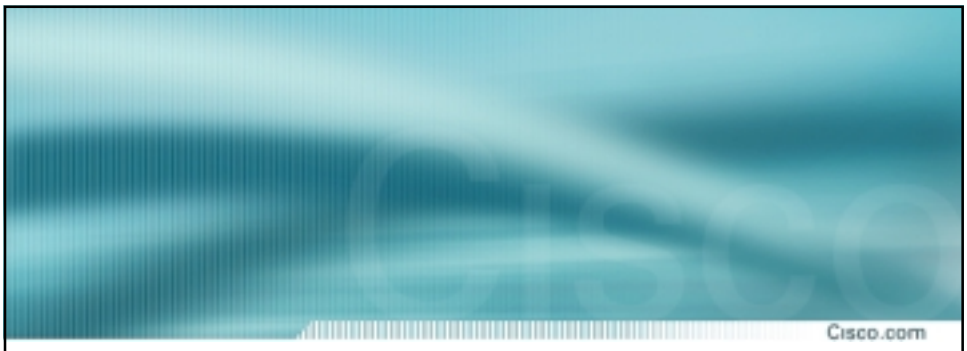

VoIP and Remote Solution

Oct. 26th

Enterprise Tech OPS Team
Lee, Jang-Won
Jwlee@cisco.com

Agenda

- ***Cisco Voice Solution***
  - ***VoIP 소개 & Application***
  - ***Cisco IP Telephony Solution***
  - ***New Product Update for Branch office***
- QoS Update for IPsec

VoIP 기술 소개

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❑ 정의

기존의 음성 통화 망의 TDM(Time Division Multiplexing)방식을 사용하지 않고 기업 또는 공중망의 Intranet/Internet을 이용하여 Voice/Data/Video통화를 제공하는 기술

Internet Phone, IP Telephony라고도 함

- 기존 방식(Circuit switching)
  PSTN망(시외/전용선)을 통해 시내, 시외전화가 이루어짐
- 변경 방식(Packet switching)
  지역간 통화가 사내 또는 ISP 사업자의 인터넷폰 게이트웨이를 거쳐 인터넷망을 통해 이루어 지며 훨씬 적은 대역폭을 사용하므로 전용선 비용의 절감과 융통성이 부여

Applications

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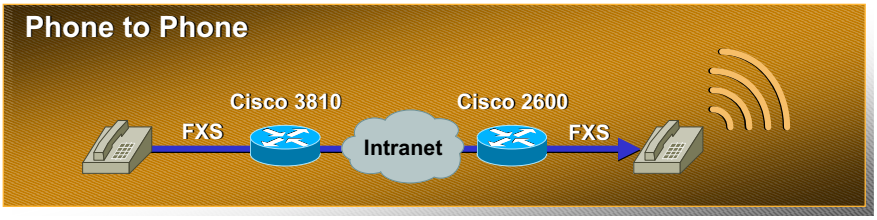
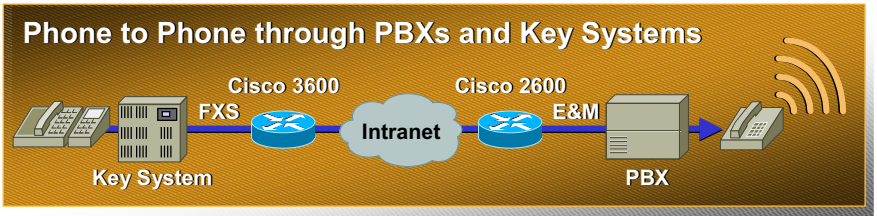
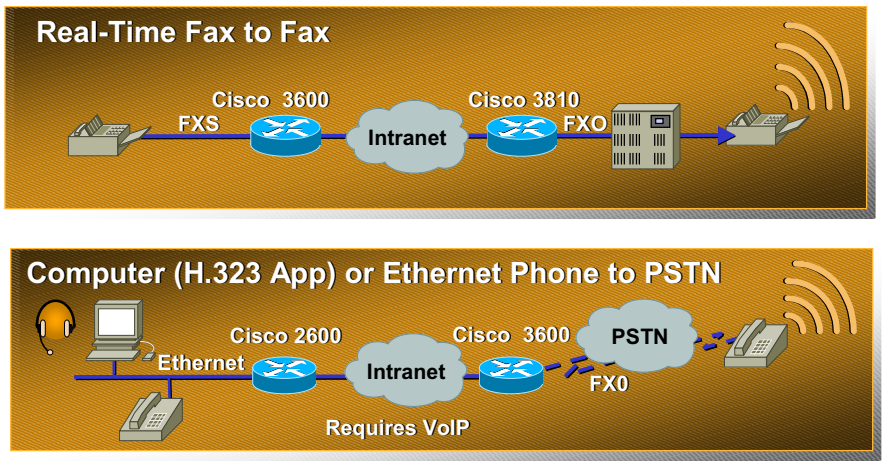
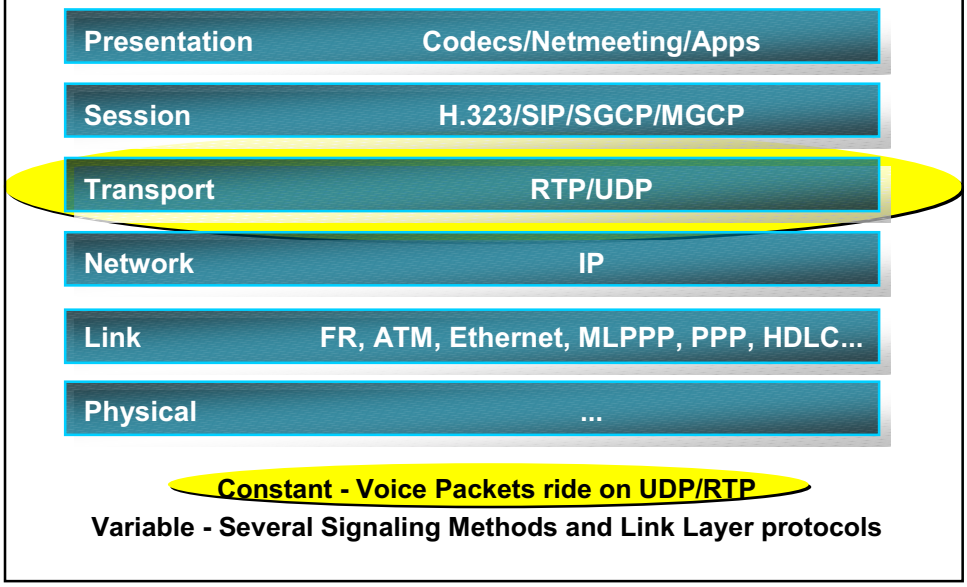




Applications



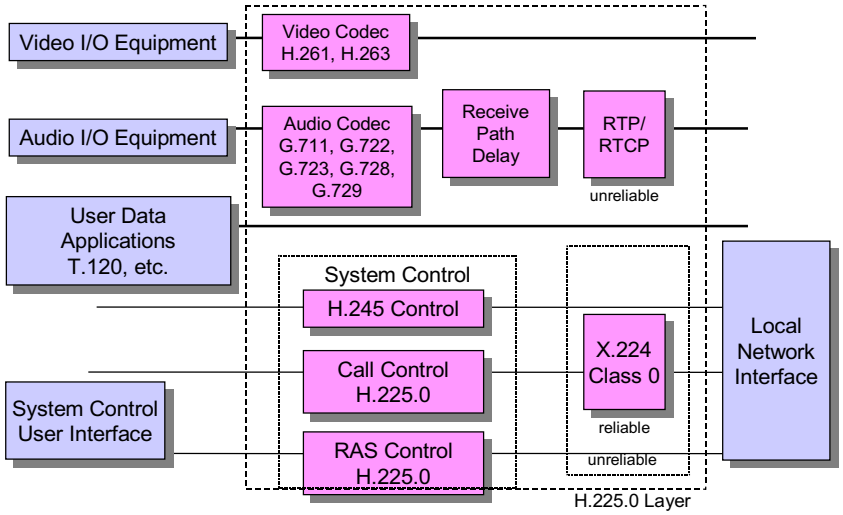
Voice over IP Protocols



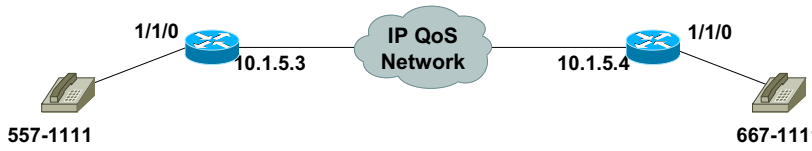
Constant - Voice Packets ride on UDP/RTP

Variable - Several Signaling Methods and Link Layer protocols

H.323 Architechture



H.323 Without RAS - Configuration

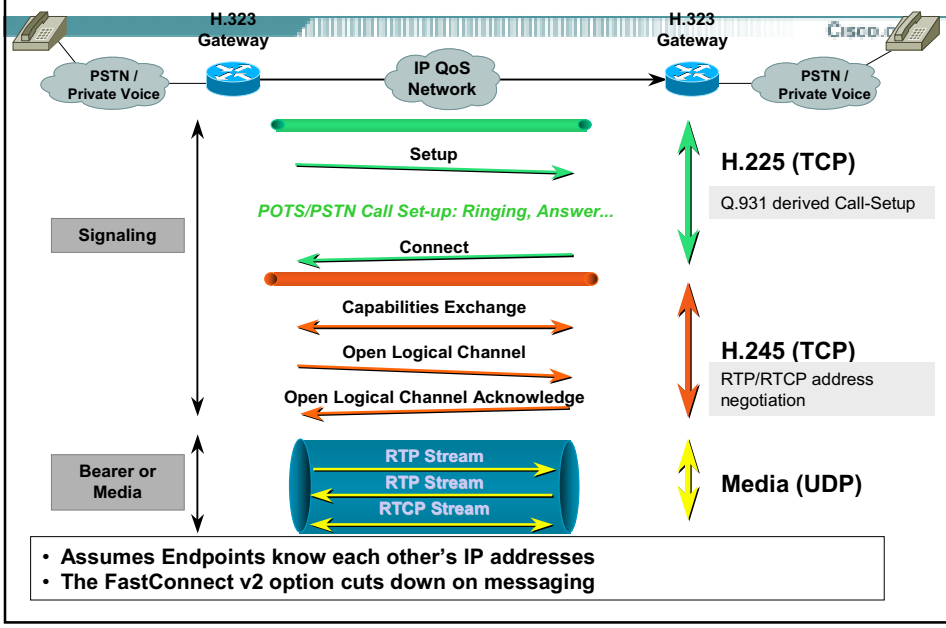


network-peer	dial-peer voice 100 voip destination-pattern 667.... session target ipv4:10.1.5.4
local-peer	dial-peer voice 4 pots destination-pattern 5571111 port 1/1/0

network-peer	dial-peer voice 104 voip destination-pattern 557.... session target ipv4:10.1.5.3
local-peer	dial-peer voice 1 pots destination-pattern 6671111 port 1/1/0

- VoIP Dial Peer points directly to the destination GW's IP address
- Scaling to large networks becomes administratively burdensome

H.323 Without RAS - Call Set-up



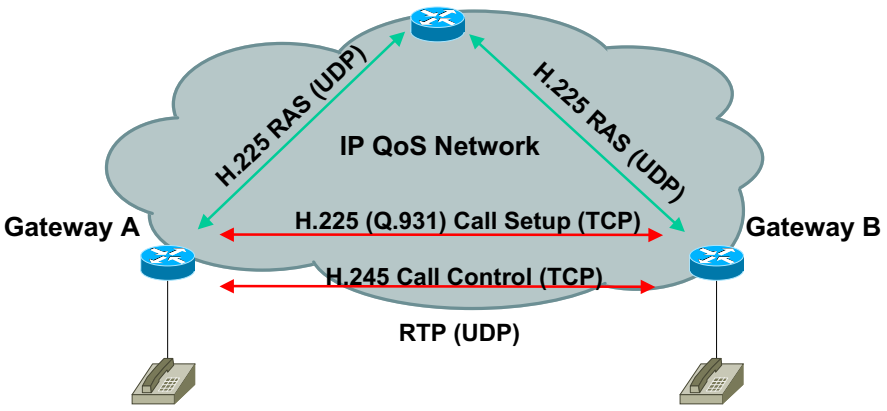
- Assumes Endpoints know each other's IP addresses
- The FastConnect v2 option cuts down on messaging

H.323 With RAS

Registration, Admission and Status

Gatekeeper

Address Translation: Every GW needs to know only about the GK, not about all other GWs



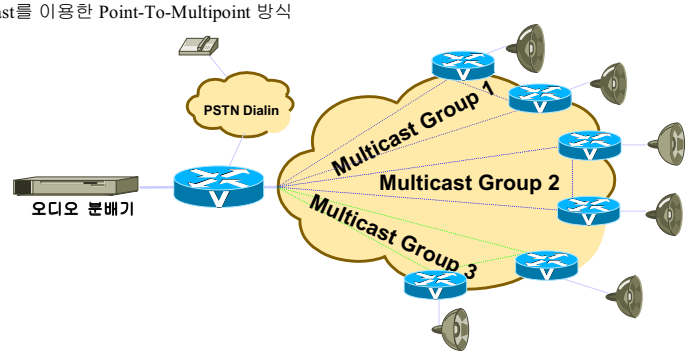
방송솔루션 소개 (Hook & Holler)

Cisco.com

❑ Cisco의 방송 솔루션 (Hook & Holler)
Cisco의 VoIP Technology는 초기에는 전통적인 PBX Toll-bypass Applications에 초점을 맞추어 개발되었으나 현재는 Data Network와 함께 Hoot & Holler Network과 조합하여 사용할 수 있다.

Cisco IOS 12.1(2)XD에서 처음 소개되었으며 Cisco의 VoIP Technology를 통하여 IP Multicasting / QoS(Quality of Service)와 함께 구현할 수 있다

Multicast를 이용한 Point-To-Multipoint 방식



방송솔루션 소개 (Hook & Holler)

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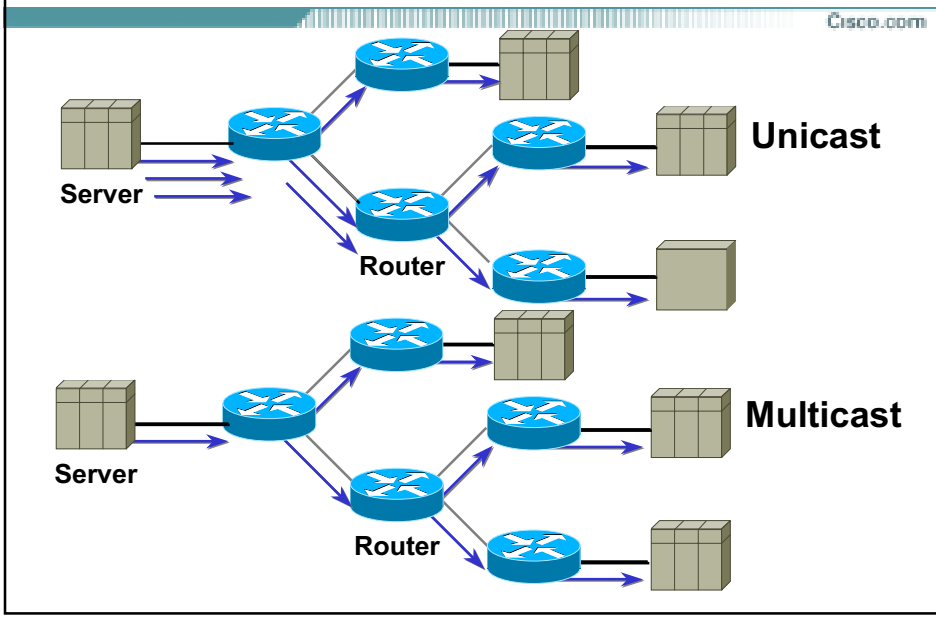
중앙의 오디오 분배기에 연결된 VoIP Router는 각 지점의 Router들과 Connection Trunk를 계속 유지 하며, 분배기에서 들어오는 방송/음악 Traffic을 IP에 실어서 다른 Voice Traffic처럼 망으로 보내는데, 다른 장비와는 달리 IP Multicast 기능을 이용하여 한번에 여러 곳으로 실제 방송하듯이 Broadcast할 수 있는 점이 장점이다. (Signaling은 H.323을 이용)
또한, Cisco Router(VoIP Gateway)에서 방송을 위한 CODEC과 음성 전화를 위한 CODEC을 분리하여 사용할 수 도 있다.

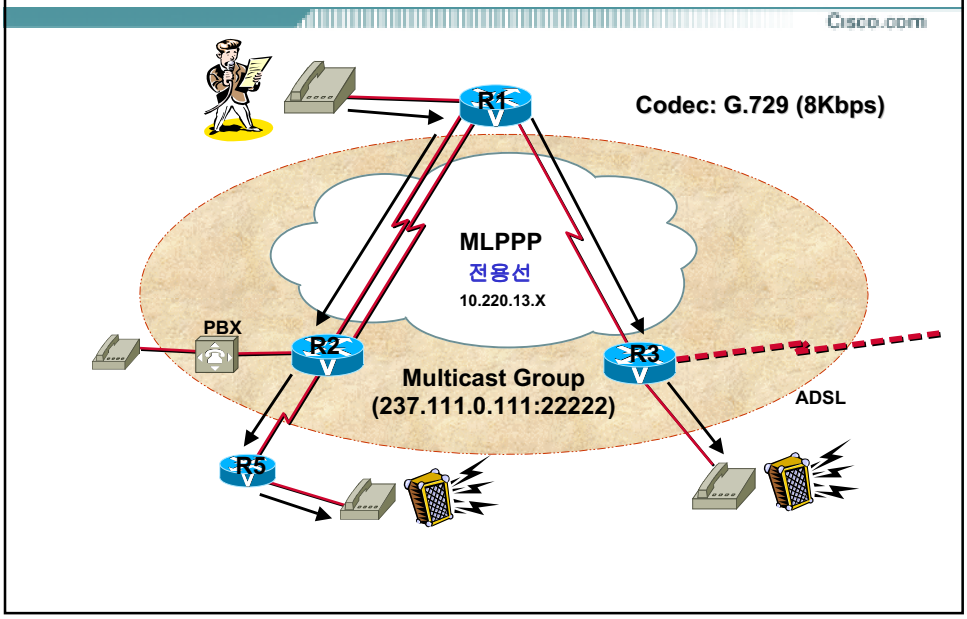
IP Multicast
기존의 Unicast와 는 달리, Point-to-Point통신이 아닌, 한번에 여러 상대와의 통신을 가능하게 해주는 기술이다.
즉, 특정한 Group을 정의 해주며 그 group에 속한 모든 member들에게 한꺼번에 packet을 전송해 주어 Unicast에서 발생하는 Packet의 양을 현저히 줄일 수 있고, TV나 음성 방송 등 현재 적용되고 있는 것과 같은 서비스를 Internet기반의 멀티미디어 서비스로 적용할 수 있도록 해준다.

방송을 하지 않는 장소는 단순히 VoIP Gateway에서 IP Multicast Group에 참가 하지 않으면 된다.
즉, 간단히 Router에서 Configuration만 바꿔줌으로써 방송을 할 수도 있고, 안 할 수도 있다.

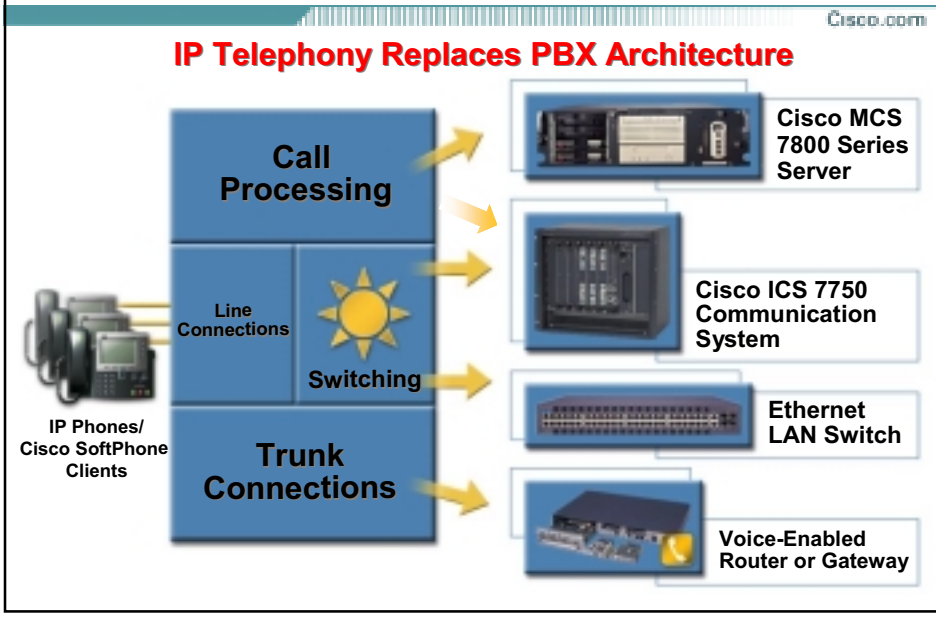
Multicast와 Unicast의 비교



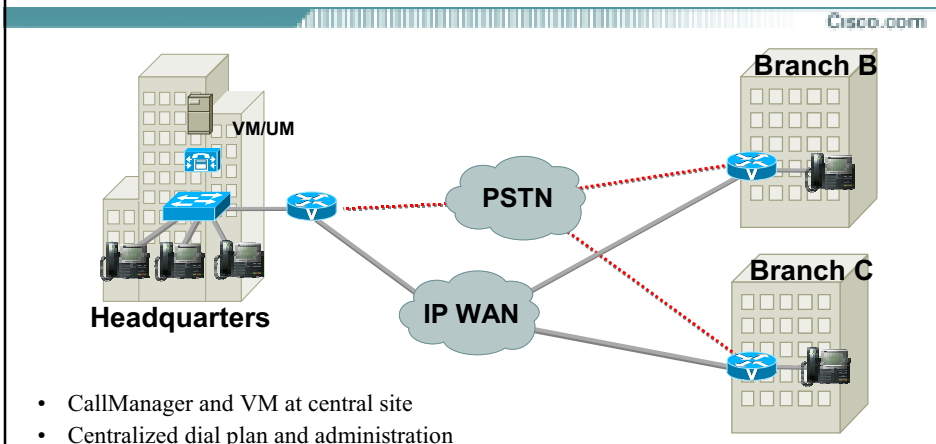
방송 Flow Example



Cisco CallManager 구조



지사/지점을 위한 IP Telephony

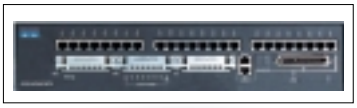


- CallManager and VM at central site
- Centralized dial plan and administration
- Supports up to 2500 users total per cluster with multiple clusters possible
- Call Admission Control implemented at central CM by limiting number of calls in or out of a site
- No service if WAN down (unless dial back-up or new IOS feature—SRST)

Catalyst® 4224 소개 (New Product)

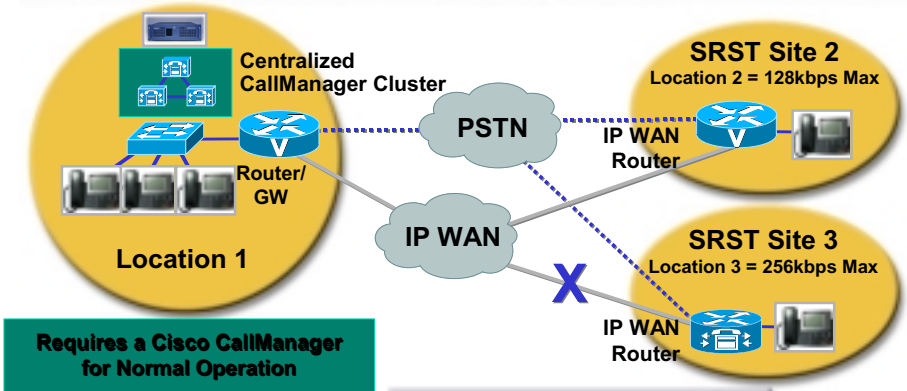
Cisco.com

- 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



Survivable Remote Site Telephony

Cisco.com



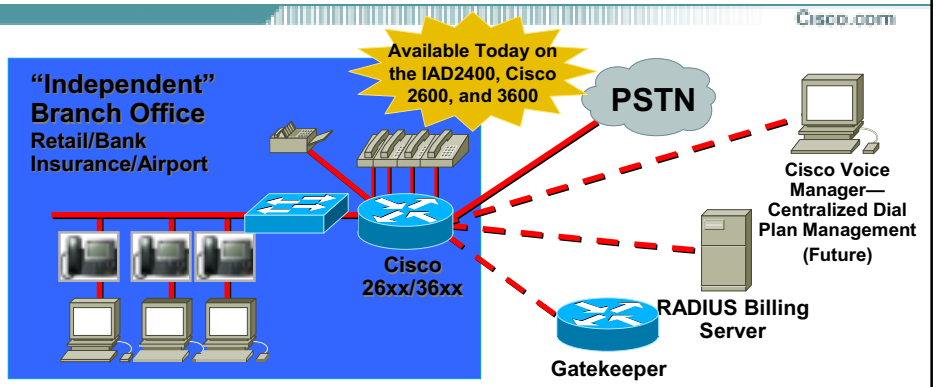
Requires a Cisco CallManager for Normal Operation

SRS Telephony in Cisco IOS Router for Fallback Call Processing

Supported on 26XX, 36XX, Catalyst 4224 with IOS 12.1.5YD

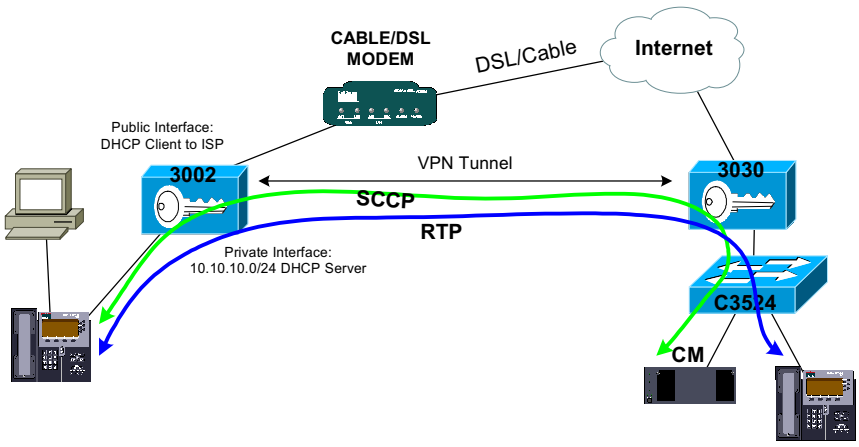
```
call-manager-fallback
ip source-address 10.2.2.1 port 2000
 max-ephones 5
 max-dn 10
 keepalive 60
```

IP Keyswitch for Branch Offices



- 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
- It is not a scaled down CallManager; has keyswitch focused features

VPN을 이용한 Configuration Example



Agenda

Cisco.com

- Cisco Voice Solution
  - VoIP 소개 & Application
  - Cisco IP Telephony Solution
  - New Product Update for branch office
- ***QoS end-to-end through SP network & CAC***

IP VPN Service—Requirements

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- Enterprise customer buying IP VPN service (MPLS or otherwise) from service provider requires 3 classes of service:
  - Gold (real-time voice): no loss, low latency, low jitter, guaranteed bandwidth (128 kbps)
  - Silver (ERP application): low loss, guaranteed bandwidth (128 kbps)
  - Bronze (other traffic): best effort
- Link to SP is 512 kbps, simple 2 site example

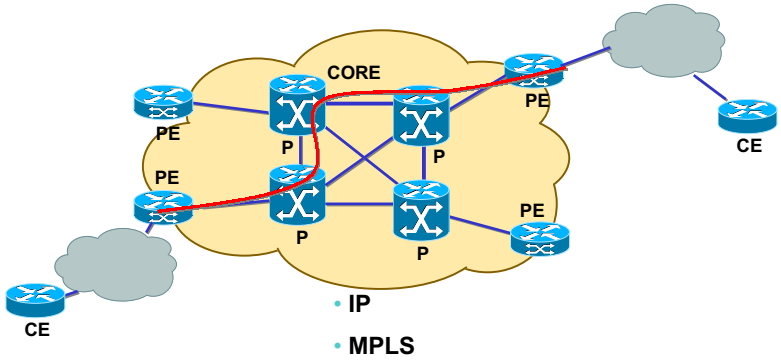
IP VPN Service—Questions to Ask

Cisco.com

- Can service provider (SP) make SLA guarantees for the 3 classes
- What happens to traffic that violates contract
- Will IP precedence or DSCP values be changed by SP network

IP VPN Service—Topology

Cisco.com



Enterprise Customer Needs IP or MPLS VPN with Guaranteed QoS for 3 Classes of Traffic

IP VPN Service—Recommended Design

Cisco.com

- It's about control—send traffic to the SP understanding how it will be treated
  - Make sure Gold class never violates contract
  - Police Silver class to agreed rate, with some bursting capability
  - Allow Bronze traffic to use rest of available bandwidth
- SP is likely to police the 3 classes and may re-mark or drop exceeding or violating packets

IP VPN Service—Configuration

Cisco.com

```
Router(config)# class-map Gold
Router(config-cmap)# match ip rtp 16384 17383
Router(config-cmap)# exit
Router(config)# class-map Silver
Router(config-cmap)# match access-group 101
Router(config-cmap)# exit
```

IP VPN Service—Configuration

Cisco.com

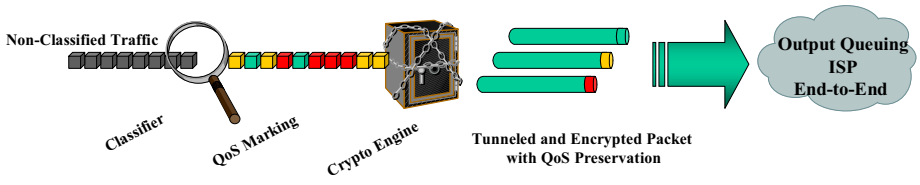
```
Router(config)# policy-map ipvpn
Router(config-pmap)# class Gold
Router(config-pmap-c)# priority 128
Router(config-pmap)# class Silver
Router(config-pmap-c)# bandwidth 128
Router(config-pmap-c)# police 128000 16000 16000
conform-action set-dscp-transmit 26 exceed-action
set-dscp-transmit 30 violate-action drop
Router(config-pmap)# class class-default
Router(config-pmap-c)# set ip dscp 0
Router(config-pmap-c)# fair-queue
```

VPN Modules for Cisco 2600 and 3600

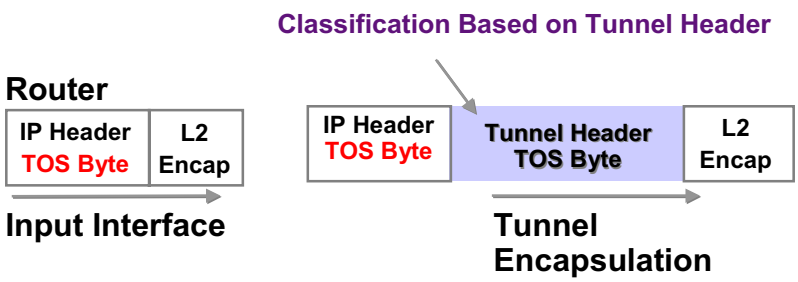
Cisco.com

Cisco QoS VPN for IPSec

- With 12.2(2)T all Cisco 2600/3600 with VPN Modules now copy ToS to front of VPN tunnel
- Works with IPSec and IPSec with GRE
- Diff-serv, NBAR, CAR—entire TOS byte is copied to the IPSec header so precedence can be applied
- Enables classification for encrypted and tunneled VPNs
- Supports ISP Differentiated Services offerings
- Preserves QoS Signaling end-to-end



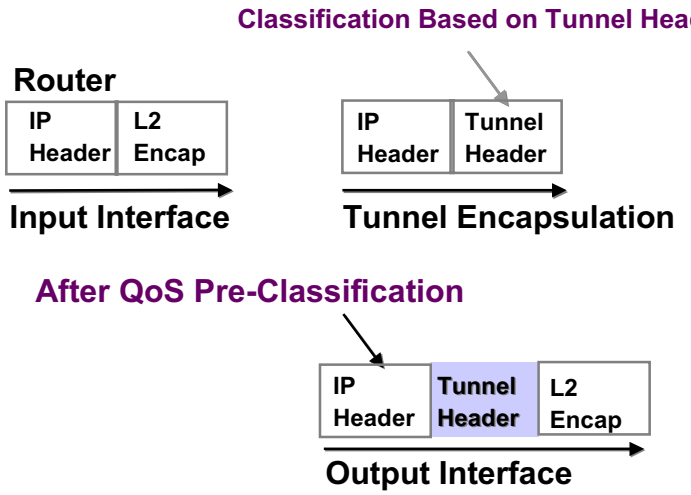
VPN QoS: ToS Field Copy



Copy ToS from original IP header to the tunnel header:

- Done by default for GRE and IPSec

VPN QoS: Pre-Classification



Configuring QoS for VPNs

Cisco.com

- **GRE and IPIP tunnels**

  Router(config)# interface tunnel0
  Router(config-if)# qos pre-classify

- **IPSec tunnels**

  Router(config)# crypto map secured-partner-X
  Router(config-crypto-map)# qos pre-classify

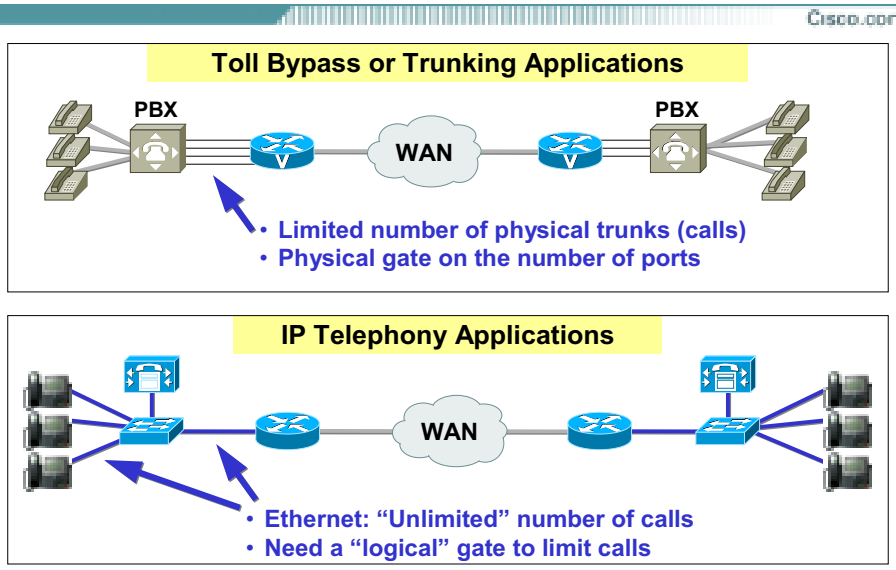
Definition of Call Admissions Control

Cisco.com

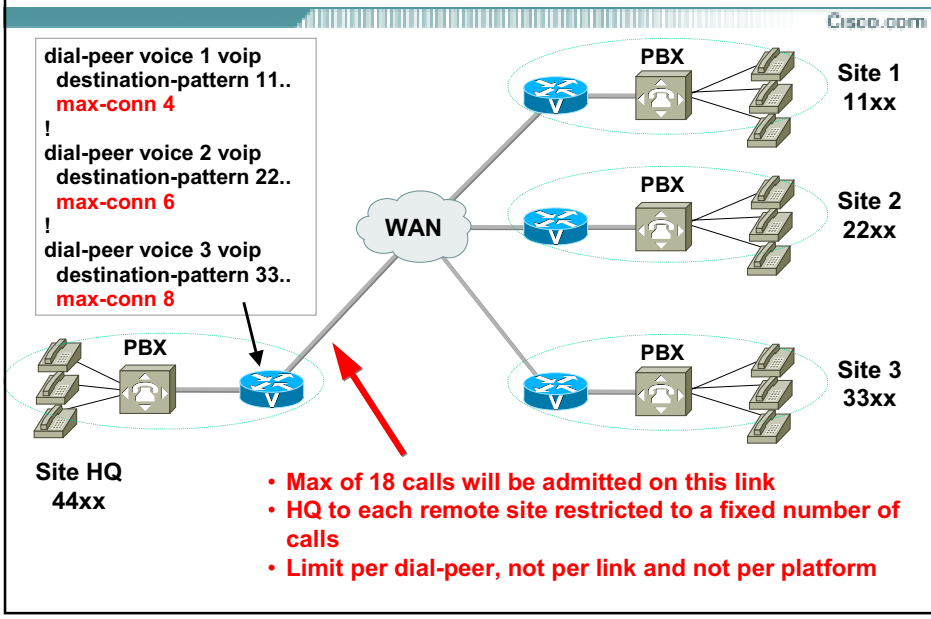
Call Admission Control (CAC) is a deterministic decision before call establishment, on whether the required network resources are available to provide QoS to the new call

- CAC features allow VoX GWs and CM to make an informed decision before admitting a new call based on the condition of the network, e.g.
  - reorder tone
  - try another VoX route
  - redirect via the PSTN
- For real-time sensitive traffic like voice, it's better to deny network access [under congestion] than to allow traffic and drop/delay it

Limit the Incoming DS0s



Max-Connections per Dial-Peer



Trunk Conditioning

12.1.3T

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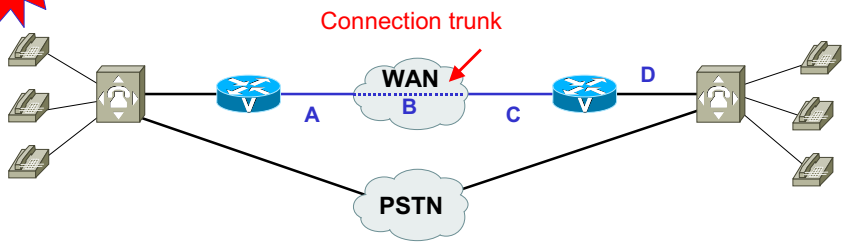


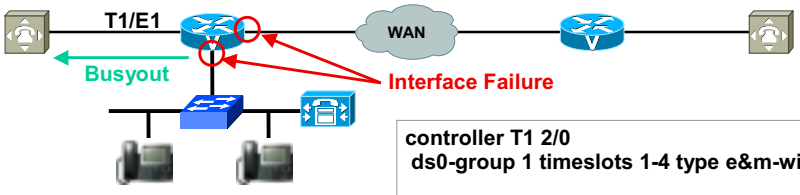
- VoIP, VoFR, VoATM "Connection Trunk" only
  - point2point connections
- Keepalives between endpoints to detect IP (A, B or C) or far-end trunk (D) failures
- Sends busy or OOS (out-of-service) on the trunks to the PBX to allow it to reroute calls

Local Busy-Out Monitor (LVBO)

12.0.7XK

Cisco.com

- **Voice-port busyout when local network interface(s) fail**
  - **Busyout based on specified network interfaces changing state**
    - **change to out of service or into service**
  - **Force individual voice ports into the busyout state**
  - **Force a trunk into busyout state (maintenance/admin reasons)**
- **Analog and CAS trunks only**
- **Up to 32 interfaces can be monitored for any voice port**




```
controller T1 2/0
 ds0-group 1 timeslots 1-4 type e&m-wink-start

voice-port 2/0:1
 busyout monitor Serial0/1
 busyout monitor Ethernet0/1
```

Security Assurance Agent (SAA) Probes

Cisco.com

- Network congestion analysis mechanism
- IP networks only
- Provides congestion information for configured IP addresses
  - delay and packet loss (ICPIF is calculated from this)
  - does NOT provide any bandwidth information, either configured or available
- Client server-protocol defined on UDP
  - Uses well-known UDP port 1976 for sending the SAA probe
  - SAA probe packets go out randomly on ports selected from within the top end of audio UDP defined port range (16384 - 32767)
  - By default the SAA probe uses the RTCP port (odd port#)
  - SAA probe can be configured to use the RTP port (even port#)
- SAA is a Cisco Proprietary protocol, first introduced on selected platforms in 12.0.7T
  - higher end IOS platforms tend to support it (e.g. 7200/7500)
  - lower end IOS platforms tend not to (e.g. 1750)
  - CM does not support SAA probes
  - IP Phones do not support SAA probes
- SAA Probe simulates a "voice packet"
  - Mimics codec sizing; RTP/UDP packet; can set IP Prec; UDP audio ports

ICPIF

Cisco.com

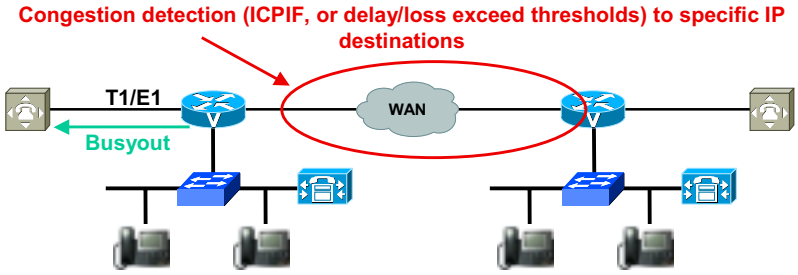
- ICPIF: Calculated Planning Impairment Factor
- ITU G.113: Transmission Impairments
- ICPIF Value interpretations given by G.113:
  - 5: Very good
  - 10: Good
  - 20: Adequate
  - 30: Limiting case
  - 45: Exceptional limiting case
  - 55: Customers likely to react strongly
- ICPIF values are calculated
  - uses SAA Probe delay and loss information

Advanced Busy-Out Monitor (AVBO)

12.1.3T

Cisco.com

- Voice-port busyout when IP network is congested
  - Individual voice ports to enter the busyout state
  - SAA probe values for predetermined IP destination(s)
  - Threshold configuration for determining “congestion”
- IP destinations only (VoIP)

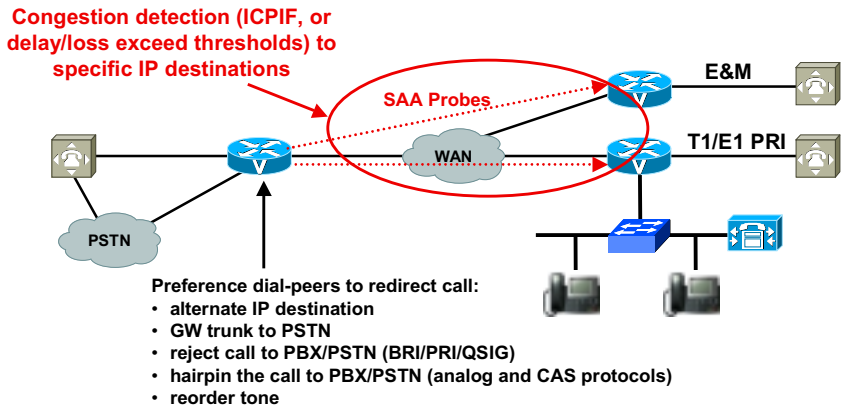


PSTN Fallback Overview

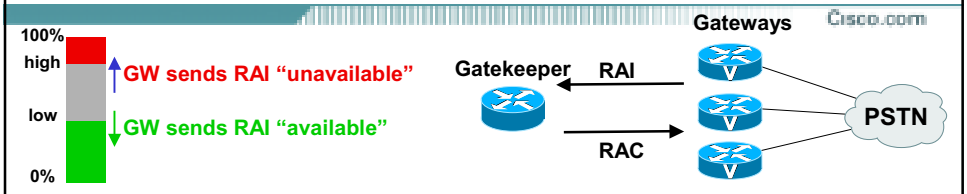
12.1.3T

Cisco.com

- Monitor (measurement-based) congestion in IP network
- Reject, or redirect, a new call based on congested conditions
- AVBO feature busies out entire PBX trunk: PSTN Fallback decides on a per-call basis whether to allow/deny the call set-up



H.323 Resource Availability (RAI)



- A GW informs the GK when it is running short on resources
  - GW says "No" to GK when resources used exceed "high water" mark
  - GW says "Yes" to GK when resources used fall below "low water" mark
- CAC decision is controlled by the terminating GW
- DS0s and DSPs are included in calculation
- Set high and low water marks several %-points apart to avoid bouncing on a per-call basis
- Allows GK to redirect calls to GWs with capacity to terminate them
  - GK takes RAI into account in GW selection process
  - GK will not assign a call to the GW low on resources, unless there is only a single GW to handle a call to the required destination
- Critical feature in SP VoIP networks
  - POPs with banks of GWs to the PSTN

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