#### SIMULATE REAL-WORLD IP NETWORKS

Impairments, Delay, Errors, Loss, Optical, Electrical...



### Company Overview – GL Communications Inc.

- Headquarters: Gaithersburg, Maryland USA
  - Multiple GL Branch Offices and Worldwide Representatives
- Founded in 1986
- Test & Measurement Equipment
  - > IP, VoIP, SONET-SDH, TDM & Wireless Solutions
  - Visualization, Capture, Storage, Portability, Cost-Effectiveness
  - Endpoints Devices & Core Network Elements
- Engineering Consulting Services (mainly for Transportation and Government agencies)



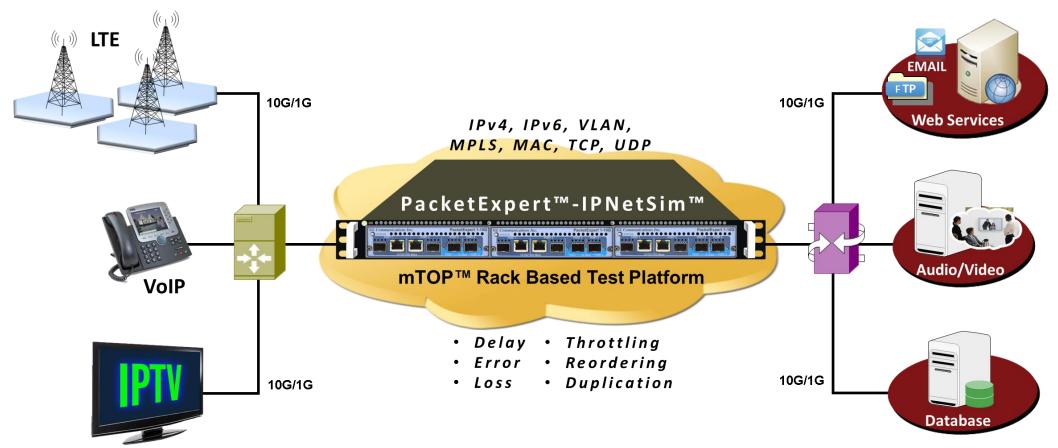


#### Overview

#### How does GL simulate real-world IP Networks? What is GL's IPNetSim?

- **♦** Lab Testing Solution application and automation
- ❖ Emulate Full Duplex 1 Gbps and 10 Gbps networks

- Real-world network conditions by imposing impairments
- Multiple streams independently configured





#### **Hardware Overview**



#### Portable Units



PacketExpert™ 10GX Standalone

- 4 x 1 Gbps Optical OR Electrical
- 2 x 10 Gbps Optical only



PacketExpert™ 1G (4 Port)

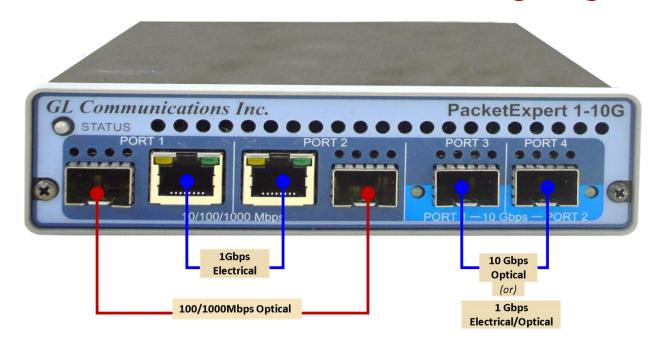


PacketExpert™ 10G Standalone

- 2 x 1 Gbps Optical OR Electrical
- 2 x 10 Gbps Optical only



#### Hardware Overview + Highlights

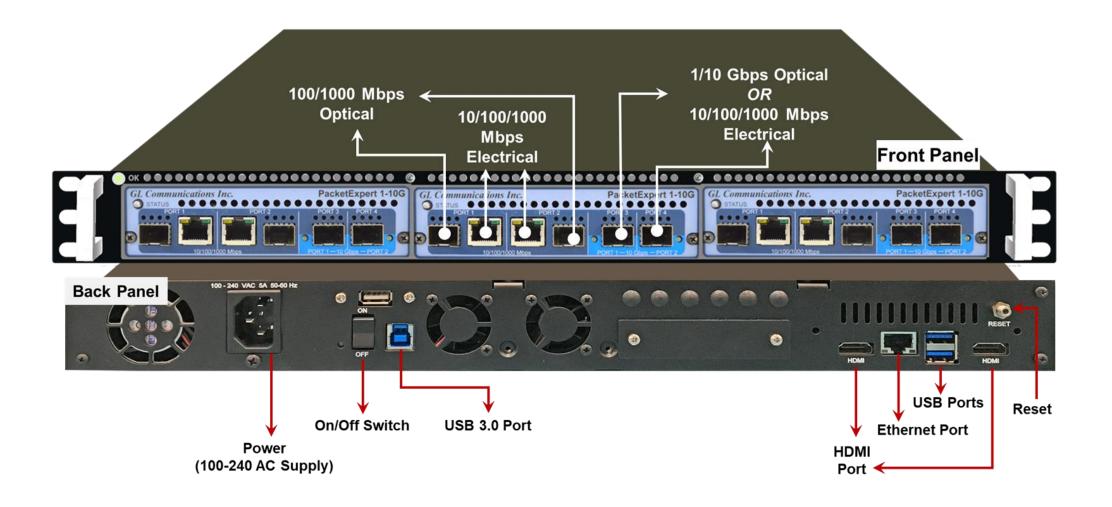


**IPNetSim™** is an optional application available within PacketExpert™ platforms (PXG100 and PXN100).

- ❖ IPNetSim™ operates in both multi-stream and single stream mode.
- ❖ IPNetSim acts as a bridge between two network segments. As long as the hardware has power it allows frames to flow freely.
- ❖ IPNetSim allows users to define up to 16 different streams of traffic. Each of these streams can have its own independent set of impairments applied to them. More to come on streams and exactly how GL defines them.
- IPNetSim is hardware-based...meaning all impairments and timing controls happen at the hardware level.



#### Hardware Interfaces



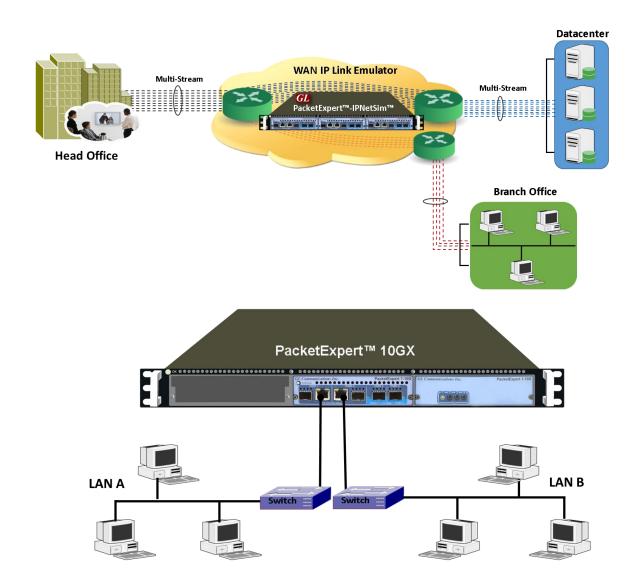


### **Application Overview**



#### **Application Overview**

- ❖ Test Enterprise and Individual-level applications...
  - ➤ Audio and video streaming (VoIP, IMS, HDT, IPTV)
  - ➤ Storage services (Critical Data Access)
  - ➤ Cloud and web services
  - > FTP / HTTP
- Simulate backhaul network
  - ➤ Static and dynamic networks
  - ➤ Satellite + other long delay networks
- Test Quality of Service (QoS) and Quality of Experience (QoE)
- Evaluate the stability of network devices (switches, VoIP Phones, VoIP PBXs, Set-top boxes and VoD Servers.

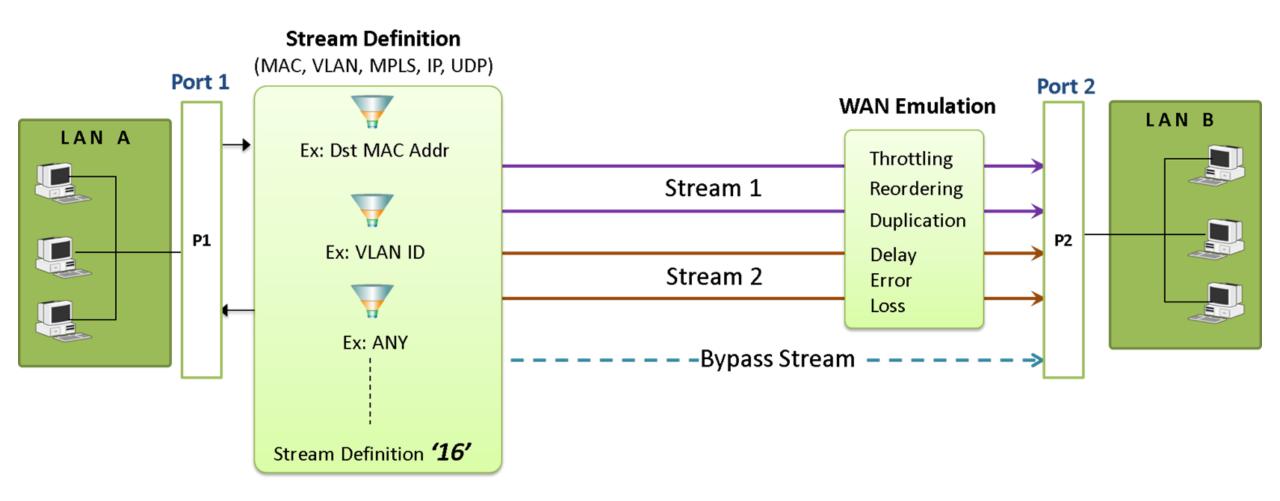




#### "Stream" Overview

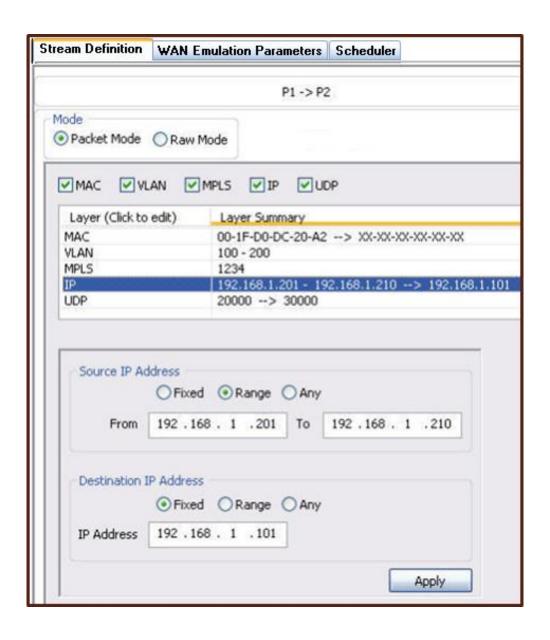


#### "Stream" Overview



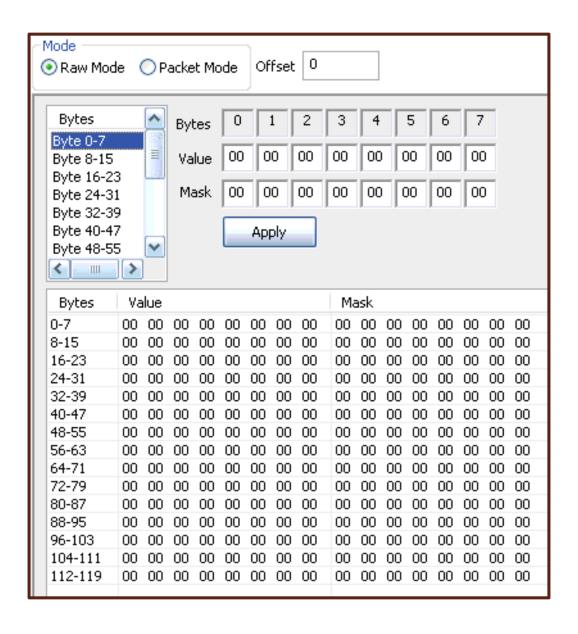


#### Define Streams in Packet Mode





#### Define Streams in Raw Mode



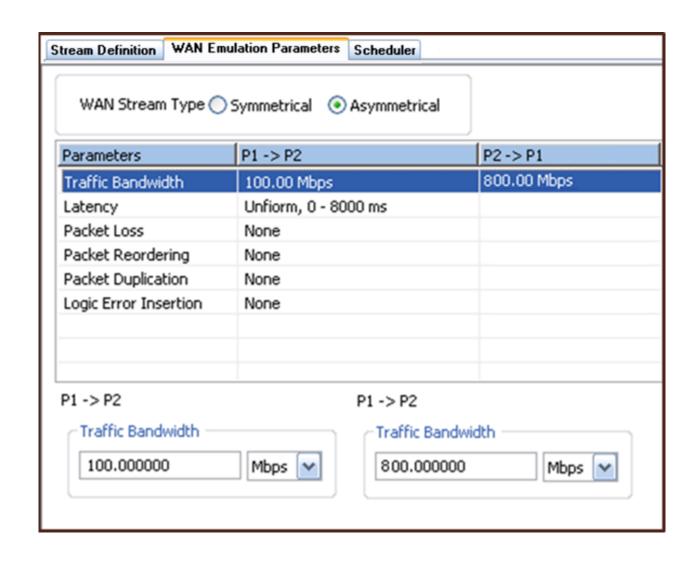


### **Impairments**



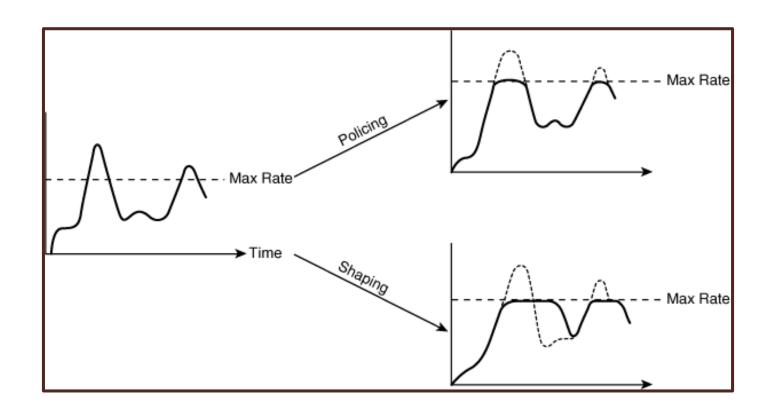
#### **Traffic Bandwidth**

- Traffic which exceeds the stated rate is silently dropped
- UDP Applications will experience dataloss
- TCP Applications should adapt via congestion-avoidance algorithms





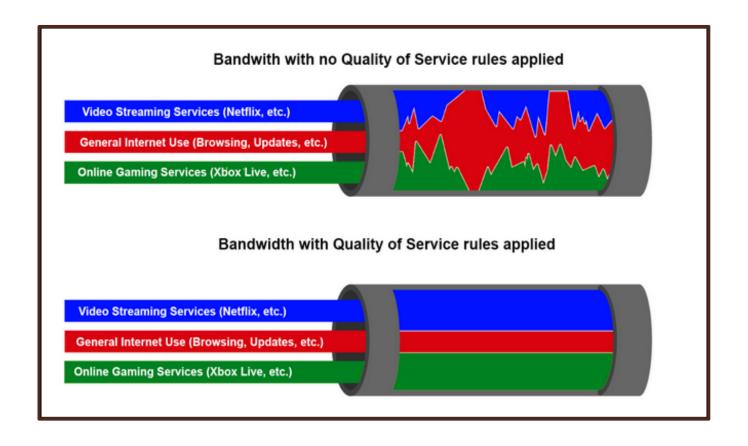
#### **Traffic Bandwidth**



Simulate WAN Applications where *Traffic Policing Policies* may be in effect, *ie Service Level Agreements* between Provider and Customer



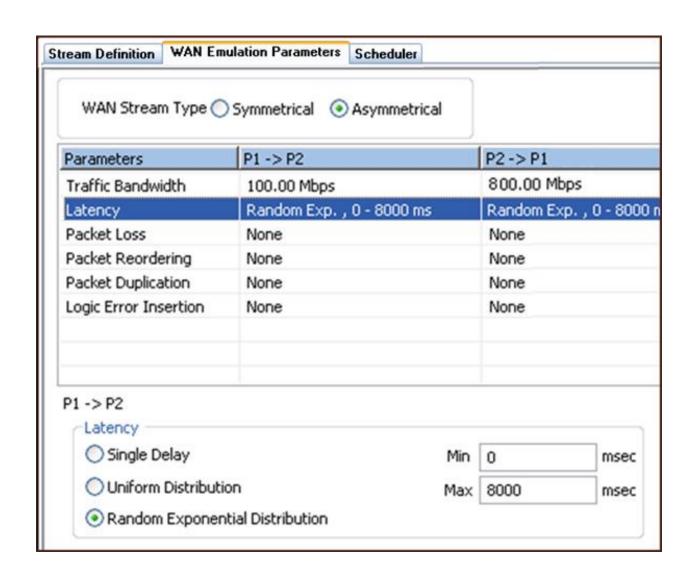
#### **Traffic Bandwidth**



**Simulate QoS settings** by setting different bandwidth caps on different ports (or port ranges)



- Apply Static Delay, or a Uniform or Exponential distribution between a minimum and maximum
- ❖ Delay a packet up to 8000 ms in 1ms increments

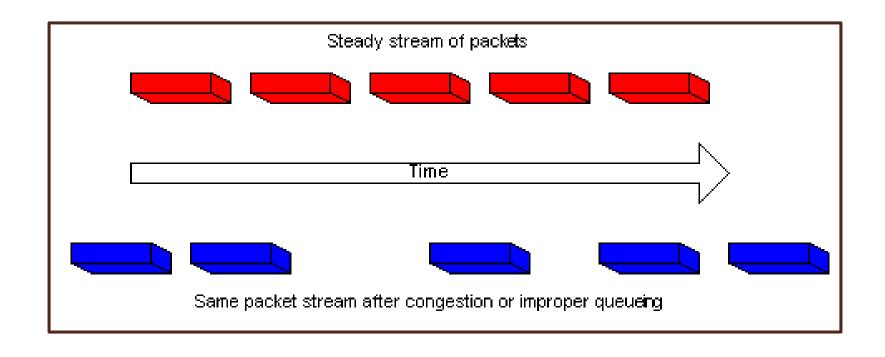






Apply a large static delay to simulate backhaul communication (satellite hops, etc.)

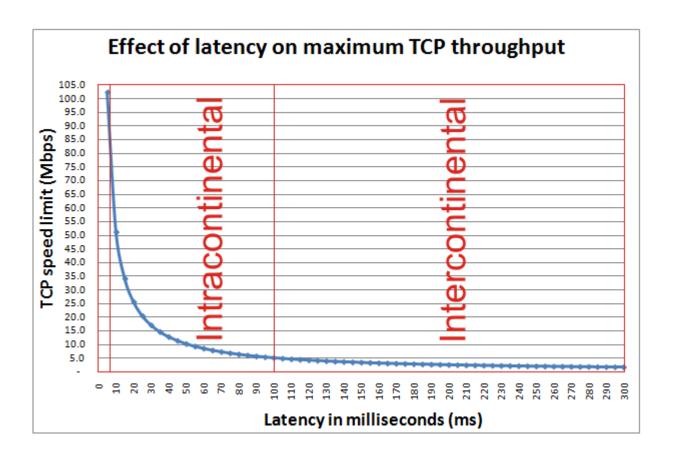




- Apply Variable delay (ie, Jitter) to simulate Traffic Shaping policies and/or Network Congestion
- Jitter leads to packet discard (and therefore data loss) in Real Time UDP Applications



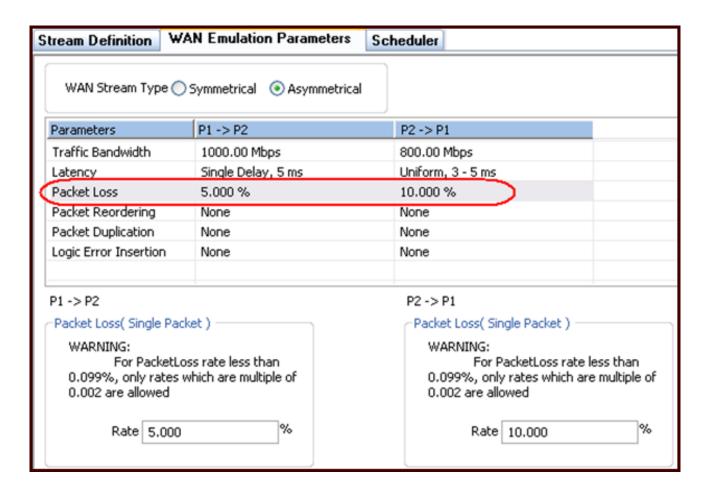
Increased Latency causes TCP
 applications to spend increasing
 amounts of time idling while waiting
 for ACKs from the far side, thereby
 throttling throughput





#### Packet Loss

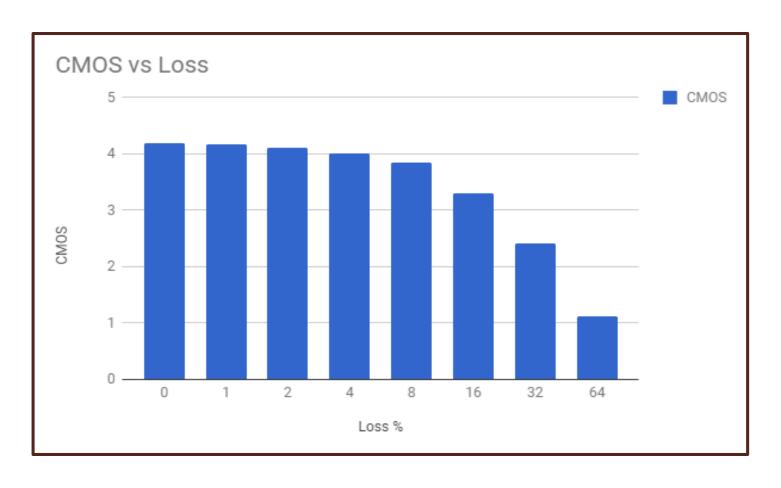
 Randomly drop from 0.01% to 100% of all Packets in the stream





#### Packet Loss

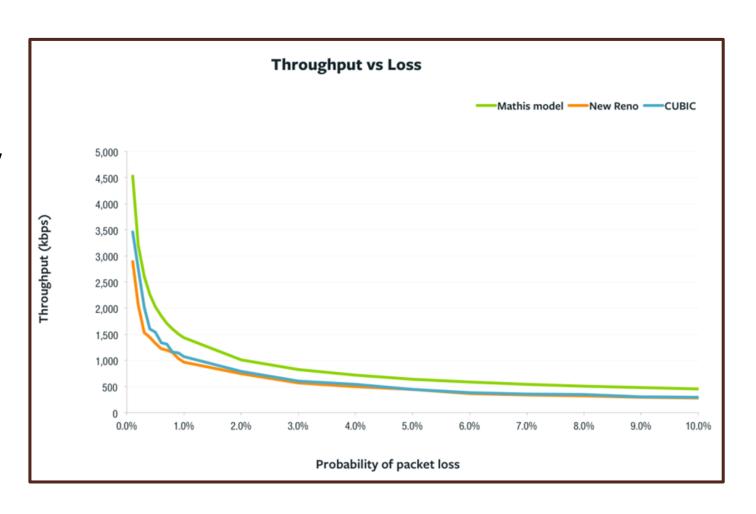
 Real Time UDP Applications are resilient to minor loss, but vulnerable to heavy loss





#### Packet Loss

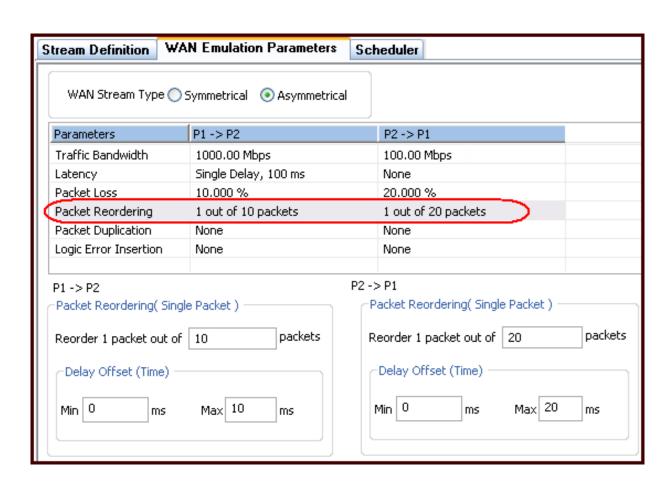
 TCP Applications are vulnerable to even very minor loss rates as every loss results in retransmissions and reduced window sizes.





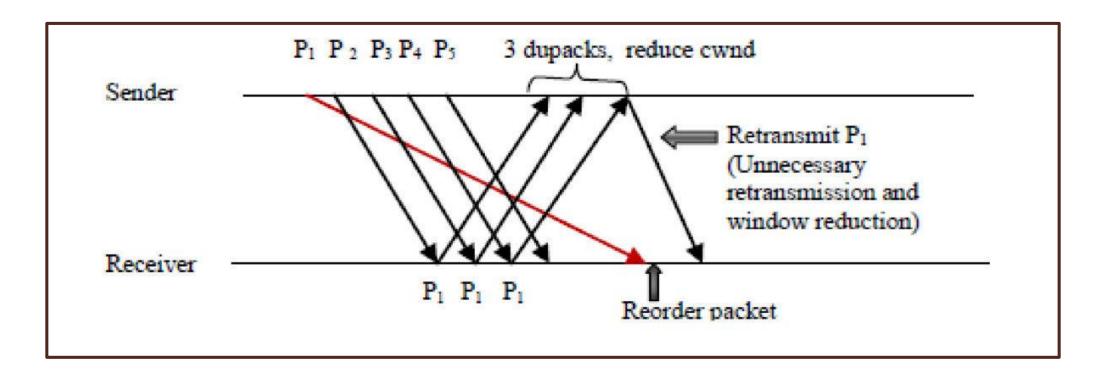
#### Packet Reordering

- Reorder 1 out of every X packets.
- Set a minimum time in ms to hold the reordered packet
- Set a maximum time in ms to hold the reordered packet





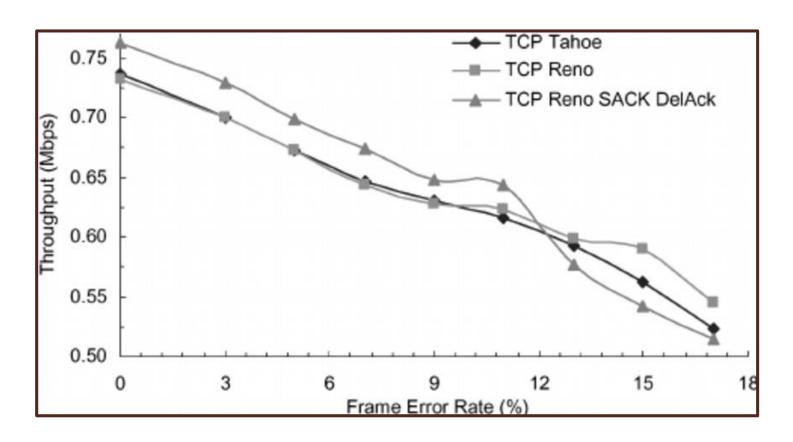
### Packet Reordering



• When a packet is out-of-order, TCP behaves exactly as though the preceding packets are lost resulting in duplicate ACKs, retransmissions and window reduction



### Packet Reordering

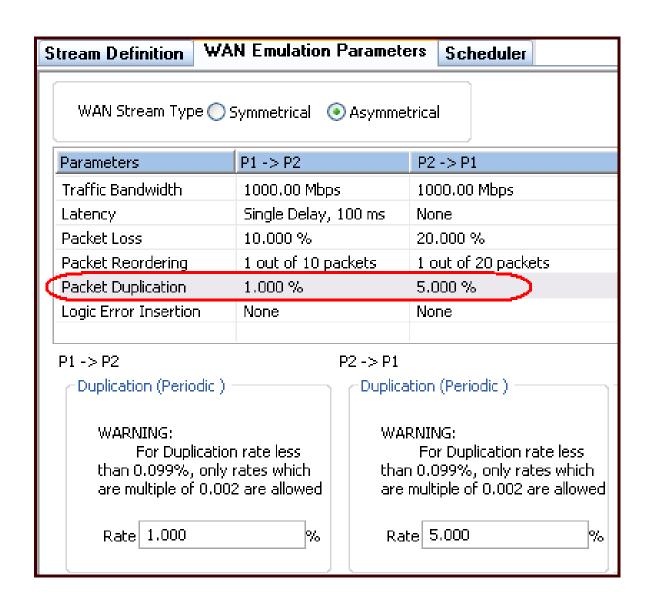


• TCP Selective Acknowledgement (SACK) can mitigate this issue by letting the receive side Acknowledge OOO packets.



#### **Packet Duplication**

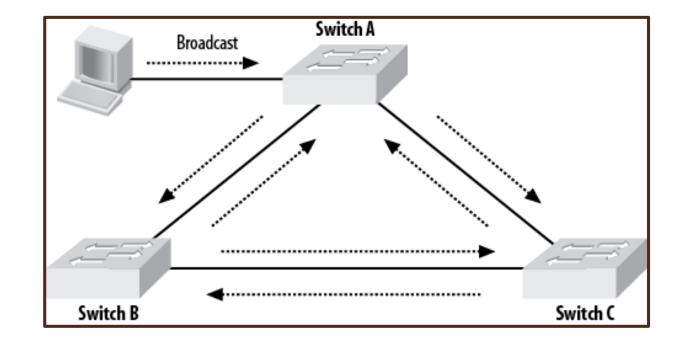
- Randomly duplicate from 0.01% to 100% of all Packets in the stream
- Emulate WAN applications where multiple paths are possible and Load Balancing may be present





#### **Packet Duplication**

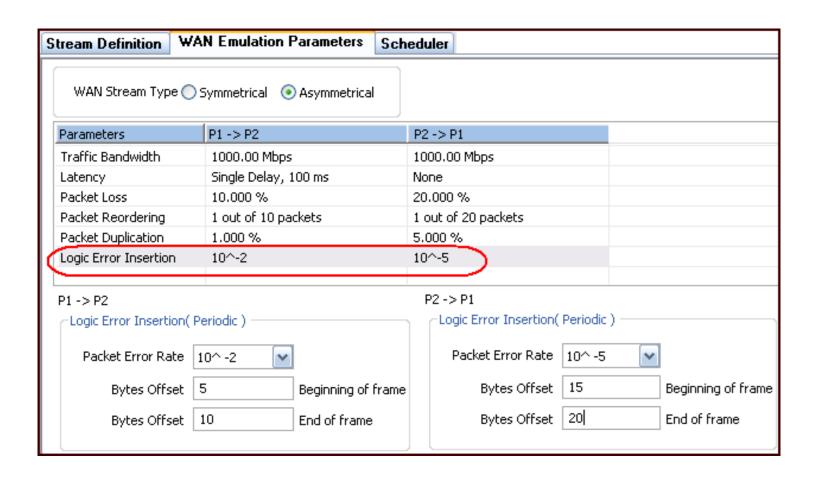
- Duplication can be fatal in broadcast situations (ie broadcast storm)
- Similarly dangerous in multicast applications where small network misconfigurations can have disproportionately large consequences
- Watch out for this in multipath Spanning
  Tree networks





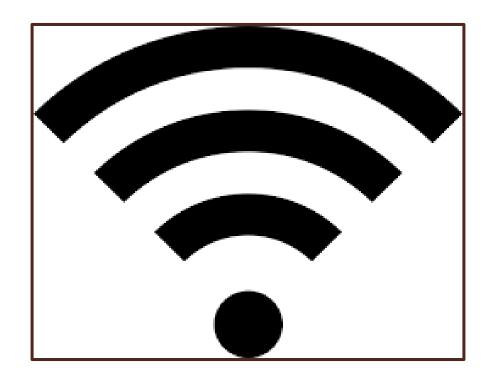
#### Logic Error Insertion

- Insert a single bit error every 10^-X frames (-1 <= X <= -9)</li>
- Use byte offsets to target particular parts of a frame





### Logic Error Insertion

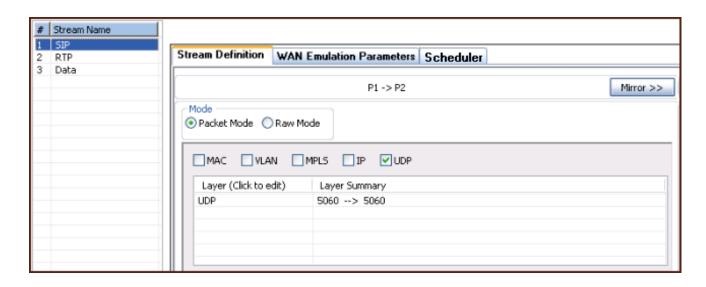


• Cellular and WiFi links are very prone to bit errors (as well as latency and bandwidth issues)

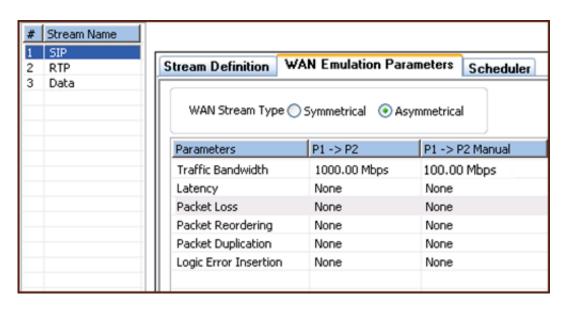
### **Application Examples**



#### Application Example: VoIP

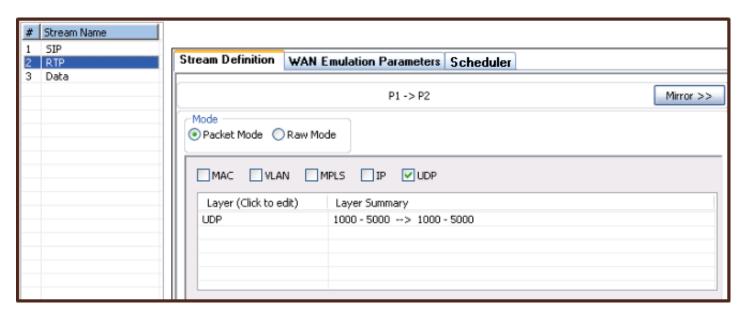


Configure SIP packets to be completely unimpaired

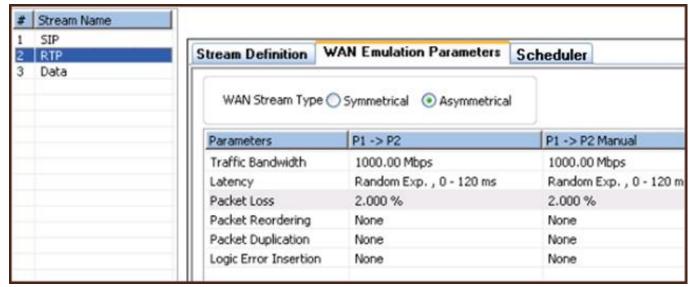




#### Application Example: VoIP

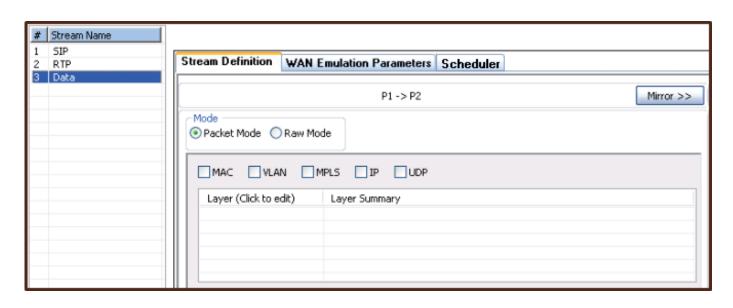


 Apply loss and jitter to RTP streams

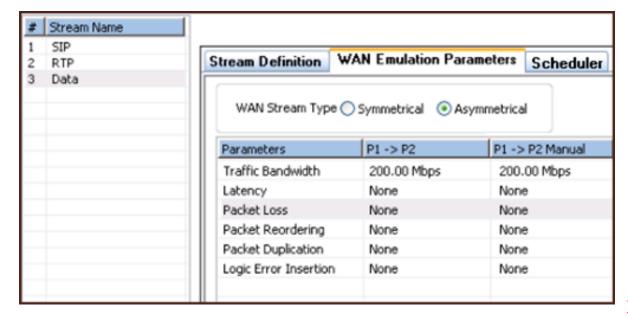




#### Application Example: VOIP

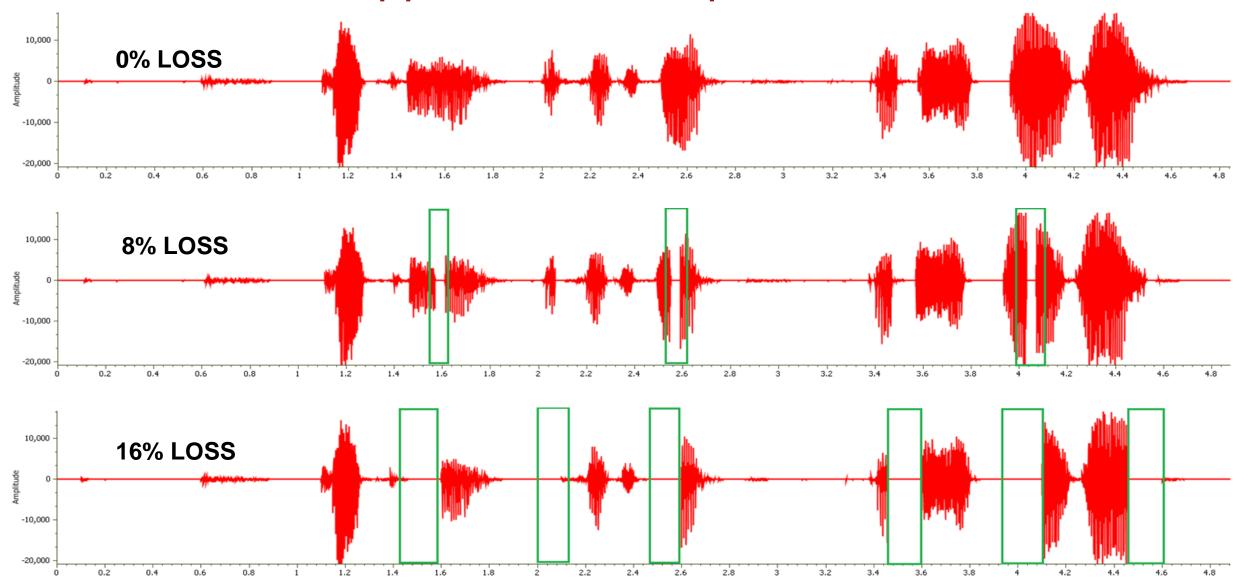


 Set an SLA style bandwidth cap on Data



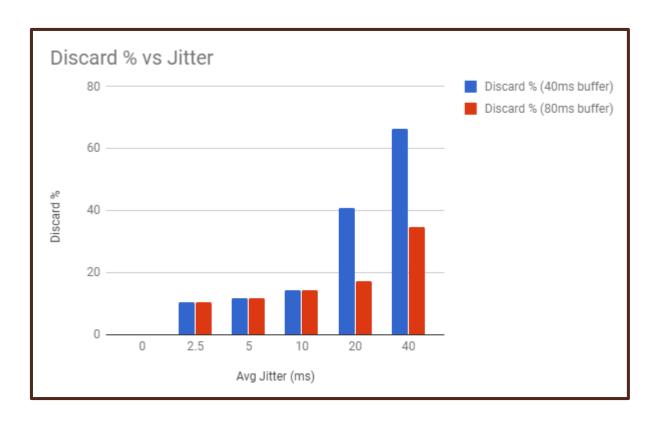


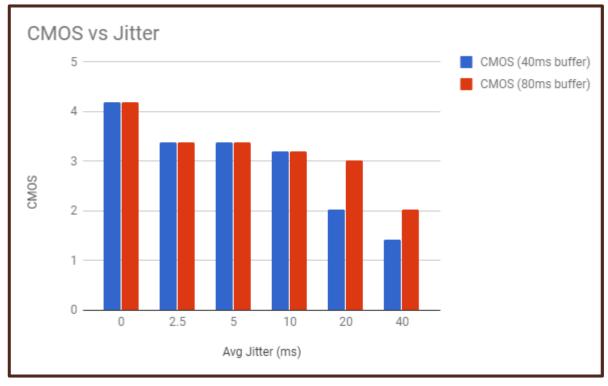
## Application Example: VoIP





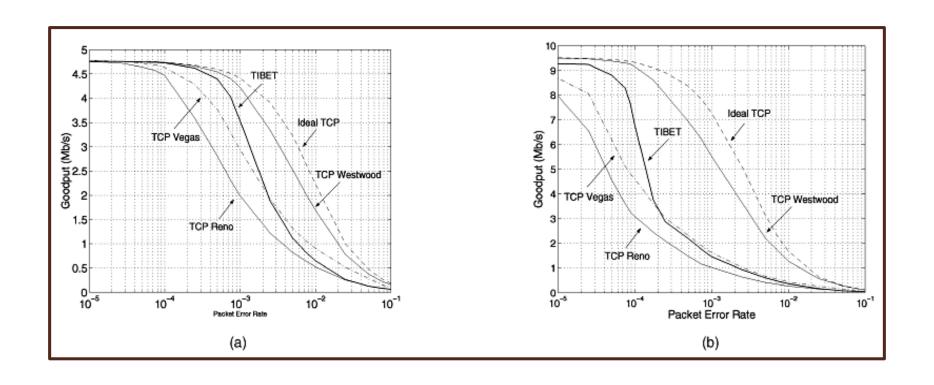
#### Application Example: VOIP





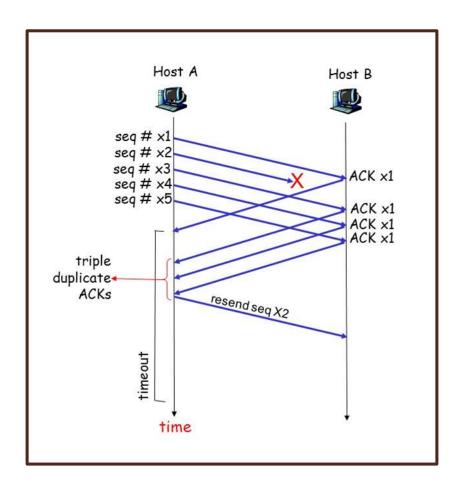
- Determine how your application will behave under expected (and unexpected) network conditions
- Determine what codecs you should use, what jitter buffers, etc.





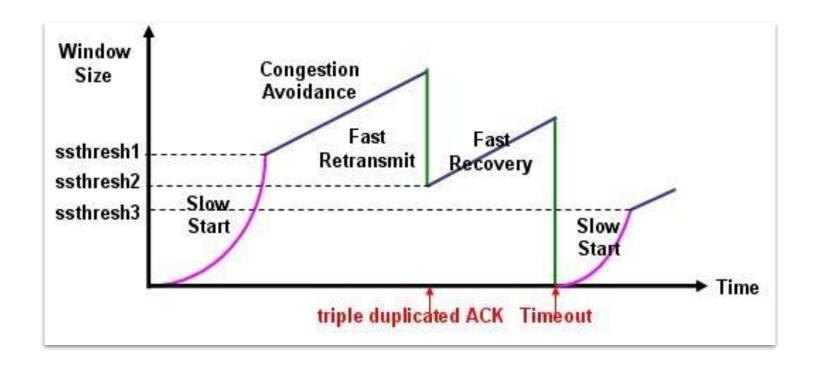
- TCP was first described in an IEEE paper written in 1974
- Ethernet was first standardized by the IEEE in 1983
- The first GSM Network went online in 1991





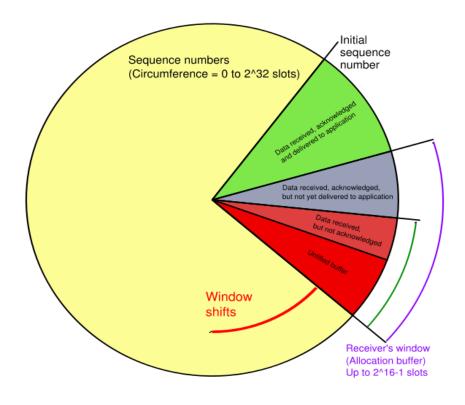
• Loss in a TCP application leads to duplicate ACKs, which lead to retransmissions





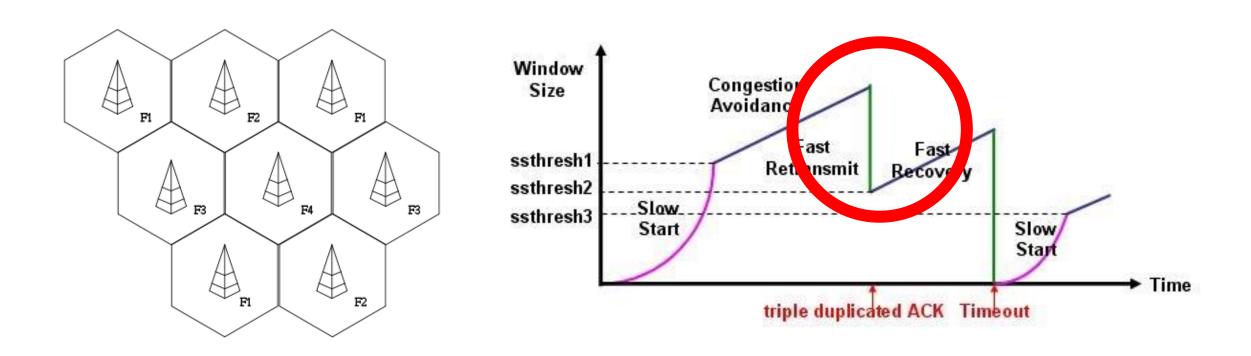
- TCP assumes that Loss is due to Congestion
- When Loss occurs TCP automatically cuts throughput to avoid congestion





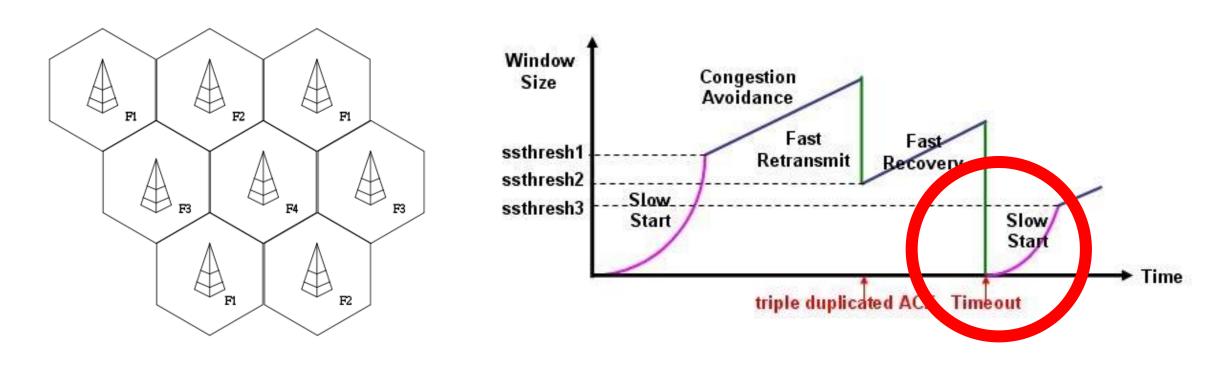
- TCP uses a Sliding Window mechanism to limit how many unacknowledged bytes can be transmitted before the sender is forced to idle
- High Latency links slow ACKs, cause forced idle, and limit throughput





- When a client moves between cells handoff will cause Loss
- TCP will interpret this as Congestion and cut throughput even if the exact same amount of bandwidth is still available (even if more bandwidth is now available)!





- TCP has Retransmission Timeout mechanism that attempts to track the RTT of the connection
- If Latency suddenly increases (ie cell handoff), this can easily cause Timeouts to trigger, immediately cutting throughput to the minimum!





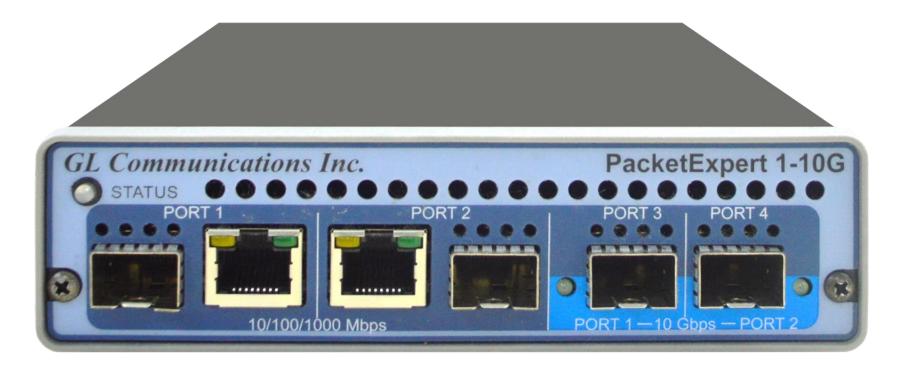
- Wireless communication is High Latency
- Wireless communication is prone to Packet Loss and Packet Corruption
- Applications that work perfectly in the lab can be crippled by a wireless link



# **Automation Options**



#### **Automation Options**



Automate with C/C#, TCL, Python and Java



## **THANK YOU - Any Questions?**

