Operational Issues in VoIP

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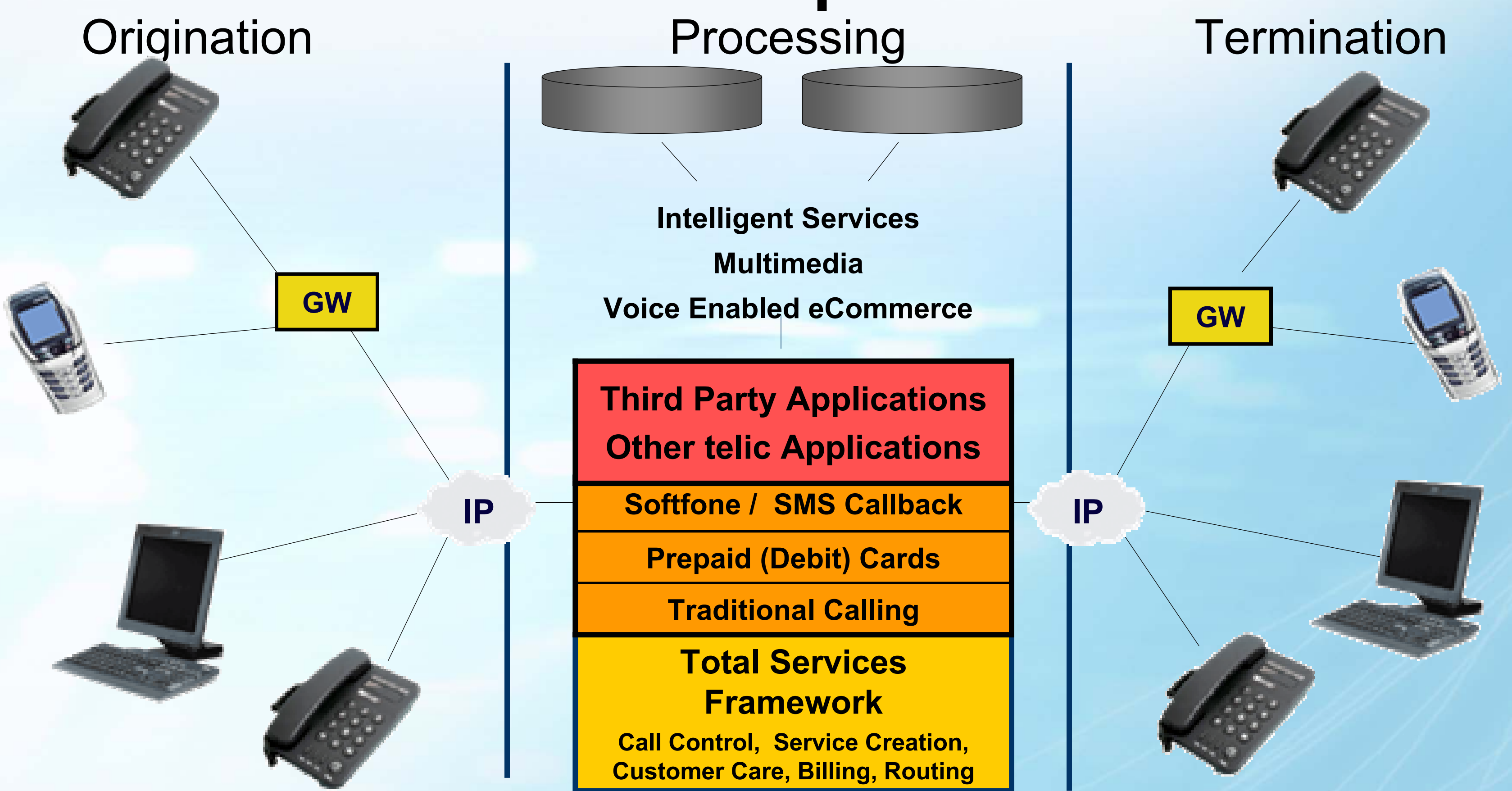
Objective

- To understand what issues and challenges organizations may face when providing Voice over Internet Protocol (VoIP) services to customers.
- Perspective: Application Service Provider (ASP)

Agenda

- Support Infrastructure
- Change Management
- Interoperability
- Quality of Service
- Legal Issues

ASP Model Perspective

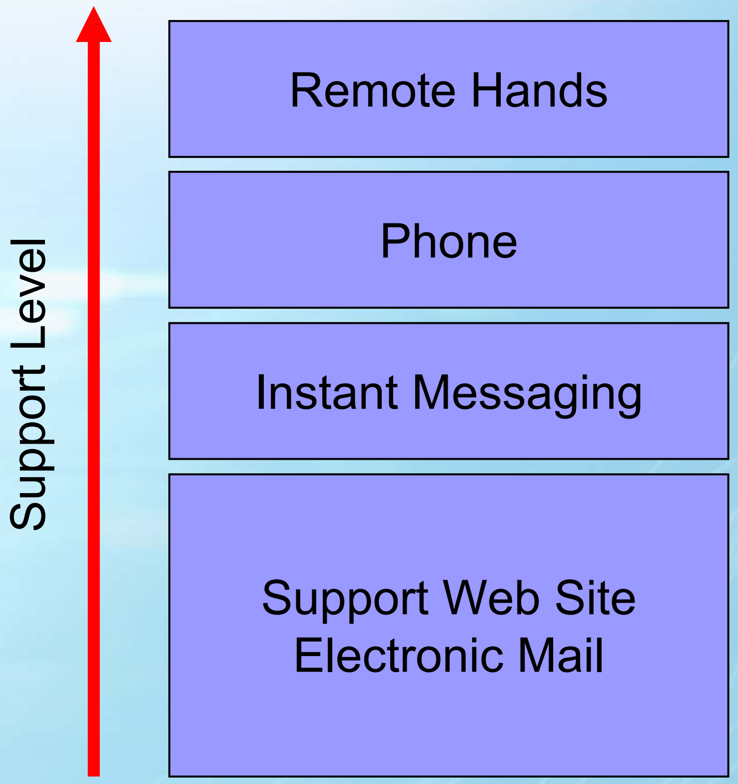


An end-to-end platform to provision and bill for IP enhanced services *in real time*

Topic: Support Infrastructure

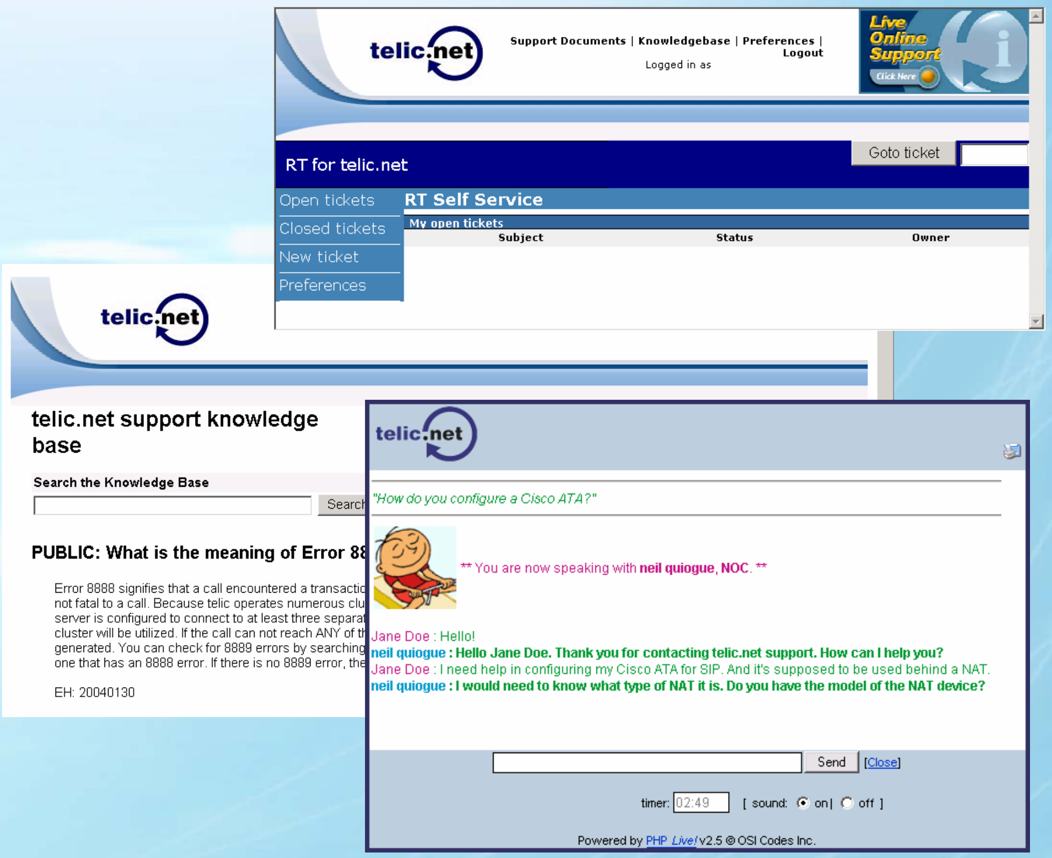
Support Infrastructure

- Support Web Site
- Electronic Mail
- Phone
- Instant Messaging
- Remote Hands



Support Web Site

- Centralized System for Communication between Parties
- Components
  - Ticketing System
  - Knowledge Base
  - Support Documents
  - Instant Messaging



Electronic Mail

- Tie up with a ticketing or help desk system for tracking of issues.
- Make sure that mail system has anti-virus and anti-spam setup.
- Handling spam e-mail
  - Do not auto reply to requests
  - Manage spam rules (e-mail from customers might be dropped)
  - Tag spam mail with appropriate headers for proper filtering

Instant Messaging

- Advantage
  - Real time communication
  - Cheaper than phone calls
  - Easier to understand since regional variations like accents and intonation are eliminated.
- Disadvantage
  - Customers expect instant responses also!
  - Customers tend to use IM versus sending electronic mail
    - Enforced by contracts or agreements
    - Management of expectations is necessary

Freely Available Instant Messengers

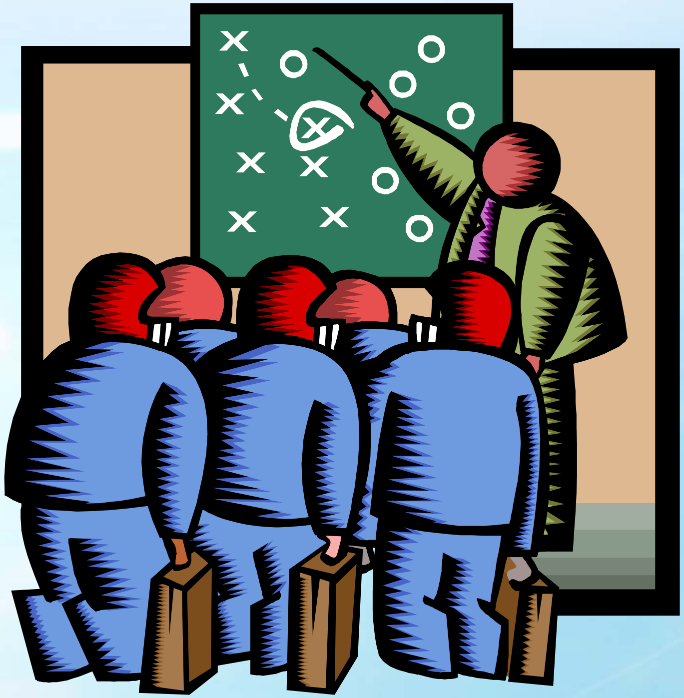
- Freely available like ICQ, Yahoo! Messenger, MSN Messenger and Jabber*
  - Advantage
    - Free!!!
    - Feature rich for single users
  - Disadvantage
    - Not really meant for support
    - Does not scale with multiple support staff and number of customers
    - Transactions are not often saved in ticketing system so history is left on user's computer
    - No control of maintenance

** Some may not be applicable to open source IM*

Commercial Instant Messengers

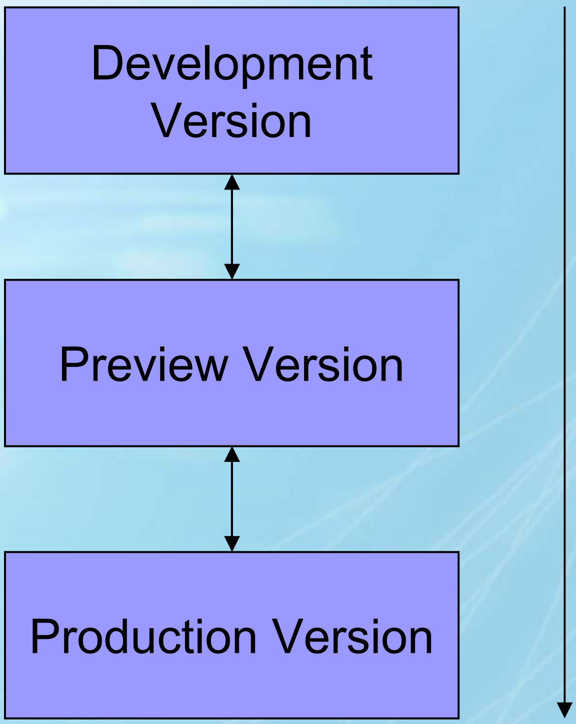
- Either pay for the application or pay for the hosted service
- Disadvantage
  - It costs money! ASP IM’s have monthly recurring charges.
  - Application On Own Server
    - Maintenance costs including bug fixes
  - Application Hosted
    - Minimal control on maintenance and availability
- Advantage
  - Customizable to needs or requirements of a service organization
    - E-mail when online support is not available
    - Transactions are saved centrally so administrators can query anytime
  - Scales with multiple support staff and customers based on a centralized environment.

Topic: Change Management



Software Change Management

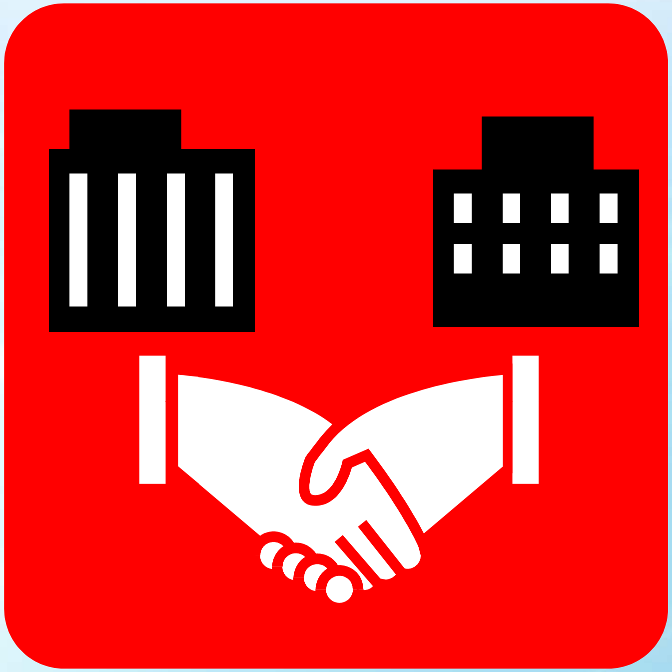
- Development Version
  - Alpha and Beta Testing of Applications
  - Users: Developers
- Preview Version
  - Release Candidate Testing of Applications
  - Users: Internal and Chosen Customers
- Production Version
  - Release Testing of Applications
  - Users: Everyone



Configuration Change Management

- Centralized Configuration Management
  - Web Based Configuration System
    - Restrictions can be imposed by interface and business rules
    - Accountability is imposed
    - Change history and revert to previous versions as needed
  - Deployment of Configuration
    - Secure Push (scp/sftp)
    - Secure Pull (rsync + ssh)
    - Configuration Maintenance (cfengine)
    - For Devices: tftp-based configuration

Topic: Interoperability



Interoperability

- Different interpretation of standards
  - Drafts are updated but applications are not
  - Applications only support part of the standards
  - Telephony services not yet fully defined
- Interoperability testing required for each device
  - Need to watch out for changes in software or firmware versions
  - Re-test as necessary

Test Scenarios

- Basic
  - Call Setup (H.323 or SIP)
    - H.323: H.225 Tunneling, Fast Start, GK Requirement?
    - SIP: SIP Registration, SIP Session Timer
  - DTMF (in-band, out-of-band, RFC 2833)
  - Interoperability with Applications (e.g., IVR, Callback)
- Behind NAT
  - Conduct previous tests behind a NAT device
  - Result is highly dependent on type of NAT

Interoperability Issues Encountered [1]

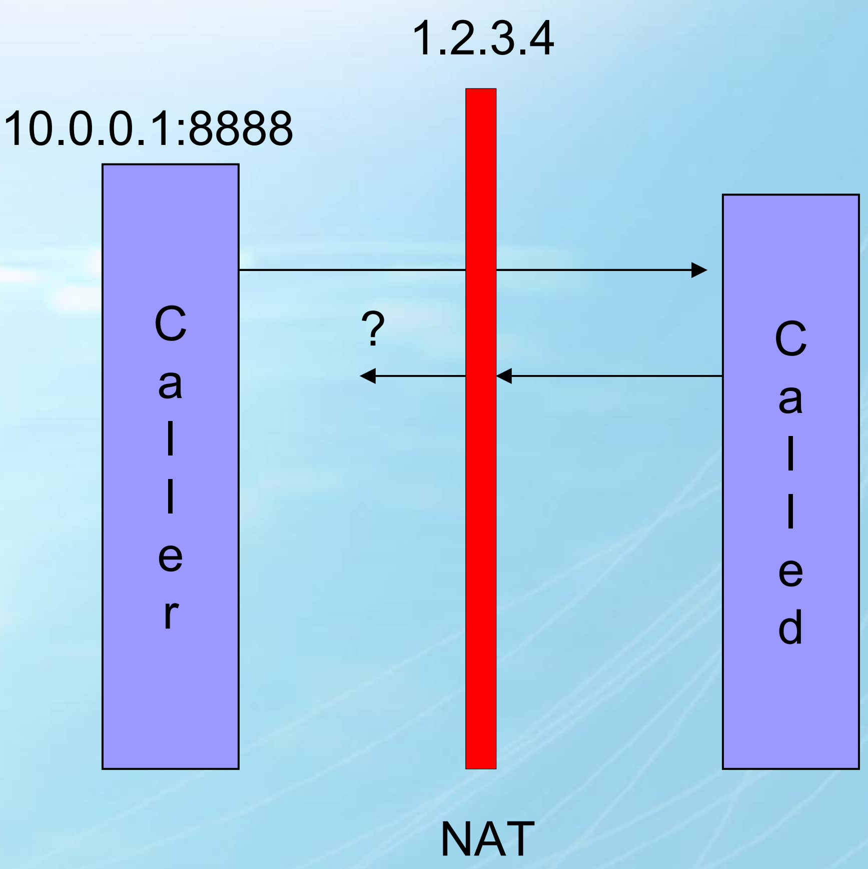
- Third Party Softswitch
  - Base RTP port was using odd number port.
  - RFC 3550 required even port for RTP and odd port for RTCP though provisions exist for an application to adjust accordingly
- Gateways With No Answer Supervision
  - All calls would be treated as seized even if it’s not answered.
  - Leads to billing issues.

Interoperability Issues Encountered [2]

- Terminating H.323 Gateway
  - Media is negotiated before sending trunk is busy error code. Results in calls not failing over.
  - A solution is to wait for “Connect” before negotiating the media.
- Runaway Calls
  - Neither origination nor termination received a BYE signal resulting in a runaway call
  - A solution is to use SIP session timers to act as keep-alive.

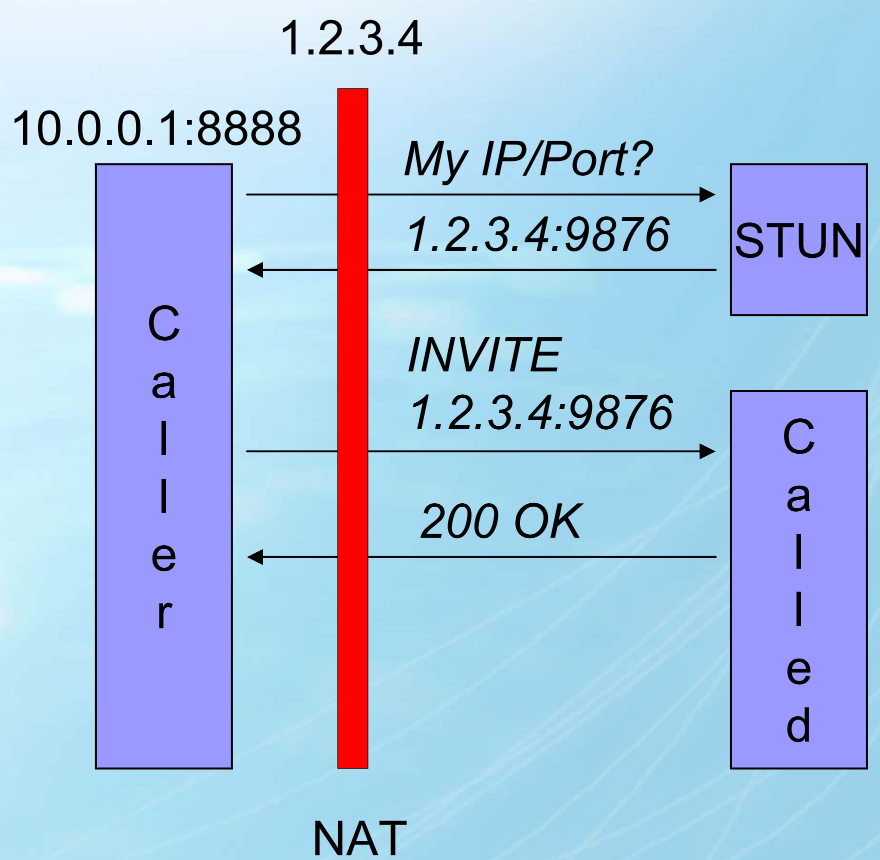
Interoperability with NAT

- NAT device needs to know to which internal IP and port to forward requests to.
- Due to differing implementations and behavior of NAT, issues occur (e.g., one-way audio).
- Very few VoIP (SIP) aware NAT deployed



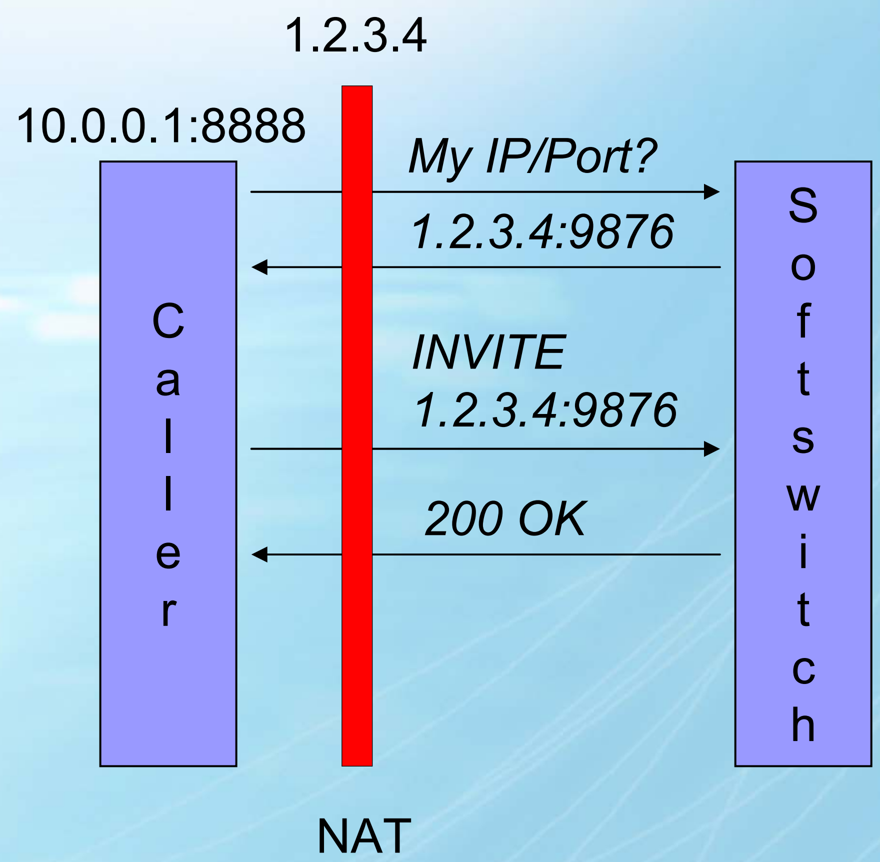
Simple Traversal of UDP over NAT (STUN)

- Operation
  1. Client sends request to STUN server
  2. STUN server copies Source Address to response
- Disadvantage
  - Does not work with Symmetric NAT
  - “Keep alive” needed for bindings to be maintained



Telic Solution for NAT

- Similar to Traversing Through Relay NAT (TURN)
- STUN + Media Proxy through the Softswitch
- Works with Symmetric NAT
- Disadvantage
  - Still needs “keep alive” for maintaining bindings.
  - Quality might be affected



Topic: Quality of Service

Quality of Service (QoS)

- QoS is normally based on the perception of the users.
- Multiple factors on running VoIP over Public Network
  - Voice Coder/Decoder Used
  - Latency or Delay
  - Packet Loss

Voice Coder To Use?

- Used to calculate the minimum bandwidth required to provide service:
  - G.729 and G.723 are commonly used
    - Mean Opinion Score (MOS) of 3.6 to 3.9
  - Based on number of samples per frame
    - G.729 (2 samples per frame): 24Kbps
    - G.729 (4 samples per frame): 16Kbps

Latency and Packet Loss

- Latency or Delay
  - ITU-T G.114 ~ 150 ms end-to-end delay
  - Actually, as long as delay is consistent and < 500 ms, it is acceptable for most users.
- Packet Loss
  - Often due to congestion
  - Minimal packet loss but tolerant to around 5%.

Topic: Legal Issues

To Regulate or Not To Regulate

- What would an industry self-regulation agreement on VoIP look like?
- Should the rules be different for different types of VoIP?
- How can the government be sure that all portions of the industry will comply with a "voluntary" agreement on VoIP?
- Who can speak for the industry on VoIP issues and don't different parts of the industry disagree?
- Would there still be political pressure to regulate?
- Will the US States step in if the FCC doesn't set the rules?

From: Warren Communications News
2004/01/13

In the US…

- December 1, 2003 VoIP Forum
- FCC: Listen and Learn approach
- Pulver: Regulation Free or Smart Regulation
  - Do not apply existing regulatory policies to VoIP (also echoed by Hodulik of UBS)
- Concern over Law Enforcement and Emergency Services like 911/E911, CALEA, and Disability Access
- Summary: The environment should encourage growth and innovation and be socially responsible.

In Other Countries…

- Legality of VoIP is often a grey area.
- Laws and regulations are normally biased towards existing telecommunication companies.
- Or the telecommunication companies themselves impose rules or policies.
- Those that are legal often require you to get a license (which range from difficult to impossible)

In the Philippines…

- National Telecommunications Commission (NTC): VoIP is allowed only if undertaken by Telcos
- Otherwise, legislative franchise and NTC permits are needed
- Incumbents are planning in deploying VoIP or partnering with VoIP providers
- US-PH is 4th largest international route with 1.7 billion minutes in 2001
- Did you know? In 01/2004, PH Telecom Executives were held in HI, US when some US firms accused PH Telecom Providers of fixing rates.

Summary

- VoIP is tagged to bloom in 2004 (aka first year of VoIP age)
- VoIP has matured enough to gain commercial viability
- But issues still remain which may hinder its widespread deployment
  - Availability, Reliability and Quality
  - Interoperability
  - Legality

Questions?

NAT Types [Reference]

- **Full Cone:** A full cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Furthermore, any external host can send a packet to the internal host, by sending a packet to the mapped external address.
- **Restricted Cone:** A restricted cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X.
- **Port Restricted Cone:** A port restricted cone NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet, with source IP address X and source port P, to the internal host only if the internal host had previously sent a packet to IP address X and port P.
- **Symmetric:** A symmetric NAT is one where all requests from the same internal IP address and port, to a specific destination IP address and port, are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host.