Voice over IP Tutorial

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Introduction
What is VoIP?

- Legacy Telephony
  - TDM/SS7 based infrastructure
  - Traditional Class 5/Class 4 switches
- Voice over IP
  - IP-based packet infrastructure for PSTN voice transport
  - New elements that collectively perform traditional functions and more
- And what is IP Telephony?
  - Voice + Messaging + Video + Data over IP networks = IP Telephony
  - Public Internet : Best Effort Service
  - Managed IP Network : SLA based Service
The Packetization Process

Analog Voice (400-4000kHz) → Filter lower/higher frequencies → Sample at 8000 samples per second → Quantization → Digital Encoding into bytes

Digital Encoding into bytes

Echo Cancellation (if reqd.) → Silence Suppression (Optional). Yields ~50% bandwidth gains

Digitized Voice Packet encapsulated within RTP packet

Sample at 8000 samples per second → RTP packet encapsulated within UDP packet → Compression (Optional). Yields further bandwidth gains

UDP packet encapsulated within IP packet

Voice Over IP Packet

Voice Samples

RTP Payload

UDP Payload

IP Payload

RTP Header

UDP Header

IP Header
Traditional PSTN Network

SS7 signaling

SS7 network

Legacy Class 4/5 Switch
Legacy Class 4/5 Switch
Legacy Class 4/5 Switch

TDM network

TDM bearer

Call Control, Signaling, Bearer/Media and Features
VoIP Network Components
The Elements

- Terminals or Endpoints
  - IP Phones
  - Soft Phones/PC Phones
- Media converter
  - Media Gateway/PSTN Gateway
- Call Processor
  - Media Gateway Controller or Gatekeeper or Proxy Server or Softswitch
- Signaling Gateway
- Application Server
- Media Server
VoIP Network Paradigms

Centralized Model

- Central Intelligence
- Master Controller
- Dumb Slave GWs
- Dumb Slave GWs
- Dumb Slave GWs

Distributed Model

- Distributed Intelligence
- Intelligent Server
- Intelligent Client
- Intelligent Server
- Intelligent Client
- Intelligent Server
VoIP Protocol Soup

- H.323
- MGCP
- Megaco/H.248
- SIP-T
- BICC

Call Control/Signaling

Gateway Control

Bearer

TRIP

SDP

ENUM

RTP

More ……
Call Control Signaling Protocols: H.323

- ITU-T defined standard
- Originally developed for ISDN based multimedia services over LAN
- Distributed protocol model
- Consists of
  - Terminals
  - Gatekeepers
  - Gateways
  - Multipoint control units
- Umbrella protocol comprising of several other protocols like H.225, H.245, T.120 etc. defining RAS, capability negotiation etc.
- Binary ASN.1 encoding
- H.323v4 currently implemented everywhere
  - Future H.323v5
• IETF RFC 3265 (obsoletes RFC 2543)
• Developed for multimedia services over IP networks based on http model
• Designed to employ existing popular Internet protocols like DNS, SDP etc.
• Distributed model consisting of User agents and Servers
• Text-based implementation is perceived to be simpler, modular, easily adaptable to the www
Call Control Signaling Protocols: SIP

- Registrar
- Redirect Server
- Location Server
- Proxy Server
- IP network
- SIP Phone
- SIP User Agent
H.323 versus SIP: Reality?

- ITU protocols more tightly defined; IETF looks for looser working code
- H.323 older and more established; SIP relatively newer but fast catching up
- H.323 widely deployed today; SIP is being widely adopted by large players
- Importance of the Internet and web-based applications increasing
- SIP capable of giving service providers greater control of services, extensibility and interoperability with the www; hence, may eventually win the race
- For a long time however, both these protocols need to co-exist
  - Robust standards must be developed to define interoperability to make things easier
Source: Hughes Software Systems
Call Control Signaling Protocols: MGCP

- IETF informational RFC 3661
- Provides call control services in a packet network
- Early implementation of Master/Slave protocol
- Consists of media gateways and call agents
  - Call Agents-> centralized intelligent entities handling call control and signaling
  - Media Gateways-> dumb devices handling media
  - Call Agent communicates with Media Gateway via MGCP
- Now a closed effort from standards perspective
  - MGCP implementations do exist today. MGCP variants NCS/TGCP are adopted by Packetcable.
Call Control Signaling Protocols: Megaco/H.248

- Enhances MGCP
- Joint effort by ITU and IETF (IETF nomenclature- Megaco/RFC 3525, ITU nomenclature- H.248)
- Provides call control services in a packet network
- Adopts the Centralized model
- Supports IP/ATM networks
- MGC-MG communication via Megaco/H.248
- Deals with contexts and terminations
  - decouples physical terminations from logical (ephemeral) ones
  - more suited to handling multimedia
- More complete and robust, standard allowing for multi-vendor interoperability
- IETF RFC 3372
- Defines a framework to interface SIP with ISUP
  - To maintain feature transparency in the SIP network w.r.t PSTN to support IN services not supported in SIP
  - To deliver SS7 information (in its entirety) to some trusted SIP elements
- Integration methods
  - Encapsulation of ISUP within SIP using MIME
  - Translation of ISUP parameters to SIP header
  - Provision to transmit mid-call ISUP signaling messages through INFO method
Controller - Controller Protocols: SIP-T

- Implemented at SIP-PSTN boundary gateways
  - Carried end to end
- SIP-T is relevant in the following scenarios
  - PSTN origination, IP termination
  - IP origination, PSTN termination
  - PSTN origination, PSTN termination with IP transit
  - IP origination, IP termination: SIP-T is not required
Controller - Controller Protocols: BICC

• Development triggered by a need for a packet-based PSTN replacement
• Functional separation of call and bearer signaling protocols in a broadband network
  • IP/ATM bearers in addition to TDM bearer
  • Uses SS7 signaling (with extensions to ISUP)
  • Binding information allows correlation between call control and bearer
• BICC defines three capability sets
  • CS1: supports ATM-based (AAL1/AAL2) bearer
  • CS2: supports IP-based bearer
  • CS3: still in works to support advanced services and interoperability with SIP
SIP-T versus BICC

**SIP-T**
- IETF defined
- Defined to maintain feature transparency across SIP networks (deliver ISUP to SIP endpoints)
- Packet-based signaling and bearer
  - SIP signaling
  - IP bearer; ATM supported through RFC 3108 (may be some issues, but defined)
- Provides a transition to pure advanced multimedia services based SIP network

**BICC**
- ITU defined
- Defined to separate call control from bearer (extends ISUP to handle packet bearers)
- SS7 signaling, Packet bearer
  - Network is SS7 (ISUP) signaled
  - TDM/ATM/IP bearers
- Intended for packet-based next-generation network supporting all existing legacy services.
Bearer Protocol: RTP

- IETF RFC 3550 (obsoletes RFC 1889)
- End-to-end network transport services for multimedia applications
- Services include payload type identification, sequence numbering, time stamping and delivery monitoring
- Control protocol (RTCP) to monitor data delivery
- Can be used with any transport protocol
  - Depends upon underlying transport layer for QoS
  - Applications typically use RTP over UDP
Protocol selection is a strategic decision depending on existing network and future services planned. Ultimately, one winner will make it easy for all!
VoIP Network Performance
The Key Parameters

- Coding Algorithms
- Echo Cancellation
- Latency
- Jitter/Jitter Buffer
- Packet loss
- Transcoding/Tandeming
- QoS
- Reliability/Availability
- Quality
Coding Algorithms

- **Compression**
  - What codec is used and their corresponding bit rates
    - Greater the compression, more the encoding delay
    - Determining appropriate packetization times and packet length
  - MOS score of codec determines perceived quality

- **VAD and CNG**
  - At the transmitter
    - Detection of voice activity
    - Suppression of silence
  - At the receiver
    - Comfort Noise generation
    - Voice playback
Echo Cancellation

- Echo detection and cancellation
- Availability and echo signal return loss quality
- Adjustments to loudness rating
- Tail length is MG role dependent
Latency/Delay

- Packetization Delay
- Propagation Delay
- Network Processing Delay
- Jitter buffer delay and speech playback
- PLCs add about 5ms delay
- PSTN benchmark for toll quality voice is 150ms RTT (ITU G.114)
  - Delay greater than 300ms is completely unacceptable for toll quality
  - An occasional packet loss is tolerable, but latency beyond 200 ms RT is not.
• Jitter = Delay Variation
• Jitter Buffer compensates for jitter on the receiver side
• Jitter buffer size should be optimally chosen
  • Rule of thumb: Jitter Buffer size = atleast 2 x speech frame size
  • Absolute jitter buffer size = end-to-end delay variation + some safety margin
    • Used by Gateways that have more processing power
Packet Loss

- Packet loss should be below 1% for acceptable quality
- Use Codecs with packet loss concealment algorithms
  - E.g G.729, G.723.1 have built in PLC; add-on PLCs have to be used with G.711 and G.726
- PLC algorithms compensate about 40 ms of missing speech.
  - Delay >40ms & <200 ms, speech is clipped
  - Delay >200ms, speech dropouts
- Packet loss is mostly bursty in nature. Hence, packet loss performance is directly related to packet size, the shorter the better
Transcoding/Tandeming

- Transcoding: Two or more encodings of a signal through different types of non-G.711 codecs separated by G.711 e.g G.726 to G.711 to G.729A
- Tandeming: Two or more encodings of a signal through same types of non-G.711 codecs separated by G.711 e.g G.729A to G.711 to G.729A
- Transcoding increases distortion and delay
- Only one transcode can be tolerated before the network performance drops to unacceptable levels for most combinations of non-G711 codecs
Quality of Service

- Means to prioritize voice packets
- Real time voice packets receive higher priority than non-real time data packets
- Helps improve performance by decreasing delay/jitter for voice packets
- Significant delay/jitter events can be avoided only by implementing a proper QoS Strategy
**QoS Strategy**

- **Best Effort**
  - A class of service in which the network provides no guarantees to the edge equipment
- **Prioritized Queuing**
  - Differentiation in the queuing of traffic for various classes of traffic
  - Assigns a priority or classification to every IP packet
  - Packets are sent in order of priority
- **Traffic Engineered Tunnels**
  - Constraint-based (traffic sensitive) connection-oriented paths through a routed network
  - MPLS Label Path
  - ATM VC
Speech Quality

Speech quality important for

• Monitoring/fault-finding
• Service level agreements
• Optimisation of network

Factors that affect quality

» Background noise
» Silence suppression
» Low bit-rate coding
» Errors (mobile, packet)
» Delay
» Echo
» Handsets/access network

Quality will remain an issue so long as bandwidth or processing power are limited

• e.g. mobile, leased capacity

• Quality measures: MOS; PSQM (Perceptual Speech Quality Measure); PAMS (perceptual analysis measurement system); PESQ (Perceptual Evaluation of Speech quality)
• End-to-end speech quality is the key measure of voice QoS
The VoIP network must:

- provide a customer’s service preferences anywhere
- securely

- adjusting to the constraints of the networks and access methods used: Wireline, Wireless, 3rd Party, Customer Owned, or Public Internet

Clearly a list of codecs, packet sizes, loss rate and jitter targets are needed to ensure voice quality to a defined level of acceptability.
VoIP Network Security
New IP Requirements

- Viewing packets as part of sessions
- Policies are required for Sessions
- New IP Services are enabled to handle this
  - Routing sessions between different networks, carriers and domains
  - Session packet flow anchoring
  - Detect failures and reroute
  - Usage based Billing/reporting at session flow level
  - Session aware borders for security
- Two way VoIP communication impeded by NAT/NAPT
  - Signaling messages can be exchanged on defined port
  - Bearer messages are a problem
- Special tags in SIP message to permit two way communication
- Simple Traversal of UDP messages over NAT (STUN)
  - Creates NAT awareness in Clients
  - To modify SDP messages
Session Border Controller

- VoIP Firewall Traversal Solution for Carrier to Carrier Peering
  - Integrated SIP Application Layer Gateway (ALG)
  - Modify signaling traffic to accommodate NAT/NAPT
  - Dynamic ‘pinhole’ opening/closing
- Topology Hiding
  - Provide an address normalization boundary
- VoIP Media Anchoring Solution
- VoIP Session QoS / Service Level Agreement Solution
  - Per session based policing
  - Guaranteed service in congested environments
- VoIP Session Admission Control Solution
VoIP Regulations
Numbering Services
- Rate Centre Association of Numbers
- Impact on Number Conservation
- Number Portability Compliance for VoIP providers?

Information service versus Telecommunications service
- Access charges at Origination and Termination points?

CALEA
- Requires North American telecommunications carriers to modify their equipment, facilities, and services to ensure that they are able to comply with authorized electronic surveillance.
- Similar requirement for the VoIP?
VoIP Network Economics
Is VoIP economical?

- **What Service Providers want**
  - Decrease expense
    - Lower capex on new infrastructure elements introduced
    - Lower opex on maintenance of existing infrastructure
  - Increase ARPU
    - Increase talk-time and use of Value added services (Centrex)
  - Ease of feature incorporation
  - Ease of Application development and innovation
  - Distributed architecture, fewer elements
In 2003, IP accounted for the majority - 58% of network traffic - TIA

International VOIP calls grew from 0.2% of telephone traffic in 1998 to 10.4% in 2002 - Broadwatch News Analysis

Total Internet protocol (IP) revenues expected to grow in 2004 by 7.8 percent, achieving a total of $13.9 billion – TIA

Major VoIP Providers
- Vonage - 70,000 subscribers
- Free World Dial-up - 80,000 subscribers
- Yahoo BB - Over 3 Million subscribers
- FastWeb - 400K subscribers
VoIP Network Applications
Application Scenarios

- IP PBX
- IP Centrex
- Enhanced IP Telephony
- Class 4 Replacement
- Class 5 Replacement
- *And more*.....
A CPE based IP telephony service that replaces a traditional TDM PBX
A network based IP telephony service that leverages a traditional Class 5 based Centrex service.
A network based IP telephony service that provides multi-media voice over an IP network, in addition to basic Centrex features.
Class 4 Replacement

• Scenario
  • ILECs, CLECs, IXCs, Large Corporations

• Benefits
  • By-pass traditional long distance toll network (Class 4) carriers and their per-minute usage rates and run their voice traffic over IP networks for a reduced cost.
  • Lower costs with higher bandwidth efficiency

• Issues
  • Traffic engineering of IP network for PSTN QoS
  • Migration from Circuit to Packet-based Network
Scenario
Out of Region and “Greenfield” deployments

Benefits
Flexibility - Enable Rapid Deployment of New Services
Distributed Architecture rather than Hierarchical Class Model

Issues
Maturity of softswitch technology
Ability to support all legacy systems supported by a Class 5 switch
Evolution to VoIP network
Voice over Broadband Network Architecture

PSTN/SS7 Network

SS7/PSTN Network

Softswitch

SIP

H.248

IP Network

H.248

TDM Network

Class 4

Class 4

Gateway

GR-303 Interface

Voice Over Broadband
DSL, Cable, Wireless

Video

Voice

Data

Web
Evolution of VoIP - Generic View

Central Office

Line GW

Trunk GW

Multi Service Edge Switch

TDM CLASS 5 HOST

IXC

PSTN

Digital Centrex

Analog Centrex

Integrated Desktop (SIP Soft Client)

Enhanced IP Telephony (SIP)

Data PC

PDA or Laptop

Video

Voice

Data

Web