Optimizing Converged Cisco Networks (ONT)

Cisco VoIP Overview

http://www.INE.com

Instructor Introduction

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Basic VoIP Components

- Phones
  - Analog
  - IP Phones (SCCP, SIP)
  - Soft Phones
  - Video Phones
- Gateways
  - Connects VoIP network and PSTN network

VoIP Signaling Protocols

- MGCP
  - Commonly used for gateways
  - Client/Server (Call Agent controls gateway)
- H.323
  - Used for Gateways/Gatekeepers
  - Umbrella for control protocols (H.225/245 etc)
- SIP
  - Trunks, Endpoints
  - Open Standards

Basic VoIP Components (cont.)

- H.323 Gatekeepers
  - Call routing
    - Name or number to IP resolution
  - Call Admission Control (CAC)
    - Are there enough resources to place the call?
- Multipoint Control Units (MCU)
  - Conference bridge
  - Multiplexes signals into a single stream

Basic VoIP Components (cont.)

- Call Agents
  - e.g. Cisco Unified Communications Manager (CUCM)
  - Call control/routing
  - Call Admission Control (CAC)
  - Bandwidth control
  - Address translation
- Application & Database Servers
  - Provide TFTP & XML services for IP phones

Basic VoIP Components (cont.)

- Digital Signal Processor (DSP)
  - Converts digital to analog signal inside gateway
  - e.g. router's Packet Voice DSP Module (PVDM)

VoIP Designs

- Analog phones over IP network
  - Gateway converts analog signal to IP packets and sends to IP network
- IP phones over analog network
  - Gateway converts IP packets to analog signal and sends to PSTN
- IP phones over IP-only network
  - No conversion needed

Analog Interfaces

- Gateway uses three main interfaces to talk to analog devices and PSTN
  - Foreign Exchange Station (FXS)
    - Connects to analog end station and provides power
    - e.g. router connection to analog phone or fax
  - Foreign Exchange Office (FXO)
    - Acts as the end station
    - Receives power from remote end
    - e.g. router connection to PSTN
  - Earth & Magneto / Ear & Mouth (E&M)
    - Analog trunk
    - e.g. PBX to PBX or PBX to PSTN

Digital Interfaces

- Basic Rate Interface (BRI)
  - 2 x 64kbps Bearer (B) channels
  - 1 x 16kbps D channel for out-of-band signaling
- T1 Primary Rate Interface (PRI)
  - 23 x 64kbps B channels
  - 1 x 64kbps D channel for out-of-band signaling
  - AKA Common Channel Signaling (CCS)
- T1 Channel Associated Signaling (CAS)
  - 24 x 64kbps B channels
  - Uses in-band signaling
  - AKA Robbed Bit Signaling (RBS)
- E1 CAS/CCS
  - 30 x 64kbps B channels
  - 1 x 64kbps D channel for out-of-band signaling

Phone Call Stages

- Call setup
  - Call routing
    - Where is the call going?
  - Call Admission Control (CAC)
    - Is there enough bandwidth?
  - Includes negotiation of port, codec, etc.
- Call maintenance
  - Monitor loss, jitter, delay, etc.
- Call teardown
  - Release the resources

VoIP Deployment Models

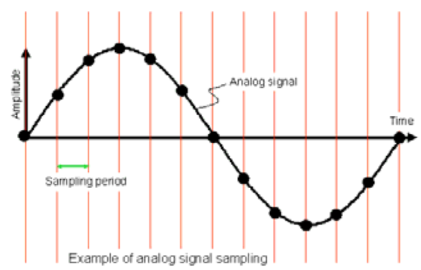
- Single site
- Multiple sites with centralized call processing
- Multiple sites with distributed call processing
- Clustering

Centralized vs. Distributed Call Control

- Single Call Agent (Centralized)
  - Smaller installations ~10-20,000 Users
  - All traffic is LAN based
- Multiple Call Agents (Distributed)
  - Larger installations ~10,000 users and up
  - Traffic traverses the WAN

Analog to Digital Conversion

- Sampling
  - Nyquist Theorem (8000 samples/second)
- Quantization
  - Digital representation of an analog waveform



Example of analog signal sampling

Analog to Digital Conversion

- Encoding
  - – Converting quantization values to binary
  - – 8 Bit designator for each sample point
  - – 8000 samples/second
  - – 8x8000 = 64000 kbps = uncompressed voice
  - – Standard 64kbps B channel
  - – Pulse Code Modulation

Analog to Digital Conversion

- Compression
  - – Reducing size of quantized bits
  - – Two Types
    - Adaptive Differential PCM (ADPCM)
      - – No longer commonly used (Quality Degradation)
      - – Lowest compression to 16 kbps
    - Conjugate Structure Algebraic Code Excited Linear Prediction (CS_ACELP)
      - – Widely used in VOIP
      - – Compresses to 8 kbps

Digital to Analog Conversion

- Decompression
  - – Expanding bit codes to full length
- Decoding
  - – Convert 8 bit binary segments to mapped points on quantization graph
- Reconstruct the signal
  - – Create analog sound wave to be played to called party

Voice Codecs & Compression

- G.711
  - – 64 kbps (Uncompressed, Highest Quality)
  - – Used within same site (same location)
- G.729
  - – 8 kbps (Compressed, Good Quality)
  - – Often used between sites (different locations)
- ***Bandwidth values do not include network overhead***

VoIP Overhead

- Packet Size *before* Layer 2 Overhead:
  - G.711: 200 bytes / G.729: 60 bytes
  - Includes: Voice Payload, IP (20 bytes), UDP (8 bytes), and RTP (12 bytes) headers
- Packet Size *after* Layer 2 Overhead:
  - G.711: 206-218 bytes / G.729: 66-78 bytes

Calculating VoIP Bandwidth

- Packet rate – Standard 50 pps
- Payload size – Depends on Codec
  - G.711 160 bytes / G.729 20 bytes
- IP overhead
  - 40 bytes Uncompressed / 2 or 4 Compressed
- Layer 2 overhead
  - Approximately 6 – 18 bytes
    - Ethernet, Multilink PPP, Frame Relay FRF.12
- Tunneling overhead

VoIP Encapsulation

- RTP: Real-Time Transport Protocol
  - More reliable protocol
  - RTCP: (Control)
    - Monitors quality of stream
      - Jitter, Loss, Delay
- cRTP: Compressed RTP
    - RTP, UDP, IP Header: 40 bytes > 2 or 4 bytes
    - Useful on slow speed WAN links:
      - G.729 Payload 20 bytes

Quality & Mean Opinion Score (MOS)

- Voice quality is measured using MOS
  - MOS Scale 1-5 1: Inaudible - 5: Perfect
    - MOS Goal: 4.5 (PSTN Quality)
- Metrics for Voice Quality:
    - Delay: (Mouth to Ear) Digitization, Packetization, Serialization
      - No more than 150msec one way
    - Jitter: Uneven arrival or packets (Uneven Delay)
      - No more than 30msec one way
    - Loss: Packet Drops
      - No more than 1 percent

Voice Activity Detection (VAD)

- On average 35% of a phone conversation is silence
- By default, even silence is sent as packets
- VAD stops voice stream each time a threshold of silence is reached
- CNG – Comfort Noise Generation
  - White Noise to eliminate "Call Drop Sound"
- VAD is not reliable, and not recommended

Hardware Resources - DSPs

- Terminating calls
  - Call enters router from PSTN (Analog) DSP converts to digital signal
- Conferencing
  - Binding multiple calls into a single conversation
- Transcoding
  - Codec Conversion
- Echo Cancellation

Call Admission Control

- CUCM or Gatekeeper
  - Bandwidth considerations
  - Rejected Calls (Dropped or Rerouted)
- CUCM Bandwidth: (Regions/Locations)
  - G.711: 80 kbps per call
  - G.729: 24 kbps per call
- Gatekeeper Bandwidth: (Zone/Sessions)
  - G.711: 128 kbps per call
  - G.729: 16 kbps per call

VoIP Q&A