Lab Introduction – software
Voice over IP
Lab Capability and Status

◆ Software used in this course installed in Engineering labs including the lab opened for students ENGR1506 - [http://labs.ite.gmu.edu/](http://labs.ite.gmu.edu/)
  - OPNET/Riverbed Modeler software
  - Asterisk softswitch and X-Lite softphone client (as part of the VM setup at GMU or using the Amazon EC2 in the cloud)
  - Wireshark

◆ Three, potentially four, labs
  - SIP and H.323 signaling and RTP analysis using a Wireshark Packet Analyzer
  - Softswitch/PBX Example – Asterisk
  - Simulation and Modeling, VoIP Performance

◆ Project requirements
Modeling and Simulation (M&S)

◆ Modeling is a process for studying the behavior of complex systems
  ➢ Abstracting important details of system
  ➢ Ignoring the rest
  ➢ Reducing system’s complexity to understand complex behaviors

◆ Capabilities
  ➢ Network design
  ➢ Protocol design
  ➢ Application design
  ➢ Network performance assessment
  ➢ Capacity planning
  ➢ Survivability and resiliency analysis
  ➢ Etc.
Modeling Approaches

- Various Approaches
  - Analytical: Equations (or other formalisms) that result in closed-form solution of system
  - Simulation: Program, physical model, or systems of equations, that describe instantaneous state of system
    - Continuous - the time step is fixed at the beginning of the simulation, time advances in equal increments, and values change based directly on changes in time
    - Discrete - The system changes state as events occur and only when those events occur; the mere passing of time has no direct effect on the model. Unlike a continuous model, simulated time advances from one event to the next and it is unlikely that the time between events will be equal
  - Emulation: combine simulation and a live network
  - Prototype: a proof of concept implementation
  - Operational system: The real stuff
Lab Simulation and Modeling Tools

- Software provided by Riverbed/OPNET Technology through University Program
- University Program packages
  - Full-feature commercial software for a small fee
    - Available to qualifying universities for academic teaching and research
    - Over 1,500 universities currently using this software world-wide
  - FREE limited edition of Modeler software based on 17.5 PL6 release
- Modeler and System-in-the-Loop module
  - Modeler – tool for network modeling and simulation that helps analyzing and designing communication networks, devices, protocols, and applications
    - Analyze VoIP calls and SIP signaling with SIP Proxy Server
  - SITL allows you to map multiple physical interfaces to different network addresses in a simulated network, thus allowing physical hardware and a simulation to interact as a unified
    - Developmental, scalability, and stress testing
    - Interoperability testing
    - Training system development
    - Model validation
    - **In our class we will analyze how network performance impacts VoIP calls**
Softphone Software

◆ X–Lite 3.0 is CounterPath’s next-generation SIP based softphone client, offering users all the productivity of a traditional telephone with desktop and mobile computer enhancements

◆ X-Lite supports a variety of headset devices to augment the modern telephony experience

◆ Standard telephone features
  - Two lines
  - Call display and Message Waiting Indicator (MWI)
  - Speakerphone
  - Mute, Redial, Hold, Do not disturb, Call ignore
  - Call history – list of received, missed, dialed and blocked calls
  - Call forward, Call record
  - Three-way audio and video conferencing
Enhanced Features and Functions

- Instant messaging and presence using the SIMPLE protocol
- Acoustic echo cancellation, automatic gain control, voice activity detection
- Support for the following audio codecs:
  - Broadvoice-32, Broadvoice-32 FEC, G.711aLaw, G.711uLaw, GSM, iLBC, L16 PCM Wideband
- Support for the following video codecs:
  - H.263, H.263 1998
- Automatic selection of the best codec based on the remote party’s capability, available bandwidth, and network conditions
  - X-Lite switches codecs during a call in response to changing network conditions
- Compliance with the RFC 3261 SIP standard
- STUN and ICE NAT traversal
- Support for DTMF (remember this one???)
  - (RFC 2833, inband DTMF or SIP INFO messages)
Asterisk

- Open-source hybrid TDM and packet voice PBX and IVR software/platform

- Sponsored by Digium, the main hardware provider for POTS interface cards
  - Digium named in the top 10 open source companies to watch by networkworld.com

- Its name comes from the asterisk symbol “*” and it works on many different platforms including Windows

- Open standards along with some proprietary protocol support (like Cisco’s Skinny and MGCP)

- Modular plugin type system

- Asterisk the PBX is designed to interface any piece of telephony hardware or software with any telephony application, seamlessly and consistently
Two X-Lite softphones will connect through the simulated network with two routers and an Internet Cloud in the middle

SIP signaling used for call establishment, maintenance and disconnect

Once the call is established RTP stream will carry voice conversation through the simulated network

Use cloud object to introduce packet loss and evaluate voice quality

Packet analyzer switch....
DEMO Part of Lab #3: Network Performance/Voice Quality

SIMULATION/EMULATION SPACE

User 1
X-Lite SIP SoftPhone
IP Address: 192.168.20.1

User 2
X-Lite SIP SoftPhone
IP Address: 192.168.10.1

Network Performance characteristics: Delay, Jitter, Packet Loss

192.168.20.2

IP Network

192.168.10.2