VoIP Troubleshooting and Monitoring

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Troubleshooting

- Provide examples of common problems
- Identify sources of problems and their symptoms
- Remediation
- Techniques you can use in your network
- Monitoring requirements
- What to monitor
- Useful metrics

The Network is the Foundation for VoIP

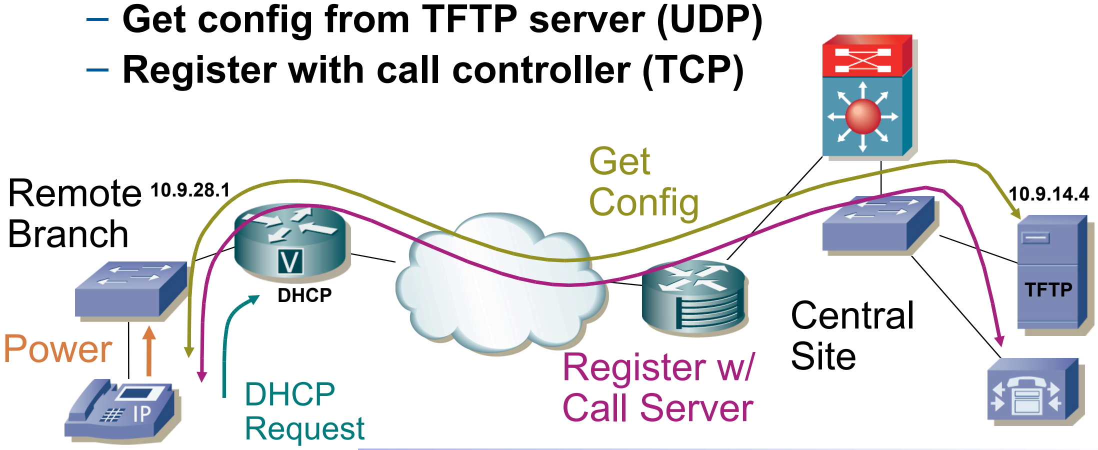
- VoIP depends upon the network
  1. Network hardware and links
  2. Network protocols (routing & switching)
  3. Transport protocols (TCP/UDP)
  4. VoIP protocols and operation
- Other features
  – QoS
  – Redundancy
- Use VoIP operational model to aid troubleshooting and monitoring

Applications (VoIP)
Communication Protocols (TCP/UDP/IP)
Routing & Switching Protocols (OSPF, STP)
Network Hardware & Links (Routers & Switches)

Call Operation	Misc Operation and Services
Call Setup	
Connectivity and Registration	

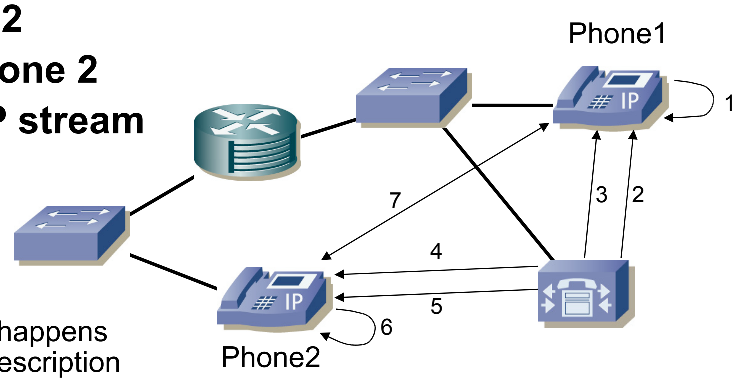
How VoIP Works

- Connectivity and Registration
  – Power requested by continuous Fast Link Pulse (FLP)
  – DHCP request & response (UDP)
  – Get config from TFTP server (UDP)
  – Register with call controller (TCP)



How VoIP Works (cont)

- **Call setup and operation**
  1. Off-hook, Dialtone, Phone 1
  2. Collect digits and call setup, Phone 1
  3. Ringback tone, Phone 1
  4. Call setup, Phone 2
  5. Ring Phone, 2
  6. Off-hook, Phone 2
  7. Connect RTP stream



* Basic steps; a lot more happens than in this high-level description

Troubleshooting Diagnostic Aids

VoIP TROUBLESHOOTER.com Online Diagnostic Tools for Network Managers

Home > Online Diagnostic Tools > Diagnose Problems from Network Statistics — Quick Links

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- Forum
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- Online Diagnosis
  - Call Setup
  - Network Symptoms
  - User Symptoms
  - Calculator
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Diagnose Problems from Network Statistics

Problem			Problem occurs		
Loss	Jitter	Out of Order	Intermittently	Periodically	Continuously
Low	Low	Low	Grounding problem		Loss Plan
Low	High	Low	LAN congestion	Route flapping	Access Link Congestion
			Access Link congestion	Softphone timing	LAN congestion
High	Low	High	Route flapping	Route flapping	
Low	High	High			Load sharing
High	Low	Low	Link Failures	Route flapping	Bad Ethernet Cable
			Bad Ethernet Cable	Router - RED	Duplex Mismatch
High	High	Low	Access Link congestion	Route flapping	LAN congestion
					Access link congestion

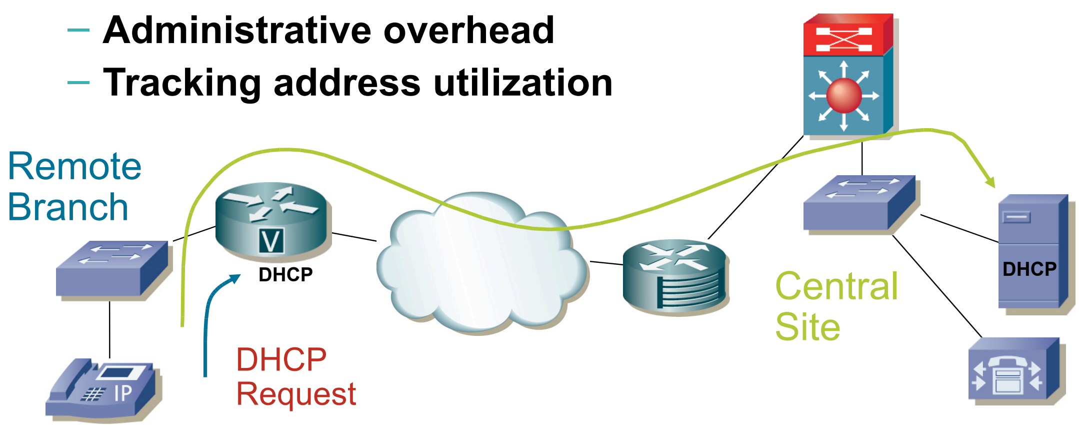
Connectivity – VLAN

- Voice VLAN mis-configured
  - Phone comes up in the wrong VLAN
  - Static configuration on phone (eBay purchase)
  - Switch misconfigured
- No Voice VLAN
  - Phone connected to data port
  - Switch misconfigured (include voice vlan)

```
interface FastEthernet0/9
 switchport access vlan 100
 switchport mode access
 switchport voice vlan 411
```


Connectivity – DHCP

- IP address assignment, default gateway, addl boot info - Cisco: option 150, Avaya: option 176
- Local vs Central DHCP server
  - Short lease vs Long lease
  - Administrative overhead
  - Tracking address utilization

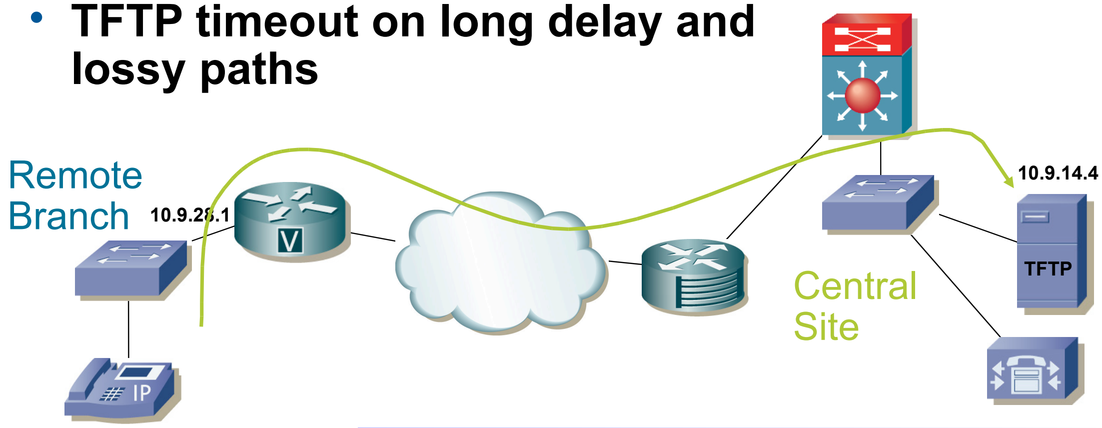


Connectivity – DHCP Location Tradeoffs

- **Central**
  - Multi-day address lease – longer than typical downtime
  - Reduces network equipment configuration
  - Good if many small branches exist
  - Handling long connectivity downtime due to disaster
- **Local**
  - Short address lease
  - Manage DHCP config at each site
  - More appropriate at larger remote sites.
  - Good if downtime is more extensive
  - Very remote offices with poor connection reliability

Connectivity - TFTP

- **Download the phone config and OS**
- **Connectivity between phone and TFTP server**
  - Co-located with central DHCP server is good
  - TFTP uses UDP – Firewall or ACL configuration
- **TFTP timeout on long delay and lossy paths**

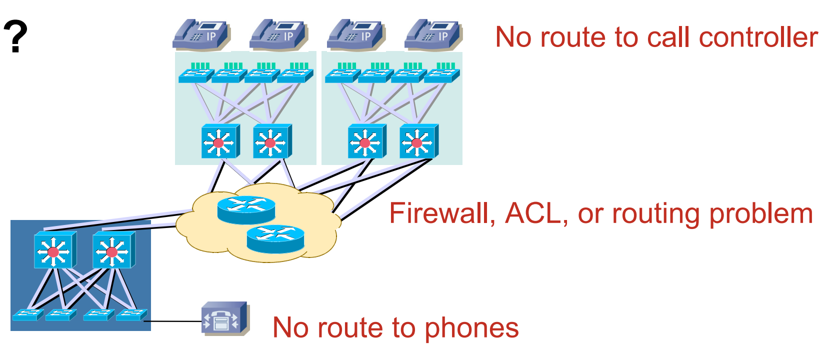


Connectivity – TFTP

- TFTP server failure
  - Address in DHCP option 150 for Cisco; 176 for Avaya
  - Redundant server specification is good
- Bad TFTP file
  - Doesn't exist – often wrong phone MAC address
  - Bad format or contains typos
- Long system boot times, due to power outage
  - Example: 20 minutes to get all phones working
  - Network infrastructure boot time
  - DHCP/TFTP/Call servers booting, then overloaded
  - Download congestion!
  - Use load balancing

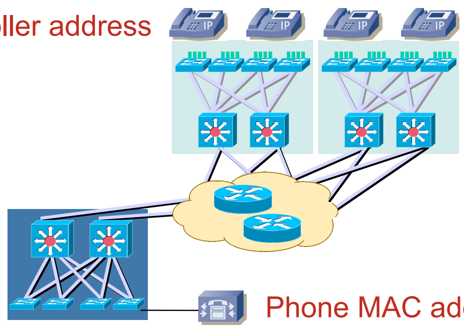
Registration

- Can’t connect to the Call Server
  - Routing problem between phone and call server
  - Incorrect firewall, or ACL configuration
- Test with ping and traceroute from call server
- Which phones are affected?
- New site?



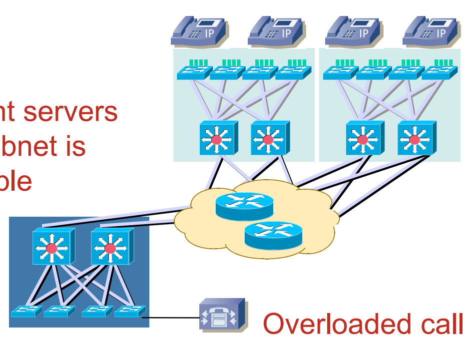
Registration

- Can’t connect to the Call Server
  - Phone not configured in Call Server
  - MAC address wrong in Call Server
  - Default TFTP config file has wrong Call Controller address



Registration

- Can’t connect to the Call Server
  - Call server capacity (e.g., after power outages)
  - Call server is down
    - Use redundant call servers on different subnets



Call Setup

- **Incorrect destination call routing**
  - **Dial plan problems**
    - **Overlapping dial spaces**

      4-digit dialing:
      736-8[0-4]XX
      355-8[5-9]XX
      Then add:
      736-85XX

    - **Incorrect dial search spaces**

      7-digit dialing:
      939XXXX (Internal)
      939XXXX (Local)
      9.939XXXX (Local)
      9.393@ (Local or LD)

  - **Troubleshoot with DNA (Dialed Number Analyzer)**

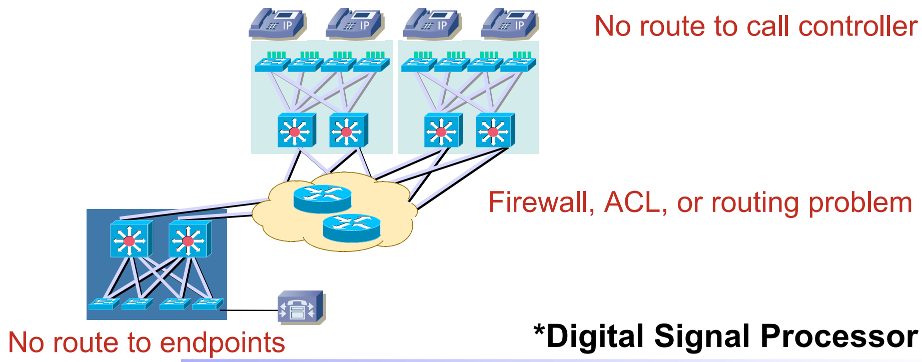
Call Setup

- **Phones get calls for other locations**
  - **Numbers and hunt groups tied to phone, not line**
  - **Phone moved but call server not updated**

- **Spend time on a good dial plan!**
  - **10-digit, multi-tenant plan**
  - **Map dial spaces onto this plan**
  - **Can still do 4-digit (or N-digit) dialing**
  - **Allows for growth, merger, acquisition**
  - **Much, much less expensive to maintain**
  - **Note: include planning to avoid toll fraud**

Call Setup

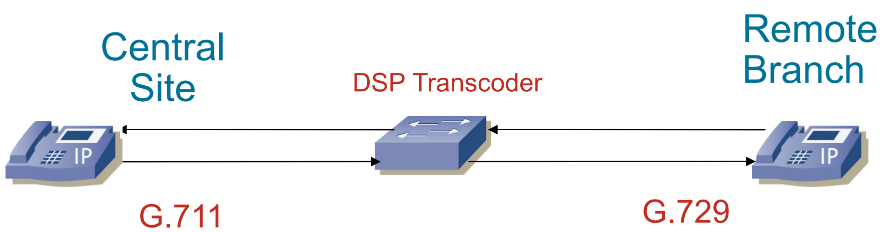
- TCP is used between call server and endpoints
  - Routing problem between call controller & endpoints
  - Typically won't get dial tone or registration
  - Ping, traceroute, ACL checks, etc (sound familiar?)
  - Endpoints include PSTN gateways and DSPs*



*Digital Signal Processor

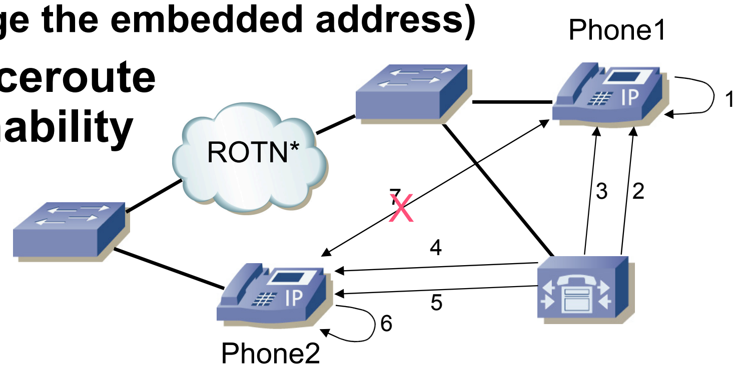
Call Setup

- DSP required to match codecs or for conf calls
- Troubleshooting
  - CUCM log: “no resources”
  - Monitor DSP pool utilization
    - Cat 6500: show port voice active
    - Command syntax and limits depend on hardware
- Solution: buy more hardware



Call Operation - No-Way Audio

- Audio RTP data sent in UDP datagrams
- Endpoints don't have connectivity
  - Routing problem
  - Firewall or ACL blocking a path
  - Cisco Skinny payload carries IP addr (NAT must know to change the embedded address)
- Use ping & traceroute to check reachability



* Rest Of The Network

Call Operation - One-Way Audio

- Check basic connectivity
  - Firewall or ACL blocking one path
  - Routing problem
- Two-way, then one-way
  - Change in routing or configuration
  - DSP crash (when transcoding or conference call)
  - Link congestion and no QoS or bad QoS
- Troubleshooting
  - What changed? (routing & configuration)
  - Who was affected?
  - Log analysis

Call Operation - Delay, Jitter, Packet Loss

G.729 Good

60ms Jitter

10% packet loss

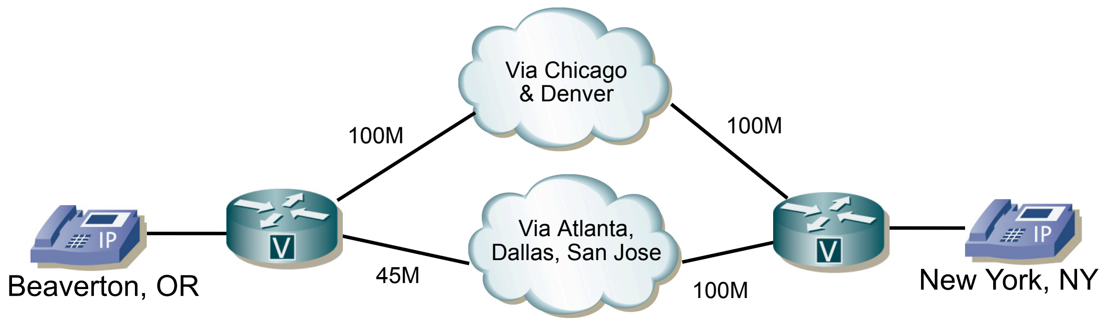
- Causes:
  - Inconsistent or no QoS
  - Duplex mismatch or bad link
  - Routing problems (loss) or multipath (jitter)
  - Oversubscribed links (congestion & loss)
- Know when it's happening
  - Be able to detect the cause of each problem
  - Monitoring depends on vendor
    - RTCP stream (Avaya, Nortel)
    - Call stats on call server (Cisco)
    - ITU specs: 150ms delay, 30ms jitter, 1% loss

Call Operation - Delay

- ITU Spec: 150ms one-way delay
- Reduces interaction of a call
  - Wait for voice to travel to the other end of the call
  - Worst case is like a push-to-talk radio (Nextel?)
  - Roughly 10ms per 1000 miles (~30ms across the US)
- Causes:
  - Sub-optimum route path selection
    - New York to Atlanta via San Francisco
  - Long delay path, e.g., satellite circuit (250ms one-way)

Call Operation - Jitter

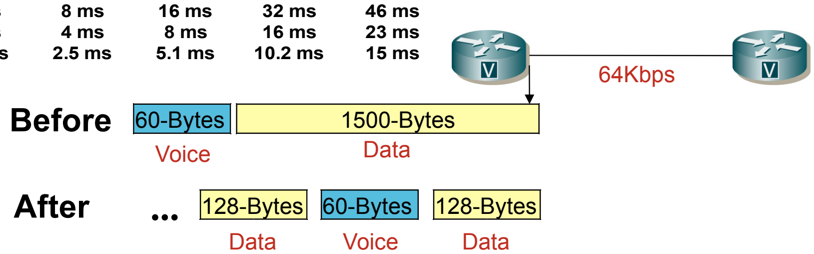
- **Phones buffer packets to handle minor jitter**
  - **Packets with large jitter arrive too late and are dropped**
  - **Route flapping**
  - **Multipath load balancing**



Call Operation - Jitter

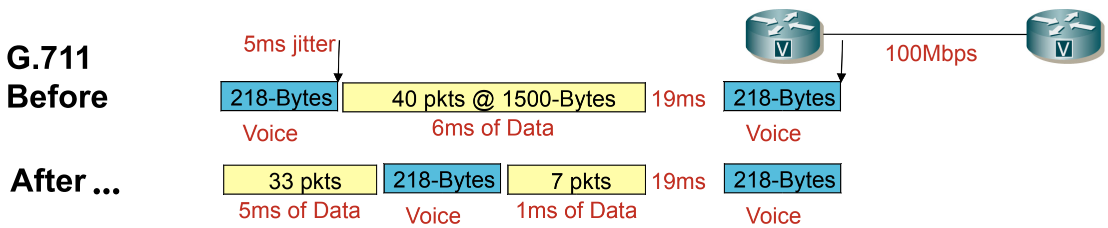
- **ITU Spec: 30ms jitter**
- **Big packets delay voice on low speed links**
- **Use Link Fragmentation and Interleaving (LFI)**
  - **Choose fragment size for delays of about 15 ms**

	Packet Size (bytes)					
Link Speed	**64**	**128**	**256**	**512**	**1024**	**1500**
64Kbps	8 ms	16 ms	32 ms	64 ms	128 ms	187 ms
128Kbps	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256Kbps	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512Kbps	1 ms	2 ms	4 ms	8 ms	16 ms	23 ms
768Kbps	0.6 ms	1.2 ms	2.5 ms	5.1 ms	10.2 ms	15 ms



Call Operation - Jitter

- **Inconsistent or no QoS implemented**
  - **Series of big packets delay voice**
  - **Only occurs when a link is oversubscribed**
  - **Priority queue moves voice to the front of the queue**
  - **Caution: Priority queue can starve lower priority queues; use policing to limit its effect**
  - **Configuration details vary among products**



Call Operation – Packet Loss

- **ITU Spec: 1% packet loss (codecs handle 5%)**
- **Incorrect or no QoS configuration**
  - **Oversubscribed priority queue with policing**
    - **Designed for 4 concurrent calls, 20ms rate**
      - **G.729 on Frame Relay: 28.14 kbps ***
      - **G.711 on Ethernet: 91.56 kbps ***
    - **Facility expands and 8 concurrent calls occur**
    - **Policing on priority queue drops excess traffic**
  - **Monitor QoS queue drops**
- **VoIP traffic not properly classified**
  - **Dropped when congestion occurs**

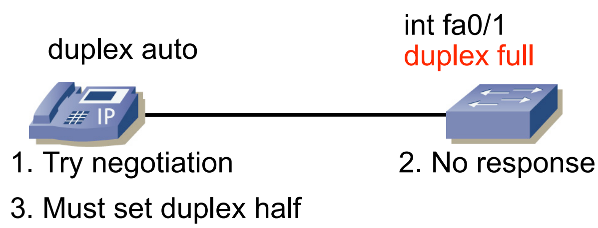
*** google: "cisco codec bandwidth" for calculators**

Call Operation – Packet Loss

- **Duplex mismatch (very common)**
  - **Fixed configuration on one end of link**
  - **The fixed configuration end doesn't negotiate**
  - **Look for errors: FCS, Runts, Late Collisions**
  - **Use Auto-negotiate for phones**

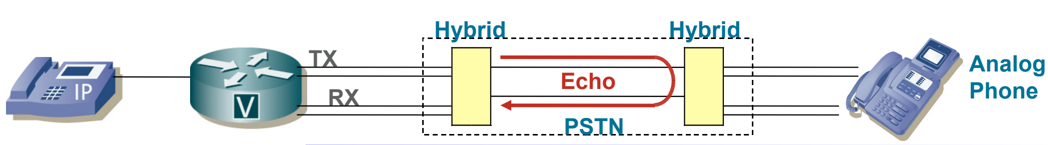
```
interface FastEthernet 0/1
 duplex auto
```

- **Bad cabling**
  - **Bad crimp**
  - **Cat 3 cable**
  - **Pinched cable**
- **Use 'duplex auto'**



Call Operation – Echo

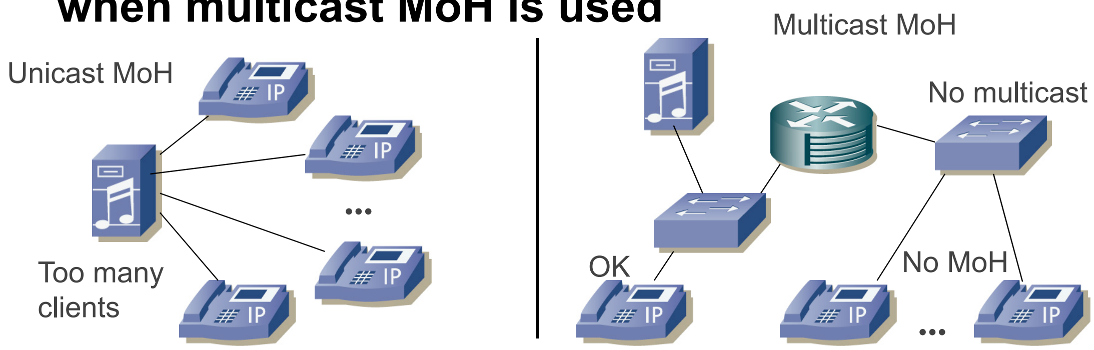
- **Symptom: Excessive talker echo (the most common)**
- **Acoustic echo - speaker output feed-back**
  - **Speaker phone or cheap earphone on remote end**
  - **Increase echo processing timer**
- **Electrical echo**
  - **Connection to analog via two-wire to four-wire *hybrid***
  - **Reduce output gain & increase input attenuation in small steps (10% - 20%)**
  - **DSP bugs**
- **Delays inherent in IP telephony accentuate echo**



Music on Hold

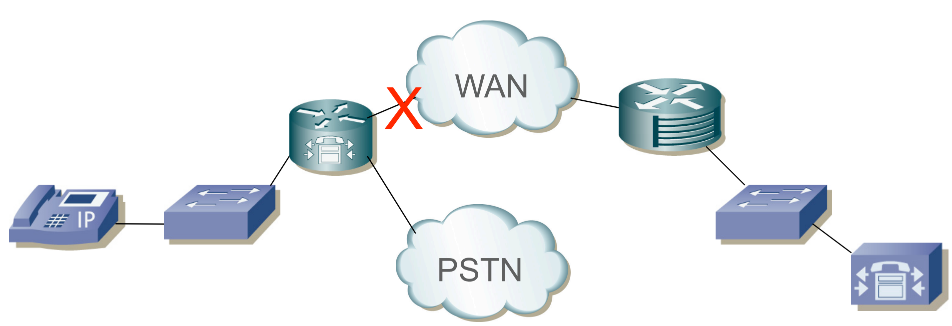
- **Symptom: Music on Hold not on some phones**

- **No MoH resource defined for the phones**
- **MoH resources exhausted, typically when unicast playback is selected**
- **Multicast routing not consistently configured when multicast MoH is used**



Survivable Remote Site Telephony (SRST)

- **Symptom: Phones can't register with SRST Router**
- **SRST not configured on phone & router**
- **More phones or directory numbers than SRST router supports**
- **Short DHCP lease (increase to 8 days)**



Summary: Troubleshooting

- **Configuration mistakes are the major cause of problems**
- **Collect data; subdivide the problem**
- **Test hypothesis and repeat**

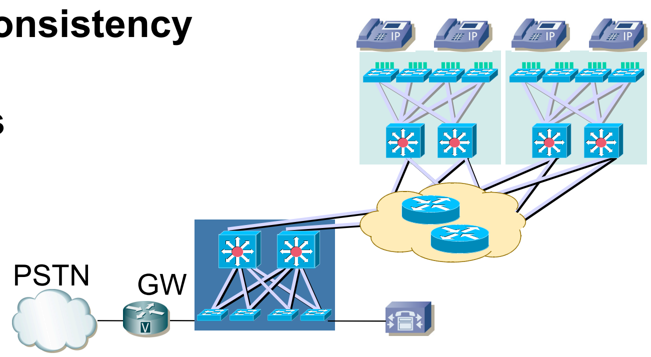
Applications (VoIP)
Communication Protocols (TCP/UDP/IP)
Routing & Switching Protocols (OSPF, STP)
Network Hardware & Links (Routers & Switches)

- **Use the Network and Operational Models to subdivide the problem and aid troubleshooting**

Call Operation	**Misc Operation and Services**
Call Setup	
Connectivity and Registration	

Manual Monitoring Doesn't Scale

- **Above 20-50 devices is too big**
- **Check system interdependencies**
  - **Root bridge depends on the switches in the STP domain**
  - **Duplex mismatch depends on connected device**
  - **Routing protocol consistency**
  - **VoIP call quality**
  - **QoS configurations**



Monitoring Requirements

- Real-time
  - Events; Performance; Error detection
- Trending
  - Historical utilization and operational data
- Configuration management
  - Saving configs and checking against policies
- Latent problem detection
  - Combining data to find potential problems

Metrics

- Measurable
  - Link, CPU, memory utilization
  - QoS queue drops
  - Interface errors
- Actionable
  - Must be usable for identifying and fixing problems
- Update frequency
  - Nyquist sampling theorem: sample at 2X the freq of the data
  - Dependent on the use
    - Trending and historical
    - Real-time & diagnostic

Realtime – Events

- Syslog & SNMP traps
  - Sent asynchronously by network gear
  - High volume (particularly firewalls)
  - UDP-based (unreliable delivery)
  - Informational through critical severity
- Log everything
  - Keep for historical reference
- Filters for different recipients
  - Network operations team
  - Unified communications team
  - Security team
- Sync device clocks with NTP
  - Correlate timestamps from multiple devices

Realtime – Event Processing

- Handling the volume
  - Filter out unimportant events
  - Tune filters over time
- Daily summary report

```
Summary of GNS Cisco syslog Messages on Wed Jan 17 23:59:00 EST 2007
Cisco Messages:
   437 DUAL-5-NBRCHANGE
   353 LINEPROTO-5-UPDOWN
   114 CRYPTO-6-IKMP_MODE_FAILURE
...
Messages sorted by frequency and source device:
   346 test1.com DUAL-5-NBRCHANGE
   114 test2.com CRYPTO-6-IKMP_MODE_FAILURE
    84 test3.com LINEPROTO-5-UPDOWN Tunnel119
    67 test4.com DUAL-5-NBRCHANGE
```

Realtime – Cisco Events

- Cisco: "System Error Messages for Cisco Unified Communications Manager"
- CCM_CALLMANAGER-CALLMANAGER-3-CallManagerFailure
- CCM_CALLMANAGER-CALLMANAGER-3-SDLLinkOOS: Cluster communications link failure
- CCM_CALLMANAGER-CALLMANAGER-4-MediaResourceListExhausted: media resource type not found
- CCM_CALLMANAGER-CALLMANAGER-3-TspError: phone registration problem
- LINK-3-UPDOWN: backbone and important links
- CDP-4-DUPLEX-MISMATCH: high utilization links
- LINK-4-ERROR: excessive link errors
- SYS-5-RESTART: device restarted
- DUAL-3-SIA: EIGRP routing protocol problem
- SYS-{1345}-SYS-LCPERR{1345}: Cat 6500 internal error

Realtime – VoIP Performance

- Delay, Jitter, Loss stat collection
  - Cisco: Call Detail Record (CDR) & Call Maintenance Record (CMR) collection
  - Avaya: RTCP stream directed to collector
- ITU specs:
  - Delay: 150ms one-way
  - Jitter: 30ms
  - Loss: 1%
- Determine your thresholds
  - Military often uses much higher values
  - 1% packet loss is terrible for data
  - NY to SF is 30ms one-way

Realtime – Triggers

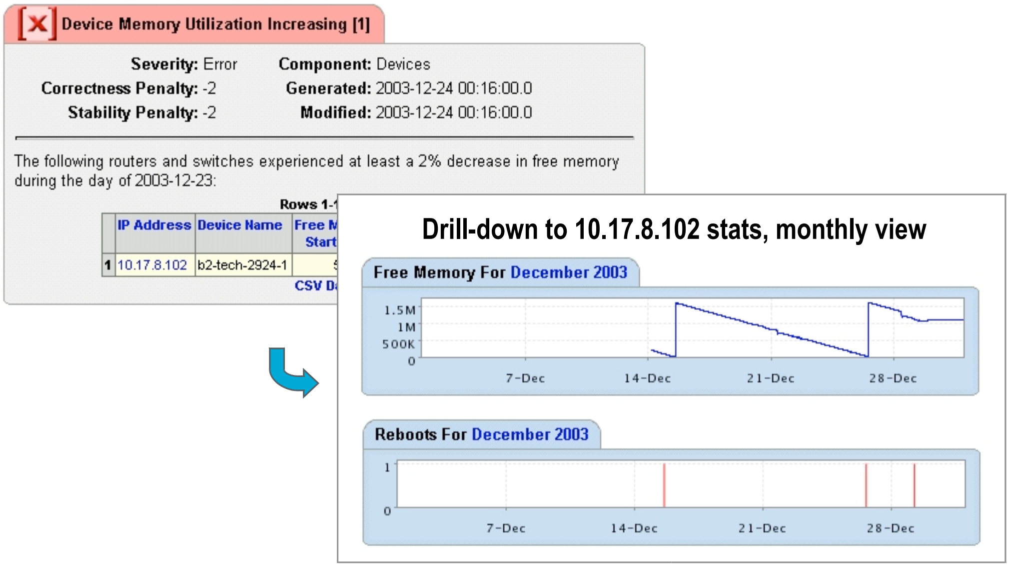
- Call completion failure codes
  - search cisco.com "Call Termination Cause Codes"
- Environmental failures other than events
  - High power supply utilization
  - Fan failure (should be an event, but uses UDP)
  - Temperature
  - UPS battery reserve, AC supply status, etc
  - Change in STP root bridge
  - Redundant router (HSRP/VRRP) change

Trending

- Correlate with configurations to find latent problems
- Trends in call quality (CDR/CMR trending)
- UPS battery life and planning replacements
- CPU & Memory utilization trends, particularly in software-based routers
- QoS queue drops

Trending Example

- **Memory leak – router crash every twelve days**



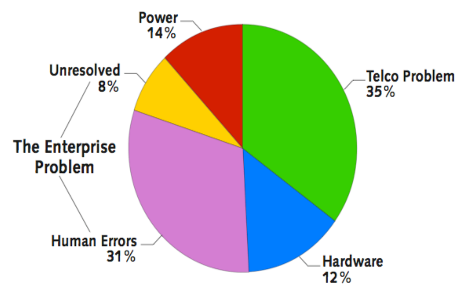
Trending – VoIP Resource Utilization

- **DSP pool utilization (CISCO-DSP-MGMT-MIB)**
  - cdspCardResourceUtilization
    - Indicates the percentage of current DSP resource utilization of the card
  - cdspCardLastHiWaterUtilization
    - Indicates the last high water mark of DSP resource utilization
  - Calculate total utilization across all cards
- **Trunk channel utilization & CUCM monitoring**
  - CISCO-CCM-MIB-V1SMI: ccmGatewayTrunkTable
  - Calculate utilization from total and in-use counts
- **Metric**
  - 70% for growing organization; 90% for no growth

Configuration Management

- **Greatest impact on network stability and faults**
  - Majority of network problems are due to configuration mistakes
  - More than 40%; amount depends on the analyst
  - Impossible to get to five-nines without it
- **What to track**
  - Who made the change
  - What changed
  - When was it changed
  - Use a AAA server (Radius or TACACS+)
- **Critical in VoIP networks**

Causes of Network Downtime
Source: Yhe Yankee Group 2002 Network Downtime Survey,



Configuration Management

- **Basic requirements**
  - Configuration archive
  - Check Running vs Saved configurations
  - Log configuration changes
  - Tools to view changes

Running Config @ 2004-01-02 09:54:30	Saved Config @ 2003-12-01 04:03:23

Change Count: 7

```
!                                                   !
version 12.0                                        version 12.0
service timestamps debug uptime                     service timestamps debug uptime
service timestamps log uptime                       service timestamps log uptime
service password-encryption                         service password-encryption
!                                                   !
hostname tr3-c-rsm-2                                hostname tr3-c-rsm-2
!                                                   !
aaa new-model                                       aaa new-model
aaa authentication login default local              aaa authentication login default local
enable password 7 1373319D035C726E98                enable password 7 1703015B015C423E98
!                                                   !
username greg password 7 015B72841B2D93             username fred password 7 11460516071630
                                                    username sally password 7 02080E57415B9A
                                                    username john password 7 006A13107D4E58
ip subnet-zero                                      ip subnet-zero
```

Configuration Management

- **Example: The Site That Lost Its VoIP**
  - Major VoIP deployment
  - No automated tools in place
  - All routers and switched updated at the site
  - Two weeks later: power outage at the site
  - VoIP is down
  - Analysis: Configurations were not saved to NVRAM

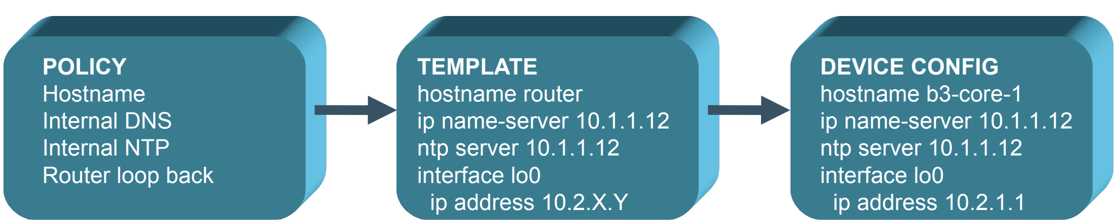
2008-03-28 00:38:03 Config Difference [4] 3 1

Config Difference [4]

	IP Address	Device Name	Device Type	Saved Differences	Running vs. Saved Differences	Diff
1	172.23.24.3	hwlab-s2	Switch (99.0%)	2	0	+
2	172.23.24.29	r2	Router (99.0%)	2	0	+
3	172.23.23.65	ts1	Router (99.0%)	0	2	+
4	172.23.24.33	vpn1	Router (99.0%)	0	2	=

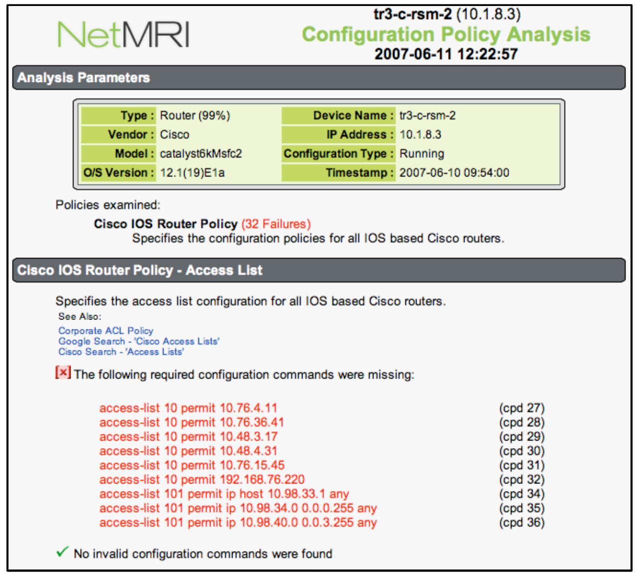
Configuration Policy

- **Policy definition process**
  1. Policy defined
  2. Template created
  3. Per-device modifications made to template
  4. Install final configuration in the device
- **Policy is infrequently reviewed afterwards**
  - Configs divert from policy as changes accumulate
  - Manual method are tedious and error-prone



Validating Configuration Policy

- Not just regulatory – check best practices
- Mechanism
  - Compare templates with device configs
  - Identify differences
  - Create an alert
- Value
  - Validate existing policies
  - Identify devices that don't match a new policy



Fixing Configuration Policy Exceptions

- Remediation
  - Some policy exceptions can be automatically fixed
    - Duplex mismatch
    - Bridge priority
    - Router ARP timer > switch CAM timer
  - Service impacting changes need manual application

- Without automated policy validation, configs become inconsistent

- QoS policies
  - Trusting QoS in the right places?
  - Correct QoS marking policies in place?

Latent Problems – No Redundancy

- **HSRP & VRRP**
  - No redundant router
  - First failure was not noticed

HSRP Not Recognizing Peer [34]

Component: Routing
Severity: Error
Correctness: -2
Stability: 0

Analysis Task: HSRP Issues
Generated: 2007-04-25 00:00:01
Analysis Start: 2007-04-24 00:00:00
Analysis End: 2007-04-25 00:00:00

The following routers are not recognizing their HSRP peers during the day of 2007-04-24. In such cases, the router may not be receving HSRP hellos from the neighbor router. Check the communications between the routers and the HSRP configuration of the routers in the HSRP group.

See Also: Cisco Search - HSRP not recognizing peer
Export Details as CSV Data File

	Adds	Changes	Same	Drops	Supp
Summary	+ 30	≈ 1	= 3	– 0	S 0

Rows 1-10 of 34
Standard View: 34/34 >>|

	S	Virtual IP	HSRP Group	Router IP	Router Name	Unknown Peer	Diff
1		10.1.217.1	217	172.27.22.5	tr3-c-rsm-1	Standby	=
2		10.17.2.1	91	10.17.8.2	B2-dist-rsm-1	Standby	+
3		10.17.8.1	30	10.17.8.2	B2-dist-rsm-1	Standby	+
4		10.17.16.1	90	10.17.8.2	B2-dist-rsm-1	Standby	+
5		10.17.48.1	110	10.17.8.2	B2-dist-rsm-1	Standby	+
6		10.17.64.1	120	10.17.8.2	B2-dist-rsm-1	Standby	+
7		10.17.80.1	130	10.17.8.2	B2-dist-rsm-1	Standby	+
8		10.17.96.1	140	10.17.8.2	B2-dist-rsm-1	Standby	+
9		10.17.112.1	150	10.17.8.2	B2-dist-rsm-1	Standby	+
10		10.17.128.1	160	10.17.8.2	B2-dist-rsm-1	Standby	+

Select All — 0 selected — Clear All
CSV Data

Latent Problems – Wrong Root Bridge

- **Root Bridge**
  - Must determine switches in spanning tree domain
  - Check bridge priority on all switches in the domain

VLANs

Rows 1-20 of 176

	VLAN ID	VLAN Name	Root Bridge	Count
1	1	default	t34-dist-6506-1	24
2	1	default	tr3-c-6509-1	5
3	1	default	t56-dist-6506-1	17
4	1	default	b1-dist-6509-1	33
5	1	default	B2-dist-6509-1	52
6	1	default	DMZ-Dist-3550	1
7	1	default	00:0A:8A:89:87:00	2
8	1	default	00:0A:8A:89:E3:80	1
9	1	default	b3-dist-6513	2
10	1	default	core-fw-3508-1	1
11	1	default	t12-dist-6506-1	19
12	2	primary-to-core	t12-dist-6506-1	1
13	2	question	tr3-c-6509-1	1

VLAN Root Details

VLAN ID :	1	Bridge Max Age :	2000
VLAN Name :	default	Bridge Hello Time :	200
Root Bridge :	tr3-c-6509-1	Bridge Fwd Delay :	1500
Root Priority :	8192	Top Changes :	0
Root Bridge ID :	0x20:00:00:0A:42:B0:B4:00		

VLAN Switches

Rows 1-5 of 5

	Device Name	VLAN Name	Priority	Bridge Address	Timers
1	rmt-sites-2950-1	default	49152	00:09:B7:F7:78:C1	OK
2	tr3-c-6509-2	default	16384	00:0A:42:B0:A4:00	OK
3	**tr3-c-6509-1**	**default**	**8192**	**00:0A:42:B0:B4:00**	**OK**
4	tr3-VoIP-2950-1	default	49153	00:0A:8A:89:E0:C1	OK
5	dist-ed-4503-1	default	32769	00:0C:CE:96:53:80	OK

CSV Data

Summary

- The network is the foundation for VoIP
- VoIP is a complex system – many interdependencies
- Monitor key parameters with automated tools
- Use the Network and Operational Models to subdivide the problem and aid troubleshooting