Grid High Performance Networking Research Group

GRID WORKING DRAFT

Document: draft-ggf-ghpn-netissues-0
Category: Informational Track

www.globalgridforum.org/6_DATA/gridhigh.htm

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June 2003

Networking Issues of Grid Infrastructures

Status of this Memo

This memo provides information to the Grid community. It does not define any standards or technical recommendations. Distribution is unlimited.

Comments

Comments should be sent to the GHPN mailing list (ghpn-wg@gridforum.org).

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1. Introduction

The Grid High-Performance Networking (GHPN) Research Group focuses on the relationship between network research and Grid application and infrastructure development. The vice-versa relationship between the two communities is addressed by two documents, each of it describing the relation from the particular view of either group. This document summarizes networking issues identified by the Grid community.

2. Scope and Background

Grids are built by user communities to offer an infrastructure helping the members to solve their specific problems. Hence, the geographical topology of the Grid depends on the distribution of the community members. Though there might be a strong relation between the entities building a virtual organization, a Grid still consists of resources owned by different, typically independent organizations. Heterogeneity of resources and policies is a fundamental result of this. Grid services and applications therefore sometimes experience a quite different resource behavior than expected. Similarly, a heavily distributed infrastructure with ambitious service demands to stress the capabilities of the interconnecting network more than other environments. Grid applications therefore often identify existing bottlenecks, either caused by conceptual or implementation specific problems, or missing service capabilities. Some of these issues are listed below.

3. End-Systems

This section describes experienced issues related to End-Systems.

3.1 Communication Protocols and their Implementation

The evolution of the Transmission Control Protocol (TCP) is a good example on how the specification of communication protocols evolves over the time. New features were introduced to address experienced shortcomings of the existing protocol version. However, new optional features also introduce more complexity. In the context of a service oriented Grid application, the focus is not on the various protocol features, but on the interfaces to transport services. Hence, the question arises whether the advanced protocol capabilities are actually available at the diverse end-systems and, if they are, which usage constraints do they imply. This section describes problems encountered with the implementation of communication protocols, with a focus on TCP.

A widely deployed interface to implementations of the TCP protocol stack is provided by the Berkeley socket interface which was developed at the University of California at Berkeley as part of their BSD 4.1c UNIX version. The fundamental abstraction of this API is that communication end-points are represented as a generic data structure called socket [RFC147]. The interface specification lists a set of operations on sockets in a way that communication can be implemented using standard input/output library calls. It is important to note that the abstraction provided by sockets is a multi-protocol abstraction of communication end-points. The same data structure is used with Unix services as files, pipes and FIFOs as well as with UDP or TCP end-points.

Though the concept of sockets is close to that of file descriptors, there are, however, essential differences between a file descriptor and a socket reference. While a file descriptor is bound to a file during the open() system call, a socket can exist without being bound to a remote endpoint. For the set up of a TCP connection sender and receiver have to process a sequence of a function-calls which implement the three-way handshake of TCP. While the sender issues the connect()-call, the receiver has to issue two calls: listen() and accept().

An important aspect is the relation between the above listed call-sequence and the protocol processing of the TCP handshake. While the listen()-call is an asynchronous operation which is related to the receipt of TCP-SYN-messages, connect() and accept() are typically blocking operations. A connect()-call initiates the three-way handshake, an accept call processes the final message.

There is, however, a semantical gap between socket buffer interface and the protocol capabilities of TCP.While the protocol itself offers the explicit use of the window scale option during the three-way handshake, there is no way in commonly used operating systems to explicitly set this option by issuing a specific setsockopt()-call.

In fact, the window scale option is derived from the socket buffer size used during the connect()- and listen()-call. Unfortunately, this selection is done on a minimum base which means that the minimum required window-scale option is used. To explain this mechanism in more detail, suppose that the used socket buffer size would be 50KB,100KB, and 150KB.

In the first case, the window scale option would be not used at all. Because the TCP protocol does not allow to update the window scale option afterwards, the maximum socket buffer size for this session would be 64KB, regardless whether socket-buffer tuning

libraries would recognize a buffer shortage and would try to increase the existing buffer space.

In the second case, many operating systems would select a window scale option of 1. Hence, the maximum socket buffer size would be 128KB. In the final case, the window scale option used is 2 which results in a maximum buffer size of 256KB.

This argumentation leads to the conclusion that any buffer tuning algorithm is limited by the lack of influencing the window-scale option directly.

3.2 Operating System Capabilities and Configuration Issues

Similarly to the above described influence of the selected socket buffer size, widely deployed operating systems do have a strong impact on the achievable level of service. They offer a broad variance of tuning parameters which immediately affect the higher-layer protocol implementations.

For UDP based applications, the influence is typically of less importance. Socket buffer related parameters such as the default or maximum UDP send or receive buffer might affect the portability of applications, i.e. by limiting the maximum size of datagrams UDP is able to transmit. More service relevant is the parameter which determines whether the UDP checksum is computed or not.

The potential impact on TCP based applications, however, is more significant. In addition to the limitation of the maximum available socket buffer size, a further limitation is frequently introduced by the congestion window as well. Here, an operating system tuning parameter additionally limits the usable window size of a TCP flow and might therefore affect the achievable goodput even though the application explicitly sets the socket buffer size. Furtheron, parameters such as delayed acknowledgements, Nagle algorithm, SACK, and path MTU discovery do have an impact on the service.

3.3 OS and system-level optimizations

The evolution of end-to-end performances hinges on the specific evolution curves for CPU (also known as Moore law), memory access, I/O speed, network bandwidth (be it in access, metro, core). A chief role of an Operating System (OS) is to strike an effective balancing act (or, better yet, a set of them) given a particular period in time along the aforementioned evolution curves. The OS is the place where the tension among curves proceeding at different pace is first observed. If not addressed properly, this tension percolates up to the application, resulting in performance issues, fairness issues, platform-specific counter-measures, and ultimately non-portable code.

To witness, the upward trend in network bandwidth (e.g., 100Mb/s, 1Gb/s, 10 Gb/s Ethernet) put significant strain on the path that data follow within a host, starting from the NIC and finishing in an application's buffer (and vice-versa). Researchers and entrepreneurs have attacked the issue from different angles.

In the early '90's, [FBUFS] have shown the merit of establishing shared-memory channels between the application and the OS, using immutable buffers to shepherd network data across the user/kernel boundary. The [FBUFS] gains were greater when supported by a NIC such as [WITLESS], wherein buffers such as [FBUFS] could be homed in the NIC-resident pool of memory. Initiatives such as [UNET] went a step further and bypassed the OS, with application's code directly involved in implementing the protocol stack layers required to send/receive PDU to/from a virtualized network device. The lack of system calls and data copy overhead, combined with the protocol processing becoming tightly coupled to the application, resulted in lower latency and higher throughput. The Virtual Interface Architecture(VIA) consortium [VIAARCH] has had a fair success in bringing the [UNET] style of communication to the marketplace, with a companion set of VI-capable NICs adequate to signal an application and hand-off the data.

This OS-bypass approach comes with practical challenges in virtualizing the network device, while multiple, mutually-suspicious application stacks must coexist and use it within a single host. Additionally, a fair amount of complexity is pushed onto the application, and the total amount of CPU cycles spent in executing network protocols is not going to be any less.

Another approach to bringing I/O relief and CPU relief is to package a "super NIC", wherein a sizeable portion of the protocol stack is executed. Enter TCP Offload Engines (TOEs). Leveraging a set of tightly-coupled NPUs, FPGAs, ASICs, a TOE is capable to execute the performance-sensitive portion of the TCP FSM (in so-called partial offload mode) or the whole TCP protocol (in full offload mode) to yield CPU and memory efficiencies. With a TOE, the receipt of an individual PDU no longer requires interrupting the main CPU(s), and using I/O cycles. TOEs currently available in the marketplace exhibit remarkable speedups. Especially with TOEs in partial-offload mode, the designer must carefully characterize the overhead of falling off the hot-path (e.g., due to a packet drop), and having the CPU taking control after re-synchronizing on the PCB. There are no standard APIs to TOEs.

A third approach is to augment the protocol stack with new layers that annotate application's data with tags and/or memory offset information. Without these fixtures, a single out-of-order packet may require a huge amount of memory to be staged in anonymous memory (lots of memory at 10Gb/s rates!) while the correct sequence is being recovered. With these new meta-data in place, a receiver

would aggressively steer data to its final destination (an application's buffer) without incurring copies and staging the data. This approach led to the notions of Remote Direct

Data Placement (RDDP) and Remote Direct Memory Access (RDMA) (the latter exposing a read/write memory abstraction with tag and offset, possibly using the former as an enabler). The IETF has on-going activities in this space [RDDP]. The applicability of these techniques to a byte-stream protocol like TCP, and the ensuing impact on semantics and layering violations are still controversial.

Lastly, researchers are actively exploring new system architectures (non necessarily von Neumann ones) wherein CPU, memory, and networks engage in novel ways, given a defined set of operating requirements. In the case of high-capacity optical networks, for instance, the Wavelength Disk Drive [WDD] and the OptIPuter [OPTIP] are two noteworthy examples.

3.4 Multi-Stream File Transfers

Moving a data set between two sites using multiple TCP sessions provides significantly higher aggregate average throughput than transporting the same data set over a single TCP session, the difference being proportional to the square of the number of TCP sessions employed. This is the outcome of a quantitative analysis using three simplifying assumptions:

1. the sender always has data ready to send
2. the costs of striping and collating the data back are not considered
3. the end-systems have unlimited local I/O capabilities.

It is well-known that 2) and 3) are not viable assumptions in real-life, therefore the outcome of the analysis has baseline relevance only.

Throughput dynamics are linked to the way TCP congestion control reacts to packet losses. There are several reasons for packet losses: network congestion, link errors, and network errors. Network congestion is pervasive in current IP networks, where the only way to control congestion is through dropping packets. Traffic engineering, admission control and bandwidth reservation are currently in early stages of definition. DiffServ-supporting QoS infrastructures will not be widely available in the near future.

Even in a perfectly engineered network, link errors occur. If we take an objective of 10**(-12) Bit Error Rate, for a 10Gbps link, this amounts to one error every 100 seconds. Network errors can

occur with significant frequency in IP networks. [STOPAR] shows that network errors caught by TCP checksum occur between one packet in 1100 and 1 in 32000, and without link CRC catching it.

TCP throughput is impacted by each packet loss. Following TCP's congestion control algorithm existent in all major implementations (Tahoe, Reno, New-Reno, SACK), each packet loss results in the TCP sender's congestion window being reduced to half of its current value, and therefore (assuming constant Round Trip Time), TCP's throughput is halved. After that, the window increases linearly by roughly one packet every two Round Trip Times (assuming the popular Delayed-Acknowledgement algorithm). The temporary decrease in TCP's rate translates into an amount of data missing transmission opportunity. As shown below, the amount of data missing the opportunity to be transmitted due to a packet loss is (see [ISCSI] for mathematical derivations relative to TCP Reno):

D(N) = E**2/(N**2)*RTT**2/(256*M)

where

D = amount of data not transmitted due to packet loss, in MB

E = Total bandwidth of an IP "pipe", in bps

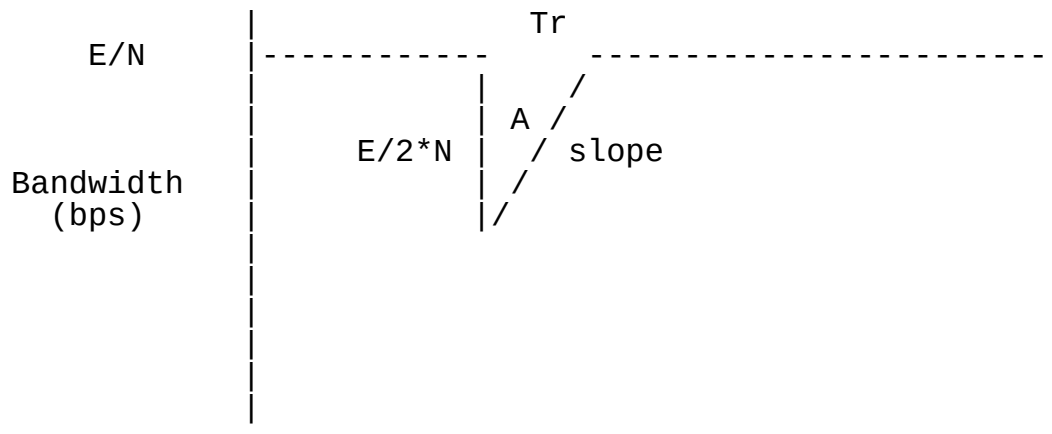
N = number of TCP streams sharing the bandwidth E, unitless

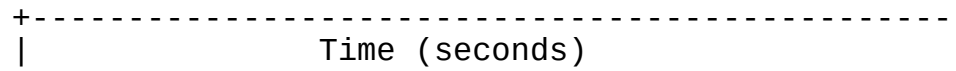
RTT = Round Trip Time, in ms

M = packet size, bytes

For example, for a set of N=100 connections totaling E=10Gbps, RTT=10ms, M=1500B, the data not transmitted in time due to a packet loss is D(N)=2.6MB.

To show this consider the following hypothetical graph of bandwidth versus time:





First, the area inside the triangle, A, is 1/2 base * height. The base has units of seconds and the height bps, and the product, bits. This represents the data not transmitted due to loss. The expression for the height is easily obtained since, as noted above, a dropped packet causes the bandwidth to be cut in half. TCP also specifies that the amount of data in-flight increases by one packet every 2 round trip times. We can calculate the corresponding increase in bandwidth from the equation for the bandwidth delay product [HIBWD].

This equation states buffer size = bandwidth * RTT, or rearranged the bandwidth = buffer size / RTT. So, our increase in bandwidth is M/RTT. We get this increase every x * RTT seconds, so the rate of recovery (the slope in the diagram) = M/RTT / xRTT or M/x*RTT^2 and has units of bps/s. We can now determine the recovery time(Tr), which is the base of the triangle, to be E/2N * x*RTT^2 / (8M). Finally, we can determine the equation for the area of the triangle. Using the units listed above and appropriate conversions:

$$= \frac{1}{2}\,\frac{E\ (Mb) * E\ (Mb) * x * RTT^2\ (ms^2)}{2\ 2*N(s) \quad 2*N(s) * \qquad\qquad (10^3\ ms)^2} * \frac{(1\ sec)^2}{}$$

$$\frac{* (byte) * 10^6\ bits * MB}{M\ (bytes) * 8\ bits * \quad Mb \qquad 8\ Mb}$$

In absence of Delayed-Acknowledgements (x=1) we get:

$$\Longrightarrow \frac{E^2 * 2 * RTT^2}{N^2 * M * 256} = \frac{(1*10^4)^2 * 2 * (10)^2}{(100)^2 * 1500 * 256}$$

Using our previous example of a set of N=100 connections totaling E=10Gbps, RTT=10ms, M=1500B, the time interval for TCP to recover its sending rate to its initial value after a packet loss is I(N)= 0.833 seconds.

If N = 1, the time to recover its rate, I(1)=83.3s, is of the same order of magnitude as the time between two packet losses due exclusively to the link Bit Error Rate. In other words, a packet loss occurs almost immediately after TCP has recovered its rate. This means that N=1 delivers on average just about 3/4 of the required 10Gb/s rate, since 1/4 of rate is lost during the time TCP rate increases linearly from 1/2 to full rate. (More precisely, the effective rate is 8.27Gb/s because 1/4 of rate is lost during

83.3s, and the time between two errors is now 120.825s due to decreased sending rate).

Consideration of this equation also reveals another major issue with TCP on high latency networks. Notice that the recovery time is directly proportional to the square of the RTT. This means that doubling the RTT will result in a 4x increase in the recovery time, making dropped packet even more problematic, and multi-stream TCP even more valuable. The impact of packet losses on multi-stream TCP settings has been analyzed in [AGGFLOW].

GridFTP [DAMAT] is a real world application that uses multiple streams to obtain high performance during file transfers. There is no adequate data available to demonstrate the performance in the face of packet loss; however, it can be clearly shown that aggregate throughput is dramatically improved with multi-stream TCP. There are, as you would expect, differences from the much-simplified scenario used above. Differences include the inability to utilize full bandwidth in a single stream, and a distinct "knee" after which additional streams provide only limited additional improvement in performance. There are a host of complicating factors that could account for these differences. One of them is clearly the simplification in the model. However, other factors could include buffer copies from kernel space to user space, bus bandwidth, disk performance, CPU load, etc..

In conclusion, from a performance point of view, transporting data across multiple TCP sessions is much more effective than tunneling through a single TCP session, and the difference is proportional to the square of the number of TCP sessions.

For more details on TCP performance see for example [SIMMOD]. For ongoing work in the context of improving the TCP performance in high-speed wide area networks see for example [QUSTART, SCATCP, FASTTCP]. [RFC2914] documents key issues related to fairness and flow granularity (and acceptable definitions thereof). For information on alternatives and variants to TCP, see [SURTRAN]. It is a survey prepared by the GGF Data Transport Research Group.

4. Access Domains

This section describes experienced issues related to access domains.

4.1 Firewalls

Firewalls pose interesting problems in grids. Since grid toolkits like Globus use non-standard ports for communication, job submission etc. configuration of both the toolkit and the firewall is required and cumbersome. Firewalls have to be configured to allow non-standard ports. To facilitate this process and avoid

allowing un-wanted traffic, toolkits have to be configured to use these ports consistently. There are two parts to the firewall configuration: client-side and server-side. For example, Globus uses callbacks to call functions on the clients. This requires the firewall to be configured to allow incoming ports [GTFWALL].

On the other hand, Grid toolkits have to be developed with firewall awareness. This may involve developing trusted proxies or other methods of secure means of tunneling. Grid protocols can be made firewall aware too.

Firewalls impact network performance and pose problems for maintaining quality of service. This is due to the overhead involved in analyzing the network traffic. It places burden on the CPU and the machines can becomes a bottleneck. There is always a trade off between performance and security.

4.2 Network Address Translators

Network Address Translators pose similar problems to firewalls as described above. Callbacks to clients form servers used by Globus, for example, require specific configuration to get through NATs. The NAT needs to be configured to allow such traffic patterns as well. Maintaining servers behind a NAT is hard if not impossible. For instance, Globus security mechanisms [GTFWALL] do not allow servers to be placed behind a NAT as they need to know actual IP address.

4.3 Middleboxes with L4-7 impact

The vision of a network agnostic to any L4-7 consideration has supported the explosive growth of IP networks over the last 15 years. The increasing relevance of security, mobility, gigabit-range throughput, streaming media, have de-facto implanted the appreciation for L4-7 issues at crucial points inside the network. The ensuing network nodes with L4-7 scope (in short, middleboxes) include: firewalls and intrusion detectors, SSL accelerators, traffic-shaping appliances, and load balancing intermediaries (often generalized as elements of a content delivery network). In more subtle ways, even the traditional L2-3 routers/switches now factor L4 considerations in the form of active queue management (e.g., RED) tailored to TCP, the dominant L4 protocol (90% of traffic over backbone extents is carried by TCP).

With middleboxes, the greater efficiencies and "hi-touch" services come with all important side-effects, which fall in two realms. Firstly, the network has built-in knowledge of some L4-7 protocols, and can show resistance to using some other L4-7 protocols, much the same way it shows resistance in upgrading from IPv4 to IPv6(as one would expect for a L3 protocol). Secondly, there is a need to

discover and signal such middleboxes to select one of several pre-defined behaviors.

As a practical consequence of the first side-effect, for the foreseeable future Grid communities will have the freedom to use any L4 protocol as long as it is TCP! Let us consider the case of a Grid infrastructure interested in using the SCTP protocol [RFC2960] for its bulk data transfers. SCTP is a standard-track L4 protocol ratified by the IETF, with TCP-like built-in provisions for congestion control, and thus safe from a network perspective. This example is not fictional, in that SCTP does bring interesting elements of differentiation over TCP (e.g., datagram delineation, multi-homing, etc.), which become especially appealing at gigabit rates. Across the end-to-end path, the points of resistance to SCTP will likely show up in a) termination points (contrast with the state-of-the-art high-performance TCP's Off-load Engines, TOE, in silicon), b) intrusion detection points (where a protocol's FSM must be statefully analyzed), c) firewalls with application-proxy capability (another instance of protocol termination or splicing, see case a), and d)content delivery networks (wherein the protocol is terminated and security processing is rendered prior to steering the data, contrast with the state-of-the-art TOEs and SSL accelerators, also in silicon). All of this warrants SCTP the risk of falling off several hot-paths, not to mention clearing all the security checkpoints along the way.

But there is more to it than just ossification around a L4 protocol called TCP. The TCP operating requirements are practically limited to using fixed destination port numbers, because firewalls and intrusion detection devices have fundamental troubles coping with dynamic ports usage (the H.323 circles first learned this lesson, the hard way). In fact, many a community resorted to the extreme point of sanctioning that their destination port number be port 80, regardless of their higher-level protocols and applications, thus de-facto voiding the very value of firewalls and intrusion detection.

As said for the second class of side-effects, an application will likely need to discover and signal "middleboxes" in order to access the QoS and security behaviors of choice. Without signaling from the application, the middlebox may even dispose of the soft-state associated with the application, and reuse the resources for other applications (this is a typical syndrome with firewalls and “silent” long-lasting TCP connections). Unfortunately, this is an area still showing a wide variety of plays. The wire protocol can be in-band (e.g., SOCKS) or out-of-band (e.g., RSVP). Furthermore, the programming model can be structured around APIs, or require a point-and-click GUI session, or a command-line-interface (CLI) script.

The common case of long-lasting TCP connections traversing one or more middleboxes is worth a special mention. It has been observed that the intervals without traffic may result in a loss of the soft-state at the middleboxes (even though the TCP flow is alive and well). To avoid this, Grid developers are often tempted to use the KEEPALIVE option in the TCP protocol (accessible through a "setsocketopt()" system call in common OSs). It must be noted that KEEPALIVE is a frowned upon option in TCP. In fact, no RFC mandates its implementation. [RFC1122] discusses its implications (while acknowledging that popular implementations went off coding it as a "premium" feature a long time ago). Developers are encouraged to build their liveness handshakes (if any) into the protocol(s) above TCP, resulting in more accurate liveness reports on the actual endpoints.

SOCKS [RFC1928] is an attempt to standardize the exchange between application and firewall, though the market adoption and degree of confidence on the overall security solution are spotty at best. This fragmented and still immature solution space does not help Grid users who, among others, would certainly benefit from a comprehensive, unified style of interaction with middleboxes.

Standard signaling protocols are expected to come from the IETF MIDCOM working group [MIDCOM]and the NSIS working group [NSIS] at the IETF (though the actual APIs are out of their scope).

4.4 VPNs

With a Virtual Private Network, a user has the experience of using dedicated, secure links of various reach (LAN, MAN, or WAN), even though in reality the actual network is built out of Metro/WAN network extents over public, insecure networks (such as the Internet). VPNs are known to scale quite well, from the consumer market (e.g., telecommuters using VPNs across WiFi and the Internet) to the large enterprise market (e.g., for branch-office to headquarters communication). Typical VPN technologies use (but are not limited to) the Ipsec protocol [RFC2401] and the Internet Key Exchange (IKE) protocol [RFC2409]. In a VPN, either the ingress point, or the egress point, or both can have portable, pure-software implementations, or come in appliance-style embedded setups.

Once a VPN is established, the VPN is meant to be entirely transparent to the user. As such, Grid applications will typically continue to use security fixtures of their own, in an end-to-end fashion, and the existence of an underlying VPN covering a portion of the end-to-end extent goes totally un-noticed. There are, however, two important exceptions.

VPN protocols have provisions for periodically renegotiating new keying material, so as to maintain the integrity of the VPN for a

very long time (possibly indefinitely). In practice, however, local security protocols must require users to periodically re-instate their credentials into the VPN console, to take into account changes in personnel's authorization. This added burden can be irksome to many a Grid user, especially when there are long-running tasks at stake, and the VPN is provisioned via an appliance that can only be operated via point-and-click sessions or command-line-interface scripts (as the vast majority of appliances are, today). This situation is clearly vulnerable to operator errors, given that application and VPN console are totally disjoint.

When a VPN has a fatal error, the application will discover it the hard way, with traffic coming to a screeching halt, and retransmission attempts going off periodically. Whenever the application and VPN console are disjoint, there is no way for the application to restart the VPN, or signal a 3rd party to do so.

In both circumstances, it would be nice if the Grid application could access the VPN console, re-affirm credentials, and register for notifications through an API like the GSS API.

5. Transport Service Domains

This section describes experienced issues related to the core.

5.1 Service Level Agreement (SLA)

Connectivity or data transport service between two geographically dispersed locations is usually provided by an independent third party, generically called a Service Provider (SP). The Service Level Agreement (SLA) is a contract agreed upon between the SP and the service consumer (in this case a grid subscriber) detailing the attributes of the service like connection uptime, scheduled downtime, unscheduled downtime, service provider liabilities among others. Since the SLA contains business-related parameters that are outside the scope of this document, the term Service Level Specification (SLS)[RFC3260] will be used to specify the technical qualities of the service.

5.1.1 Grids and SLS

Grids are built by user communities using resources that are typically geographically dispersed, even if they belong to the same administrative organization. Grid applications utilizing the distributed compute and storage resources depend on the underlying network connectivity provided by the transport service provider for successful and timely completion. There is a high likelihood that the remote members of the grid virtual organization have different transport providers for their service. It is also possible that each grid location has different service and physical layer connectivity combinations at the network access i.e. IP over SONET

leased line service, or a L2 Ethernet/Frame Relay/ATM service. All these factors lead to different SLS's at each location and can cause a grid application to get inconsistent end-to-end Quality of Service especially in case of failures. For example, if a grid application requiring transport level performance requires resources at a location with SLS for Layer 3 (IP) service, it has to derive through unspecified means the transport layer service equivalent to ensure compatible service levels.

Each Grid Application may have different Quality of Service requirements of the network. For example, a visualization grid application may require high bandwidth, low latency Fiber Channel service for storage access while a computationally-intensive grid application may require just a best-effort IP service for data movement. The Grid resource allocation algorithm may not be able to allocate the proper grid resources without having the knowledge of network services and SLS parameters available at each grid location.

A common template to specify Grid SLS with measurable performance parameters related to grid applications will be needed for the grid application to work seamlessly across diverse geographical locations. The parameters of SLS can then become a great tool for grid users to measure the quality and reliability of the offered service over time.

It should be noted that even though a SP provides an SLS compliant service, the grid application may not get the right QoS due to performance of network owned by the grid organization. The grid organization needs to provide similar SLS for its own internal networks in order for guaranteed end-to-end application QoS.

5.1.2 SLS Assurance

Currently, the transport service provider provides the mechanisms to monitor the network and assures the user of compliance to the negotiated SLS requirements and parameters. The grid user does not have any means to independently measure and verify the SLS negotiated or determine if the network QoS needed by the application is being met at each location and thus, cannot guarantee grid application performance. Providing mechanisms to the Grid applications to monitor network SLS parameters and have access to network alerts, errors and outages will result in better resource selection and also assure end-to-end service quality to the grid application. There are cases where the SP is not able to provide customers access to network information for SLS monitoring and assurance purposes. In that case, the SP should be able to measure and monitor end-to-end application performance and keep a real-time log accessible by customers to ensure SLS compliance.

5.1.3 On-demand SLS

One of the major values of the grid is the ability to form grid virtual organizations dynamically to access the resources need for a particular application. The compute and storage resources are dynamically allocated from an available pool. For example a compute intensive, high-energy physics application can use the majority of grid compute resources for a few weeks and then a data intensive data-mining application, can leverage the same compute/data resources with different network requirements. Currently, the SLS's are negotiated at time of service, and do not change through the length of service contract. Providing dynamic network resources with associated dynamic SLSs will help deliver a quality of service based on application needs as well as provide efficient use of available network resources.

5.2 Overprovisioned networks

The challenging network requirements of Grid applications are often associated with the demand to access an overprovisioned network. In assuming a network capabilities without limitations, the demand of Grid application would clearly be satisfied. However, the assumption of offering nearly unlimited bandwidth capabilities is not always true.

The costs of deploying optical networks are affected by a mixture of link and equipment costs. While link costs are typically sub-linear to the capacity, the equipment costs for beyond 2.5 Gbps interfaces are still super-linear. Furtheron, the amount of parallel wavelength multiplexed within a single fiber is also still limited, either caused by the limited capabilities of the existing fibre itself, or by the dimension of the optical cross connects. A network supporting hundreds of lambdas on a particular fiber is not emerging within a reasonable time scale.

On the other hand, end-systems can be expected to be attached by Gigabit Ethernet interfaces now, and 10GiGE in the near future. Also, the Grid is about to deploy applications which aim for the actual use of the available bandwidth capabilities. This leads to an environment in which the classical onion model, i.e. an increase of bandwidth capabilities when moving towards the core, is problematic. The concept of overprovisioning might therefore not scale with the deployment of Grid applications. Meshing, i.e. the use of multiple fibers, could be an economic solution to this. Here, however, one has to consider that Grid users are not really concerned about capacity, but about goodput. Mis-ordered packets must be avoided when meshing is implemented. On the other hand, mashing nicely fits to the concept of parallel file transfers introduced in section 3.3.

6. General Issues

So far, we listed issues which were related to a particular region of the interconnecting network. This section lists the remaining issues.

6.1 Service Orientation and Specification

In some sense, the Grid can be compared with the World Wide Web. While the original goal of the World Wide Web was to offer location independent access to information resources by using a simple protocol, a common name space (the Universal Resource Locator), the integration of the name space into a standardized hypertext language, and a graphical user interface supporting these hypertext documents, the Grid is about to offer individuals and institutions the opportunity to build virtual organizations which facilitates the access to the problem solving services of the community. A Grid infrastructure facilitates the composition of existing services to build more advanced, higher-level services.

Of course, when following this service oriented view of the Grid, the question arises about network related services. Low-latency and high-throughput communication are performance-critical in most distributed environments. For Grid applications, the demanded level of value-addedd services is basically as follows:

- Access to a premium service which offers low-latency communication between the two end-points. This service assures that the individual packets which were conformant to a given traffic profile (token/leaky bucket constrained) were transported to the destination within a given delay boundary. In addition to the classical real-time traffic, such as voice over IP or video conferencing, the Grid introduces more challenging communication demands, for example in the context of a distributed VR-environment in which the haptic is remotely driven.

- Access to a guaranteed rate/bandwidth on-demand service. This service follows the assurance of the Premium service with respect to the avoidance of packet drops, but does not have to state strong delay boundaries.

While a guaranteed rate service allows for the implementation of deadline data transfers, a less-than best-effort service, i.e. the scavenger service, is of particular interest to support high-throughput communication of single applications in order to allow for fairness among competing best-effort transfers.

The Optical Internetworking Forum (OIF) has published an implementation agreement for interfacing to services in optical networks. This optical User Network Interface (UNI) offers [OIFUNI] a GMPLS-compatible way to implement bandwidth on-demand services. It thus has a strong relation to the service oriented view of the

Grid. However, the current UNI 1.0 version does not fully cover the functionality required by a Grid infrastructure.

Assuming that network services can be used by Grid applications to compose higher level services, the question arises whether there are particular provisioning capabilities which are of benefit. The coordinated allocation of multiple resources is a challenge. The start up of the individual service requests somehow has to be synchronized without wasting potentially scarce and thus expensive resources by an allocated service request which has to wait for the allocation of related tasks. One potential solution to this is given by the ability to reserve resources in advance. Within the Grid Resource Allocation Agreement Protocol (GRAAP) Working Group of the Global Grid Forum, the term advance reservation was defined as follows:

> An advance reservation is a possibly limited or restricted delegation of a particular resource capability over a defined time interval, obtained by the requester from the resource owner through a negotiation process.

6.2 Programming Models

The GFD-E.5 Advance Reservation API document describes an experimental interface to advance reservation capabilities. The API can be considered a remote procedure call mechanism to communication with a reservation manager. A reservation manager controls reservations for a resource: it performs admission control and controls the resource to enforce the reservations.

The document describes a C-binding of this API which allows for a uniform programming model which is capable of making and manipulating a reservation regardless of the type of the underlying resource. It thereby simplifies the programming when an application must work with multiple kinds of resources and multiple simultaneous reservations. The document defines a set of reservation related functions and their parameters. Resource specific service parameters are encoded in a particular resource description language.

6.3 Support for Overlay Structures

Overlay structures provide a way of achieving high-performance using existing network infrastructure. Resilient overlay networks[RON] allows applications to detect and recover from path outages and other routing problems. Features like application-controlled routing, multi-path routing and QoS routing can have great impact on performance of grid applications. Though this has promising implications, placing of overlay nodes can be a tricky problem.

6.4 Multicast

The ever growing needs for computation power and accesses to critical resources have launched in a very short time a large number of grid projects. The very basic nature of a grid is to allow a large community of people to share information and resources across a network infrastructure. Most of the grid usages nowadays consist in (i) database accesses, sharing and replication (DataGrid, Encyclopedia of Life Science), (ii) distributed data mining (seti@home for instance) and, (iii) data and code transfers for massively parallel job submissions. For the moment, most of these applications imply a rather small number of participants and it is not clear whether there is a real need for very large groups of users. However, even with a small number of participants, the amount of data to be exchanged can be so huge that the time to complete the transfers can rapidly become unmanageable! More complex, fine-grained applications could have complex message exchange patterns such as collective operations and synchronization barriers.

Multicast [DEERING] is the process of sending every single packet from the source to multiple destinations in the same logical multicast group. Since most of communications occurring on a grid imply many participants that can be geographically spread over the entire planet, these data transfers could be gracefully and efficiently handled by multicast protocols provided that these protocols are well-designed to suit the grid requirements. Motivations behind multicast are to handle one-to-many communications in a wide-area network with the lowest network and end-system overheads while achieving scalability.

In contrast to best-effort multicast, that typically tolerates some data losses and is more suited for real-time audio or video for instance, reliable multicast [SRELMUL] requires that all packets are safely delivered to the destinations. Desirable features of reliable multicast include, in addition to reliability, low end-to-end delays, high throughput and scalability. These characteristics fit perfectly the need of the grid computing and distributed computing communities. Embedding a reliable multicast support in a grid infrastructure would not only optimize the network resources in term of bandwidth saving, but also increase both performances for applications, and interactivity for end-users, thus bringing the usage of grids to a higher level than it is at the moment (mainly batch job submission).

Here is some necessary background on main multicasting protocols and mechanisms in IP networks. Internet Group Management Protocol (IGMP) is used by hosts to join or leave a multicast group. RFC 3376 describes IGMPv3. As regards multicast forwarding algorithms, there are two main families of algorithms: reverse path forwarding (RPF) and center-based tree (CBT). The former yields two advantages

because of fastest delivery of multicast data and different tree creation for different source node resulting in more efficient utilization of network resources. The latter utilizes another method to calculate optimum paths and its main disadvantage consists in creating suboptimal path for some sources and receivers.

Based on these two main algorithms, there were developed several multicast routing protocols as Distance Vector Multicast Routing Protocol (DVMRP), Multicast OSPF (MOSPF) and Protocol Independent Multicast (PIM). DVMRP was initially defined in RFC 1075 and it uses RPF algorithm. Multicast Extensions to OSPF (MOSPF) is defined in RFC 1584. It is not a separate multicast routing protocol as DVMRP. This protocol forwards datagrams using RPF algorithm and it does not support any tunneling mechanism. Unlike MOSPF, PIM is independent of any underlying unicast routing protocol and has two different ways of operation: dense mode (PIM-DM) and sparse mode (PIM-SM) defined in RFC 2362. The former implements the RPF algorithm. The latter uses a variant of CBT algorithm. PIM-DM should be used in contexts where the major part of hosts inside a domain needs multicast data but also in contexts where senders and receivers are relatively close, there are few senders and many receivers, multicast traffic is heavy and/or constant. PIM-DM does not support tunnels as well. One of the main benefits of PIM-SM is the capability to reduce the amount of traffic injected into the network because of multicast data are filtered from network segments unless a downstream node requires them. Furthermore pruning information is maintained only in equipments connected to the multicast delivery tree. PIM-SM is well suited for those situations in which there are a large number of multicast data streams flowing towards a small number of the LAN segments and also in those environments in which there are few receivers in a multicast group or when senders and receivers are connected through WAN links or the stream is intermittent.

With regard to inter-domain routing, there are two approaches to multicast domains interconnection: Multicast Source Discovery Protocol (MSDP) and Border Gateway Multicast Protocol (BGMP). They both are not currently IETF standards.

Nowadays, MBONE is still operational but multicast connectivity is natively included in many Internet routers. This trend is growing and will eliminate the need for multicast tunnels. Current MBONE environment is only a temporary solution and will be obsolete when multicasting is fully supported in every Internet router.

Recently, development of multicast systems has accelerated thanks to new and improved applications such as many grid applications: teleimmersion, data distribution, gaming and simulation, real-time data multicast. Many of these applications use UDP instead of usual TCP because of reliability and flow control mechanisms have not

been optimized for real-time broadcasting of multimedia data. In some contexts, it is preferred to loose few packets instead of having additional TCP delays. In addition to UDP, many applications use Real-Time Transport Protocol (RTP).

Another open issue is concerned with multicast security, that is, securing group communications over the Internet. Initial efforts are focused on scalable solutions with respect to environments in which there are a single sender and many recipients. Initially, about multicast data delivery, IP-layer multicast routing protocols are principally considered (with or without reliability) such as those exposed before. Typically, each group has its own trusted entity (Group Controller) that manages security policy and handles group membership. Some minimal requirements are group members' admission and source/contents authentication; DoS attacks protection is desirable as well. Considering that many applications fall in one to many multicast category, each one with its own requirements, it is not a feasible way to think of a "one size fits all" solution. So it is going to define a general framework characterized by three functional building blocks: data security transforms, group key management and group security association, group policy management. With regard to large multicast groups, see for instance [MSEC]. Actually, there are no standards. Some working groups inside IETF and IRTF are actively working on this very crucial topic.

Besides the routing layer discussed previously, multicast at the transport layer mainly provides the reliable features needed by a number of applications. Many proposals have been made during these past 15 years and early protocols made usage of complex exchanges of feedback messages (ACK or NACK) [XTP95][FLO97][PAU97][YAV95]. These protocols usually take the end-to-end solution to perform loss recoveries and usually do not scale well to a large number of receivers due to the ACK/NACK implosion problem at the source. With local recovery mechanism, the retransmission of a lost packet can be performed by any receiver in the neighborhood (SRM) or by a designated receiver in a hierarchical structure (RMTP, TMTP, LMS [PAP98], PGM [GEM03]). All of the above schemes do not provide exact solutions to all the loss recovery problems. This is mainly due to the lack of topology information at the end hosts and scalability and fairness with TCP still remain open issues.

Given the nature of the information exchanged on a grid, reliable multicast is the best candidate for providing an efficient multipoint communication support for grid applications. The objectives are ambitious: extending the current grid capabilities for supporting fully distributed or interactive applications (MPI, DIS, HLA, remote visualization...). With the appropriate reliable multicast facilities, grid infrastructures would be more efficient to handle a larger range of applications.

There are however a number of factors that seriously limit the availability of multicast on large scale networks such as the Internet or a grid infrastructure. Some are technical, others are more politic.

If we consider a dedicated grid infrastructure with all participants and ISPs willing to move forward (unfortunately this is not the case), then issues related to interdomain routing, security or firewalls could be fixed quite easily with the current tools and protocols (MBGP and MSDP for interdomain routing and for controlling sources for instance, PIM-SSM for security), especially when the size of the group is not very large. What's left is the core problem of reliable multicast: how to achieve scalability of recovery schemes and performances? As stated previously in the brief background, there is no unique solution for providing multicast facilities on an internetwork: end-to-end, with local recoveries, with router assistance... To this long list, should be added the alternative solutions to IP multicast based on overlays and host-based multicast that scale quite well up to some hundreds of receivers [STO00][HUI00][SAY03]. In this context, it seems very reasonable to consider all possibilities and to have specific solutions for specific problems. One example could be to have an overlay-based multicast for small groups of computing sites and a fully IP multicast scheme for larger groups. The main difficulties are then to provide a multicast support for high throughput (job and data transfers) and low latency (for distributed/interactive applications).

Regarding how the multicast support should be presented to the user or the application, there are several design choices that we believe can coexist (and are fully complementary): a separate program 'a la' ftp or a separate library to be linked with the application or a fully integrated solution with high interaction with the grid middleware, this last solution being the more transparent one for the end-user, but also the most difficult to achieve.

6.5 Sensor Networks

TBD.

7. Security Considerations

Network security can be implemented at the link level (i.e., L2, as in WEP or FrameRelay security), at the network level (i.e., L3, as in IPsec), and at the application level (i.e., at layers above 4, like TLS). These approaches have well-known strengths and weaknesses, re-enforcing the concept that there isn’t a “one size fits all” network security solution. Additionally, these approaches are not mutually exclusive. They can coexist quite nicely and can be applied incrementally, as the traffic flows from private

enclaves to the public, insecure Internet. While “the more, the merrier” argument typically holds when dealing with security, there are important issues in computing overhead, packet header overhead, high-availability, and policy.

7.1 Security Gateways

GGF’s Grid Security Infrastructure (GSI) qualifies as application-level security. As any other application-level security schema, it targets true end-to-end security, thus removing the annoying problem (as well as vulnerability) of trusting network intermediaries. When properly configured, GSI is “good to go” over any network extent, regardless of its level of security.

In many a scenario, however, it is expected that local policies will dictate the use of a security gateway (e.g., an Ipsec device) between private and public enclaves. Most security gateways do not discriminate between traffic that needs security versus traffic that is already secured in an end-to-end fashion. To an operator, the gateway’s appeal is that it is a fixed point of transit between private and public enclaves, and its well-being can be easily audited. A GSI user can argue that the gateway needlessly adds meta-data overhead to the packet, and likely represents a bottleneck (e.g., heavy duty crypto processing) if the gateway insists in applying another layer of authentication and confidentiality. The Ipsec tunnel-mode protocol (commonly used by security gateways) inserts a new IP header and a AH/ESP header, and there may be a chance that the new packet comes to exceed the link MTU (e.g., the Ethernet maximum frame size). The problem is further exacerbated by the fact that IP fragmentation is a deprecated feature (i.e., all firewalls reject IP fragments nowadays), and Path MTU discovery may fail to detect the actual MTU available.

Given that local policies are neither necessarily reasonable nor flexible, a GSI user can relax the security stipulations at her end, and, for instance, skip encrypting traffic if the security gateway is known to do so already, and she can live without confidentiality across the limited network extent between the Grid application and the security gateway. With state of the art technology, this type of reasoning cannot be automated in any way, and the GSI user is left with ad-hoc interpretation of her local policies, intervening security gateways, topologies, and the likes.

Some network gateways may attempt to compress traffic prior to its traversing a limited-bandwidth network extent. The composition of encryption and compression raises an issue of temporal dependence amongst the two. Compression is likely to yield gains when performed before encryption. Conversely, compression results in no gains and gratuitous overhead if performed after encryption. In fact, an encrypted set cannot be compressed, because the bit distribution operated by the encryption algorithm voids all known

compression techniques, which thrive on regular patterns. Should data be encrypted at the GSI level, any attempt to compress data past that point will produce no benefit, and will rather add overhead; data must be compressed prior to the GSI layer. If the GSI user delegates encryption to a security gateway, then there will be solid opportunities to compress the data at the NIC level or inside the network.

Section 4.3 details other coordination issues between GSI users and legacy VPNs devices (failures, timeouts).

7.2 Authentication and Authorization issues

Authentication (AuthN) and Authorization (AuthZ) are typically implemented as network services. That is, they reside in the network and are implemented within a consolidated locus for directories and access policies. This way, revocation of privileges, auditing, and any other management operation are particularly efficient. AAA (Authentication, Authorization, and Accounting) is a widely used denomination for this class of network services.

It is imperative that the AuthN and AuthZ services be available at all times, else the end-systems' security fixtures that depend on them will come to a screeching halt (while caching of earlier AuthN and AuthZ decisions at the end-systems level is not a good idea, in that it circumvents revocation actions that may have happened meanwhile). This availability requirement poses a burden on the server(s) implementing AAA functionality (typically a fault-tolerant cluster of servers), as well as the network paths connecting end-systems to AAA services. The latter may all of a sudden become unreachable due to slow router convergence after partial failures in the network, inadvertent SLA breaches, or outright malicious intrusion and DoS attacks underway.

The centrality of AAA services and their unexpected unavailability thus warrant the syndrome that Butler Lampson aptly described as: "A distributed system is one in which I can't get my work done because a computer I've never heard of has failed".

In GSI, the security mechanism are accessed through an indirection layer called GSS API, which hides to the user the fact that, for instance, Kerberos is being used instead of PKI. While GSS is a sophisticated and useful programming model, there is a flip side to it in case of failures. Should the Kerberos server(s) become unreachable, the troubleshooting of the ensuing failures may turn out to be cumbersome (the Kerberos server playing the role of the computer never heard of in Lampson's citation). Whereas other systems requiring an explicit Kerberos login by a user (e.g., the Andrew distributed File System) are more amenable to track down the

failure (though the failure will still be fatal until the Kerberos service comes back on line).

7.3 Policy issues

The sites forming a Virtual Organizations may very well live by different security standards. While one site has established a sophisticated certificate practice statement, at another site of the same VO the passwords are written on the back of keyboards, and private keys are unprotected. The wide variety of crypto parameters creates a host of potential pitfalls. In fact, the vast majority of security exploitations leverages the weakest policy definitions and especially their implementations. Exposure to these risks is inherent to the way Grids work. Hence the ongoing effort in the Security Area, as in the GGF GCP WG.

Security gateways enact a Layer 3 overlay (i.e., based on IP, Ipsec)that suffers similar vulnerabilities. In this space, the IETF is actively working on IP-level security policies (IETF IPSP WG). It will take a while before the outcome of this work will be widely available in the security gateway marketplace.

Due to the different nature of application-level security and network-level security, the former and the latter can coexist while using entirely different mechanisms and policies. In many organizations, however, it becomes attractive for the two security approaches to share in on some of the AAA fixtures, and on the hefty costs incurred by organizations to make these fixtures work dependably (e.g., high availability, policy stipulations, certificate authorities, auditing, etc.). The implementation of the PKI infrastructure is a potential point of convergence. GSI can leverage PKI infrastructure through the GSS API, while the Internet Key Exchange (IKE) protocol can perform certificate-based peer authentication (i.e., via X.509v3) using digital signatures.

It has been noted earlier on (section 7.1) that a GSI user can delegate some of the security protection to a legacy security gateway, thus eliminating the overhead of security measures being applied twice to the same data. There is no way, however, for the GSI user to get a quantitative, objective measure of the relative strength in application-level and network-level security, when considering both security mechanisms and the policies involved. The finalization and market adoption of the outcomes of GGF GCP WG and IETF IPSP WG will go a long way towards providing a framework upon which automated evaluation tools can be built.

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