Cisco Voice Solution

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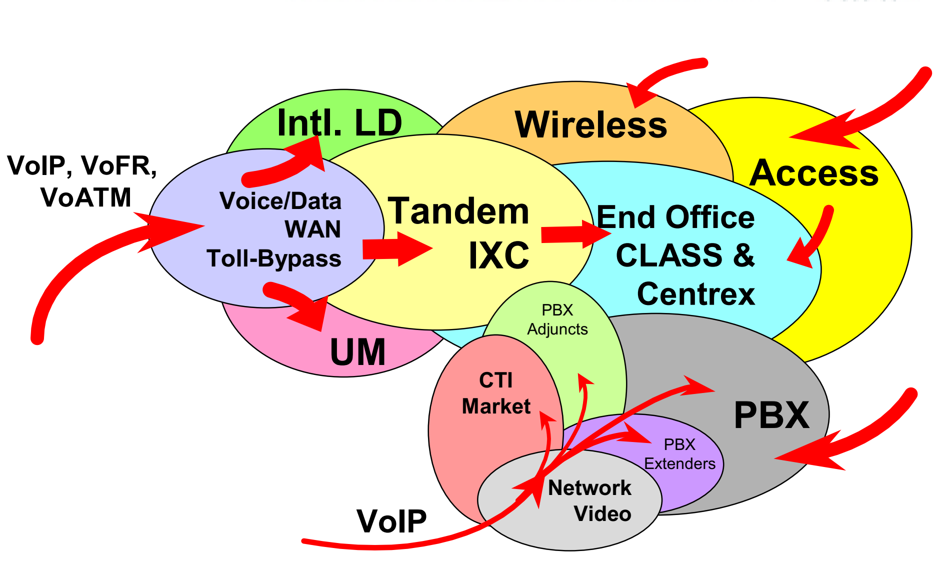
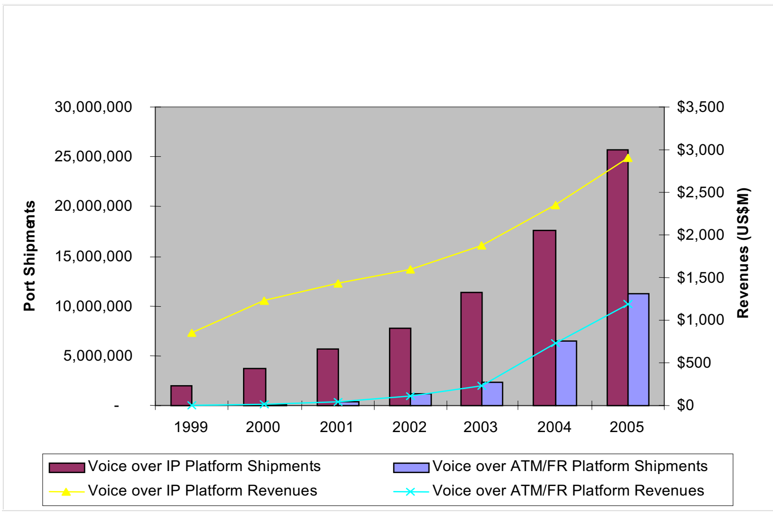
목 차

- ***Voice Market Trend and Status***
- Cisco VoIP (Toll Bypass) Solution
  - ✓ VoIP 개요 및 Technical Issues
  - ✓ Design & Application
  - ✓ Technical & Product Update
- Cisco IP Telephony Solution
  - ✓ Cisco IP PBX 소개
  - ✓ Product & Function Update
  - ✓ Design & Application
  - ✓ Case Study

Cisco VoIP Market 방향



VoIP 와 VoATM/FR 이용 현황 및 예측 비교



Probe Research, 2000

Cisco is Leading Company in VoIP world

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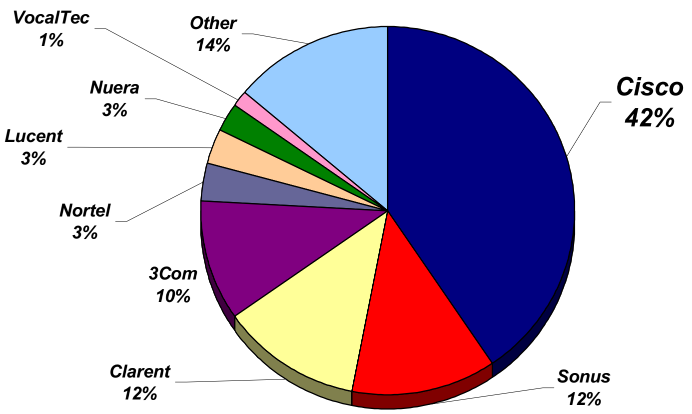
CY01 Q2 WW VoIP Equipment
(includes LAN Telephony, Enterprise & Service Provider)
Total Revenue Estimate = $400M for Q2
Source: "Voice-over-IP Equipment", Synergy Reserach, August 2001



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Cisco VoIP (Toll Bypass) Solution

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VoIP 개요

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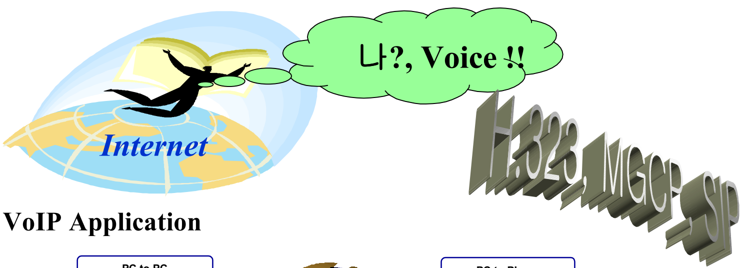
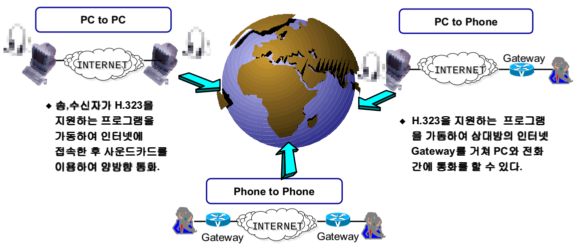
❑ **IP (Internet Protocol)기반의 음성 전송 기술**



❑ **VoIP Application**



Technical Issues
음성 품질의 영향 요소 (Losses, Delay, Echo)

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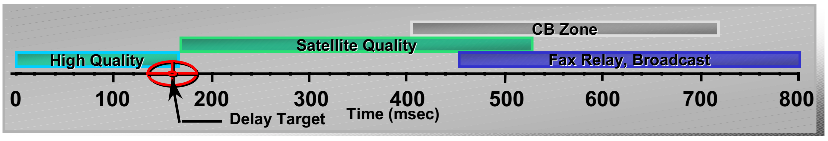
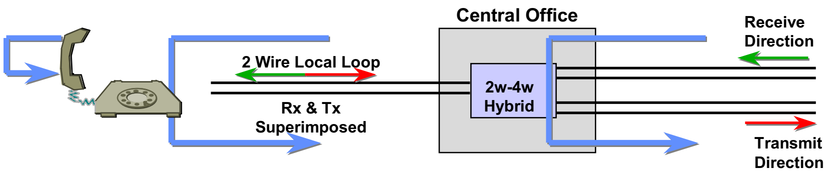
1. Packet Delay (최대 150 ~ 200ms 허용)



2. Packet Losses (약 3개까지의 Packet Loss는 MOS값을 2.4까지 내림)

3. Echo Problem



Technical Issues – Codec & Bandwidth

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✓ Wan 대역폭을 최소화하면서 Data, Voice통합을 위한 음성 압축 기술

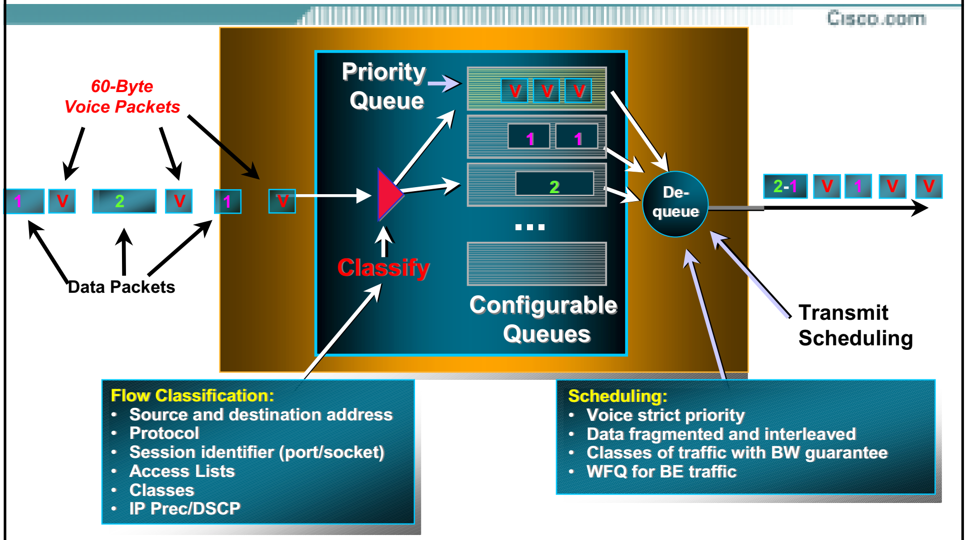
✓ 각 Codec별 Bandwdth

CODEC	Sampling Rate	Voice Payload in Bytes	Packets per Second	IP Bandwidth per conversation
G.711	20 msec	160	50	80 kbps
G.711	30 msec	240	33	53 kbps
G.729A	20 msec	20	50	24 kbps
G.729A	30 msec	30	33	16 kbps

✓ WAN 오버헤드까지 계산 했을 경우의 Bandwidth

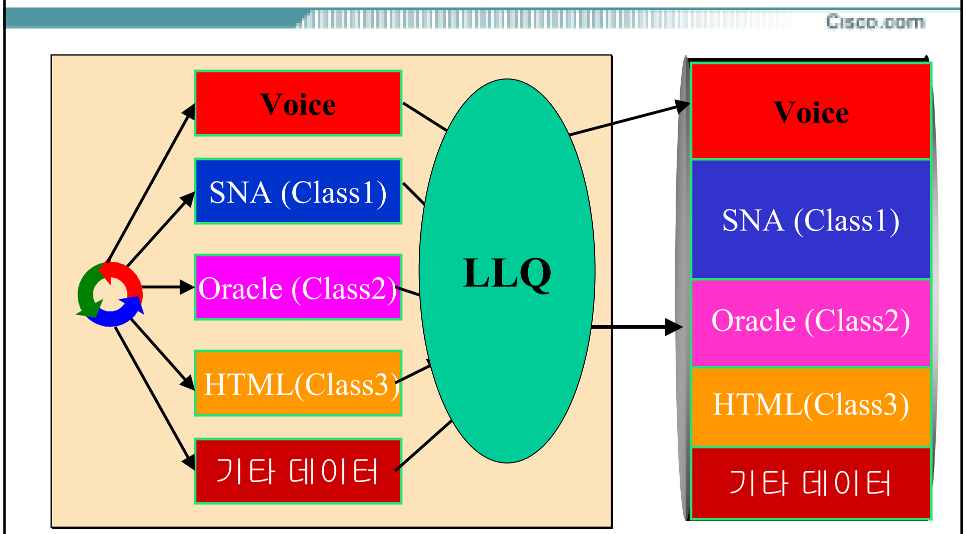
CODEC	Ethernet 14 Bytes of Header	PPP 6 Bytes of Header	ATM 53 Bytes Cells with a 48 Byte Payload	Frame-Relay 4 Bytes of Header
G.711 at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps
G.711 at 33 pps	56.5 kbps	54.4 kbps	70 kbps	54 kbps
G.729A at 50 pps	29.6 kbps	26.4 kbps	42.4 kbps	25.6 kbps
G.729A at 33 pps	19.5 kbps	17.4 kbps	28 kbps	17 kbps

Technical Issues – QoS (LLQ)

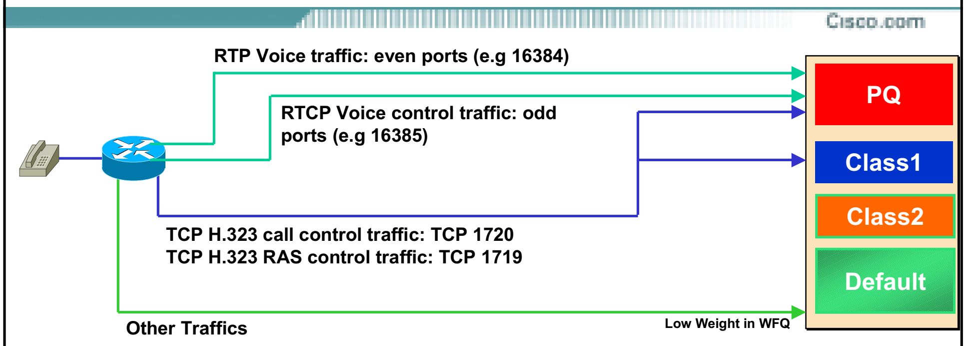


• *BW and latency guarantee - sufficient for voice*

LLQ동작 원리 (Voice + Class별 Data)



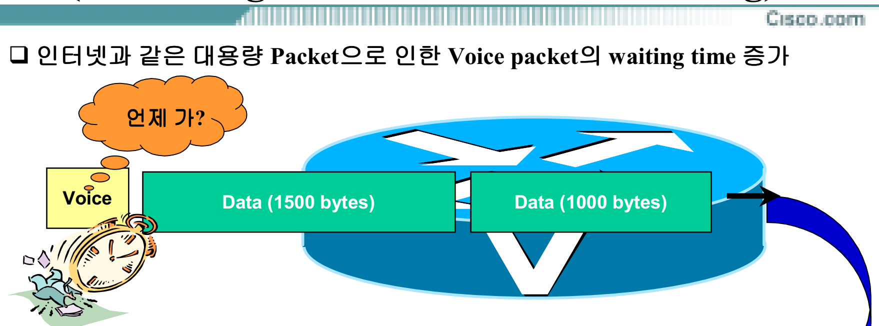
Voice Bearer and Signaling용 Port 번호



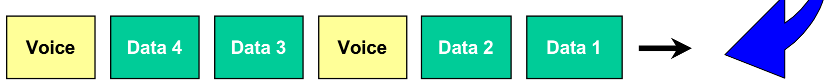
- Bearer: RTP traffic of the media stream (UDP 16384 ~32767)
- Signaling:
  - RTCP: RTP의 홀수 포트
  - TCP: H.225/H.245 Traffic (RAS:1718~1720, H245:11xxx)
  - Skinny = TCP 2000-2002
  - MGCP = UDP 2427, TCP 2428

Technical Issues
LFI (Link Fragmentation and Interleaving)

- 인터넷과 같은 대용량 Packet으로 인한 Voice packet의 waiting time 증가



- Data Packet을 Voice와 같은 크기로 맞춤



- 적용 기술 (Link speeds up to ~ 1.5M)
  - MLPPP LFI
  - FRF.12

CAC (Call Admissions Control)

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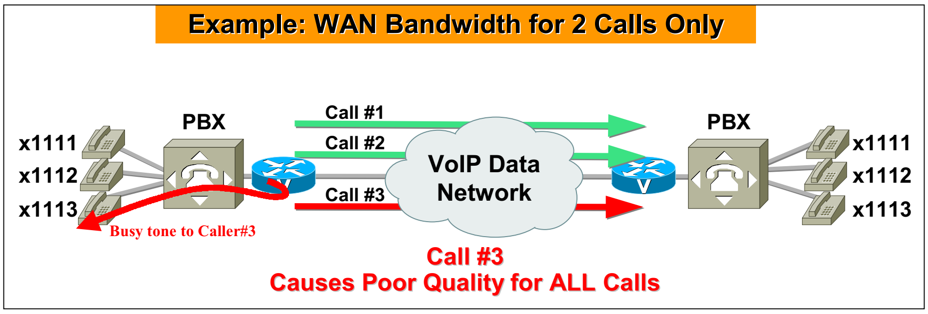
Call Admission Control (CAC) 는 QoS와 같이 VoIP의 음질을 높이기위해 네트웍의 Resource에 따라 Cisco에서 제공하는 Voice Control 기술



Cisco의 Voice QoS Summary

Low Speed WAN Links(2Mbps이하)

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Frame Relay

Classification & Marking
- IP Precedence/DSCP
- Access Lists (TCP/UDP ports)

Traffic Shaping
- Frame Relay traffic shaping
- Shape to CIR on PVC carrying voice

Queuing (and Scheduling)
- LLQ priority class

Voice QoS
- FRF.12
- CRTP (If desired)
- VAD (If desired)
- DTMF Relay (depending on codec)

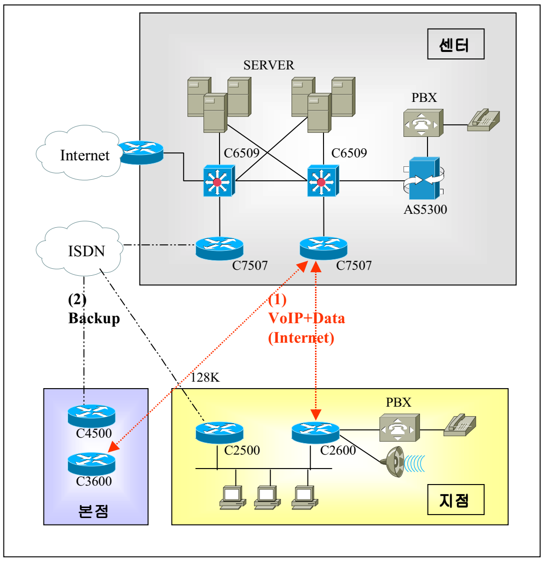
Point-to-Point (MLPPP)

Classification & Marking
- IP Precedence/DSCP
- Access Lists (TCP/UDP ports)

Queuing (and Scheduling)
- LLQ priority class

Voice QoS
- MLPPP LFI
- CRTP (If desired)
- VAD (If desired)
- DTMF Relay (depending on codec)

Case Study for QoS



Combined Internet data & Voice using each line

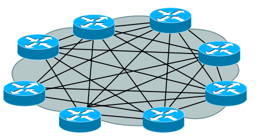
-Data, Voice 동시 수용

- R1에 국내* 지점 적용,
- R2에 국내* 지점, 해외* 개 지점 적용
- R3에 국내* 지점, 해외* 개 지점 적용

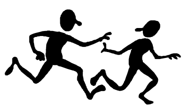
- 적용된 QoS
  - o dCRTP
  - o FRF.12
  - o LLQ (PQ-CBWFQ)

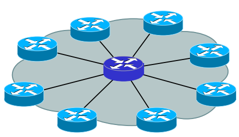
VoIP Design (Gatekeeper사용)

- 기능 (Functions)
  - Address Translation (Call Routing)
  - Admission Control (Gateway registration)
  - Zone Management
- Platform (26/36xx, 3810,72xx)



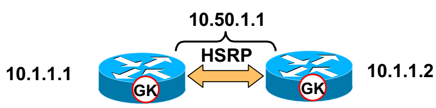
Small Network - Gateways only



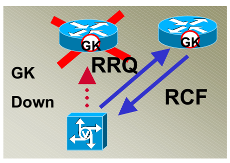


Small Network - simplified with a Gatekeeper

- Fault Tolerance
  - HSRP GateKeeper



- Alternative GateKeeper



VoIP Design시 고려 사항

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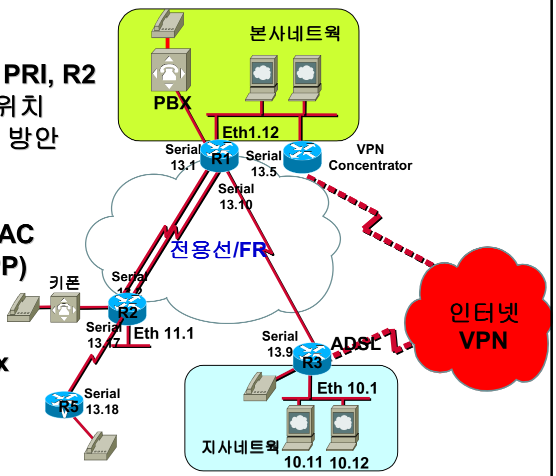
VoIP Design시 고려 사항

- PBX와 연결: FXS/FXO, E&M, PRI, R2
- Call control: GK사용여부 및 위치
  Dialplan의 적용 방안
- Codec: G.711/G.729/G.723
  (Call당 대역폭 결정)
- QoS: LLQ, cRTP사용 여부, CAC
- WAN구간: LFI (FRF.12/MLPPP)
- 장비 선정
  - ✓ Branch:1750/2611/3620
  - ✓ HQ:3660/3640/72xx/75xx
- Service: FAX, IVR사용 여부
- Voice Backup: VPN or PSTN



VoIP Application (방송 솔루션)

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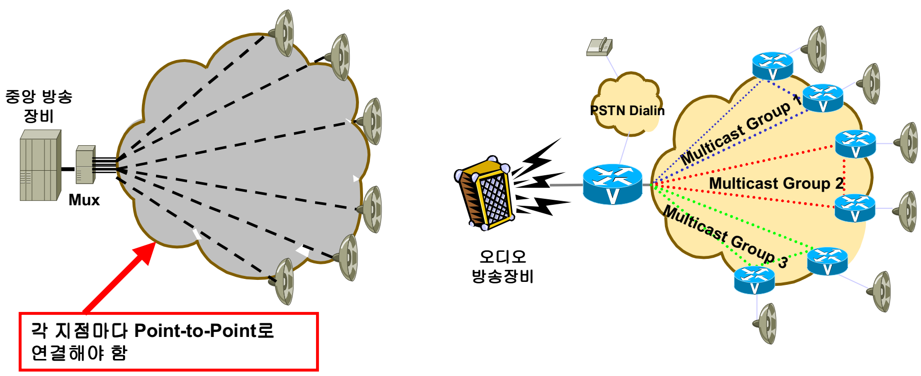
- ❑ IP Multicasting을 이용한 방송 솔루션 (금융기관, 관공서, 등등)
- ❑ 기존의 **Point-to-Point**방식보다 회선사용의 **Cost saving** 및
  관리 유지 비용 감소

Multicast를 이용한 Point-To-Multipoint 방식



Cisco의 방송 솔루션 Summary

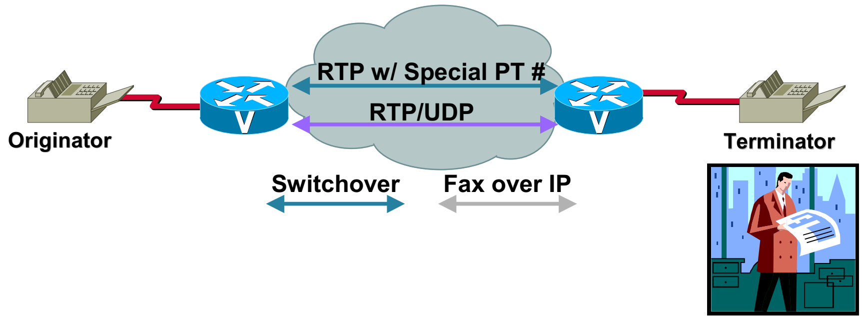
Traditional Problems	Cisco's Solution
Expensive multi-drop circuits	Cisco의 IP Multicasting을 이용한 Point-to-Multipoint
Difficult to manage	Cisco Network Management
Proprietary solution	Standards such as IP, RTP, G.729, VAT etc.

❑ Cisco방송 솔루션의 장점

- 방송을 위해 기존의 VoIP에서 쓰는 여러 Codec을 사용 가능 -> 다양한 압축
- 하나의 회선으로 여러 지점에 동시에 방송 가능 (Cost Saving)
- 완전한 End-to-End방송이 가능 함으로 음질 향상 및 관리 편의성
- IP Multicasting을 사용함으로써 WAN Bandwidth의 절약

Fax Over IP (Fax Relay)

- Fax relay을 위한 두가지 방법: Cisco Protocol and T.38
- Ingress GW receives T.30 & T.4 fax signals from PSTN
- Fax signals get demodulated by DSP, packetized, and transmitted across VoIP network as data
- Egress GW decodes the data stream and re-modulates back to fax signals to be transmitted to the PSTN



Technical & Product Updates

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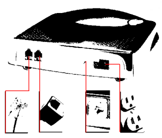
➢ **Cisco 7500 시리즈에서의 VoIP QoS 향상**
- IOS 12.2.x부터 VIP Module에서의 dCRTP/dLLQ 적용 가능으로 VoIP 음성품질 및 Performance 향상 (현재 은행 권에서 기존의 7500 라우터에 VoIP기능을 적용 사용 중)
- **1Port Voice Module → (PA-VXA-1TE1)**

➢ **VoIP망에서의 Redundancy 향상을 위한 기술 - 안전성**
- Gatekeeper에서 HSRP적용 가능
- Alternate Endpoint 기능 (Gateway가 죽었을 경우 다른 Gateway로 호를 보냄)
- Alternate Gatekeeper 기능
- PSTN Fallback 기능 (Hairpin 기능)

➢ **Performance 향상을 위한 H323의 새로운 기술 적용**
- H323 FastStart 기능 (call setup시간을 단축)
- H245 Tunneling 기술 (불필요한 signaling을 제거)
- Gateway와 Gatekeeper을 한 platform에서 동시에 사용가능

➢ **NM-HDA With 12FXS+4FXO for C2600/3600 Series**
- 현재의 아날로그 포트숫자를 증가시킨 모듈로 환경에 맞게 FXS/FXO의 숫자를 조절가능

➢ **Small Office용 전용 Gateway (ATA Series)**
- Analog 2 Port 지원
- H323, SIP 지원



Cisco VoIP Summary - 이점

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- **기존 IP Network Infra (Router)의 이용으로 인한 장비 사용 효율의 증가**
  - ☞ 빠른 ROI로 Cost Saving

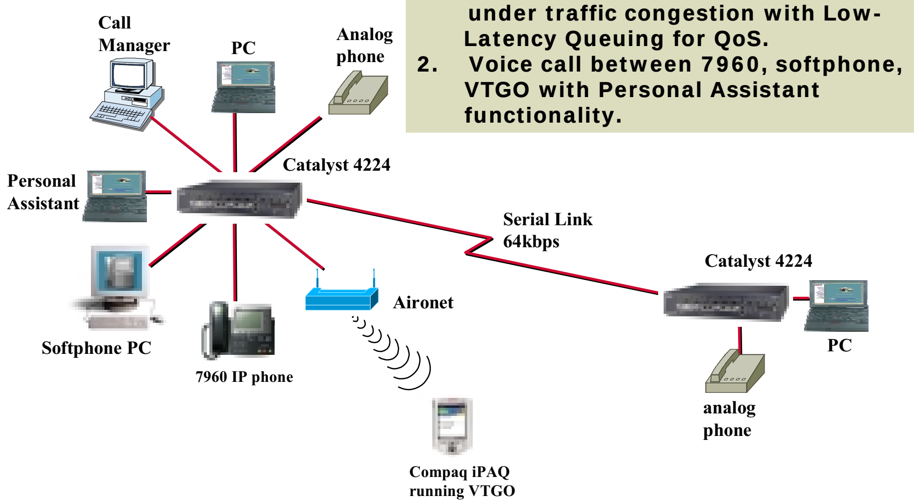
- **확장성 및 호환성 제공**
  - Open standard 을 이용 (PRI, MFC R2, H323/MGCP/SIP)
  - 전세계에 VoIP Gateway Market에서 인정된 호환성 및 안정성
  - 대부분의 3rd party들은 Cisco의 GW와 호환성 테스트를 하고 있음
- **음질 향상을 위한 다양한 tool제공**
  - QoS: 전용 장비를 사용할 필요 없이 자체의 다양한 tool사용 가능
    -> Voice에 QoS를 적용할 경우 QoS 전용 장비를 이용할 수 없음
  - 앞선 IP기술을 이용 (MLPPP/FRF.12/cRTP/RTR(음질테스트packet))
- **TDM과 같은 다양한 Call Control기술 제공**
  - Gatekeeper의 다양한 표준 지원 및 backup 기능 제공
    -> GUP, LLQ, Alternate GK, Alternate Endpoint
  - API를 공개 함으로써 자신에 맞는 control/billing application을 개발 가능
    -> GKTMP, RADIUS
- **다양한 Application 지원**
  - IP Multicast를 이용한 방송 솔루션, gateway자체의 IVR 솔루션 등,

Cisco Voice Gateways

Demonstration Overall Topology

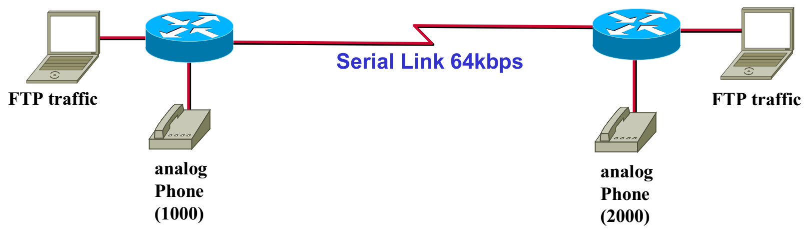
Demo plan:

1. Voice call between analog phones under traffic congestion with Low-Latency Queuing for QoS.
2. Voice call between 7960, softphone, VTGO with Personal Assistant functionality.



VoIP Call With QoS over Low Speed Wan Link (Low-Latency Queuing)

1. Catalyst 4224 (2)
   * VIC-2FXS voice/fax interface card
   * WIC-1T
2. Analog phone (2).
3. PC for traffic generation (2)



Cisco IP Telephony Solution

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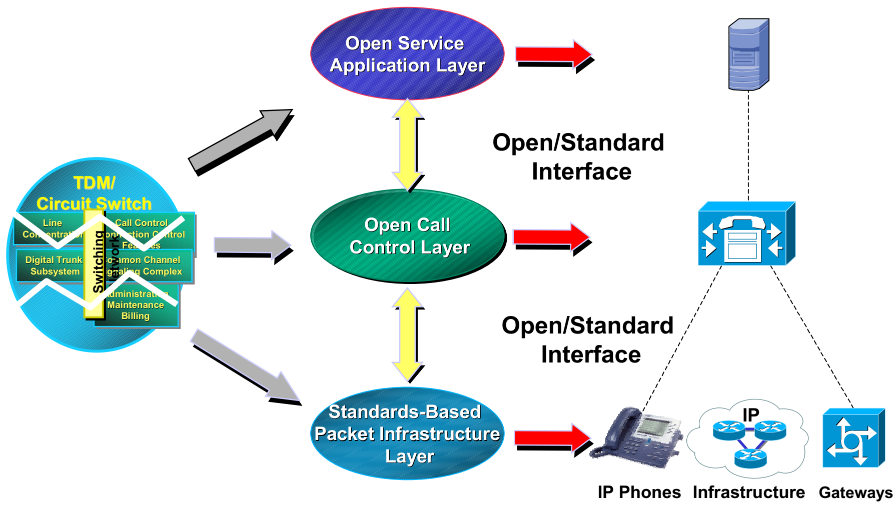
Voice기술의 새로운 접근 방향



Cisco IP PBX 소개 – Call Manager

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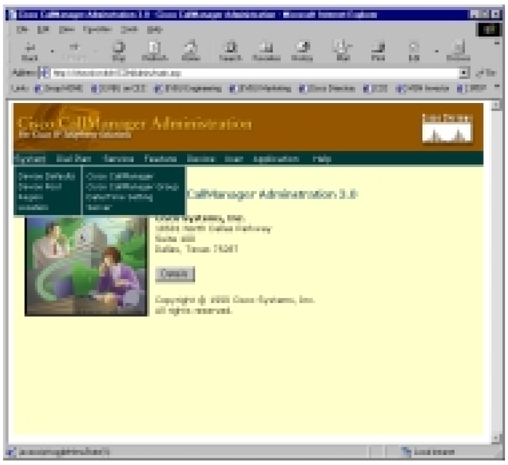

- Standard HTML을 이용하여 설치, 구성, 관리 가능
- 지원 Protocol (Skinny, H.323, MGCP)
- Media Convergence Server platforms에 기본으로 설치되어 제공되며, 제공되는 CD를 이용하여 손쉽게 다시 구성을 할 수 있음
- 일반적으로 사용되는 Telephone관련 기능은 물론, Standard를 따르는 타사의 Application 사용 가능 (TAPI, JTAPI)
- 시간당 50,000 call이상 지원
- MCS 한대 당 2500 device 지원
- Clustering를 이용하여 Load Sharing 및 Fail over 지원 – 8 대
- **한 cluster당 10,000 device 지원**

Cisco IP PBX 소개 -IP Phone

Cisco.com

7910

Expansion Module 7914

7940

7935

7960

SoftPhone

Cisco IP PBX 소개 – Inline Power

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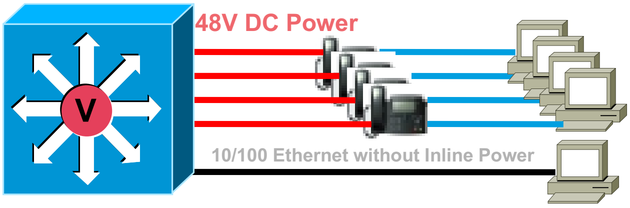
Catalyst Inline Power—Provides DC Power over Standard UTP Ethernet

Catalyst 6000
Catalyst 4000
Catalyst 3500



Catalyst inline power Ethernet modules:

- Catalyst 6000 10/100 modules can be field-upgraded
- Catalyst 6000 support RJ-45 and RJ-21
- Provides power over the signal pairs
- Catalyst power patch panel for other switches

Cisco IP PBX 소개 – Voice VLAN기능

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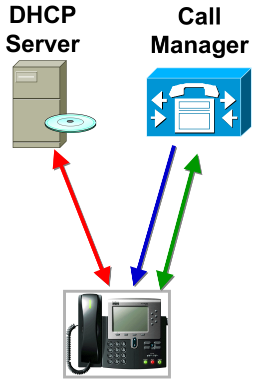
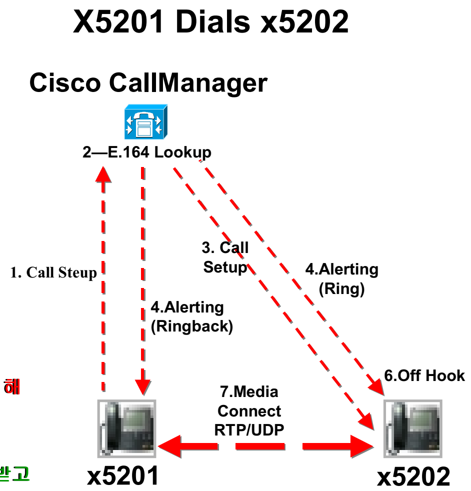
This Feature Provides Automatic Phone VLAN Configuration

- No end-user intervention required
- Extends the benefits of VLANs to IP phones
- Preserves existing IP address topology
- Uses 802.1Q technology between switch and phone

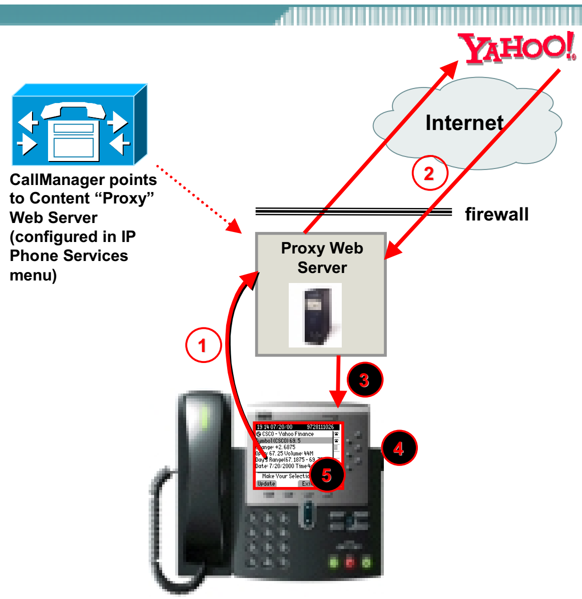
Cisco CallManager Signaling 절차





1. IP Phone은 IP address, gateway정보를 보내기 위해 DHCP에 요청을 한다
2. TFTP에 CM의 IP address을 얻기 위해 정보 요청
3. IP Phone은 CM에 등록하고 Display Template을 받고 Call을 받을 준비를 함.

Cisco IP PBX 소개 – Applications (XML)

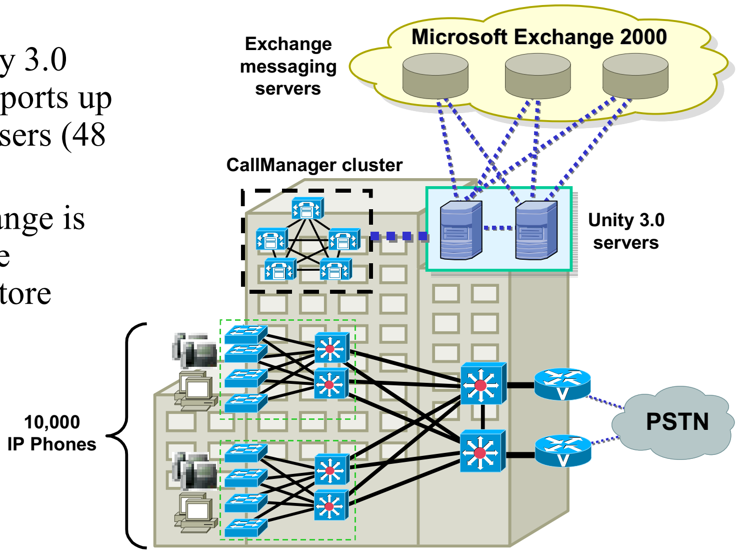


1. Cisco IP Phone HTTP Client performs an "HTTP GET" for a specified URL
2. HTTP Web server processes the request and formats the data returned
3. HTTP Web server returns the HTTP response of XML objects or plain text to the phone
4. Phone parses the HTTP response header for ContentType of "text/xml"
5. Phone presents data and options to the use per the server response

Cisco IP PBX 소개 – Applications : (VMS/UMS)

Cisco.com

- Each Unity 3.0 server supports up to 5,000 users (48 ports)
- MS Exchange is used as the message store



Cisco IP PBX 소개 – Applications :IP-IVR, IP-ICD, PA

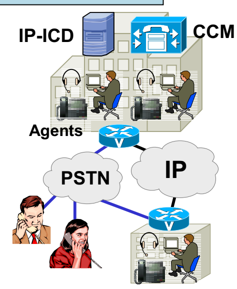
Cisco.com

Interactive Voice Response

- Can be co-resident with CallManager
- Built-in Auto Attendant
- Can be used as part of IP-CC

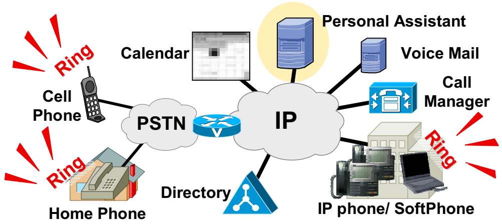
Intelligent Call Distribution

- Basic call queuing
- Agents can be distributed across the WAN
- For larger deployments, use IP-CC



Personal Assistant

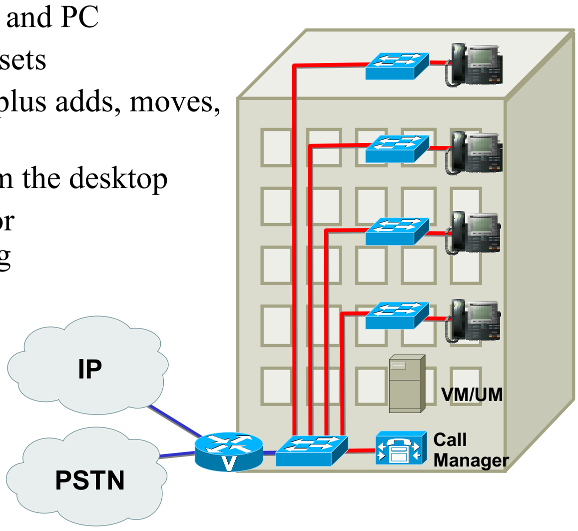
- Route calls according to time of day, calendar, caller ID
- Dispatch calls to user-defined locations (office, cellular, home)
- Integrates with speech recognition system



IP Phone 구성 예1 – Campus 솔루션

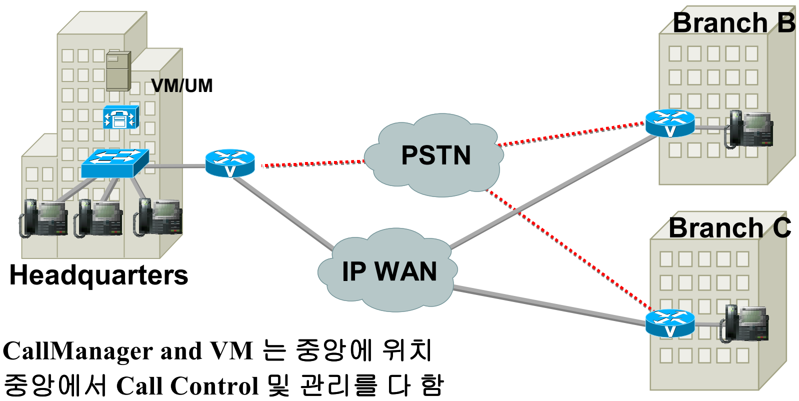
Cisco.com

- Single cable for phone and PC
- Inline power to phone sets
- Ease of IP addressing plus adds, moves, and changes
- Quality of Service from the desktop
- CallManager cluster for redundancy and scaling
- Support for 10,000 users per cluster
- Multiple clusters allowed via H.322



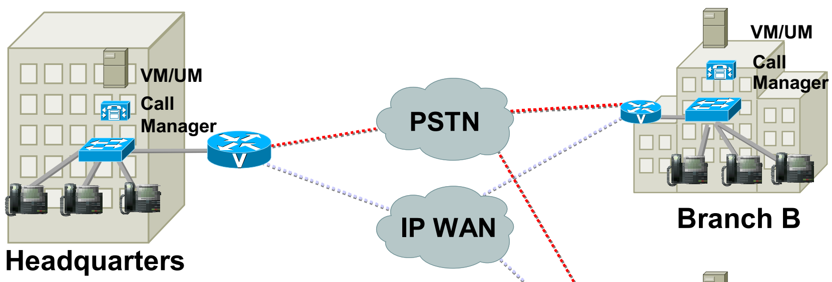
IP Phone 구성 예2 – 중앙 호처리 방식

Cisco.com

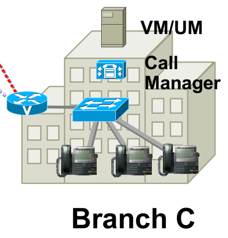


- **CallManager and VM** 는 중앙에 위치
- 중앙에서 **Call Control** 및 관리를 다 함
- 중앙의**CM**에서 **2500**대의 **IP Phone**을 관리 가능
- 지점에서 자체 **Call Control**을 할 필요가 없는 경우
- **50-User**사용자 이하의 지점에 사용

IP Phone 구성 예3 – 분산 호처리 방식

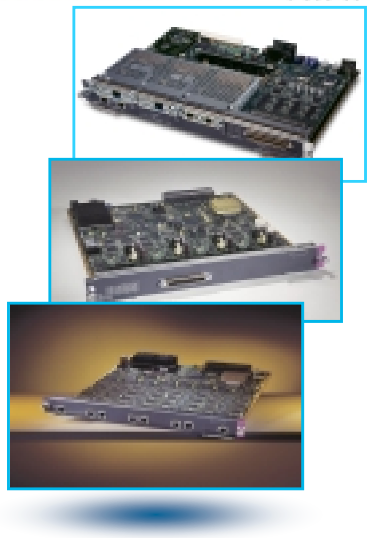


- **CallManager and voice messaging** 는 각 사이트에 위치 하면서 서로 통신 가능
- **Gatekeeper-based admission control** 이 가능
- 각 사이트간에는 압축**codec**을 이용해서 통신 가능
- **IP WAN**이 다운일 경우, **PSTN**으로 **Back-Up**가능
- 지점에서 자체 **Dial Plan**을 가지고 운영해야 할 경우
- 사용자가 **100**명 이상의 사이트에서 사용



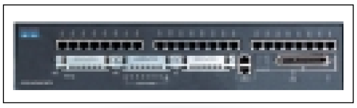
IP PBX Product Update – Gateway (Catalyst)

- Catalyst 6000 – Enterprise Core
- Catalyst 4000 – Regional Core
- T1/E1 PSTN/PBX gateway
- Analog PSTN/PBX gateway (Catalyst 4000/6000)
- Voice Network Service
  - Audio Conferencing
  - Audio Transcoding



IP PBX Product Update – Gateway (4224)

- The Catalyst 4224 is a single-box small branch office solution for offices with less than 24 users
- The Catalyst 4224 provides:
  - –IP Routing, but only IP
  - –PSTN and PBX voice gateway
  - –Onboard FXS connectivity and DSPs
  - –24 Ports 10/100 Ethernet switch with Inline power
  - –VPN and Encryption options
  - –Cisco IOS Survivable Remote Site Telephony
  - –Shares modular VIC and WIC interfaces with the Cisco 1600, 1700, 2600, 3600, and Cat 4K AGM platforms



IP PBX Product Update – Gateway (VG248)

The VG248 is a high density gateway for using analog phones, fax machines, and modems with a Cisco CallManager system

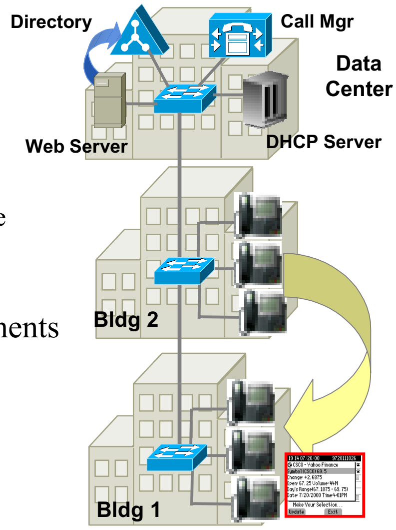
The VG248 provides

- 48 port FXS connectivity
- 1u 19” stackable unit
- 2 RJ-21 telco connectors
- Cisco CallManager 3.1 support

IP PBX Product Update – 새로운 기능 (Extension Mobility)

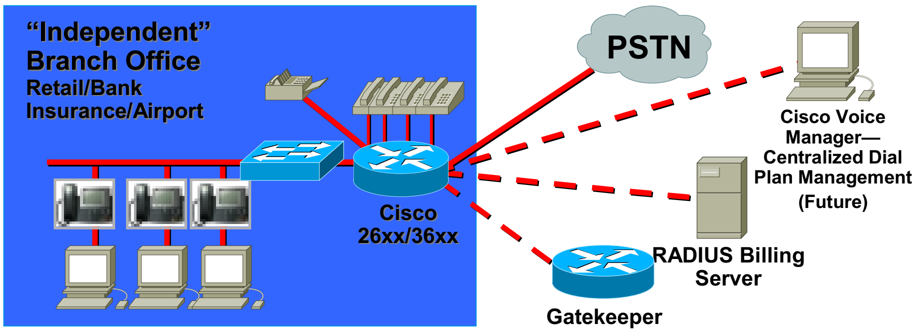
Cisco.com

- Directory based phone log-in
- Application XML based
  - -authenticates to directory
  - -Updates CallManager with user extension
  - -User's extension is pushed to phone where login occurred
  - -User receives all personal settings
- Applicable for mobile office environments
- Offered with CallManager 3.1
- Substantial cost savings benefits



IP Keyswitch for Branch Offices

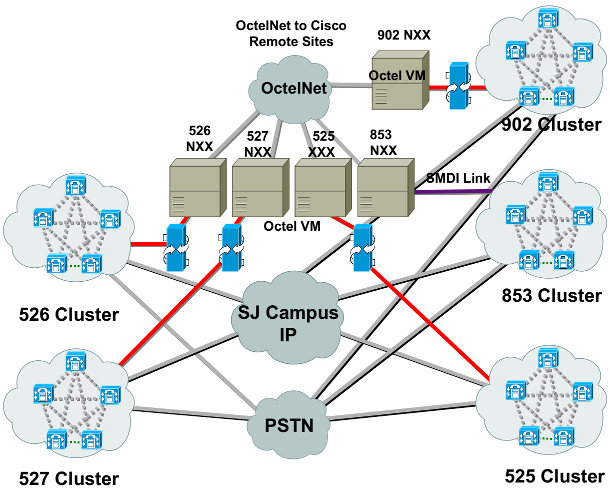
Cisco.com



- **Perfect solution for small "independently" run offices with up to 48 phones**
- Provides call processing on the local router for Cisco 7910/7940 and 7960
- Provides many features for Cisco IP Phones—Xfer, hold, FWD, shared line, multi-line appearance, POTs phones
- Leverages many voice features currently available in IOS such as **DID, DOD, Caller ID, ANI, Calling Name Display, T1 CAS, Analog FXS, FXO**

IP Phone Case Study- Cisco (캠퍼스솔루션)

Cisco.com



(현재의 IP Telephony 환경)

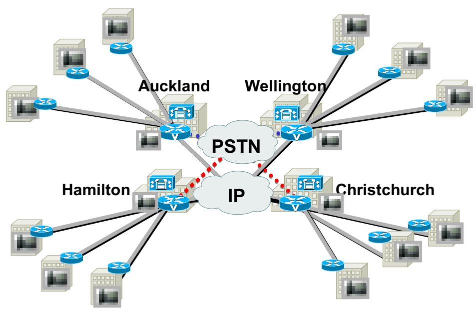
[Next Step]

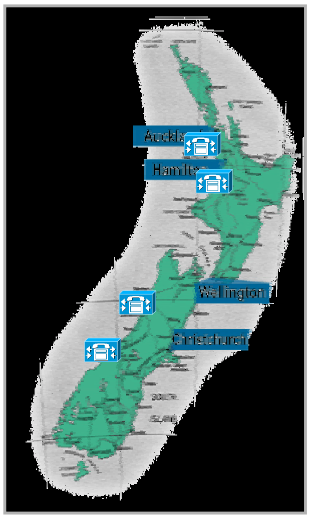
- **Trials of *telecommuter IP phones***
- **Upgrade LAN Infrastructure for *QoS***
- ***Unified messaging* to replace Octel**
- **Replace call-centres with *IP contact center***

IP Phone Case Study- 뉴질랜드 (본사/지사 솔루션)

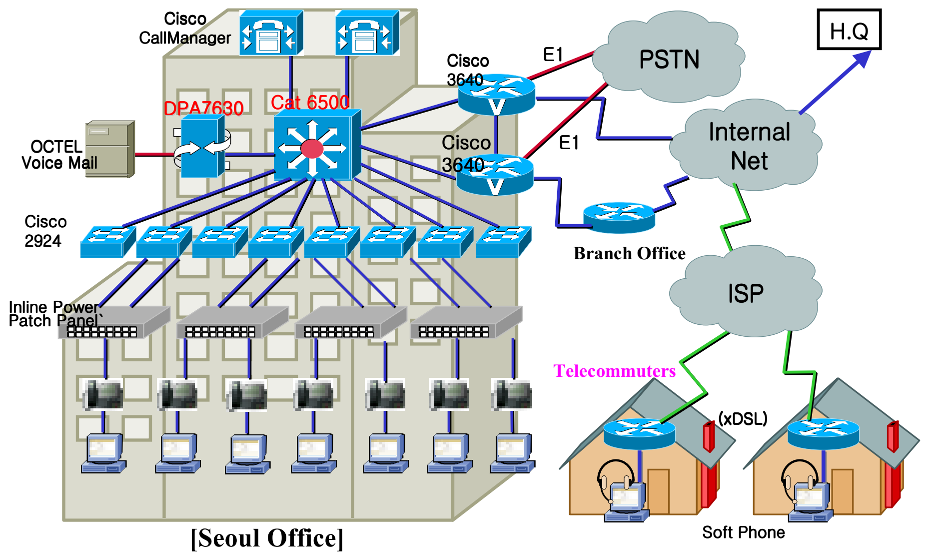
Cisco.com

- 10 MCS-7835 CallManager servers replaced 130 PBX's
- 20 VM servers for 8000 people
- Call processing only at four major hub sites
  - Auckland, Wellington, Hamilton, Christchurch
- Centralized call processing and administration





IP Phone Case Study- 한국 Reference



Cisco IP PBX Summary - 이점

<IT관리자의 고민>

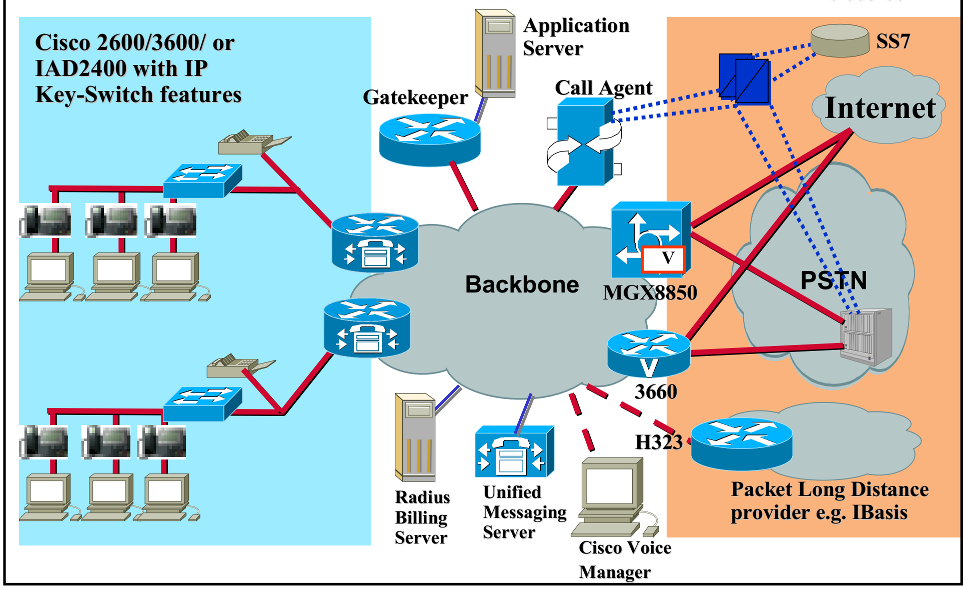
- 사무실의 이전 및 설립 (LAN 및 전화공사에 대한 고민)
- 새로운 근무 환경 조성 (재택 근무에 대한 환경등)
- 전화 및 데이터망유지 보수비 증가
- 영업조직 및 팀간의 이동이 많아 이동에 따른 전화공사가 많고 그로 인한 작업량 증가
- 전사적인 정보 공유의 신속화와 원활한 의사소통으로 인한 생산성 향상을 지향

<Cisco IP PBX 솔루션>

- 설치 비용 및 시간의 감소
- VPN을 이용한 IP Phone (재택근무자에 대한 전화비 감소)
- 유지 및 관리 Resource절감 (data/voice 통합)
- Cisco IP Phone의 Mobility 및 Extension Mobility이용
- Cisco IP Phone의 XML 및 UMS/ VMS를 통한 정보공유의 원활화

VoIP Vision & Target

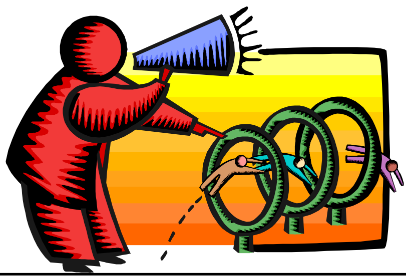
Cisco.com



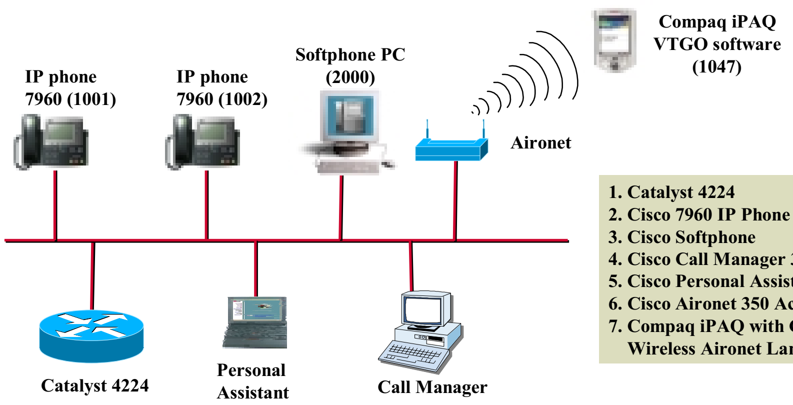
It's Time for VoIP

Cisco.com

- 인식의 전환
  - ☞ Voice는 *IP기반의 Application*
- 향후 IMT-2000등의 *멀티* *Packet Voice에*
- 기존 IP 인프라를 이용한 *다양한 Value a e 서비스로 VoIP 영역의*
- MicroSoft Windows XP의 SIP를 기반으로 한 VoIP 서비스를 계기로 각 모든 End-User 까지 VoIP Application 확장
- *무한한*
  - ☞ 현재 PSTN의 01.% ~ 5%정도의 Traffic 만을 담당하고 있음.



Integrated IP Telephony



1. Catalyst 4224
2. Cisco 7960 IP Phone
3. Cisco Softphone
4. Cisco Call Manager 3.1
5. Cisco Personal Assistant
6. Cisco Aironet 350 Access Point
7. Compaq iPAQ with Cisco Wireless Aironet Lan Card