantees that data flows are fully protected against intrusion, along with the physical security of equipment and administration procedures. The norm has since been adopted by the International Standards Organization as the ISO 15408 standard. We provide customers with security solutions adapted to their needs, addressing all issues, from transport to access gateways and

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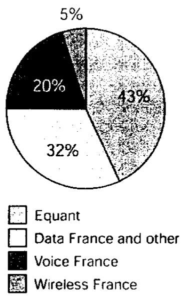
France Telecom provides support in 191 countries, assisting customers from France to Tokyo 1 or Munich 2_.

Breakdown of business services revenues



Although data services account for the largest share of revenues, voice remains a key market, reflecting the importance of personal contacts.

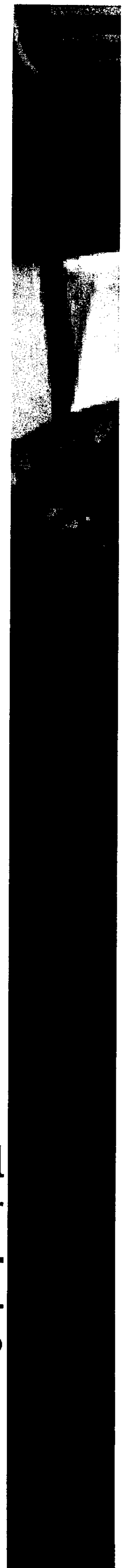
authentication. We also propose consulting and expertise to help customers improve information systems security, and propose managed security that includes firewalls, caches, filters and virus protection. Our security services include authentication, Public Key Infrastructure (PKI) and Internet Protocol Security (IPSEC).
Our security solutions also protect transactions and applications, including remote payment (online VAT payment, for example), exchanges for closed user groups, and micropayment (using W-HA technology).
Our Inter-SAN solution is designed for large businesses that require real-time backup of data transactions. This service is now available in all major French cities and has won over such firms as consumer credit leader Cetelem.

Taking customer relations to a new level

Customers expect businesses to listen to their needs and provide personalized answers, leading businesses to multiply the number of contact channels. One of the key challenges for efficient Customer Relationship Management (CRM) is enabling people to contact a business however and whenever they want, regardless of the means – fixed or wireless phones or Internet.
In 2002, our "Solutions i-contact" range broke new ground by making convergence of remote contact channels a reality. In addition to letting customers know what types of contact services are available, this range includes qualification of contacts to route them to the appropriate teams, contact management to make sure customers talk to the right specialist, and distribution of contacts for optimized processing. Every aspect of customer relationship management is covered and integrated within the operational organization of each individual business.
"Click to Contact", for example, lets websurfers select one of seven different immediate or delayed means to contact an agent – callback by phone, Internet phone or Internet chat, for example.
Another "i-contact" solution is Qualimail, which simplifies and optimizes processing email-based customer relations. The application provides companies with tools to automate, improve and simplify processing of messages, such as reception acknowledgement receipts, message qualification, standard or personalized reply messages, archiving, etc. Mail order firm Quelle, for instance, deploys this solution to handle burgeoning email traffic from its four million customers in France. The company has cut response times to a maximum of 24 hours, outside limited peak periods.
All-IP call centers are another forward-looking solution to maximize customer relationships. Mail order leader La Redoute has deployed one of these multimedia contact centers with access via its website. The 12 agents at the alloweb® platform process all customer requests, whether they arrive by email, phone or via the online forum. They can call back customers with records of orders instantly displayed on their screen. The entire platform is hosted by France Telecom, with remote administration using a standard Web browser.
These few examples are emblematic of our commitment to bringing enterprises the best technology to let them do what they know best: their business.

CONTINUED FROM PAGE 55

// november 2002 > Argonaute 3D lets medical specialists across France jointly consult 3D images for a more reliable diagnosis. **// december 2002 > TotalFinaElf** chooses France Telecom and Equant to interconnect 1,500 sites in 75 countries via an Equant IP VPN network.



IP Telephony Design Guide

AN ALCATEL WHITE PAPER

April, 2003

ARCHITECTS OF AN INTERNET WORLD ALCATEL

IP Telephony Design Guide

Overview

With the availability of today's new convergence technology more and more people are planning to deploy voice traffic over existing data networks. The practice of bandwidth sharing between voice and data traffic over a single network is not new. In the 1980's, time division multiplexing (TDM) made short work out of this task by carving up the bandwidth required to share a single, wide area network (WAN) connection between multiple locations. Although statistical multiplexing or packet-based networking was more effective for transporting voice, you still needed to maintain independent networks - one for data-/LAN-based traffic and one for voice.

With the Internet explosion and advanced PC applications that use more and more bandwidth, the data network volume has increased dramatically and is now the dominant bandwidth consumer. Therefore, it now makes sense to use the data network to transport voice instead of the voice network to transport data traffic.

This design guide covers the issues related to designing an IP telephony or voice over IP (VoIP) network for transporting voice and data over a common LAN or WAN infrastructure. Understanding the underlying technology used to transport voice traffic is important in designing an IP telephony network. Design principles used to deploy a successful LAN-based VoIP network will not necessarily work when you apply them to a WAN configuration. This document discusses the major hurdles that need to be addressed when designing either a LAN or WAN based VoIP network.

IP Telephony vs. VoIP

Let's define what IP telephony and VoIP mean in this document. IP telephony is the combination of voice, data, video, and wireless applications into an integrated enterprise infrastructure that offers the reliability, interoperability, and security of a voice network, the benefits of IP, and the efficiencies, mobility, and the manageability of a single network. IP telephony is based on circuit-switched and TCP/IP technologies and protocols, it removes the limitations of proprietary systems and provides increased productivity, scalability, mobility, and adaptability.

Voice over IP (VoIP) is the technology that is used to transmit voice over an IP network, which can be either a corporate network or the Internet.

Voice Bandwidth Requirements

In the traditional voice world a single T1 leased line is used to carry 24 toll quality telephone calls from the Public Switched Telephone Network (PSTN). Those with a private point-to-point T1 connection can compress the voice to less than 8 Kbps for more efficiency, but the quality may be sacrificed. The three most common modulation schemes for encoding voice are:

64 Kbps (PCM)/1.544 Mbps = 24 simultaneous calls on a T1

32 Kbps (ADPCM)/1.544 Mbps = 48 simultaneous calls on a T1

8 Kbps (CELP)/1.544 Mbps = 120 simultaneous calls on a T1

Efficiency is the primary WAN connection issue. Bandwidth use and voice compression play an important role in provisioning the WAN.

Voice over IP Bandwidth Requirements

What does it take to support traditional voice on data networks? The concept of combining voice on the data network is simple because voice traffic uses a lot less bandwidth than traditional LAN based computer networks. A single toll quality phone call over the public network uses 64 Kbps in each direction, that's only 0.0625% of a 100 Mbps full duplex link.

On a 100 Mbps Ethernet network each voice call takes up to 85.6 Kbps (64 Kbps + IP header + Ethernet header) in each direction supporting up to 1,160 calls over a full duplex link. On a Gigabit backbone up to 11,600 simultaneous calls can be handled.

If bandwidth were the only issue, LAN-based IP telephony networks would have been deployed years ago. But other elements, such as bandwidth hungry business applications, advancements in telephone technology, and network congestion have been the major stumbling blocks. Most of those issues have been resolved with newer VoIP technology, QoS, and the use of bandwidth managers or complex queuing schemes deployed on the LAN and WAN.

Voice Quality

Over the years voice quality has been very subjective: picking up the phone and listening to the quality of the voice. If you had two different users on the same call you may even receive reports of varying results. After years of research, human behavioral patterns have been recorded and scored, establishing an objective measurement of call quality.

The leading subjective measurement of voice quality is the Mean Opinion Score (MOS) as defined in the International Telecommunications Union (ITU) recommendation P.800. Mapping between network characteristics and quality score make MOS valuable for doing network assessments and tuning.

A MOS score can range from 5 (very satisfied) to 1 (not recommended), but keep in mind that each voice codec has a benchmark score based on several factors, including packetization delay and the inherent degradation that occurs when converting the voice to a digital signal. The highest MOS rating any codec could receive is 4.5. Each codec is given a MOS value based on any known impairments for the speed of the conversion, speech quality, and data loss characteristics. Below is a listing of the most common codecs used today for VoIP and their theoretical maximum MOS value.

Codec	Default data rate	Time between packets	Packetization delay	Default jitter buffer delay	Theoretical maximum MOS
G.711u	64 kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.711a	64 kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.729	8 kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07
G.723.1 MPMLQ	6.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87
G.723.1 ACELP	5.3 kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.69

Source: Voice Over IP 2nd edition

Each network will have a different MOS value based on QoS, delay and codec that is deployed in the IP network. When deploying an IP telephony network the goal is to get the network to support the maximum MOS value and to achieve the best quality for voice traffic. All MOS values above 4.0 are considered to be toll quality speech.

Converting Voice into Data Packets

Digital signal processors (DSP) – the engines for voice coders – are making their way into IP telephony systems. The DSP is a specialized processor that has been in use for many years in other telephone applications such as mobile wireless networks. The DSP needs to be very fast due to the computation intensive operations required to process a typical telephone call. In essence, the DSP is what converts analog voice signal into data packets so they can be transported over an IP-based network.

In this document, DSP refers to the combined efforts of DSPs and codecs to perform the conversion of analog and digital signals into IP communication flows. DSP works by clarifying or standardizing the levels or states of a digital signal. A DSP circuit is able to differentiate between human-made signals, which are orderly, and noise, which is inherently chaotic.

Typically, the voice-coding algorithm used for an IP telephony or VoIP network in a LAN environment is G.711, which divides a voice stream up into 64 Kbps packet increments. It is regarded as toll quality. Some of the other more widely available voice coding algorithms/compressors on the market are the G.729a and G.723 codecs. The G.729a and G.723 codecs are normally used for WAN connections where bandwidth is at a premium and voice compression is a requirement. The majority of vendors who support IP telephony recommend the G.729a codec due to its superior quality over G.723, making it the de facto standard for WAN connections running IP telephony.

The chart below shows the bandwidth calculation for each codec.

Voice coder	Voice bandwidth Kbps	MOS	Codec delay	Packet size (bytes)	IP/UDP/RTP headers (bytes)	cRTP'	L2 header (bytes)	Total BW	BW with silent suppression
					Ethernet				
G.711	64	4.1	1.5	160	40		14	85.6	42.8
G.711	64	4.1	1.5	160		2	14	70.4	35.2
G.729	8	3.9	15	10	40		14	29.6	14.8
G.729	8	3.9	15	10		2	14	14.4	7.2
					PPP				
G.711	64	4.1	1.5	160	40		6	82.4	41.2
G.711	64	4.1	1.5	160		2	6	67.2	33.6
G.729	8	3.9	15	10	40		6	26.4	13.2
G.729	8	3.9	15	10		2	6	11.2	5.6
G.723	6.3	3.9	37.5	30	40		6	16	8
G.723	6.3	3.9	37.5	30		2	6	8	4
					Frame Relay				
G.711	64	4.1	1.5	160	40		4	81.6	40.8
G.711	64	4.1	1.5	160		2	4	66.4	33.2
G.729	8	3.9	15	10	40		4	19.7	9.9
G.729	8	3.9	15	10		2	4	9.6	4.8
G.723	6.3	3.9	37.5	30	40		4	15.5	7.8
G.723	6.3	3.9	37.5	30		2	4	7.6	3.8
					ATM				
G.711	64	4.1	1.5	160	40		5 cells	106	53
G.711	64	4.1	1.5	160		2	4 cells	4	42.4
G.729	8	3.9	15	10	40		2 cells	2.3	14.1
G.729	8	3.9	15	10		2	1 cell	14.1	7.1
G.723	6.3	3.9	37.5	30	40		4	22.3	11.1
G.723	6.3	3.9	37.5	30		2	4	11.1	5.6

Table 1 - Bandwidth calculation by voice code

Buffering and Error Checking

Due to the bursty nature of business applications, data networks have large buffers built into them to sustain large bursts of traffic over a short period of time.

Large buffers in a voice network will only increase the delay of time sensitive traffic and cause poor call quality. Voice is very similar to constant bit rate (CBR) traffic – it requires a predictable, reliable throughput.

The majority of the LAN protocols used to transport data traffic include end-to-end error checking. So, if a packet is delayed or lost, the originating station will retransmit a copy of the frame. The end station will wait for the acknowledgement, then reassemble the packet stream and pass it on to the application. This is usually transparent to the user.

Voice transmissions on the other hand are very time sensitive. The originating station does not copy the transmitted frame into a buffer, since it would only increase the delay and degrade quality. With voice, if you lose a frame, it is lost. Both error and frame sequence checking is done at the upper level of the Real Time Protocol (RTP), but due to the time sensitive nature of the voice stream, if the frame is out of sequence it will be discarded and the next frame will be processed, thus affecting the quality of the call.

The majority of voice codecs can support minor frame loss, but the conversation will be choppy and of poor quality. Some of the IP telephony equipment manufacturers have tried to compensate for poor line quality by playing the preceding voice frame a second time, but this does not resolve the issue, it only makes it tolerable. This is why it is so important to understand the inherent behavior of voice running on a data network and the additional requirements like QoS and predictive delay that a network must meet.

IP Telephony/VoIP Audit

An IP telephony/VoIP audit should be performed for every proposed LAN/WAN segment prior to the addition of IP telephony traffic. The key to designing an IP telephony network is an understanding of the underlying technology used to transport the IP telephony traffic. The design principles used to deploy a successful LAN based VoIP network will not necessarily work when you apply them to a WAN configuration, due to a number of factors including limited bandwidth. QoS and traffic isolation are the key factors for the LAN, but bandwidth, priority and delay are important to the WAN. This can make a significant impact on the installation.

The most common cause for poor voice quality during a VoIP installation is inadequate WAN bandwidth to support both voice and data traffic. If an audit was performed prior to the installation, corrective action could have been taken to resolve the issue prior to deployment.

In some cases, a poorly designed WAN can be fixed by lowering the delay with fewer router hops, setting up QoS on the routers or increasing the amount of available bandwidth prior to the installation of voice. In other cases the solution may be too expensive or too complex and other products like bandwidth managers must be deployed prior to the addition of voice.

Bandwidth Management

If the MOS value is not in an acceptable range after completing the IP audit and tweaking the installed vendor's suggested parameters, a bandwidth manager may be needed for a successful installation. Bandwidth managers allow the end user to define how much bandwidth is going to be used by each application and guarantee what percentage of the WAN bandwidth is going to be used by voice applications.

What is QoS and why is QoS needed?

Voice quality is directly affected by many factors that can be divided into five QoS dimensions that impact the end user experience:

1) Availability

2) Throughput (both committed and burst)

3) Delay or latency

4) Delay variation, including jitter and wander

5) Packet loss

Availability

Availability is the percentage of time that the network is up. The traditional benchmark for a voice network is 99.999% ("five 9s"), or about 5.25 minutes of downtime per year. Availability is achieved through a combination of equipment reliability and network survivability. Availability is a probability calculation, so it is not simply calculated by summing the MTBF figures.

Throughput

Throughput is the amount of traffic – or bandwidth – delivered over a given period of time. Generally speaking in the LAN environment, more throughput is better.

For the majority of WAN users, throughput depends on the amount of money paid to lease carrier facilities. So efficiency, compression, and bandwidth management play key roles in designing an IP telephony network.

Delay

Delay or latency is the average transit time of a service from the ingress to the egress point of the network. Many services – especially real-time services such as voice communications – are highly intolerant of excessive or unnecessary delay. Interactive conversation becomes very cumbersome when delay exceeds 100-150 ms, when it exceeds 200 ms users find it disturbing and describe the voice quality as poor. To provide high quality voice, the VoIP network must be capable of guaranteeing low latency. The ITU-T G.114 recommendation limits the maximum acceptable round trip delay time to 300 ms between the two VoIP gateways (150 ms one-way delay).

There are many components of delay in a network that must be understood, including packetization delay, queuing delay, and propagation delay.

- ***Packetization Delay*** is the amount of time it takes the codec to complete the analog to digital conversion. Realize that IP telephony/VoIP always creates some measure of delay, as the algorithm specifies to "listen" or sample the voice for a specified period, followed by packetization.
- ***Propagation Delay*** is the amount of time it takes information to traverse a copper, fiber, or wireless link. It is also a function of the speed of light, the universal constant, and the signaling speed of the physical medium. For example, if a call has to pass through a transit node more delay is introduced.
- ***Queuing Delay*** is imposed on a packet at congestion points when it waits for its turn to be processed while other packets are sent through a switch or wire. For example, as previously stated ATM mitigated queuing delay by chopping packets into small pieces, packing them into cells, and putting them into absolute priority queues. Because the cells are small, the highest priority queue can be serviced more often, reducing the wait time for packets in this queue to deterministic levels. At gigabit speeds, however, the waiting time for high-priority traffic is very small even under the worst conditions, due to the speed of the links and available processing power.

Delay Variation

Delay variation is the difference in delay exhibited by different packets that are part of the same traffic flow. High-frequency delay variation is known as jitter, while low-frequency delay variation is called wander. Jitter is caused primarily by differences in queue wait times for consecutive packets in a flow, and is the most significant issue for QoS. Certain traffic types—especially real-time traffic such as voice, are very intolerant of jitter. Differences in packet arrival times cause choppiness in the voice. All transport systems exhibit some jitter. As long as jitter falls within defined tolerances, it does not impact service quality.

Excessive jitter can be overcome by buffering, but this increases delay, which can cause other problems. With intelligent discard mechanisms, IP telephony/VoIP systems will try to synchronize a communication flow by selective packet discard, in an effort to avoid the "walkie-talkie" phenomenon caused when two sides of a conversation have significant latency. Jitter must be less than 60ms (60ms = average quality, 20ms = toll quality).

Packet Loss

Loss – either bit errors or packet drops – has a bigger impact on IP telephony/VoIP services than on data services. During a voice transmission, loss of multiple bits or packets of stream may cause an audible pop that will become annoying to the user. In a data transmission, loss of a single bit or multiple packets of information is almost never noticed by users. In contrast, during a video broadcast, consecutive packet loss may cause a momentary glitch on the screen, but the video then proceeds as before. However, if packet drops become epidemic, then the quality of all transmissions degrades. Packet loss rate must be less than 5% for minimum quality and less than 1% for toll quality

Class of Service

The main objective of Resource Reservation Protocol (RSVP) is to guarantee end-to-end QoS throughout the network by reserving bandwidth unicast and multicast applications on an individual flow basis.

Differentiated Services (DiffServ) is designed to group all flows with the same service requirement into a single aggregate. For example: RSVP would reserve bandwidth for a single VoIP call, while DiffServ would group all VoIP traffic together in the same flow. This aggregated flow would then receive its class of service based on the application priority.

When a QoS mechanism like DiffServ is enabled, it will provide complete flexibility in defining service classes that can be provisioned in a converged voice and data network. This means that the network management system provides access to the mechanisms that allow the end user to create customized service classes for each application.

Most networks are deployed with some level of QoS at layer 3 that supports the following classes of service:

- **Expedited forwarding (EF) for control frames like RTCP**
- **Assured forwarding (AF) for VoIP traffic**
- **Best Effort (BE) for all other data traffic**

It is possible to map different QoS parameters to one another (i.e., 802.1p to ToS or ToS to DiffServ) to enable the network designer to provision an "end-to-end" class of service for voice, video and data traffic.

Deploying IP Telephony in a Converged Alcatel Network

Today's business depends on scalable network communications that allow future expansion of business options and facilities. The groundbreaking OmniSwitch family (6600 series, 7000 series, and the 8800) and OmniPCX Enterprise voice products target that future networking and business solution. The OmniSwitch family series is a new line of data infrastructure switches that spans the core, edge, and desktop of networking. The design combines Alcatel's experience and expertise building carrier and enterprise network equipment with all of the company's cutting-edge convergence technologies.

e-Business solutions must provide availability, security, intelligence, and manageability. These values are both essential to successful modern business and fundamental to appropriate new technology.

The OmniSwitch family offers carrier-class availability throughout all networking components to deliver the infrastructure mandatory for IP telephony and mission-critical applications. A multi-layered approach to security is offered, securing traffic to, through and between switch nodes, preventing unauthorized access to business traffic and ensuring privacy. Intelligence mandates that all switching decisions are distributed and performed at wire-rate. Alcatel's implementation is wire-rate into, through the backplane, and out all network interfaces without performance bottlenecks. Manageability involves both networking and management system features. OneTouch QoS means that complex QoS policies are implemented consistently with a simple point-and-click interface.

Deploying IP Telephony and VoIP in a Multi-Vendor Environment

Even though IP telephony and VoIP technology have made some vast reliability and quality improvements over the past couple of years, customers and network designers still struggle with implementing the technology in a multi-vendor network. There are many reasons for this such as: inter-operability issues, proprietary protocols, and just plain old finger pointing. Please check with the manufacture of your installed equipment for their recommendations on how to design and deploy an IP telephony or VoIP network in a multi-vendor setting.

Design Recommendations

One of the most important recommendations that can be made is to pay close attention to the infrastructure that the VoIP network is built on. The foundation must be solid otherwise there will be ongoing quality issues until the network design issues are resolved. The more time spent upfront investigating and verifying the design of the LAN and/or WAN will make a more successful ending. Verification is critical, and although it may seem reasonable to believe that the "network is new and should support QoS" it's important to check. In some cases, like running VoIP over a WAN, an audit is a must. For example, the total end-to-end delay to support a quality voice conversation must not exceed 200 ms and can only be verified by an IP audit. Remember, the longer the delay the worse the quality.

After a VoIP audit is preformed the designer must engineer the network to support the worst-case scenario, even if it happens only 1 % of the time. Engineering the network for peaks not averages maintains the highest quality of voice traffic while the network is performing at its maximum potential.

When designing a VoIP WAN, the designer is required to calculate the amount of available bandwidth for all applications required to transit the link. In most cases the link traffic is miss-calculated or the IP audit is not performed prior to the installation and the quality of the VoIP calls suffer. As previously sated, a good rule of thumb for a WAN link is to keep at least 25 % of the bandwidth available for routing table and administrative updates.

As in most architecture's, the more redundancy and availability options designed into the network the better the odds are for a successful installation. The designer must also understand that engineering all of the redundancy options available into the system could adversely affect the performance of the network. For example, adding IP redundancy into the network could increase the jitter because the VoIP packets might take multiple paths to reach the end point. This is not a major concern, but it must be evaluated prior to deploying the VoIP network.

Redundancy features cost real money, so the main task of the design engineer is to make sure the product meets the customer's requirements and at the same time keeps the proposal price competitive. In some cases this could be the difference between winning and losing the opportunity.

The following is a list of questions, thoughts, and ideas that should be considered and reviewed with customers/prospects when designing a VoIP network. It is unlikely that a network configuration will implement every feature on this list, but it's a good checklist to review prior to completing the final design.

VoIP Design Guide Check List

- ❑ **Is the LAN equipment designed to support 99.999% availability?**
  - Is the LAN configured with the following redundancy options?
    - Management modules
    - Links
    - Protocols (i.e., Fast Spanning Tree)
    - Power supplies
    - UPS system (in the event of a power outage) in the wiring closet
  - How are the IP phones going to be powered?
    - Does the LAN switch support in-line power (802.3af)?

      Is it connected to a UPS system?

      Does the IP phone model support in-line power?
    - Is an external power patch panel required?

      Is it connected to a UPS system?
    - Are you using local power?

      Is it connected to a UPS system?

      What is the ratio of IP phones with UPS to IP phones without UPS?

      Are digital/analog terminals intermixed with the IP phones in geographic layout to provide for "emergency dialing" in the event of power or network outages?
  - Is the PBX configured with the following redundancy options?
    - Management modules
    - Redundant IP modules
    - Are the VoIP links connected to multiple LAN switches?
    - Is the switch configured to support battery back-up power?
    - Is there a back-up signaling path configured for all networked sites?
- ❑ **Does the installed LAN equipment support QoS?**
  - Do you know the speed and performance of the installed equipment?
    - Manufacture
    - Product type
    - Link speeds and WAN protocols
    - Routing Protocols
  - What is the QoS design strategy?
    - 802.1p/Q
    - DiffServ
    - Is the priority set and respected on every LAN switch in the network?
  - ToS (type of service) or CoS (class of service) for the WAN
  - Do you have a current local area network diagram? This is a must.
    - When was the network diagram last updated? If it's older then 45 days, ask for an up to date diagram.
    - Has the cable plant been verified to support 100 Mbps Ethernet? (i.e., Cat 5 cable)

❑ **Isolation**

- Do you have an isolated VLAN configured just for VoIP phones?
- Has the excess broadcast traffic been removed from VoIP VLAN
  - Is IP multicast support enabled on the LAN?

❑ **Does the installed WAN support QoS?**

- Do you have a current wide area network diagram? This is a must.
- Has the packet forwarding latency and jitter been verified not exceed the maximum tolerance of the 200 ms. An IP audit is a requirement for all WAN connections.
- Is guaranteed bandwidth, packet forwarding rate and capacity specified for all WAN links? A good rule of thumb is to have a 25% available for overhead and routing table updates. Please refer to Table 1 for the bandwidth required for each codec.
  - Let's look at a simple calculation using the 25% rule, using a T1 (1.536 Mbps) as the line speed.
    - 1.536 Mbps – 25% = 1.152 Mbps, so this means that both voice and data must share the available bandwidth.
    - Is a bandwidth manager required?

Acknowledgments

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ARCHITECTS OF AN INTERNET WORLD ALCATEL

Sogecable

Sogecable evolves to IP telephony

Alcatel OmniPCX 4400 systems assure the evolution to IP telephony that the Company wants to achieve in the future

Sogecable, Spain's leading pay television group, has taken advantage of its new headquarters based in Tres Cantos (Madrid) to create a new voice communications environment based on the integration of Alcatel OmniPCX 4400 systems.

The consolidation of Sogecable as an enterprise group was definitive for the high technical executives of the company to reorganise its communication and information technologies (CIT) under a project frame of global streamlining. A big part of this strategy was designed in the year 2000 through a deep analysis of the existing data and voice systems and the infra-structures situation, supp-orted wholly by an enterprise from the CATSA group, under an outsourcing project.

During this analysis, the problems associated to the CIT infrastructures of emerged from different points: high production costs, difficulties with the distribution of the solutions across a complex network of enterprises, the expensive maintenance and the presence of multiple critical points in the voice and data networks. The situation was not alarming but problematic due on the one hand to the necessity of supporting communications networks and lines integrating a huge variety of services such as RDSI, Frame Relay, point-to-point lines or ADSL; on the other hand, the installation of multi-vendor communication elements and, of course, the heterogeneous environment management with many breaking points.
With this thought in mind, the new project was focused to achieve four big objectives: reduce the production costs by the homogenisation of the central process architectures and the electronics to serve the voice and data networks; give high availability to the systems; count with basic infrastructures of high added-value and automating the management of all the systems.

Infrastructures

The new headquarters allowed Sogecable to start its CIT project from a more basic foundation, cabling and network electronics needed to give a high bandwidth able to provide a great quality of service.

The tender was opened and after discussions with the suppliers, the customer chose Alcatel for the new voice infrastructure.

Other factors taken into account were the Quality of Service and the technological future of the products submitted by the suppliers. According to Jon Gabilondo, communications area director, "the enterprise communications systems from Alcatel assured the evolution to IP telephony required for the future".

Voice management

In the evolution to the IP world, Sogecable took a strategic decision: to renew its enterprise PBX. As Jon Gabilondo confirms, "We wanted to continue with the existing services but also to benefit from the functionalities offered by the IP telephony in terms of costs and management capacity. For this reason, the chosen systems should guarantee an efficient process of voice and data with the objective of taking advantage of their integration".

Other factors were also taken into account, such as the messaging integration and the wireless telephony given in the DECT terminals provided by Alcatel, as well as the incorporation of a management software to speed up the configuration changes, the service statistics and the administration of incidents and alarms over the network.

After the analysis of certain offers, the Alcatel OmniPCX 4400 system was the one with best performance.

"Basically, the two key factors in making the final decision, besides the costs, were the management system, which significantly reduces the integration time in the customer's own environment and the experience and knowledge of Alcatel in telephony and data", added Jon Gabilondo.

computing

And his choice paid off! Sogecable recently was awarded by Computing, a Spanish publication, in the category of "Infrastructure Streamlining".

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Crédit Agricole

Bank leader chose to converge voice and data networks onto a single platform to greatly improve client care.

«Our choice for Alcatel was based upon an open and distributed communication platform supporting IP, while leveraging our legacy Token Ring infrastructure and avoiding costly cabling upgrades required by

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buyer.»

Jean Pierre CORDEIRO,
Telecom Director,
Cr檢it Agricole Pyr輔榮s Gascogne

Challenges

- Replacing separate voice and data networks in a mission critical banking environment to reduce costs, operational complexity and streamlining vendor support issues.
- Giving 150 branch locations the same service level quality and voice service functions available at corporate headquarters.

Solution

- An Alcatel VoIP network integrating separate networks, key cost savings in several areas (e.g., cabling, PSTN line cost, staff reduction), simplifying network management and establishing a future-proof platform for evolving applications and technologies..

Benefits

- 46% GSM and PSTN line Cost savings
- Homogeneous voice features upgrade.
- Increased network manageability.

Crédit Agricole Pyrénées Gascogne

The leading French bank in customer accessibility thanks to its decentralized management structure and high-density branch office network, Credit Agricole is the first bank of France with more than 16 million customers. The group is a decentralized having three organizational structures; the parent holding company, 45 regional banks, and reporting into these are 9,000 branch offices.

Crédit Agricole Pyrénées Gascogne is one of the largest regional banks in the CA group and one of the largest financial institutions headquartered in Pau, France (100 millions annual revenue). Through its branch network and Internet banking division (IT budget: 10 millions – Telecom budget: € 2 millions), CA provides its customers with a full range of commercial and individual banking products and services.

With 152 locations and some 1800 people, Crédit Agricole offers the most diverse banking services found anywhere in the Pyrénées/Gascogne region.

Such responsibilities is one of many reasons Crédit Agricole selected a converged voice and data network to improve reliability, manageability and information access of its communication network.

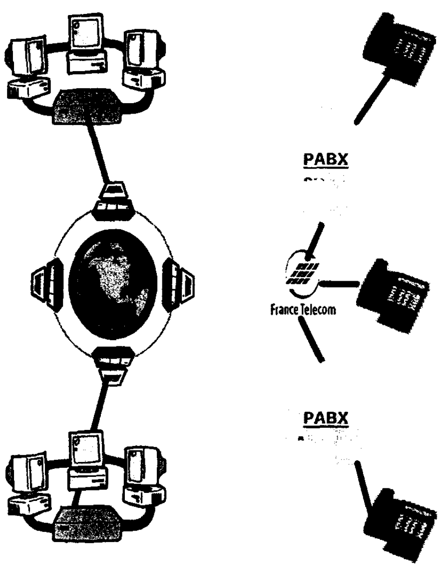
Network aches and pain
Separate voice and data networks are costly and difficult to manage.

Information technology at Crédit Agricole is essential to efficiently delivering banking service. According to Jean Pierre Cordeiro, Telecom director at Crédit Agricole, "It's important that necessary information is made available when and where it's needed to facilitate business decisions. This means providing a variety of communications services among our several sites and agencies, teams as well as to our clients ".

With a communications network playing such a critical role in meeting the needs of clients and staff, it's not difficult to understand why Cordeiro and member IT team found themselves concerned and frustrated with their existing network infrastructure.
Because of its steady growth over the last six years, CA sought to create a unified network capable of providing standardized services to all of its branches. Disparate telephony systems needed to be replaced, and essential features such as voice mail, automated attendants, and four-digit dialing needed to be implemented while maintaining branch independence.
Crédit Agricole's separate voice and data networks covered 152 sites with disparate voice features making the infrastructure costly and cumbersome to manage and change. The headquarters were equipped with 4 Alcatel OmniPCX 4400 while the sites hosted 112 Alcatel 4200 PBXs. The numerous Headquarter-site calls increased the phone subscription fees up to € 305 K and the access line link charges accounted for 11% of the total telecom budget.

"What's more" said Cordeiro, "we found that 30% of our customer calls were abandoned due to discrepancies in voice features levels". These discrepancies were created due to budgetary constraints forcing work around and short cuts that over time resulted in a complex telecom network of PBXs with different software releases, configurations and delays in change implementation".

On the data side, discontinued service support for our existing private IP network was in serious jeopardy. It was absolutely necessary to replace our network end-to-end or be without vendor support".



2 separate networks

"Our goal", Cordeiro continued, "has been to put in place a common infrastructure for all voice, data and video services provided and then take advantage of converging technologies as industry standards became established and adopted. When we took a serious look at Alcatel products line, it was obvious that it was not only a nice fit for our organization but ultimately and most importantly would greatly reduce the cost of providing services to our staff and community".

CA decided to explore Voice over IP (VoIP) as a way of putting excess bandwidth to work, improving operating efficiencies, and reducing operating expenses.

The vendor/Integrator of choice
Alcatel solution delivers more functionality and overall value

Crédit Agricole launched a comprehensive selection process evaluating Vendors and Integrators to determine the best means of meeting their long-term network objectives. "In the final analysis, Alcatel (and NextiraOne) not only delivered more options than we had asked for, but the company did so with a single server configuration, full end-to-end manageability and a very simple-to implement network plan", Cordeiro explained.

The anatomy of a converged voice over IP (VoIP) network
Crédit Agricole and Alcatel partner for an end-to-end network migration.

To provide the infrastructure and quality of service to support advanced voice communications, Alcatel simplified Crédit Agricole's network with an open and distributed appliance sever platform versus a monolithic closed PABXs architecture.
The Alcatel OmniPCX Communication server supports a wide spectrum of leading

application and telephony features over IP networks and traditional time division multiplexed (TDM) networks for slower IP transformation projects requiring support of existing PBX platforms.

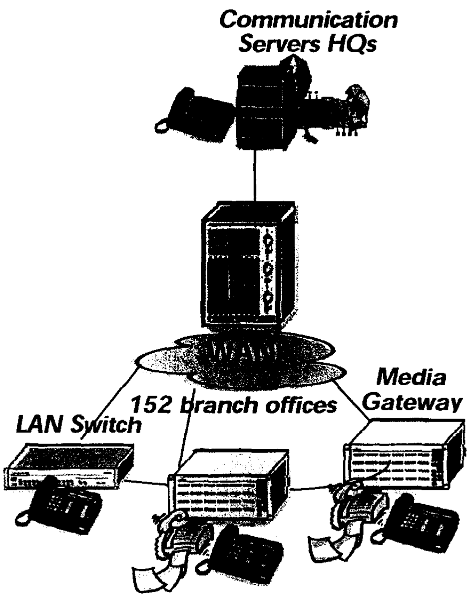
Crédit Agricole chose a fast IP transformation path resulting in seamless network integration, simplified network management, greater flexibility in feature deployment, and reduced costs for supporting its users across multiple branches and remote access locations.

With more than 1,800 telephone nodes on its network Crédit Agricole is also migrating its voice mail functionality to the latest converged voice and data technology available.

Results

With the deployment currently in progress, CA has begun to realize the following voice and data convergence benefits. Operating expense reductions

- Streamlined, any to any corporate communications.
- Full bandwidth utilization of the data network.
- Standardization of voice mail and Auto Attendant services across the network.
- Independent operation of business-critical telephony systems at each branch.
- Cost-effective centralized management of 152 branches from the central site eliminating the need for third-party managers
- Replacement of costly dial-up connections with cost-effective connections over the corporate WAN
- Reductions in station-to-station long distance charges



Abundant network benefits
Converged voice and IP network puts Crédit Agricole Telecom in control, impress users with advanced functionality, and prepares for the future

During Crédit Agricole one-year deployment process, many benefits of the Alcatel solution emerged. Cordeiro says Crédit Agricole continues to discover new advantages.

"Our biggest benefit was the immediate feeling and realization that we – our IT staff – are now in control of the network" said Cordeiro. The network is easy to implement, configure, manage / administer and recover. "The management of a single converged voice and data network is freeing so much of our time. We used to fight fires and do crisis management. Now we are actually able to be proactive".

The project results were as follows:
The investment payback was 18 months.
The project costs and benefits were as follows (in euros):

	Before	After
Access Line Subscriptions	€ 305 K	€ 214 K
Communications HQ/Sites	€ 107 K	€ 0 K *Free !*
Communications Fixed/Mobile	€ 229 K	€ 153 K *Centralized Gateway*
Communications Others	€ 732 K	€ 610 K *Site to site communications*
TOTAL	**€ 1373 K**	**€ 977 K**

[redacted]

Equipment related costs	
Hardware:	€ 350 K
Software:	€ 122 K
Consultancy	
Installation & support:	€ 218 K
Communication charges	€ 0 K
Staff costs, system admin	€ 80 K
Maintenance & MACs:	€ 70 K
Total investment:	€ 690 K (€ 5 K/ site).

"Centralize the PSTN access lines for the decentralized outgoing calls " that was the main cost benefit said Cordeiro.

Cordeiro cites other important benefits in addition to substantial savings in costs over the original solution:

Integrated infrastructure. The solution lets Crédit Agricole operate a converged voice and data network and manage it as one 'flat' network. It will make possible the delivery of integrated services, and positions Crédit Agricole to fully leverage next-generation applications such as unified messaging, mobility, centralized directories and other emerging services.

Ease of administration. The network management solutions make monitoring, troubleshooting, managing configurations "a true networking lifesaver" says Cordeiro.

Effective, satisfied users. "I have people lined up and waiting for some of the applications our new network makes possible" said Cordeiro.

VoIP Communications Costs	39 671 €
Communications costs without VoIP	73 521 €
Savings	**46.04%**

November 2002 Telecom costs

Convergence project Pre-requisite

The key success factor cited by Cordeiro was the obligation to succeed data/voice teams convergence before the network convergence. The organization needs to control and be sure of its data network. A network managed by a third party is not trustworthy.
It is also necessary to invest patience and communicate with the users.

For further information see
www.alcatel.forum/

At a glance

- Crédit Agricole Pyrénées Gascogne.
- Pau, France.
- www.credit-agricole.fr
- 152 sites.
- 1 800 employees.
- Provides banking services.
- Aims to be France's leading banking services.
- Project ROI: 18 months
- Project investments: € 600,000
- Savings: € 400, 000 annually
- Providers: Alcatel/NextiraOne

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