

# Quality of Service for Voice Over IP (QoS for VoIP)

**Presented by:**  
**Dr. Peter J. Welcher**



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## About the Speaker

- **Dr. Pete Welcher**
  - Cisco CCIE #1773, CCSI #94014, CCIP
  - Network design & management consulting
    - Stock quotation firm, 3000 routers, TCP/IP
    - Second stock quotation firm, 2000 routers, UDP broadcasts
    - Hotel chain, 1000 routers, SNA
    - Government agency, 1500 routers
  - Teach many of the Cisco courses
- **Enterprise Networking Magazine articles**
  - <http://www.netcraftsmen.net/welcher/papers>



## Objectives

- **Upon completion of this seminar, you should be able to:**
  - Explain the need for QoS for VoIP
  - Describe factors affecting network voice quality
  - Explain some ways QoS can improve the quality of VoIP in a network

## Topics

- **Why QoS for Voice?**
- **QoS Review**
- **Techniques for QoS for VoIP**
- **A Little Call Admission Control**

## Network Convergence

- **Run data, voice, video, new applications over the same networks**
  - High availability network
  - New strategic applications for mixed media
- **Reduced costs**
  - One infrastructure to maintain
  - Aggregated bandwidth is (much) cheaper
  - PBX and trunking costs
  - Old + new skills to manage VoIP

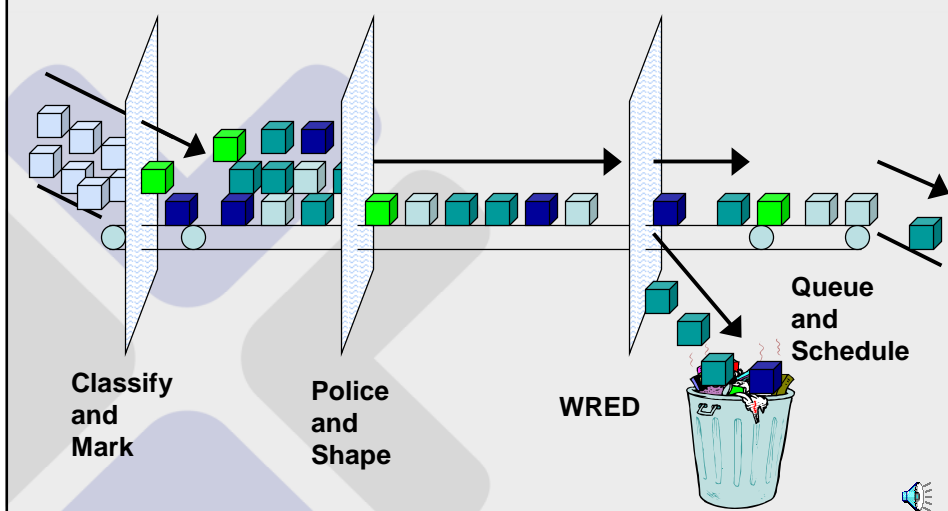
## Where Does QoS Fit In?

- **QoS protects VoIP on shared media**
  - The culprit: bursty data applications
  - Lose 2 VoIP packets and get lower quality VoIP
- **QoS can prioritize VoIP**
  - VoIP is sensitive to delay and jitter (fluctuations in delay)
  - Prioritization minimizes those effects

## Topics

- Why QoS for Voice?
- QoS Review
- Techniques for QoS for VoIP
- A Little Call Admission Control

## The Operations of QoS







## What Can QoS do for VoIP?

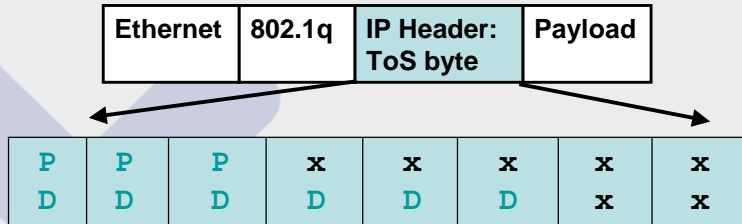
- **Classify and mark**
  - Identify and set Layer 2 or 3 header bits to mark classes of traffic and indicate desired treatment to downstream devices
- **Just setting the IP Precedence or DSCP bits (coming slide) can provide preferential treatment for a class of traffic**
  - When used with WFQ or CBWFQ
  - WFQ is on by default on E1 or slower serial links

## Marking: IP Precedence and Diff-Serv

- **IP Precedence marks packets into six classes (2 reserved)**
- **Diff-Serv uses 64 classes (DSCP), retro-compatible**

Number	Name
0	Routine 
1	Priority
2	Immediate 
3	Flash 
4	Flash-override
5	Critical 
6	<i>Internet-reserved</i>
7	<i>Network-reserved</i>

## Layer 3 Marking



- **IP header ToS byte used for IP marking**
- **IP Precedence: 3 bits, 0-7**
  - 6 & 7 reserved for system use
- **Diff Serv: 6 bits, 0-63**
  - Backwards compatible

## IP Precedence and DiffServ

Precedence	DiffServ	DSCP
7	111xxxxx	Class Selector 7
6	110xxxxx	Class Selector 6
		Expedited Forwarding
5	101xxxxx	Class Selector 5
		Assured Forwarding 4
4	100xxxxx	Class Selector 4
		Assured Forwarding 3
3	011xxxxx	Class Selector 3
		Assured Forwarding 2
2	010xxxxx	Class Selector 2
		Assured Forwarding 1
1	001xxxxx	Class Selector 1
0	000xxxxx	Best Effort
		CS7
		CS6
		EF
		CS5
		AF4
		CS4
		AF3
		CS3
		AF2
		CS2
		AF1
		CS1
		BE
		111000xx
		110000xx
		101110xx
		101000xx
		100dd0xx
		100000xx
		011dd0xx
		010dd0xx
		001dd0xx
		001000xx
		000000xx
		56
		48
		46
		40
		34,36,38
		32
		26,28,30
		24
		18,20,22
		16
		10,12,14
		8
		0

## What Can QoS do for VoIP?

- **Police and shape**

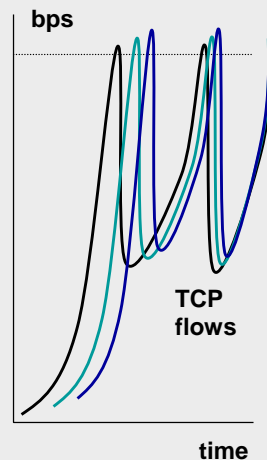
- Control bursts of excess traffic and bandwidth used by high bandwidth applications
- Business high-bandwidth applications:
  - Network server backups
  - Business Video on Demand (VoD)
- Non-business high-bandwidth applications:
  - Napster, Kazaa, Gnutella
  - Non-business Video on Demand (news, ads, web sites, etc.)



## What Can QoS do for VoIP?

- **Congestion Avoidance**

- Weighted Random Early Detection (“WRED”)
- When queue average depth over time rises, start dropping less important packets to preserve queue space (bandwidth) for more important packets
- Slow down TCP flows to prevent “group slowdown”



## What Can QoS do for VoIP?

- **Queue and schedule**

- Prioritization gets VoIP or IP Video Conferencing (IPVC) out of the router first
- Intelligent queuing and scheduling with Class-Based Weighted Fair Queuing deliver a mix of policy techniques
- Co-existence for all the applications running on shared media

## What Can QoS do for VoIP?

- **RSVP**

- Bandwidth reservation for Call Admission Control
  - See later slides
- Each RSVP flow of traffic is automatically assigned a weight to guarantee bandwidth with WFQ or CBWFQ
- Currently there is no bandwidth enforcement (limitation) mechanism tied to RSVP



## CBWFQ / MQC

- The preferred technique for configuring QoS has become Modular QoS CLI (MQC), also called Class-Based Weighted Fair Queuing (CBWFQ)
- There are 3 steps to configure this
  - Define your classes of traffic (classify)
  - Define your policies using those classes
  - Apply to one or more interfaces (in, out)

## CBWFQ / MQC Example

```
class-map match-any voice
  match ip dscp EF
policy-map SlowLink
  class voice
    priority 120
  class tunes
    police 8000 conform-action drop
    exceed-action drop
interface serial 0
  service-policy output SlowLink
```

1: Define classes

2: Define policies

3: Apply

## Topics

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## Step 1 Towards QoS

- **Good stable hierarchical WAN and hierarchical campus network**
  - Too many WAN hops can lead to high latency
- **Redundant, high-availability design**
- **Protocols configured for quick failover**
- **Some network management in place**
  - Some bandwidth reporting and capacity planning
  - Reporting and resolution of outages
- ***If the network is down, that's REALLY bad QoS!***

## Expectations and QoS

- **QoS can do good things, but it can't do the impossible**
  - QoS can help protect critical traffic on occasionally congested links
  - QoS is not a cure for really inadequate bandwidth
  - VoIP needs its fair share of bandwidth too

## QoS, VoIP, and Speeds

- **64-256 Kbps**
  - Rather low bandwidth, need to do as much as we can to optimize QoS
- **T1 speed and above**
  - Don't need to protect against some of the low-speed issues
  - Can't use some of the tools: CPU impact
- **Higher speeds**
  - Use QoS as insurance, protect VoIP

## Overview of How QoS Helps Voice

- **Reduce latency**
- **Get voice packets out first!**
  - Prioritize voice traffic
  - Prioritize voice on the campus network too
- **Use low bandwidth efficiently**
- **Avoid VoIP packet loss**
- **Don't forget your signaling traffic...**
  - Slow signaling = slow call completion
- **Call Admission Control**

Latency...

Latency...

Latency...



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## QoS for VoIP Techniques

- **Slow links**
  - Use bandwidth wisely: cRTP
  - Reduce latency: LFI
- **All links**
  - Prioritize voice: PQ-WFQ or LLQ



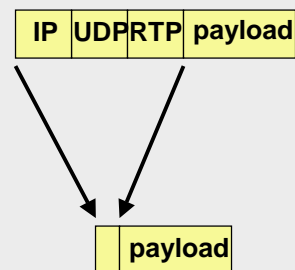
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## Adding Bandwidth

- **Adding bandwidth reduces serialization delay**
- **Adding bandwidth usually reduces queue size, hence reduces queuing delay**
  - Unless link is badly oversubscribed
- **Adding bandwidth costs money**
  - Especially when you do it across a lot of leased line, FR, or ATM sites

## Use Low Bandwidth Efficiently: CRTP

- **Compressed RTP**
  - IP + UDP + RTP header = 40 bytes
  - Payload is typically 20 bytes
  - cRTP reduces that to 2-4 bytes by assigning each connection an ID
  - cRTP sends the connection ID + changing bits
  - Use 1/3 to 1/2 the bandwidth! (E.g. 8K vs 24K)
- **Consider CPU impact**
  - Should be OK up to 1/2 Mbps or so



## cRTP -- 2

```
ip rtp header-compression
```

- **Interface command**
- **cRTP is faster in newer Cisco IOS code:**
  - Version 12.1 and later: fast-switched or CEF-switched
    - Beware CPU impact in prior releases
    - Depends on what else the CPU is doing

## CRTP on Frame Relay

- Usually configured with on the physical interface
- Can be configured for selected PVC's only as part of a frame-relay map command

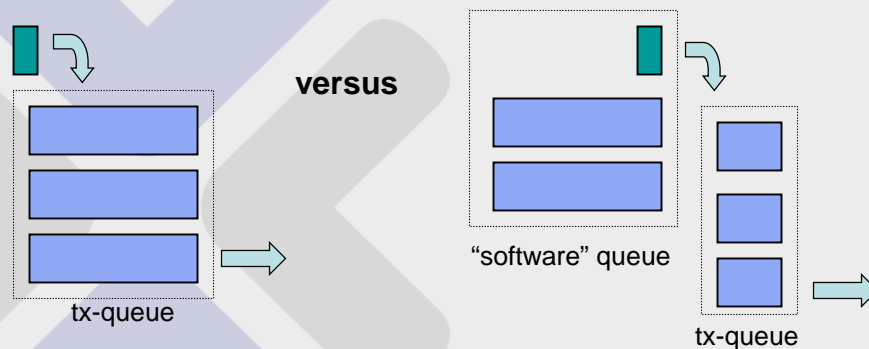
```
frame-relay ip rtp header-compress [passive]
```

# Latency

- The voice courses or Networkers presentations go into latency budget at length
- The main causes of delay are:
  - Codec
  - Queuing \*
  - Wait for packet being transmitted \*
  - Serialization \*
  - Jitter buffer
  - Other
  - \* = the causes of delay that we can do something about
- Can also consider **signaling** delay
  - Time to initiate ring, time after phone answered to complete call

# Link Fragmentation and Interleaving

- Fragment large data frames, slip voice frames ahead of queued packets for large frame under transmission



## Link Fragmentation and Interleaving

- **Ways to do Fragmentation and Interleaving:**

- FRF.12 for Frame-Relay links, VoIPoFR & VoFR
- VoFR: use FRF.11, esp. if speed very low
- MLPPP/LFI for PPP links
- ATM -- not applicable (generally higher speed)
  - Can use PPP over ATM if desired
  - E.g. FR to ATM service interworking
- MTU size reduction
  - Avoid if at all possible, may make congestion worse due to retransmissions



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## Link Fragmentation and Interleaving -- 2

- **Useful up to  $\frac{1}{2}$  T1 to T1 speed**
  - CPU impact
  - Not really needed above that speed
- **Do use the same fragment size on both ends of the link**

Link Speed	# Bytes in 10 msec
56 K	70
64 K	80
128 K	160
256	320
512	640
768	960



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## LFI: ML-PPP

- **Create multilink virtual interface**
- **Configure ML-PPP LFI:**

```
interface multilink 1
  ip address ...
  ppp multilink
  ppp multilink fragment-delay 20
  ppp multilink interleave
```

## LFI with FRF.12

- **Newer versions of this also prioritize VoIPoFR as well as VoFR traffic**
  - You must however set the fragment size bigger than the VoIP packet size

## Configuring FRTS FRF.12

```
interface serial 0
encapsulation frame-relay
ip address ...
frame-relay traffic-shaping
class FrameRules

map-class frame-relay FrameRules
frame-relay cir 128000
frame-relay mincir 128000
frame-relay bc 2000
frame-relay fragment 160
frame-relay fair-queue
```



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## FR or ATM Tip

- **If you can buy prioritized service for Voice from your Service Provider, consider dual PVC's**
  - One for voice, one for data
  - This can be especially effective when data bandwidth is low, e.g. 56-256 Kbps or so
- **Get a good SLA (if you can)**
- **Be careful, with some providers this doesn't buy you much**



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## QoS on FR

- **Some of the features like cRTP and LFI may require configuring FRTS on subinterfaces**
- **Modular QoS CLI / CBWFQ is configured:**
  - On the FRTS pseudo-interface (class map) in earlier Cisco IOS versions
  - On the FR subinterface in newer versions

## Using Bandwidth Efficiently

- **Other considerations:**
  - Consider using a high-compression codec
  - Consider VAD
    - **Off** by default (for a reason)
  - Control echo
  - Observe resulting voice quality
- **Personally observed in lab: an 8 Kbps codec gave mediocre voice quality, but it took a sincere effort to congest the 56K link before it got worse!**
- **Trade-offs!**

## Get VoIP Out of the Router Fast

- **Priority for VoIP Packets**
- **PQ-WFQ**
  - Prioritizes specified range of UDP ports, RTP packets
  - `ip rtp priority` command
- **Low Latency Queuing: LLQ**
  - Also called PQ-CBWFQ
  - `CBWFQ / MQC priority` command
  - More flexible prioritization

## RTP Priority vs. LLQ

- **LLQ is the recommended technique going forward**
  - Works well, flexible, powerful
- **LLQ does police (drop) excess traffic to prevent starvation of lower priority queues**
  - You do need to track how much VoIP you have in your network and manage traffic levels

## Priority for VoIP — LLQ

```
Router(config)# policy-map premium_policy
Router(config-pmap)# class voice
Router(config-pmap-c)# priority 128
Router(config-pmap)# exit
```

```
Router(config)# policy-map premium_policy
Router(config-pmap)# class voice
Router(config-pmap-c)# priority percent 25
Router(config-pmap)# exit
```

## Signaling Traffic

- **H.323, SIP, MGCP**
- **Used for call setup (busy, ring, handset pickup and call completion)**
- **User perception of quality may be affected if call setup is slow**
- **Consider setting signaling traffic to IP Precedence level 3 or DSCP AF31**

## Classifying Signaling Traffic

Protocol	Transport & Ports
<b>Skiny (Cisco phone to Call Manager)</b>	<b>TCP 2000-2002</b>
<b>H.323</b>	<b>TCP 1720, 11xxx</b>
<b>MGCP</b>	<b>UDP 2427, 2428</b>
<b>SIP</b>	<b>UDP 5060</b>

## Jitter

- **LFI should reduce jitter on slow links**
- **Can use Traffic Shaping on bursty classes of traffic**
- **LLQ or prioritization may reduce the impact of data traffic bursts**
  - VoIP traffic gets sent ahead of the data
  - But: queuing in the FR or other WAN network?

## ATM VC-Bundles

- Requires PA-A3 on 7200/7500 with right processor, OC-3 or IMA not AIP on 2600/3600
- Can create bundle of VC's with one address
- Can map precedence levels to VC's in bundle
- If one VC in bundle fails, can automatically shift its traffic to another ("bumping")
- Per-VC WFQ, CBWFQ, LLQ support
- See also  
[http://www.cisco.com/en/US/partner/products/sw/ios\\_swrel/ps1835/products\\_configuration\\_guide\\_chapter\\_09186a00800bd9ea.html](http://www.cisco.com/en/US/partner/products/sw/ios_swrel/ps1835/products_configuration_guide_chapter_09186a00800bd9ea.html)

## Topics

- Why QoS for Voice?
- QoS Review
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- A Little Call Admission Control

## What is Call Admission Control?

- **Call Admission Control denies additional calls when a resource (bandwidth) runs out**
- **“Protect voice from voice”**
  - If there is too much voice, LLQ will drop packets from all conversations
  - Better to force excess calls through PSTN
- **This applies, only more so, with IPVC**



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## Local Call Admission Control Techniques

- **Local CAC techniques**
  - Physical DS0 count
  - Max-connections
    - Configure per dial peer, works well in some topologies
  - Voice-bandwidth for FR
  - Trunk conditioning (busy or OOS)
    - Keepalives detect connection trunk failures
    - Triggers busy or out-of-service to PBX
  - LVBO, Local Busy-Out Monitor
    - Busyout when specified interfaces down
    - Force ports or tunks into busyout state



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## Measurement or Resource-Based CAC Techniques

- **Measurement-based CAC techniques**
  - AVBO (based on SAA, busy out whole trunk)
  - PSTN fallback (based on SAA, per call)
- **Resource-based CAC techniques**
  - Resource Availability Indicator (RAI)
    - At call-terminating gateway
    - DS0 and DSP availability
  - Gatekeeper Zone Bandwidth
    - Also useful with H.323 video
  - RSVP (with H.323)
    - Guarantees QoS for call duration (vs. at time of call)



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## RSVP

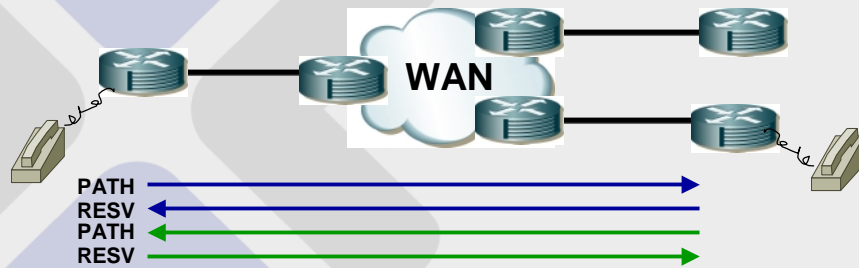
- In general, application traffic receivers can reserve bandwidth along the path back to the server, using RSVP
- Routers allocate bandwidth from pool upon request, or signal an error
- RSVP gives a scalable topology-and application-independent way to administer and reserve bandwidth



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## RSVP

- **Cisco H.323 v1 and v2 gateway call setup is now synchronized with RSVP (Call Manager is not)**
  - Call will fail unless RSVP reservation succeeds
  - The RSVP reservation guarantees the requested bandwidth, using weights to work with the WFQ and CBWFQ schedulers



## IP Video-Conferencing

- **IP Video-Conferencing (VC):**
  - Current Best Practice is to set Video Conferencing traffic to IP Precedence 4 (DSCP AF41)
  - Set signaling and control traffic to precedence 3 (DSCP AF31)
  - You don't want a burst of video disrupting voice at Precedence level 5 (DSCP EF).
    - Can do LLQ for IPVC
    - Can do CBWFQ "bandwidth" command instead
    - I personally don't want to mix voice and video in priority queues
  - [Strongly consider H.323 gatekeeper, RSVP, and CAC](#)

## Streaming Video

- **Streaming video is more tolerant of delay and jitter**
  - Delay is evened out by the jitter buffer
  - Non-interactive so 1-5 seconds delay not a real problem
- **Send streaming video as AF13 traffic**
  - Medium priority application grade traffic
  - High drop priority when there's congestion

## Campus Switches and VoIP

- **Prioritize VoIP in switches**
  - The 6500 line cards have been adding more hardware queues as they evolve
  - The newer ones tend to have a hardware priority queue
- **Why QoS? Have lots of bandwidth...**
  - Still need to protect VoIP from tail drops when traffic burst occurs

## VoIP and Special Access Media

- **Cable, DSL, Wireless LAN QoS ...**
  - Need media-specific access link support for QoS
  - DOCSIS for cable
  - New Cisco WAP's
- **See Networkers 2002 presentations for details**
  - There's lots of details
- **The main point: apply the above techniques where available**

## Summary

### In this talk we have

- Reviewed QoS basics and CBWFQ
- Talked about cRTP, LFI
- Seen how to configure LLQ
- Discussed markings for VoIP, Signaling, IPVC, and streaming video

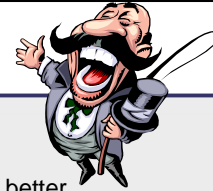
## Where to Get More Info

- **Cisco DQOS or IQOS courses**
- **Cisco Telephony QoS Design Guide**
  - [http://www.cisco.com/univercd/cc/td/doc/product/voice/ip\\_tele/avvi\\_dqos/index.htm](http://www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/avvi_dqos/index.htm)
- **Networkers 2002 slides**
  - <http://www.cisco.com/networkers/nw02/presos/docs/NSC-215.pdf>
  - <http://www.cisco.com/networkers/nw02/presos>
  - <http://www.cisco.com/networkers/nw02/presos/pws/docs>
- **For more info (and links), see my CiscoWorld articles**
  - <http://www.netcraftsmen.net/welcher/papers>

## Questions

**Any Questions?**

## A Word From Us ...



- **We can provide**
  - Network design review: how to make what you have work better
  - Periodic strategic advice: what's the next step for your network or staff
  - Network management tools & procedures advice: what's right for you
  - Implementation guidance (your staff does the details) or full implementation
- **We do**
  - Small- and Large-Scale Routing and Switching (design, health check, etc.)
  - IPsec VPN and V3PN (design and implementation)
  - QoS (strategy, design and implementation)
  - IP Telephony (preparedness survey, design, and implementation)
  - Call Manager deployment
  - Security
  - Network Management (design, installation, tuning, tech transfer, etc.)



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## Cisco Certifications

### **Chesapeake Netcraftsmen** **is certified by Cisco in:**

- IP Telephony
- Network Management
- Wireless
- Security
- (Routing and Switching)



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