e-VLBI Overview

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Agenda

- e-VLBI Basics
  - Modes
  - Advantages
  - Expectations
- Network Basics
  - Architecture
  - Transport Protocols
  - Tuning
- Future for e-VLBI

e-VLBI Modes

- 3 Modes for e-VLBI transfers
  - Real-time mode
  - Non-real time mode
  - Psuedo real-time mode
- Real-time mode
  - Data rate generated at the source data acquisition system must be sustained end-to-end on the network to an end system, e.g. the correlator, on the network
    - Data stream must have the same characteristics as if correlating data from disks locally

e-VLBI Modes (cont)

- Non real-time mode
  - Data is buffered at the source and electronically transferred to the destination.
    - Best effort transport
- Pseudo real-time mode
  - Data is recorded at the station and played back from a remote site, via the network, at a later time
    - Data stream must have the same characteristics as if correlating data from disks locally

e-VLBI Advantages

- Rapid processing turnaround
  - Astronomy
    - Ability to study transient phenomena with feedback to steer observations
  - Geodesy
    - Higher-precision measurements for geophysical investigations
    - Better Earth-orientation predictions, particularly UT1, important for military and civilian navigation

e-VLBI advantages (cont)

- Can increase the sensitivity of VLBI observations
  - For most VLBI observations, sensitivity increases as sqrt(bandwidth)
  - Increasing bandwidth is usually the most cost-effective way to increase sensitivity
  - The growth of network bandwidth data rates far exceeds the recording growth capability data rates

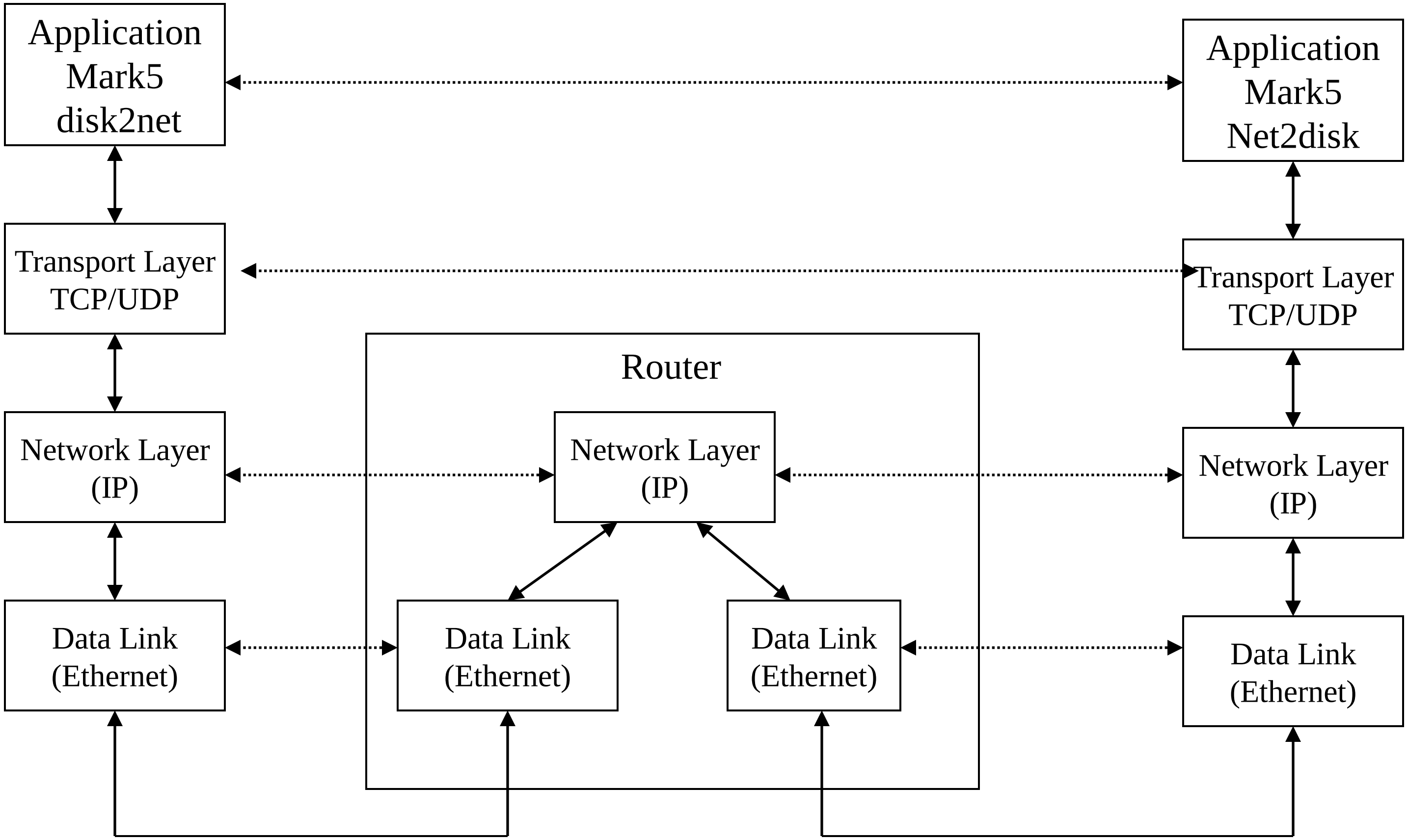
e-VLBI Expectations

- I have a 1 Gbps network connection at the station
  - I want 1Gbps performance or as close to it as possible
  - Will I always be able to achieve it?
  - Yes, no, maybe
- Yes
  - Over short to medium distances
  - Networks are clean
    - No errors between end points
    - Allow aggressive transport protocols
      - Links are under utilized

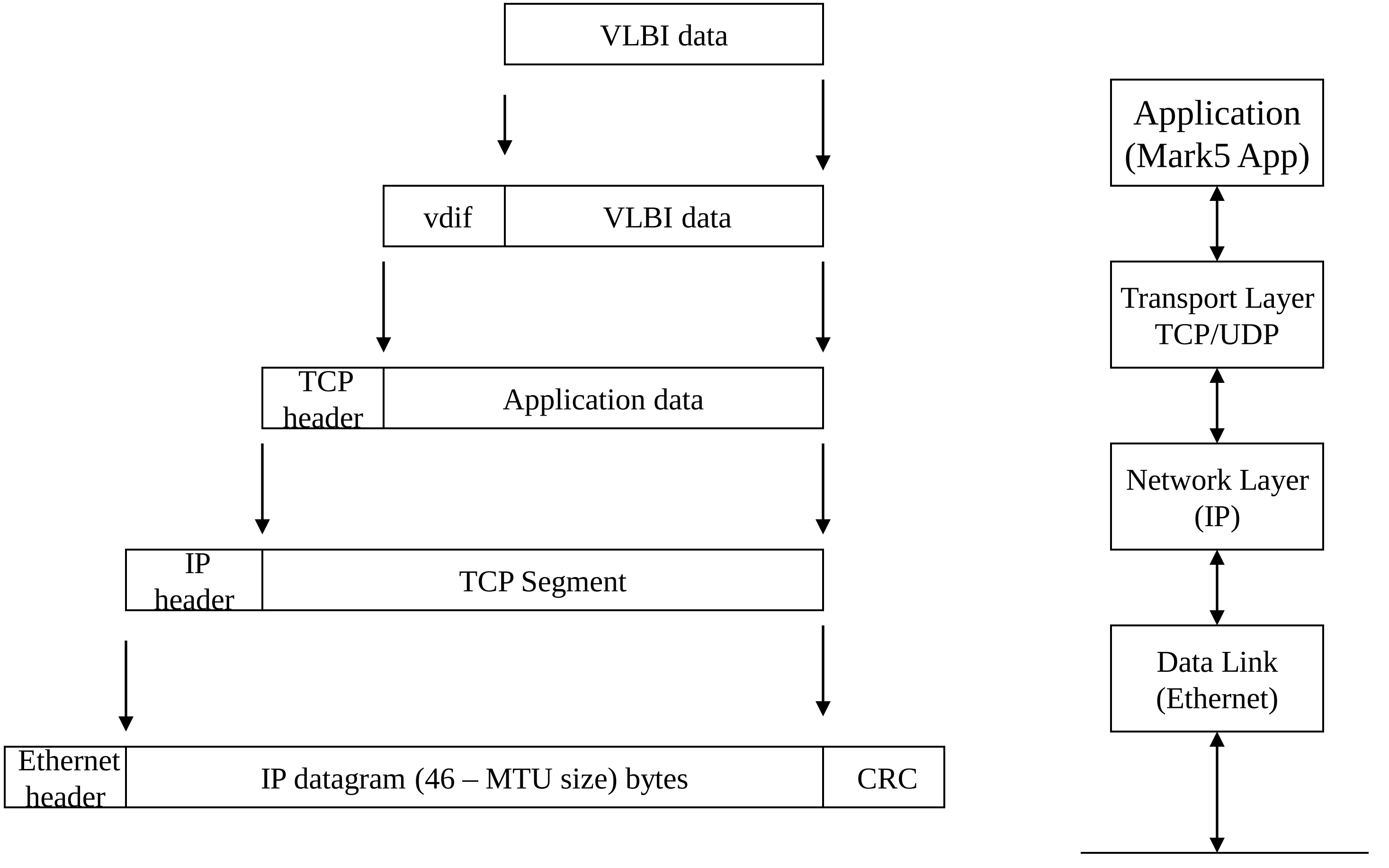
e-VLBI Expectations (cont)

- No
  - Long haul networks
    - Jumbo frames are not supported
  - Traversing multiple network domains
    - Equipment in path may prohibit performance
    - Errors on one network
  - Are not allowed to use aggressive protocols
    - Shared networks
- Maybe
  - When usage is low, allowed to be more aggressive
    - Between 12AM – 6AM

Network Protocol Stack



Data Framing



* MTU (Maximum Transmission Unit) – Normal length 1500, Jumbo frames 9000 bytes

End-to-End Connections

- Circuit Switching
  - Old Phone system
    - Dedicated resources between sender / receiver
    - Whether you send data or not.
- Packet Switching
  - Ability to statistically multiplex multiple data streams on a single channel
  - Goal – efficient utilization of the channel

Quality of Service (QoS)

- Ability to have the packet switched networks
  - Emulate circuit switch services
- Heavily researched but extremely complex to implement and support
- Each network layer can provide some type of QoS

Transport Layer

- Provides end-to-end communication between two or more hosts.
- Isolates application from changes in the underlying hardware
  – IP routing / ATM / SONET
- Provides a number of services to upper layers
  – e.g. Reliable or unreliable delivery of data

Transport Layer

- Protocol
  - How the sender / receiver cooperate to provide that service
- Reliable service
  - TCP (Transmission Control Protocol)
  - Connection based protocol
  - 100 % guaranteed ordered delivery of data
  - Handshaking (acknowledgements)

Transport Layer

- Unreliable service
  - UDP (User Datagram Protocol)
  - Connectionless protocol
- Protocols built on top of UDP
  - Provide reliable transport
  - Congestion control added
    - In Linux user space - outside the kernel
    - In the kernel
  - Overcome short comings of pure TCP protocols

TCP

- Connection based protocol
- Window Based
  - Transmits so much data and wait for acknowledgments that data was received
  - Retransmits if missing data
  - Transmits next window
- Performance is best
  - when the network pipe between sender /receiver is full of data
  - no errors on transmitted packets

TCP

- Originally designed with certain assumptions.
  - When a data segment is lost it is caused by congestion
  - Congestion refers to too many segments competing for network resources
    - Buffer overflows
      - Routers
      - Switches
      - NIC Cards
  - Responsive to congestion notification
    - Reducing sending transmission rate

TCP (cont)

- Performance is dependent on
  - Bandwidth Delay Product (BDP)
    - Transfer rate * Round trip time delay
  - Original TCP does not scale well for
    - High bandwidth networks
    - Long haul networks
- Newer TCP protocols developed
  - Goal: meet the short comings of the original design
  - More aggressive congestion control

TCP Performance

- BDP
  - Need to know the slowest link between sender/receiver
  - Round Trip Time (RTT)
    - Use ping or traceroute
- Example
  - Round trip time 170 msecs
    - (NyAlesund – Haystack)
  - Assume you have a 1 Gbps link

    BDP = (1Gbits/sec) * (1byte/8bits) *(170msec)

    BDP = 21.25 MBytes
    - In order to fill the network pipe require 21.25 MB of data on the line
- Default TCP buffer size is 64KBytes
  - Adjust the kernel buffer size allocation
  - Application sets the window size based on BDP

UDP

- Characteristics
  - Connection-less protocols
    - No connection needs to be established to start
    - Start transmitting data
  - Transmit data at whatever rate you wish
  - No feedback
    - Congestion in network
    - Flow control
      - Receiver telling sender to slow down rate
  - No guaranteed delivery
    - Order not guaranteed

UDP (cont)

- Overcome TCP's shortcomings
  - approach a more efficient approach to utilizing the channel
- Class of protocols created using UDP as the base protocol but add:
  - Guaranteed delivery of data
    - selective acknowledgments
  - Congestion control
    - How to react to errors in transmission
  - Order guaranteed
  - Flow control
    - Slow the rate down, speed up

UDP (cont)

- NOTE:
  - Some domains do not allow UDP traffic
    - Denial of Service (DoS) attack
    - Lack of congestion control
  - Network protocol stack is optimized for TCP not UDP
    - TCP used in 99.99 % of applications

Newer Transport Protocols

- “TCP Friendly”
  - Flows behave under congestion
  - Responsive to congestion notification
  - In steady-state does not use more bandwidth than a conformant TCP protocol
    - Drop rate
    - Round trip time (RTT)
    - Maximum transmission unit (MTU) size
- Not all protocols are TCP friendly
  - Usually ones designed for
    - Long haul
    - high bandwidth
- UDP and TCP can fall into this category

Tsunami UDP

- Developed by Indiana University
- Advantages
  - No slow start up for transmission
    - Maximum rate at the start
  - Specify speedup / slow down recovery when errors are seen
  - Data transmission
    - default priority for data integrity
  - Data rate priority
    - disables retransmissions
    - maximizes bandwidth
- Adopted by astronomy community for ftp application -

Tsunami UDP

- Disadvantages
  - Assumes you have
    - a priori knowledge about the quality of your network connection
    - For a long haul network, do you truly know the utilization of all links connecting you end-to-end
      - over many domains?
  - If configured incorrectly
    - Poor performance for user
    - Poor performance for other users sharing the same network link

Tuning

- How to tune a Linux distribution
  - Earlier kernels required manual configuration
  - Based upon the BDP
  - Tune both the sender and receiver
- Linux 2.6.18 and greater have full auto-tuning
  - 4MB maximum buffer sizes
  - Manual tuning is not generally recommended
  - Sender tuning has been enabled for years

Tuning (cont)

- Autotuning adjusts the receive buffer size
  - Autotunes for each connection
  - cat /proc/sys/net/ipv4/tcp_moderate_rcvbuf
    - 1
- Per connection memory space defaults in
  - /proc/sys/net/ipv4/tcp_rmem -TCP receive buffer
  - /proc/sys/net/ipv4/tcp_wmem - TCP send buffers

Tuning (cont)

- tcp_rmem and tcp_wmem
  - Three element array
    - minimum, default, maximum buffer size
  - Balances memory usage
    - Not just TCP window size
    - Actual memory usage
      - socket data structures
  - Sets bounds on auto-tuning
  - Must tune both end systems for optimal performance

Tuning (cont)

- Maximum > BDP
  - Dependent on the amount of memory you have in your system
    - e.g. 2MB of memory, the maximum value will be 2MB
      - Can never achieve more than what is physically available.
- Middle value = Initial buffer size
  - Set for typical usage
    - small flows
  - If set to large = maximum value
    - with autotuning will waste memory
    - Can hurt performance
  - typical values
    - tcp_rmem = 87380
    - tcp_wmem = 16384

Tuning (cont)

- rmem_max, wmem_max
  - Maximum buffer size an application can request
    - setsocketopt()
      - SO_SNDBUF
      - SO_RCVBUF
- You do not need to adjust these unless your are planing to use some form of application tuning.
- **NOTE: Manually adjusting socket buffer sizes with setsockopt() disables autotuning. Application that are optimized for other operating systems may implicitly defeat Linux autotuning.**

Tuning (cont)

- Data Link Layer Configuration
  - MTU – Maximum Transmission Unit
    - Normal user 1500 bytes is sufficient
      - Accounts for 95% of the users of the Internet
    - Present maximum is 9000 bytes
      - Not all equipment on the WAN can support this
      - Tests to determine if the path supports jumbo frames
      - Included in the 2.6.18 kernel, needs to be enabled
    - Use “ifconfig” command
      - determine the interfaces settings
      - to change MTU size if possible
  - Autotuning available
    - /proc/sys/net/ipv4/tcp_mtu_probing (disabled by default)

New Services

- Internet2 / Geant
  - Now offer the ability to dynamically request
    - dedicated bandwidth over the long haul network
  - Advantages
    - Use of aggressive transport protocols
    - Maximize end-to-end capacity
    - Errors are not congestion over the long haul network
    - Perfect for real-time and time critical transfers
  - Disadvantages
    - $ required for service

Questions / Comments?

Thank you.