Network Working Group
Request for Comments: 1363

C. Partridge
BBN

September 1992

A Proposed Flow Specification

Status of this Memo

This memo provides information for the Internet community. It does not specify an Internet standard. Distribution of this memo is unlimited.

Abstract

A flow specification (or "flow spec") is a data structure used by internetwork hosts to request special services of the internetwork, often guarantees about how the internetwork will handle some of the hosts' traffic. In the future, hosts are expected to have to request such services on behalf of distributed applications such as multimedia conferencing.

The flow specification defined in this memo is intended for information and possible experimentation (i.e., experimental use by consenting routers and applications only). This RFC is a product of the Internet Research Task Force (IRTF).

Introduction

The Internet research community is currently studying the problems of supporting a new suite of distributed applications over internetworks. These applications, which include multimedia conferencing, data fusion, visualization, and virtual reality, have the property that they require the distributed system (the collection of hosts that support the applications along with the internetwork to which they are attached) be able to provide guarantees about the quality of communication between applications. For example, a video conference may require a certain minimum bandwidth to be sure that the video images are delivered in a timely way to all recipients.

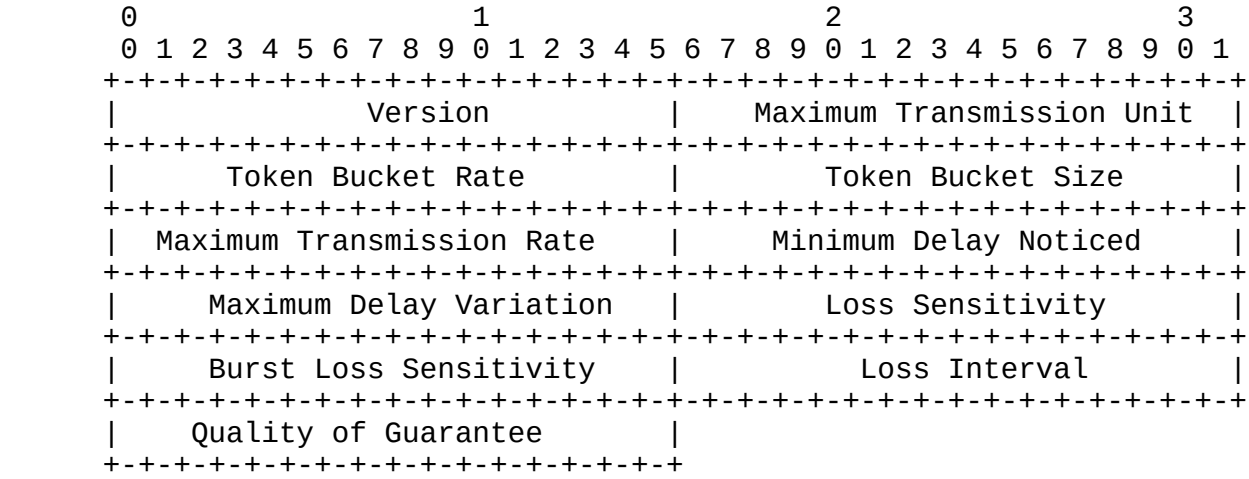
One way for the distributed system to provide guarantees is for hosts to negotiate with the internetwork for rights to use a certain part of the internetwork's resources. (An alternative is to have the internetwork infer the hosts' needs from information embedded in the data traffic each host injects into the network. Currently, it is not clear how to make this scheme work except for a rather limited set of traffic classes.)

There are a number of ways to effect a negotiation. For example a negotiation can be done in-band or out-of-band. It can also be done in advance of sending data (possibly days in advance), as the first part of a connection setup, or concurrently with sending (i.e., a host starts sending data and starts a negotiation to try to ensure that it will allowed to continue sending). Insofar as is possible, this memo is agnostic with regard to the variety of negotiation that is to be done.

The purpose of this memo is to define a data structure, called a flow specification or flow spec, that can be used as part of the negotiation to describe the type of service that the hosts need from the internetwork. This memo defines the format of the fields of the data structure and their interpretation. It also briefly describes what purpose the different fields fill, and discusses why this set of fields is thought to be both necessary and sufficient.

It is important to note that the goal of this flow spec is to able to describe *any* flow requirement, both for guaranteed flows and for applications that simply want to give hints to the internetwork about their requirements.

Format of the Flow Spec



Discussion of the Flow Spec

The flow spec indicates service requirements for a single direction. Multidirectional flows will need to request services in both directions (using two flow specs).

To characterize a unidirectional flow, the flow spec needs to do four things.

First, it needs to characterize how the flow's traffic will be injected into the internetwork. If the internetwork doesn't know what to expect (is it a gigabit-per-second flow or a three kilobit-per-second flow?) then it is difficult for the internetwork to make guarantees. (Note the word "difficult" rather than "impossible." It may be possible to statistically manage traffic or over-engineer the network so well that the network can accept almost all flows, without setup. But this problem looks far harder than asking the sender to approximate its behavior so the network can plan.) In this flow spec, injected traffic is characterized as having a sustainable rate (the token bucket rate) a peak rate (the maximum transmission rate), and an approximate burst size (the token bucket size). A more precise definition of each of these fields is given below. The characterization is based, in part, on the work done in [1].

Second, the flow spec needs to characterize sensitivity to delay. Some applications are more sensitive than others. At the same time, the internetwork will likely have a choice of routes with various delays available from the source to destination. For example, both routes using satellites (which have very long delays) and routes using terrestrial lines (which will have shorter delays) may be available. So the sending host needs to indicate the flow's sensitivity to delay. However, this field is only advisory. It only tells the network when to stop trying to reduce the delay - it does not specify a maximum acceptable delay.

There are two problems with allowing applications to specify the maximum acceptable delay.

First, observe that an application would probably be happy with a maximum delay of 100 ms between the US and Japan but very unhappy with a delay of 100 ms within the same city. This observation suggests that the maximum delay is actually variable, and is a function of the delay that is considered achievable. But the achievable delay is largely determined by the geographic distance between the two peers, and this sort of geographical information is usually not available from a network. Worse yet, the advent of mobile hosts makes such information increasingly hard to provide. So there is reason to believe that applications may have difficulty choosing a rational maximum delay.

The second problem with maximum delays is that they are an attempt to quantify what performance is acceptable to users, and an application usually does not know what performance will be acceptable its user. For example, a common justification for specifying a maximum acceptable delay is that human users find it difficult to talk to each other over a link with more than about 100 ms of delay. Certainly such delays can make the conversation less pleasant, but it

is still possible to converse when delays are several seconds long, and given a choice between no connection and a long delay, many users will pick the delay. (The phone call may involve an important matter that must be resolved.)

As part of specifying a flow's delay sensitivity, the flow spec must also characterize how sensitive the flow is to the distortion of its data stream.

Packets injected into a network according to some pattern will not normally come out of the network still conforming to the pattern. Instead, the pattern will have been distorted by queueing effects in the network. Since there is reason to believe that it may make network design easier to continue to allow the networks slightly distort traffic patterns, it is expected that those applications which are sensitive to distortion will require their hosts to use some amount of buffering to reshape the flow back into its original form. It seems reasonable to assume that buffer space is not infinite and that a receiving system will wish to limit the amount of buffering that a single flow can use.

The amount of buffer space required for removing distortion at the receiving system is determined by the variation in end-to-end transmission delays for data sent over the flow. If the transmission delay is a mean delay, D, plus or minus a variance, V, the receiving system needs buffer space equivalent to 2 * V * the transmission rate. To see why this is so, consider two packets, A and B, sent T time units apart which must be delivered to the receiving application T time units apart. In the worst case, A arrives after a delay of D-V time units (the minimum delay) and B arrives after a delay of D+V time units (the maximum delay). The receiver cannot deliver B until it arrives, which is T + 2 * V time units after A. To ensure that A is delivered T time units before B, A must be buffered for 2 * V time units. The delay variance field is the value of 2 * V, and allows the receiver to indicate how much buffering it is willing to provide.

A third function of the flow spec is to signal sensitivity to loss of data. Some applications are more sensitive to the loss of their data than other applications. Some real-time applications are both sensitive to loss and unable to wait for retransmissions of data. For these particularly sensitive applications, hosts may implement forward error correction on a flow to try to absolutely minimize loss. The loss fields allow hosts to request loss properties appropriate for the application's requirements.

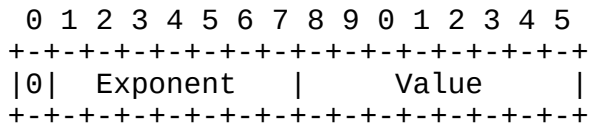
Finally, it is expected that the internetwork may be able to provide a range of service guarantees. At the best, the internetwork may be asked to guarantee (with tight probability bounds) the quality of

service it will provide. Or the internetwork may simply be asked to ensure that packets sent over the flow take a terrestrial path. The quality of guarantee field indicates what type of service guarantee the application desires.

Definition of Individual Fields

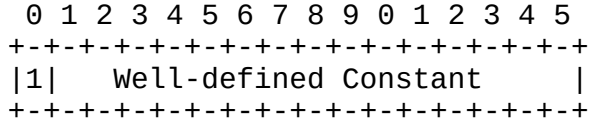
General Format of Fields

With a few exceptions, fields of the flow spec are expressed using a common 16-bit format. This format has two forms. The first form is shown below.



In this format, the first bit is 0, followed by 7 bits of an exponent (E), and an 8-bit value (V). This format encodes a number, of the form V * (2**E). This representation was chosen to allow easy representation of a wide range of values, while avoiding over-precise representations.

In some case, systems will not wish to request a precise value but rather simply indicate some sensitivity. For example, a virtual terminal application like Telnet will likely want to indicate that it is sensitive to delay, but it may not be worth expressing particular delay values for the network to try to achieve. For these cases, instead of a number, the field in the flow spec will take the following form:



The first bit of the field is one, and is followed by a 15-bit constant. The values of the constants for given fields are defined below. Any additional values can be requested from the Internet Assigned Numbers Authority (IANA).

Version Field

This field is a 16-bit integer in Internet byte order. It is the version number of the flow specification. The version number of the flow specification defined in this document is 1. The IANA is responsible for assigning future version numbers for any proposed

revisions of this flow specification.

This field does not use the general field format.

Maximum Transmission Unit (MTU)

A 16-bit integer in Internet byte order which is the maximum number of bytes in the largest possible packet to be transmitted over this flow.

This field does not use the general field format.

The field serves two purposes.

It is a convenient unit for expressing loss properties. Using the default MTU of the internetwork is inappropriate since the internetwork have very large MTU, such the 64Kbytes of IP, but applications and hosts may be sensitive to losses of far less than an MTU's amount of data -- for example, a voice application would be sensitive to a loss of several consecutive small packets.

The MTU also bounds the amount of time that a flow can transmit, uninterrupted, on a shared media.

Similarly, the loss rates of links that suffer bit errors will vary dramatically based on the MTU size.

Token Bucket Rate

The token bucket rate is one of three fields used to define how traffic will be injected into the internetwork by the sending application. (The other two fields are the token bucket size and the maximum transmission rate.)

The token rate is the rate at which tokens (credits) are placed into an imaginary token bucket. For each flow, a separate bucket is maintained. To send a packet over the flow, a host must remove a number of credits equal to the size of the packet from the token bucket. If there are not enough credits, the host must wait until enough credits accumulate in the bucket.

Note that the fact that the rate is expressed in terms of a token bucket rate does not mean that hosts must implement token buckets. Any traffic management scheme that yields equivalent behavior is permitted.

The field is in the general field format and counts the number of byte credits (i.e., right to send a byte) per second which are

deposited into the token bucket. The value must be a number (not a well-known constant).

The value zero is slightly special. It is used to indicate that the application is not making a request for bandwidth guarantees. If this field is zero, then the Token Bucket Size must also be zero, and the type of guarantee requested may be no higher than predicted service.

Token Bucket Size

The token bucket size controls the maximum amount of data that the flow can send at the peak rate. More formally, if the token bucket size is B, and the token bucket rate is R, over any arbitrarily chosen interval T in the life of the flow, the amount of data that the flow sends cannot have exceeded B + (R * T) bytes.

The token bucket is filled at the token bucket rate. The bucket size limits how many credits the flow may store. When the bucket is full, new credits are discarded.

The field is in the general field format and indicates the size of the bucket in bytes. The value must be a number.

Note that the bucket size must be greater than or equal to the MTU size.

Zero is a legal value for the field and indicates that no credits are saved.

Maximum Transmission Rate

The maximum transmission rate limits how fast packets may be sent back to back from the host. Consider that if the token bucket is full, it is possible for the flow to send a series of back-to-back packets equal to the size of the token bucket. If the token bucket size is large, this back-to-back run may be long enough to significantly inhibit multiplexing.

To limit this effect, the maximum transmission rate bounds how fast successive packets may be placed on the network.

One can think of the maximum transmission rate control as being a form of a leaky bucket. When a packet is sent, a number of credits equal to the size of the packet is placed into an empty bucket, which drains credits at the maximum transmission rate. No more packets may be sent until the bucket has emptied again.

The maximum transmission rate is the rate at which the bucket is emptied. The field is in the general field format and indicates the size of the bucket in bytes. The value must be a number and must be greater than or equal to the token bucket rate.

Note that the MTU size can be used in conjunction with the maximum transmission rate to bound how long an individual packet blocks other transmissions. The MTU specifies the maximum time an individual packet may take. The Maximum Transmission Rate, limits the frequency with which packets may be placed on the network.

Minimum Delay Noticed

The minimum delay noticed field tells the internetwork that the host and application are effectively insensitive to improvements in end-to-end delay below this value. The network is encouraged to drive the delay down to this value but need not try to improve the delay further.

The field is in the general field format.

If expressed as a number it is the number of microseconds of delay below which the host and application do not care about improvements. Human users only care about delays in the millisecond range but some applications will be computer to computer and computers now have clock times measured in a handful of nanoseconds. For such computers, microseconds are an appreciable time. For this reason, this field measures in microseconds, even though that may seem small.

If expressed as a well-known constant (first bit set), two field values are accepted:

0 - the application is not sensitive to delay

1 - the application is moderately delay sensitive
e.g., avoid satellite links where possible).

Maximum Delay Variation

If a receiving application requires data to be delivered in the same pattern that the data was transmitted, it may be necessary for the receiving host to briefly buffer data as it is received so that the receiver can restore the old transmission pattern. (An easy example of this is a case where an application wishes to send and transmit data such as voice samples, which are generated and played at regular intervals. The regular intervals may be distorted by queueing effects in the network and the receiver may

have to restore the regular spacing.)

The amount of buffer space that the receiving host is willing to provide determines the amount of variation in delay permitted for individual packets within a given flow. The maximum delay variation field makes it possible to tell the network how much variation is permitted. (Implementors should note that the restrictions on the maximum transmission rate may cause data traffic patterns to be distorted before they are placed on the network, and that this distortion must be accounted for in determining the receiver buffer size.)

The field is in the general field format and must be a number. It is the difference, in microseconds, between the maximum and minimum possible delay that a packet will experience. (There is some question about whether microsecond units are too large. At a terabit per second, one microsecond is a megabit. Presumably if a host is willing to receive data at terabit speeds it is willing to provide megabits of buffer space.)

The value of 0, meaning the receiving host will not buffer out delays, is acceptable but the receiving host must still have enough buffer space to receive a maximum transmission unit sized packet from the sending host. Note that it is expected that a value of 0 will make it unlikely that a flow can be established.

Loss Sensitivity

This field indicates how sensitive the flow's traffic is to losses. Loss sensitivity can be expressed in one of two ways: either as a number of losses of MTU-sized packets in an interval, or simply as a value indicating a level of sensitivity.

The field is in the general field format.

If the value is a number, then the value is the number of MTU-sized packets that may be lost out of the number of MTU-sized packets listed in the Loss Interval field.

If the value is a well-known constant, then one of two values is permitted:

0 - the flow is insensitive to loss

1 - the flow is sensitive to loss (where possible choose the path with the lowest loss rate).

Burst Loss Sensitivity

This field states how sensitive the flow is to losses of consecutive packets. The field enumerates the maximum number of consecutive MTU-sized packets that may be lost.

The field is in the general field format.

If the value is a number, then the value is the number of consecutive MTU-sized packets that may be lost.

If the value is a well-known constant, then the value 0 indicates that the flow is insensitive to burst loss.

Note that it is permissible to set the loss sensitivity field to simply indicate sensitivity to loss, and set a numerical limit on the number of consecutive packets that can be lost.

Loss Interval

This field determines the period over which the maximum number of losses per interval are measured. In other words, given any arbitrarily chosen interval of this length, the number of losses may not exceed the number in the Loss Sensitivity field.

The field is in the general field format.

If the Loss Sensitivity field is a number, then this field must also be a number and must indicate the number of MTU-sized packets which constitutes a loss interval.

If the Loss Sensitivity field is not a number (i.e., is a well-known constant) then this field must use the well-known constant of 0 (i.e., first bit set, all other bits 0) indicating that no loss interval is defined.

Quality of Guarantee

It is expected that the internetwork will likely have to offer more than one type of guarantee.

There are two unrelated issues related to guarantees.

First, it may not be possible for the internetwork to make a firm guarantee. Consider a path through an internetwork in which the last hop is an Ethernet. Experience has shown (e.g., some of the IETF conferencing experiments) that an Ethernet can often give acceptable performance, but clearly the internetwork cannot

guarantee that the Ethernet will not saturate at some time during a flow's lifetime. Thus it must be possible to distinguish between flows which cannot tolerate the small possibility of a failure (and thus must guaranteed at every hop in the path) and those that can tolerate islands of uncertainty.

Second, there is some preliminary work (see [2]) that suggests that some applications will be able to adapt to modest variations in internetwork performance and that network designers can exploit this flexibility to allow better network utilization. In this model, the internetwork would be allowed to deviate slightly from the promised flow parameters during periods of load. This class of service is called predicted service (to distinguish it from guaranteed service).

The difference between predicted service and service which cannot be perfectly guaranteed (e.g., the Ethernet example mentioned above) is that the imperfect guarantee makes no statistical promises about how it might mis-behave. In the worst case, the imperfect guarantee will not work at all, whereas predicted service will give slightly degraded service. Note too that predicted service assumes that the routers and links in a path all cooperate (to some degree) whereas an imperfect guarantee states that some routers or links will not cooperate.

The field is a 16-bit field in Internet byte order. There are six legal values:

0 - no guarantee is required (the host is simply expressing desired performance for the flow)

100 (hex) - an imperfect guarantee is requested.

200 (hex) - predicted service is requested and if unavailable, then no flow should be established.

201 (hex) - predicted service is requested but an imperfect guarantee is acceptable.

300 (hex) - guaranteed service is requested and if a firm guarantee cannot be given, then no flow should be established.

301 (hex) - guaranteed service is request and but an imperfect guarantee is acceptable.

It is expected that asking for predicted service or permitting an imperfect guarantee will substantially increase the chance that a

flow request will be accepted.

Possible Limitations in the Proposed Flow Spec

There are at least three places where the flow spec is arguably imperfect, based on what we currently know about flow reservation. In addition, since this is a first attempt at a flow spec, readers should expect modifications as we learn more.

First, the loss model is not perfect. Simply stating that an application is sensitive to loss and to burst loss is a rather crude indication of sensitivity. However, explicitly enumerating loss requirements within a cycle is also an imperfect mechanism. The key problem with the explicit values is that not all packets sent over a flow will be a full MTU in size. Expressed another way, the current flow spec expects that an MTU-sized packet will be the unit of error recovery. If flows send packets in a range of sizes, then the loss bounds may not be very useful. However, the thought of allowing a flow to request a set of loss models (one per packet size) is sufficiently painful that I've limited the flow to one loss profile. Further study of loss models is clearly needed.

Second, the minimum delay sensitivity field limits a flow to stating that there is one point on a performance sensitivity curve below which the flow is no longer interested in improved performance. It may be that a single point is insufficient to fully express a flow's sensitivity. For example, consider a flow for supporting part of a two-way voice conversation. Human users will notice improvements in delay down to a few 10s of milliseconds. However, the key point of sensitivity is the delay at which normal conversation begins to become awkward (about 100 milliseconds). By allowing only one sensitivity point, the flow spec forces the flow designer to either ask for the best possible delay (e.g, a few 10's of ms) to try to get maximum performance from the network, or state a sensitivity of about 95 ms, and accept the possibility that the internetwork will not try to improve delay below that value, even if it could (and even though the user would notice the improvement). My expectation is that a simple point is likely to be easier to deal with than attempting to enumerate two (or three or four) points in the sensitivity curve.

Third, the models for service guarantees is still evolving and it is by no means clear that the service choices provided are the correct set.

How an Internetwork is Expected to Handle a Flow Spec

There are at least two parts to the issue of how an internetwork is expected to handle a flow spec. The first part deals with how the flow spec is interpreted so that the internetwork can find a route which will allow the internetwork to match the flow's requirements. The second part deals with how the network replies to the host's request.

The precise mechanism for setting up a flow, given a flow spec, is a large topic and beyond the scope of this memo. The purpose of the next few paragraphs is simply to sketch an argument that this flow spec is sufficient to the requirements of the setup mechanisms known to the author.

The key problem in setting up a flow is determining if there exist one or more routes from the source to the destination(s) which might be able to support the quality of service requested. Once one has a route (or set of candidate routes) one can take whatever actions may be appropriate to confirm that the route is actually viable and to cause the flow's data to follow that route.

There are a number of ways to find a route. One might try to build a route on the fly by establishing the flow hop-by-hop (as ST-II does) or one might consult a route server which provides a set of candidate source routes derived from a routing database. However, whatever system is used, some basic information about the flow needs to be provided to the routing system. This information is:

* How much bandwidth the flow may require. There's no point in routing a flow that expects to send at over 10 megabits per second via a T1 (1.5 megabit per second) link.

* How delay sensitive the application is. One does not wish to route a delay-sensitive application over a satellite link, unless the satellite link is the only possible route from here to there.

* How much error can be tolerated. Can we send this flow over our microwave channel on a rainy day or is a more reliable link required?

* How firm the guarantees need to be. Can we put an Ethernet in as one of the hops?

* How much delay variation is tolerated. Again, can an Ethernet be included in the path? Does the routing system need to worry if the addition of this flow will cause a few routers to run

at close to capacity? (A side note: we assume that the routers are running with priority queueing systems, so running the router close to capacity doesn't mean that all flows get long and variable delays. Rather, running close to capacity means that high priority flows will be unaffected, and low priority flows will get hit with a lot of delay and variation.)

The flow spec provides all of this information. So it seems plausible to assume it provides enough information to make routing decisions at setup time.

The flow spec was designed with the expectation that the network would give a yes or no reply to a request for a guaranteed flow.

Some researchers have suggested that the negotiation to set up a flow might be an extended negotiation, in which the requesting host initially requests the best possible flow it could desire and then haggles with the network until they agree on a flow with properties that the network can actually provide and the application still finds useful. This notion bothers me for at least two reasons. First, it means setting up a flow is a potentially long process. Second, the general problem of finding all possible routes with a given set of properties is a version of the traveling salesman problem, and I don't want to embed traveling salesman algorithms into a network's routing system.

The model used in designing this flow spec was that a system would ask for the minimum level of service that was deemed acceptable and the network would try to find a route that met that level of service. If the network is unable to achieve the desired level of service, it refuses the flow, otherwise it accepts the flow.

The Flow Spec as a Return Value

This memo does not specify the data structures that the network uses to accept or reject a flow. However, the flow spec has been designed so that it can be used to return the type of service being guaranteed.

If the request is being accepted, the minimum delay field could be set to the guaranteed or predicted delay, and the quality of guarantee field could be set to no guarantee (0), imperfect guarantee (100 hex), predicted service (200 hex), or guaranteed service (300 hex).

If the request is being rejected, the flow spec could be modified to indicate what type of flow the network believes it could accept e.g., the traffic shape or delay characteristics could be adjusted or the

type of guarantee lowered). Note that this returned flow spec would likely be a hint, not a promised offer of service.

Why Type of Service is not Good Enough

The flow spec proposed in this memo takes the form of a set of parameters describing the properties and requirements of the flow. An alternative approach which is sometimes mentioned (and which is currently incorporated into IP) is to use a Type of Service (TOS) value.

The TOS value is an integer (or bit pattern) whose values have been predefined to represent requested quality of services. Thus, a TOS of 47 might request service for a flow using up to 1 gigabit per second of bandwidth with a minimum delay sensitivity of 100 milliseconds.

TOS schemes work well if the different quality of services that may be requested are both enumerable and reasonably small. Unfortunately, these conditions do not appear to apply to future internetworks. The range of possible bandwidth requests alone is huge. Combine this range with several gradations of delay requirements, and widely different sensitivities to errors and the set of TOS values required becomes extremely large. (At least one person has suggested to the author that perhaps a TOS field combined with a bandwidth parameter might be appropriate. In other words, a two parameter model. That's a tempting idea but my gut feeling is that it is not quite sufficient so I'm proposing a more complete parametric model.)

Another reason to prefer parametric service is optimization issues. A key issue in flow setup is trying to design the the routing system to optimize its management of flows. One can optimize on a number of criteria. A good example of an optimization problem is the following question (expressed by Isidro Castineyra of BBN):

> "Given a request to establish a flow, how can the internetwork accept that request in such a way as to maximize the chance that the internetwork will also be able to accept the next flow request?"

The optimization goal here is call-completion - maximizing the chance that requests to establish flows will succeed. One might alternatively try to maximize revenue (if one is charging for flows).

The internetwork is presumably in a better position to do optimizations if it has more information about the flow's expected behavior. For example, if a TOS system says only that a flow is

delay sensitive, the routing system must seek out the most direct route for the flow. But if the routing system is told that the flow is sensitive only to delays over 100 milliseconds, there may be a number of routes other than the most direct route which can satisfy this delay, thus leaving the most direct route available for a later flow which needs a far lower delay.

In fairness, it should be noted that a danger of a parametric model is that it is very easy to have too many parameters. The yearn to optimize can be overdone. The goal of this flow spec is to enumerate just enough parameters that it appears that essential needs can be expressed, and the internetwork has some information it can use to try to manage the flows. Features that would simply be nice or useful to have (but not essential) are left out to keep the parameter space small.

An Implication of the Flow Spec

It is important to observe that the there are fields in the flow spec that are based on information from the sender (such as rate information) and fields in the flow spec that are based on information from the receiver (such as delay variation). There are also fields that may sender and receiver to negotiate in advance. For example, the acceptable loss rate may depend on whether the sender and receiver both support the same type of forward error correction. The delay sensitivity for a voice connection may depend, in part, on whether both sender and receiver support echo cancelling.

The implication is that the internetwork must permit the sender and receiver to communicate in advance of setting up a flow, because a flow spec can only be defined once both sender and receiver have had their say. In other words, a reserved flow should not be the only form of communication. There must be some mechanism to perform a short exchange of messages in preparation for setting up a flow.

(Another aside: it has been suggested that perhaps the solution to this problem is to have the sender establish a flow with an incomplete flow spec, and when the receiver gets the flow spec, have the receiver send the completed flow spec back along the flow, so the internetwork can "revise" the flow spec according to the receiver’s desires. I have two problems with this approach. First, it is entirely possible that the receiver’s information may lead the internetwork to conclude that the flow established by the sender is no good. For example, the receiver may indicate it has a smaller tolerance for delay variation than expected and force the flow to be rerouted over a completely different path. Second, if we try to avoid having the receiver’s information cause the flow to fail, then we have to over-allocate the flow’s during the preliminary setup.

But over allocating the resources requested may lead us to choose better quality paths than we need for this flow. In other words, our attempts to optimize use of the network will fail.)

Advance Reservations and Flow Duration

The primary purpose of a flow specification is to provide information to the internetwork so the internetwork can properly manage the proposed flow's traffic in the context of other traffic in the internetwork. One question is whether the flow should give the network information about when the flow is expected to start and how long the flow is expected to last.

Announcing when a flow will start is generally of interest for advance reservations. (If the flow is not be reserved substantially in advance, the presentation of the flow spec to the internetwork can be taken as an implicit request for a flow, now.) It is my view that advance reservation is a distinct problem from the describing the properties of a flow. Advanced reservations will require some mechanism to maintain information in the network about flows which are not currently active but are expected to be activated at some time in the future. I anticipate this will require some sort of distributed database to ensure that information about advanced reservations is not accidentally lost if parts of the internetwork crash. In other words, advance reservations will require considerable additional supporting baggage that it would probably be better to keep out of the average flow spec.

Deciding whether a flow spec should contain information about how long the flow is expected to run is a harder decision to make. Clearly if we anticipate that the internetwork will support advance reservations, it will be necessary for elements of the internetwork to predict their traffic load, so they can ensure that advance reservations are not compromised by new flow requests. However, there is a school of thought that believes that estimating future load from current behavior of existing flows is more accurate than anything the flows may have declared in their flow specs. For this reason, I've left a duration field out of the flow spec.

Examples

To illustrate how the flow spec values might be used, this section presents three example flow specs.

Telnet

For the first example, consider using the flow spec to request service for an existing application: Telnet. Telnet is a virtual

terminal protocol, and one can think of it as stringing a virtual wire across the network between the user's terminal and a remote host.

Telnet has proved a very successful application without a need to reserve bandwidth: the amount of data sent over any Telnet connection tends to be quite small. However, Telnet users are often quite sensitive to delay, because delay can affect the time it takes to echo characters. This suggests that a Telnet connection might benefit from asking the internetwork to avoid long delay paths. It could so so using the following flow spec (for both directions):

```
Version=1
MTU=80 [40 bytes of overhead + 40 bytes user data]
Token Bucket Rate=0/0/0 [don't want a guarantee]
Token Bucket Size=0/0/0
Maximum Transmission Rate=0/0/0
Maximum Delay Noticed=1/1 [constant = delay sensitive]
Maximum Delay Variation=0/0/0 [not a concern]
Loss Sensitivity=1/0 [don't worry about loss]
Burst Loss Sensitivity=1/0
Loss Interval=1/0
Quality of Guarantee=1/0 [just asking]
```

It is worth noting that Telnet's flow spec is likely to be the same for all instantiations of a Telnet connection. As a result, there may be some optimizations possible (such as just tagging Telnet packets as being subject to the well-known Telnet flow spec).

A Voice Flow

Now consider transmitting voice over the Internet. Currently, good quality voice can be delivered at rates of 32Kbit/s or 16Kbit/s. Assuming the rate is 32Kbit/s and voice samples are 16 bit samples packaged into UDP datagrams (for a data rate of about 60 Kbyte/s), a flow spec might be:

```
Version=1
MTU=30 [2 byte sample in UDP datagram]
Token Bucket Rate=0/10/59 [60.4 Kbytes/s]
Token Bucket Size=0/0/30 [save enough to send immediately
                          after pauses]
Maximum Transmission Rate=0/10/59 [peak same as mean]
Maximum Delay Noticed=0/10/100 [100 ms]
Maximum Delay Variation=0/10/10 [keep variation low]
Loss Sensitivity=1/1 [loss sensitive]
```

```
Burst Loss Sensitivity=0/0/5 [keep bursts small]
Loss Interval=1/0
Quality of Guarantee=1/201 [predicted service and I'll accept
                            worse]
```

A Variable Bit-Rate Video Flow

Variable bit-rate video transmissions vary the rate at which they send data according to the amount of the video image that has changed between frames. In this example, we consider a one-way broadcast of a picture. If we assume 30 frames a second and that a full frame is about 1 megabit of data, and that on average about 10% of the frame changes, but in the worst case the entire frame changes, the flow spec might be:

```
Version=1
MTU=4096 [big so we can put lots of bits in each packet]
Token Bucket Rate=0/20/1 [8 Mbits/s]
Token Bucket Size=0/17/2 [2 Mbits/s]
Maximum Transmission Rate=0/20/30 [30 Mbits/s]
Maximum Delay Noticed=1/1 [somewhat delay sensitive]
Maximum Delay Variation=0/10/1 [no more than one second of
                                buffering]
Loss Sensitivity=0/0/1 [worst case, one loss per frame]
Burst Loss Sensitivity=0/0/1 [no burst errors please]
Loss Interval=0/0/33 [one frame in MTU sized packets]
Quality of Guarantee=1/300 [guaranteed service only]
```

The token bucket is sized to be two frames of data, and the bucket rate will fill the bucket every 250 ms. The expectation is that full scene changes will be rare and that a fast rate with a large bucket size should accommodate even a series of scene changes.

Disclaimer

In all cases, these examples are simply to sketch the use of the flow spec. The author makes no claims that the actual values used are the correct ones for a particular application.

Security Considerations

Security considerations definitely exist. For example, one might assume that users are charged for guaranteed flows. In that case, some mechanism must exist to ensure that a flow request (including flow spec) is authenticated. However I believe that such issues have to be dealt with as part of designing a negotiation protocol, and are not part of designing the flow spec data structure.

Acknowledgements

I'd like to acknowledge the tremendous assistance of Steve Deering, Scott Shenker and Lixia Zhang of XEROX PARC in writing this RFC. Much of this flow spec was sketched out in two long meetings with them at PARC. Others who have offered notable advice and comments include Isidro Castineyra, Deborah Estrin, and members of the End-to-End Research Group chaired by Bob Braden. All ideas that prove misbegotten are the sole responsibility of the author. This work was funded under DARPA Contract No. MDA903-91-D-0019. The views expressed in this document are not necessarily those of the Defense Advanced Research Projects Agency.

References

1. Parekh, A., "A Generalized Processor Sharing Approach to Flow Control in Integrated Services Networks", MIT Laboratory for Information and Decision Systems, Report No. LIDS-TH-2089.

2. Clark, D., Shenker, S., and L. Zhang, "Supporting Real-Time Applications in an Integrated Services Packet Network: Architecture and Mechanism", Proceedings of ACM SIGCOMM '92, August 1992.

Author's Address

Craig Partridge
BBN
824 Kipling St
Palo Alto, CA 94301

Phone: 415-325-4541

EMail: craig@aland.bbn.com