IPCOMM2006

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IP Telephony Fundamentals:
What You Need to Know to “Go To Market”

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What Will Be Covered

- What is Voice over IP?
- VoIP Technology Basics
- How Do I Know if We’re Ready?
- What “Real” Cost Savings Should I Expect?
- Putting it All Together
- Conclusion, Q&A

What is Voice Over IP?

- The Simple Answer – It's your "traditional" voice services transported across a common IP infrastructure.
- The Real Answer – It's the convergence of numerous protocols, components, and requirements that must be balanced to provide a quality voice experience.

Recognize the Reality of IP Telephony

- IP is the catalyst for convergence of technology and organizations
- There are few plan templates for convergence projects
- Everybody seems to have a strong opinion
- Requires an educational investment in the technology (learning curve)
  - Requires an up-front investment in the technology that can be leveraged for subsequent deployments
- Surveys indicate deployment is usually more difficult than anticipated
- Most implementations are event driven (that means there is a broader plan)

IP Telephony vs. VoIP

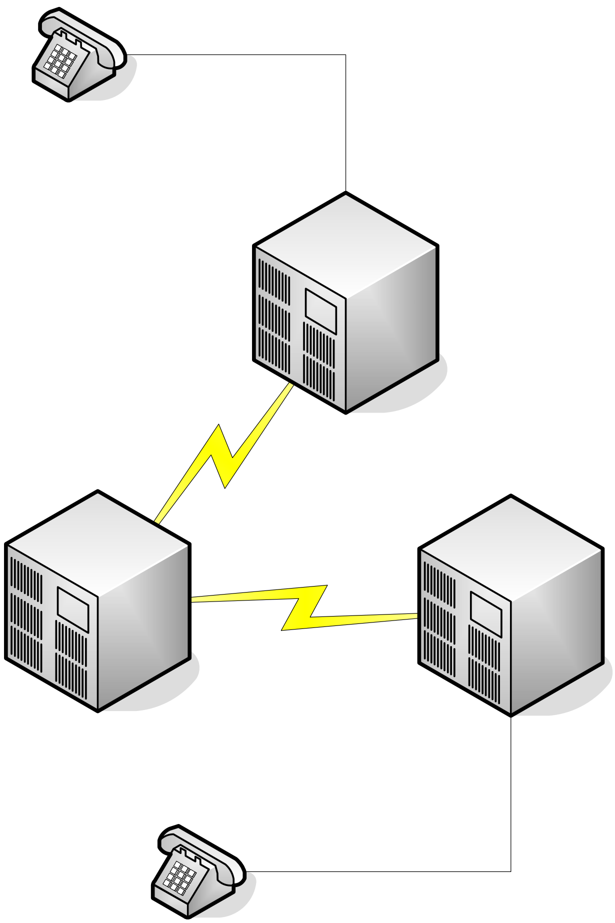
- Voice over IP – A broad technology that encompasses many, many facets.
- IP Telephony – What you're going to implement to actually deliver services across your network
  - Focuses more on features than possibilities
  - Narrows focus to specific implementations and requirements
  - Sets appropriate context for discussions

Why Does Convergence Matter?

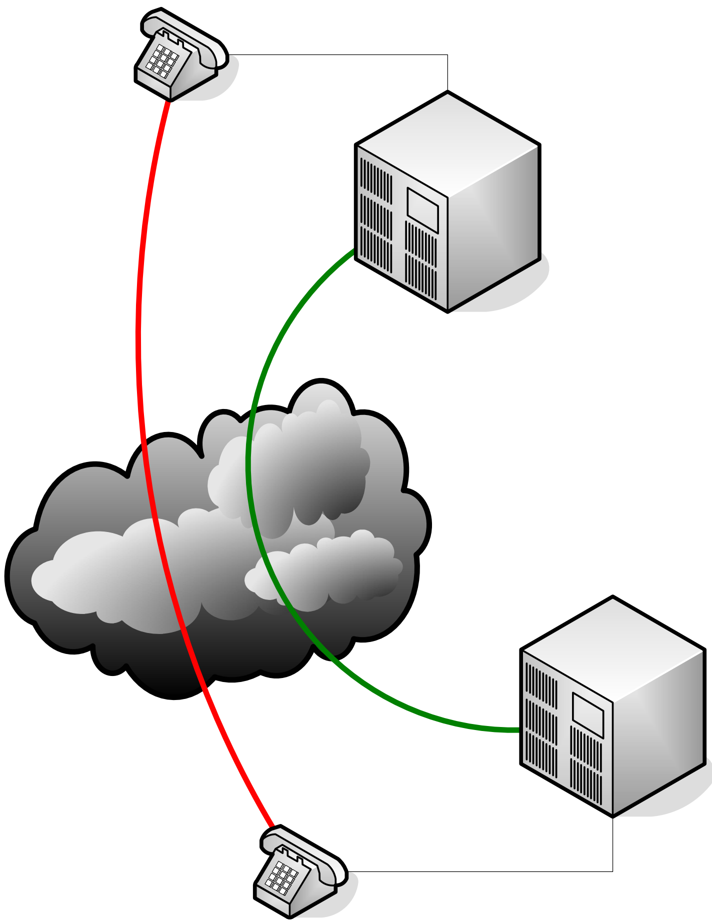
- Converged networks provide a means to simplify support structures and staffing.
- Converged networks create new opportunities for a “richer” communications environment
  - Improved Unified Messaging
  - Unified Communications
  - The Promise of Video
- Converged networks provide methods to reduce costs (if you do things right)

The Basics – TDM (vs. VoIP)

- TDM – Time Division Multiplexing
- Traditional phone Networks “Nail Up” a Line from End-to-End
- Provided Dedicated Resources, But Creates “Forced Idle Time”
- Quality Ensured By Engineering for Single Standard, Volume



The Basics (TDM vs.) VoIP



- With VoIP, Traffic Flows Across Some Common IP Network
- Call Control Servers Direct/Control Traffic to Varying Degrees
- Call Flows (Normally) Go Directly From Endpoint to Endpoint
- Call Path is not fixed and may change dynamically

So What's The Advantage?

- VoIP can enable more efficient use of circuits deployed through effective CODEC selection
- When no voice traffic is present, the line is not automatically idle
- With high bandwidth connections, extremely high calling volume supported on a single interface (1Gbps Ethernet = approx. 12,000 calls)
- If multiple routes are available, connection can re-route without interruption.

IP Telephony Devices and Roles

- Call Server/Processor
  - Databases of information
    - Phone Configurations
    - Dial Plans
    - Phone Number to Handset Mappings
    - Feature Configurations
  - Provides “the Brains” of the Operation
    - Directs inbound calls
    - Determines routing for outbound calls
    - Determines features available to users and handsets

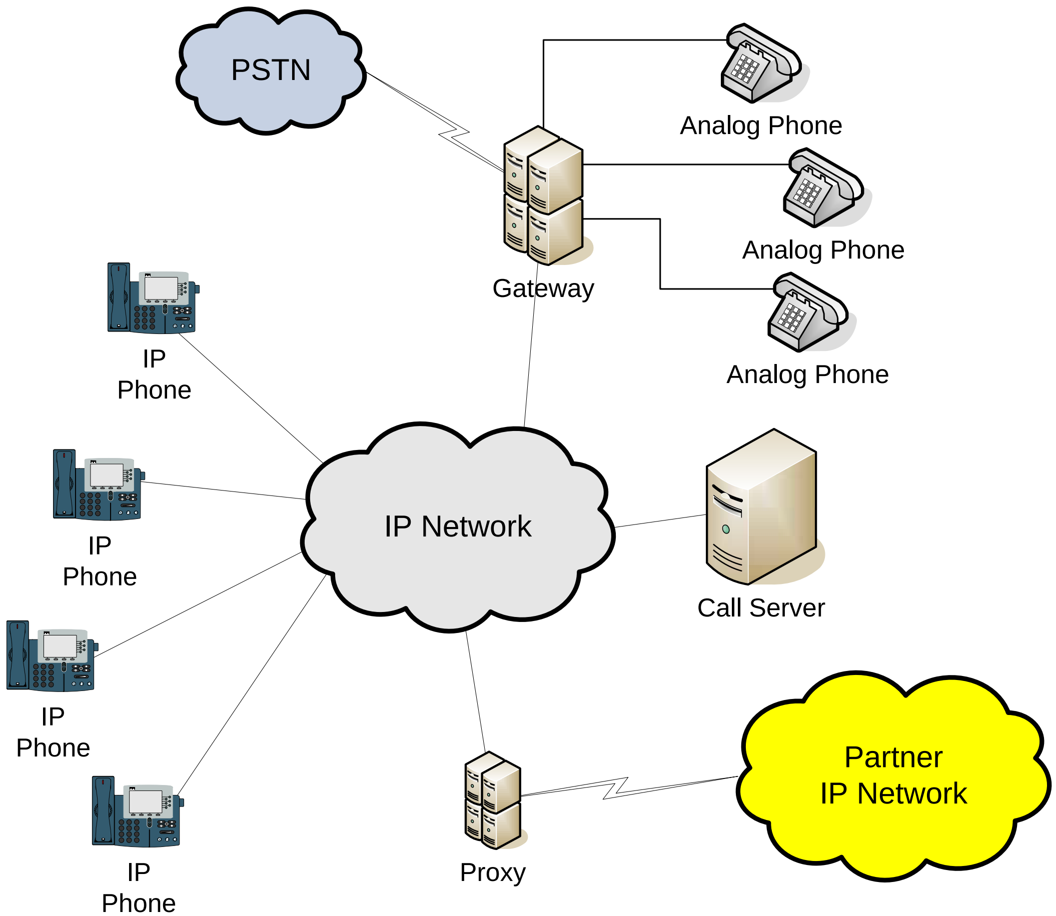
IP Telephony Devices and Roles

- Gateway (or Media Gateway)
  - Defines boundaries between “types” of calls
    - IP Telephony to PSTN
    - Translation between different Codecs
- Proxy
  - A Server, Device, or Gateway that acts as a “Middle-man” in a conversation
  - Typically used to provide security between a trusted and untrusted domain
  - Can be used to increase visibility
- Analog Telephone Adapter
  - A “basic” gateway that supports an analog phone on an IP system
  - Typically used for remote sites and provides limited options if service unavailable (think Vonage)

IP Telephony Devices and Roles

- Handset or "Hard Phone"
  - Exactly what it says
    - May be fully IP capable
    - May be digital/analog set supported by a Gateway
  - Features available will depend on combination of phone and IP Telephony system
- "Soft" Phone
  - Software run on a computer
  - May use built-in microphone/speakers, could support variety of headsets
  - Becoming more and more capable of leveraging other connection types (cellular, existing analog connections)

IP Telephony Devices – What's it Look Like?



So Why Isn't This "Everywhere?"

- Not everyone has the right hardware in place
- We need a single, authoritative database or way to translate Phone Numbers into IP addresses
- Everyone needs to choose to "open holes" to their networks to accept inbound calls
- Common hardware capabilities are needed to create and sustain the necessary connections
- Movement is in this direction, but still some time before we get there…

How Does VoIP Impact Data Networks

- Multiple Impacts on Many Aspects
  - Reliability of Data Networks Often not “Voice Ready”
  - Voice Communications Subject to Sensitive (and Finicky) Quality Issues
  - Bandwidth for applications often varies, usually in “spikes” – How does this impact voice support?
- Data Networks Require Re-engineering to Support Voice
  - Reliability improvements through diversity or technology change.
  - Quality of Service must be configured to preserve voice quality
  - Traffic analysis required for end-to-end connections to maintain quality under load
  - Management Platforms Must Adapt.

Targeting Reliability: Availability = Uptime

Availability Target	Total Downtime Per Year
99.9999%	32 Seconds
99.999%	5 Minutes 15 Seconds
99.99%	52 Minutes 36 Seconds
99.95%	4 Hours 23 Minutes
99.9%	8 Hours 46 Minutes
99.5%	1 Day 19 Hours 48 Minutes
99.0%	3 Days 15 Hours 40 Minutes

How Can We Improve Reliability?

- Increase the number of routes to any endpoint
  - Redundant Ethernet connections between switches
  - Redundant WAN/MAN connections or dial-backup capabilities
  - Change technologies (e.g. Private Line to MPLS)
- Avoid deploying “SPOFs” for critical infrastructure
  - Can you afford two call control servers?
  - Can you deploy two voice gateways?
- Stick to standard protocols and interfaces

Critical Data Resources for IPT

- Dynamic Host Control Protocol Server
  - Automatically assigns IP addresses to stations
  - Must include appropriate information for IP Phones
- Domain Name Services
  - Translate names (www.plannet.net) into IP addresses (10.20.20.101)
  - Some IPT systems may require dynamic registration of phones to DNS servers
  - If possible, best to use names for core IPT servers for configuration purposes

Critical Resources for IPT

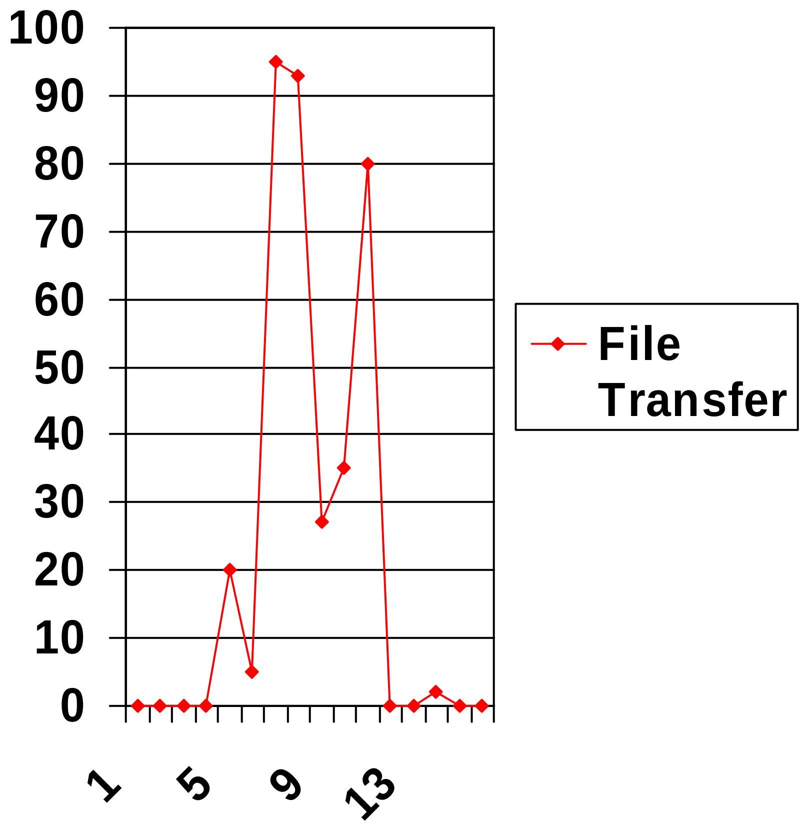
- Switches and Virtual LANs
  - Virtual LAN – a logical grouping of devices to isolate traffic (security, efficiency, or bandwidth reservations)
  - Prevents Voice and Data traffic from “mixing” in a single logical network
    - Enhances security
    - Simplifies Quality of Service activities
    - Can minimize sharing of bandwidth
  - Important note – most IP Phones have a two port switch built-in

Critical Resources for IPT

- Power
  - Most PBX's provide power to phone over phone line
  - IPT Systems can support this in a variety of ways
    - Phone with dedicated power cord (not typical)
    - Phone using proprietary Power over Ethernet (not typical)
    - Phone using standard Power over Ethernet (typical)
  - Industry standard is IEEE 802.3af
    - Requires that network switches support standard
    - Switches may be limited in ability to power all jacks under certain conditions
    - What do you put on a UPS and where?
  - Requires that you think about how to handle power outages
    - Do you power <u>every</u> phone?
    - Do you power strategically placed phones?
    - What UPS capacity do you need and where?

How Does My Data Network Impact Voice?

- Data traffic flows tend to be very erratic and "jagged."
- Aggregating traffic loads like this tends to "smooth" the peaks over time.
- Voice can be very sensitive to sudden congestion this can cause
- Networks must be designed to permit both to coexist.



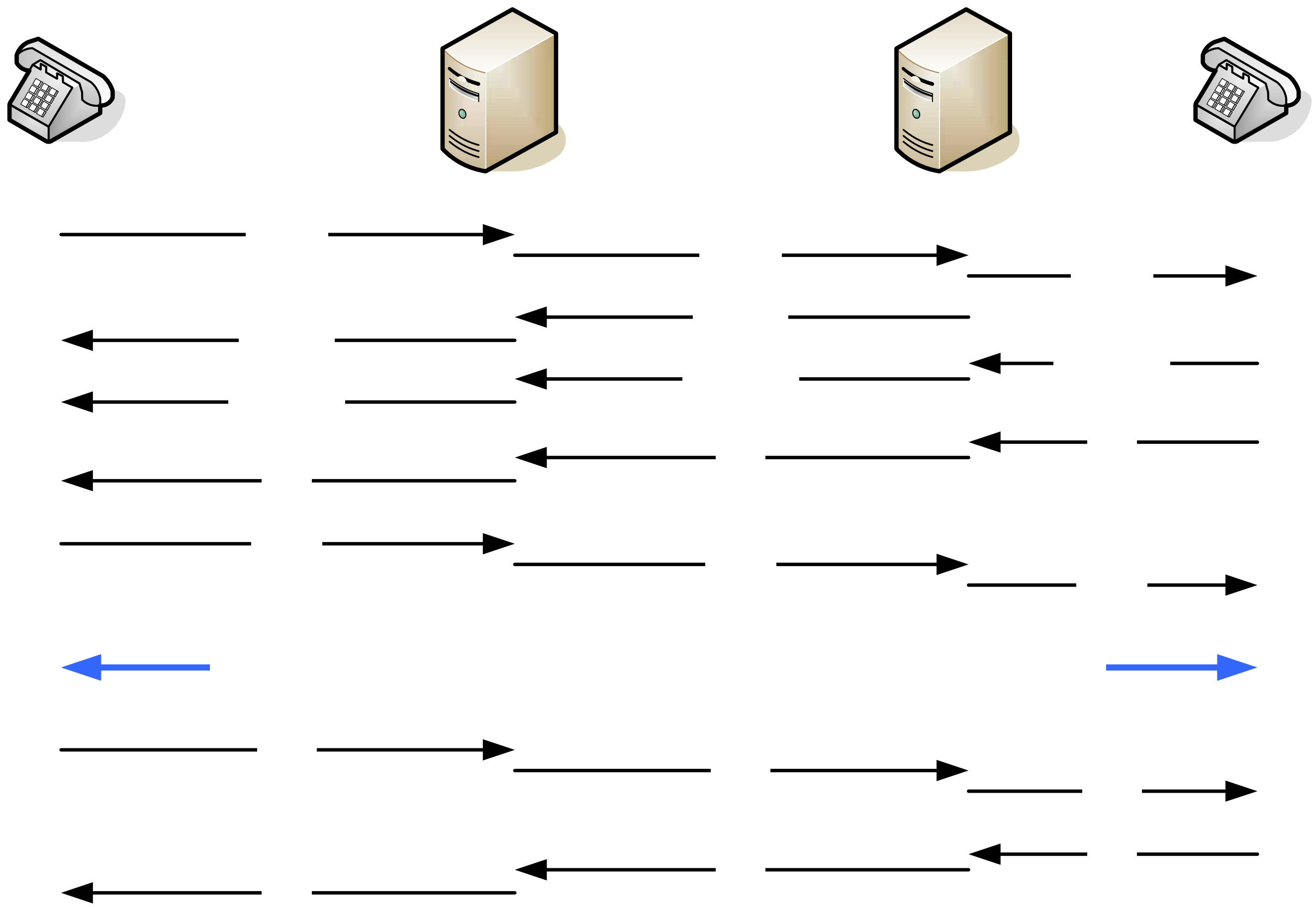
One Key Difference to Understand

- Data Traffic Typically Uses a “Reliable” Protocol (Transmission Control Protocol)
  - Every packet has a sequence number
  - If the receiving station doesn’t “see” a packet number, it asks for retransmission
  - Protocol “throttles” connection when loss is detected
- Voice Traffic Uses an “Unreliable” Protocol (User Datagram Protocol)
  - No sequence numbers, no retransmission
  - If a loss occurs, there’s no value in sending the packet again
- Result – Data Transmissions correct dynamically, Voice “Just” Loses Pieces of the Transmission

What's SIP (and Why Do I Care)?

- SIP – Session Initiation Protocol
  - Standards-based method of controlling calls
  - Supports end-point to end-point and end-point to server/proxy/gateway communications
  - Provides "Building Blocks" to Establish Calls, Video-conferences, Text Messaging Sessions
  - Can be "Extended" to Include Context (Presence)
- SIP competes with proprietary signaling protocols, but do you care?
  - Yes <u>if</u> a SIP system provides all required features
  - No <u>if</u> a SIP system leaves holes you can't close
  - Maybe <u>if</u> you can use any handset you can purchase with full feature compatibility

A Basic SIP Call



What SIP Really Is

- SIP is a Standard Published by the Internet Engineering Task Force (IETF)
- SIP Defines “Primitives” That Systems Leverage to Perform Functions
  - Make a Call
  - Receive a Call
  - Transfer (to Station or Voice Mail)
  - Music on Hold
- SIP is Designed to be “Vague” and Flexible

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Critical Things About SIP

- SIP is not the protocol that actually carries voice traffic (usually, RTP does this)
- While SIP is a standard, different vendors have "extended" the standard to enhance features
  - SIP defines features implemented on calling servers
  - SIP defines the features available on end-points
  - Features provided <u>must</u> match up between the two for system to work
- In some implementations, SIP does not provide feature parity with proprietary protocols

What's "Presence?"

- Presence is the Ability of a PBX to:
  - Leverage Multiple Communications Methods (Voice, Video, Text Messaging, E-mail)
  - Direct Calls Intelligently to Users
  - Provide Users Ability to Create Schedules
- Presence is Often (wrongly) "Linked" to SIP
  - SIP Provides an Open Standard that Simplifies Many Aspects of Presence
  - Presence is Really an Application That Leverages IPT, Not Necessarily a "Core Component"

What's My CODEC (And Why Do I Care)?

- CODEC – Coder/Decoder
  - A set of rules that defines how voice (analog) will be coded and decoded to and from digital medium
  - Typically based on International Telecommunications Union (ITU) standards (i.e. G.711, G.729)
  - Required to allow VoIP to work at all
- Why do I care what my CODEC is?
  - CODEC defines bandwidth used and quality demands
  - CODEC sets Maximum MOS Score
  - Both end-points must use the same CODEC (or something has to translate)

Codecs and Voice Quality

Codec Characteristics				
Standard	Total Payload	Payload/ Packet (Bytes)	Bandwidth Used (Ethernet)	Maximum MOS
G.711	64Kbps	160	87.2 Kbps	4.4
G.726	32 or 24 Kbps	80 or 60	55.2 Kbps or 47.2 Kbps	4.2
G.728	16Kbps	60	31.5 Kbps	4.2
G.729a	8Kbps	20	31.2 Kbps	4.2
G.723.1	6.3 and 5.3Kbps	24,20	21.9 Kbps or 20.8 Kbps	4.0 or 3.5

Bait & Switch?

- Payload and Bandwidth are not equal! Why?
- Data Network Protocols "Stack" – Each With a Different Function
- Each Protocol Has a "Header" – Overhead to Communicate Protocol Data
- Payload + Headers = Bandwidth Demanded

PPP Header	IP Header	UDP Header	Real Time Protocol – G.711 Data

Ethernet Header	IP Header	UDP Header	Real Time Protocol – G.711 Data

FR Header	IP Header	UDP Header	Real Time Protocol – G.711 Data

I Have MOS?

- MOS = Mean Opinion Score
  - Measurement of voice quality
  - Originally based on actual user samples/surveys
  - Now calculated based on detected network parameters
  - Score from 1.0-5.0 (4.0 is toll quality)
- Often Reported by Management Systems From Recorded Measurements
- Useful Tool to Find Users' Sensitivities, Report on Issues

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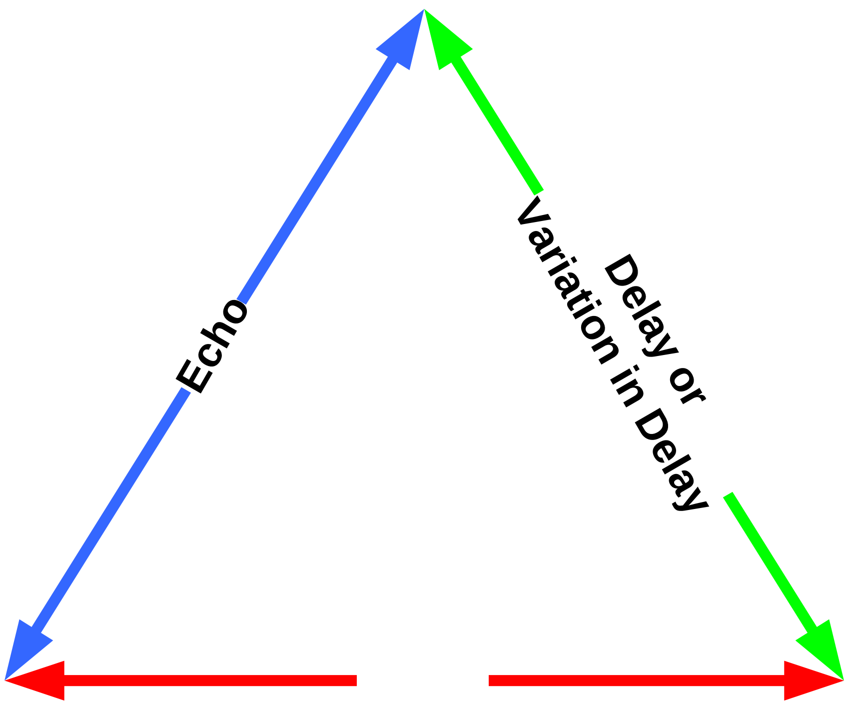
So How Do I Ensure Quality?



- Minimize Delay (Latency)
  - Total Delay End-to-End
  - Often Dictated By Distance
- Minimize Variation in Delay (Jitter)
  - Voice expects one payload every *X* ms.
  - Buffering can inject unacceptable jitter
- Minimize Loss (Discard)
  - Data Networks Handle Congestion by Dropping Data
  - Examine Congestion, QoS
- Control Echo

Latency

- Total Delay to Transmit a Packet Between Endpoints
- Sources of Latency:
  - Distance
  - Codec (Time to Digitize/Compress Voice Sample)
  - Serialization Delay (Time to Put Packet on Wire)
  - Buffering/Queueing
- Of the Latency Sources, Only Buffering/Queuing Can Be Handled With Network QoS

<u>Jitter</u>

- Variation in Delay
- Codecs Expect to Encode/Decode One Packet Every *X*ms
- Buffering/Queueing Can Occur At Every “Network Hop”
  - Many Traffic Streams Converging from One Sources
  - Many Traffic Streams Converging from Different Sources
- By Making Voice Packets “Wait in Line,” Delay Between Packets Can Vary
- Network QoS Required for <u>Every</u> Connection/Link

Discard

- Data Networks Address Congestion by Discarding Packets
  - Typical Algorithm is Relatively Random
  - Modern Algorithms Weight Packet Lengths, Other Factors
- Loss Causes Speech Clarity to Degrade (Static, Choppy Speech)
- Network QoS Settings Can Reduce/Minimize Loss, but Required for Every Link

Echo

- Some Echo is Necessary to Maintain Call Quality
- Additional Echo May be Injected
  - Digital to Analog Conversions
  - Inappropriate Configuration of Calling Platforms
  - Bad Headset Selection
- Usually Controlled Through Selection and Configuration of Devices

An Important Safety Tip

- Increasing Bandwidth Rarely Results in Long-Term Quality Improvements
  - Typically only Addresses Issues with Loss Effectively
  - Data Applications Tend to “Discover” Available Bandwidth and Find Ways to Fill It
  - Will Not Resolve (And May Actually Worsen) Processor or Memory Issues
- Quality of Service Usually Handles Issues More Effectively, But…
  - …Don’t View QoS as a Guarantee
  - …Look at QoS as an “Insurance Policy”

When You Address Quality

- Don't Get Lost in the Metrics – It's About the User Experience
  - Network Parameters can Provide Threshold Alerts, Which Are Useful
  - If You Haven't Correlated Performance Parameters With User Opinions, You May Not Be Watching for the Right Things
- Make Sure to Budget for Updated Network Management Tools That Can Report On Voice Quality and Metrics
- Carefully Evaluate the Tools That are Right for You:
  - If Voice is a "Mission Critical" application, you probably want flow-based monitoring of some form
  - If Voice is important, but not critical, network metrics may be sufficient
  - Remember that the platform is as important as the data gathered and reported.

One Last QoS Thought

- Under the Right Conditions, One More Call Degrades All Communications
  - Heavily Congested Links
  - “Hard” Bandwidth Reservations
  - Overloading Buffers Causes Excessive Packet Loss
- Evaluate Availability, Utility of Call Admission Control (CAC)
  - Only allows calls to be placed when conditions are “right.”
  - May redirect calls to alternative routes
  - Based on the theory that quality is more important than quantity
- Engineer for Busy Hour Before Enabling CAC

How Can I Tell We're Ready – Data

- Before Moving Forward With Implementation, Evaluate Existing Environment
  - How old is your network hardware?
  - What are the capabilities of operating systems?
  - How much traffic are you currently carrying?
  - What are the capabilities of WAN connections?
- Assess Existing Capabilities as Best You Can Before Going to the Market

Basics of Data Network Assessments

- Manufacturers/VARs Normally Require an Assessment Prior to Implementation
  - Verify QoS Capabilities of Hardware
  - Verify Configurations
  - Confirm PoE Capabilities
  - Upsell Possibilities
- Recommendation – Don’t Turn Down the Assessment
  - Provides a “Get out of Jail Free” card to the Manufacturer/VAR
  - Often gets to a level of detail that’s greater than internal assessments
  - Frequently identifies new and useful features you don’t know about or don’t have (but really do want)

What Your Data Network Should Support

Key Hardware Needs

- Redundant Links to Call Servers
- PoE (Preferably 802.3af)
- UPS Systems for Some Coverage
- Hardware that is Currently Supported/Shipping

Key Software Needs

- Support for DiffServ (IETF)
- Support for 802.1q (IEEE)
- Support for 802.1p (IEEE)
- Multiple Queues per Port (to address QoS needs)
- Ability to load-balance, re-route around outage
- Reliable, recent, supported software version

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General Rules for Data Network Assessments

- Don't Get Up-sold Without a Reason
  - Most hardware purchased within 3-5 years can support IPT
  - Lack of PoE normally drives significant upgrade costs
  - Verify that software upgrades won't fix identified issues
- Be Prepared for Some Capital Costs
  - Uninterruptible power supplies
  - End of life hardware/software
- Don't Waste That Capital
  - Gigabit Ethernet to the Desktop is Almost Always <u>Unnecessary</u>
  - Gigabit Ethernet to the Core is Almost Always <u>Useful</u>
  - 10Gbps Ethernet Purchases Should Probably be Out of the Question

General Rules for Data Network Assessments

- When You Get the Report
  - Read it – many of our clients don't review in detail
  - Review it – decide which points might be valid and which need more information
  - Research it – verify points of contention and/or findings
- Act on the Findings
  - Many of our "firefighting" clients didn't follow the recommendations provided
  - Many of the problems encountered early in implementation are a result of failing to follow assessment recommendations
- Remember that Assessments Tend to be (IPT) Vendor Specific

I Need a Voice Assessment?

- Most “Boilerplate” IPT Plans Don’t Include a Voice Assessment
  - Assumed that voice needs will be addressed during implementation
  - Focus is on proving out overall network to support IPT in general
- Why You Want a Voice Assessment
  - You want to understand your call flows and busy hour needs
  - You want to make sure you understand all of the impacted devices
  - You want to understand the features that you currently use so they don’t get missed.
  - You want to verify all of the “special situations” in your network

General Rules for Voice Assessments

- Make Sure You Analyze Calling Patterns
- Verify Phone Configurations and Line Appearances
- Understand What Connects to Your Existing Phone System, Why, and How
- Understand User Groups and Needs
- Evaluate Call Flows Through Your Voice Network
  - What is working today?
  - What do people want to change?
- Verify Call Coverage Requirements and Needs (Particularly for High Profile Users)

When You Get the Voice Assessment

- Translate Features from "Vendor-Speak" to Functional Requirements
  - Don't use the marketing term, use a functional description
  - Prepare scenarios that your environment requires
- Make Sure Calling Patterns are Well Described
- Leverage the Assessment to do a Feature Survey
- Break Feature Requirements into Prioritized Groups

Now That You've Got Both Assessments

- Merge the Data as Best You Can
  - Will calling patterns overload certain links?
  - Does overlaying the data reveal new single points of failure?
  - Are you planning reliability "right" for what the assessments reveal?
- With Sufficient Detail, Look for Ways to Leverage Technology Better
  - Do you need strong multicast capabilities?
  - Do call durations indicate a use for unified communications?
  - Does an overlay indicate anything "interesting" about particular user groups?

What's "Real" About Cost Savings?

- Most IPT Vendors Will Focus on Cost/TCO
- Take a "Trust but Verify" Approach to TCO Tools/Analysis
  - Tools tend to be focused on particular areas
  - Don't trust a tool if you can't review the formulas it's using
  - Set your own TCO requirements/parameters and run an independent analysis
- The Truth – You Can See Positive TCO, But It's Not Automatic

What's Real About MAC Costs

- Most Frequently Cited Cost Savings
- Real Savings for Many MAC Activities
  - User can relocate phone without technician <u>required</u>
  - Depending on system, adding new instruments may also be significantly simplified
- MAC Activities Can Still Incur Costs
  - Moves often involve new/changed line appearances
  - Moves often involve new/upgraded handsets
  - Moves may require updates to features
- Reality (That We've Seen) – MAC Costs Drop, but Often by Less Than Forecast

What’s Real – Operations Cost Savings

- Promise is Greater Efficiency and Utility from Technical Staff and Systems
  - Merged support staff assumed to be smaller than different groups
  - Expectation that “self-service” applications will reduce reliance on staff
  - Assumption that centralized network monitoring interfaces will simplify management and problem solving
- In the First Year of Many Deployments, These Savings Often Aren’t Realized

What's Real – Operations Cost Savings

- To Realize Operating Cost Savings
  - Approach merging support groups carefully
    - Don't let Data Group "Win by Default"
    - Don't let all your voice talent walk out the door
  - Update processes as you plan deployment
    - Planning processes (in particular) are often quite different
    - Take yourself from the Help Desk call to closing the ticket
- Most Important – Be Prepared to Invest
  - Staff Training (Formal and OJT)
  - Network Management Tools
  - Time to Establish Effective Team Communications

Please Don't Expect

- Don't Count on Any Reduction in Phone Bills Unless
  - Very Heavy International Usage
  - Very Heavy Inter-Office Calling Involving Long Distance
  - You Have Exceptional Opportunities for Toll Bypass
  - You've Negotiated a Terrible Contract for Voice Services
- Current Costs from Carriers Make Significant Savings on Usage Extremely Unlikely

What's Real – Application Cost Savings

- If You Only Think of IPT Deployment, You'll Miss the Benefits of New Applications
  - Will Unified Messaging Benefit You?
  - Are you currently using an Auto Attendant?
  - Are enhanced applications for your system available?
  - Can you leverage third-party systems more effectively?
  - How can you leverage Presence?
- Don't "Just" Replace the Box – Take the Time to Look at How You Can Use It Better

So Why Change to IP Telephony?

- Doing IP Telephony Right Can Reduce Costs
  - Operational Costs
  - Greater End-User Efficiency
  - Access to New Applications and Communications Methods
- Doing IP Telephony Creates New Communications Opportunities
  - Presence is Real, and Only Getting Better
  - Usually Simplifies Developing New Applications That Leverage Voice
  - Great Flexibility and Scalability
  - New Carrier Services Coming

Keep These Things In Mind

- Don’t Focus on the Technology, Focus on How You Can/Will Use It
- To Recognize TCO, Don’t Forget Processes and Organizational Issues
- Have Assessments Done and Don’t Ignore the Findings
- If the Data Network Isn’t “Voice Ready,” Your Deployment Will (Eventually) Have Issues
- Remember – You’re Doing This to Provide Service to Your Users

QUESTIONS?

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