Internet Multicast Tomorrow

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June 10, 2002

1 Introduction

This article is part of a pair, the first of which looked at the state of play in IP multicast routing (Ole: IPJ Citation?). In this article, we look at the broader problems and future activities with multicast. We divide the areas into routing, addressing, transport, security, operations, and research.

There has been quite a bit of debate about the nature of compelling applications for multicast recently[44]. It is certainly the case that we do not completely understand the "market" for multicast - this is at least in part because multicast does not yet provide a complete set of functions for all the applications and services we might imagine. This is a typical chicken and egg situation, though: To put an extreme version of the argument, the application writers don't see any multicast deployed; the ISPs don't see any multicast applications; the router vendors don't see any multicast service demand from ISPs.[1].

As we discussed in the part I of this article, this vicious circle has been broken by streaming applications for audio and video from the classical content providers in the entertainment and news industries. However we are still seeing some teething problems. However, we are also seeing broader interest and development.

The next section presents recent work on routing and addressing. After that we look at transport. Subsequently, we discuss security. Then we look at operations and management. Finally, we take a glance at some of the research ideas that are out there.

2 Routing and Addressing

The single biggest step recently in multicast routing and and addressing has been the recognition that the demand for large scale multicast is largely for one-to-many, or single source. Combined with the ability to select sources at the receiver (as a means to prevent denial-of-service attacks) in IGMPv3[42], this has made a significant improvement to ISPs willingness to deploy the service.

[1] The same problem afflicts IPv6, Integrated and possible Differentiated Services and mobile IP, of course

2.1 Source Specific, and Single Source Multicast

The origins of the idea were thesis work at Stanford by Hugh Holbrook on Express multicast [43]. This is a specialised multicast architecture for one-to-many multicast groups. In this way, Express is a subset of the current multicast model in that it allows only a single sender to a multicast group. The advantages of Express are that certain aspects of multicast routing and addressing are easier solved by ignoring the many-to-many case. Many feel that the most likely large scale applications of multicast are one-to-many which is why Express becoming popular as a short-term solution.

Express addresses are *channels* which are 64-bit addresses (i.e. source address plus group address). Express sources transmit to a channel and advertise that channel. Receivers learn about these channels through advertisements or through a other means (i.e. URL) and initiate a Express join. Routers propagate these joins directly towards the source building a source rooted multicast forwarding tree.

There are two primary benefits of the Express model. The first is that Express simplifies the complexity of multicast routing. The second is that Express simplifies the assignment of multicast addresses for IPv4. Because Express channels are 64-bits, a source can select any lower 32-bits (any group address) for its channel and not collide with another.

In order to implement Express with IPv4 multicast protocols, a special range of multicast addresses was defined. 232/8 has been allocated by IANA for single-source multicast experimentation. In this range, an address only has meaning when "coupled" with a source address. Another way to explain it is that this address range is reserved for the lower 32-bit of Express addresses. With this scheme, Express does not require any modification to multicast data packets.

As for the protocols which implement Express, it can be implemented with two protocols which have already been developed: IGMPv3 [42] and PIM-SM. IGMPv3 extends IGMP to allow source specific joins to a multicast address. This capability can be used to carry 64-bit (S,G) joins to a router. Once a router receives the IGMPv3 join, it must be able to build the source specific tree with a multicast routing protocol. PIM-SM, widely deployed in service provider networks, already possesses this capability. The combination of IGMPv3 and PIM-SM allow Express to be implemented without creating more protocols; this is one of the most powerful benefits of the Express model.

2.2 Interdomain Multicast

At this point in time, there are four fairly widely deployed multicast routing protocols: PIM-DM, PIM-SM/SSM, MOSPF, and DVMRP. Because of the different properties of these protocols, there are many difficulties in connecting heterogenous routing domains together. [38] In general, most problems arise when connecting explicit join type protocols with flood-and-prune protocols. With service providers rolling out multicast using PIM-SM, connecting DVMRP and PIM-DM flood-and-prune is becoming common.

In order to connect two multicast routing domains, a multicast border router (MBR) needs to exist between the two domains. This router must implement a shared forwarding cache architecture [39]. In this model, each multicast routing protocol running on a

MBR submits its forwarding cache entries to a shared cache. This cache is the "bridge" between the trees in the different domains.

In order that the appropriate trees are created in each domain (on either side of a MBR), signaling must exist to bring sources from one domain to receivers in the other domain. This is part of the complication in connecting flood-and-prune protocol domains to explicit join protocol domains. In an explicit join protocol such as PIM-SM, joins are sent by edge routers to either a source or an RP when a host joins. A flood-and-prune protocol works quite differently, in a sense assuming that packets are desired; trees are pruned when edge routers receive new source packet but have no local listeners.

The signaling aspect of joining two domains can be accomplished with a variety of means. There are many options, but two stand out as providing the best methods of connecting domains. The first is to use Domain Wide Reports [36] in flood-and-prune domains. DWRs are similar to IGMP reports except are sent on a domain wide basis. When a border router receives a DWR report, it can join a group on behalf of an entire domain. The second solution is to use the Multicast Source Discovery Protocol (MSDP) [37]. MSDP is currently used to send source lists between PIM-SM domains. It can also be used to connect domains by having the MBR also participate in MSDP. Sources can then be learned from an explicit join protocol domain; the MBR can then join the sources and flood them into attached flood-and-prune protocols domains.

2.3 Address Allocation

The schemes to provide dynamic distributed address allocation have not been a great success to date. But with many multicast services being either limited to a single domain, or single source, the pressure is off. Instead, source specific addresses are unique in any case. For many-to-many multicast (sometimes known as "ISM" or Internet Standard Multicast"), the problem has also been alleviated by the use of GLOP[61], which allocates sections of the address space by mapping AS numbers of a provider into class D prefixes. This is potentially inefficient, but solves the contention, collision, revocation or resolution problem that MASC/MALLOC[60] attempt to do in a distributed dynamic manner.

In the longer term this address allocation, as well as scalable solutions to many-to-many multicast in the local domain and interdomain await further development on bi-directional trees ("Bi-dir PIM" and BGMP) which we discuss next. It is likely that these will need IPv6 to scale to serious usage.

2.4 Bidirectional PIM-SM

The PIM-SM multicast routing protocol builds both source and shared trees for the distribution of multicast packets. PIM-SM shared trees are rooted at special routers called Rendezvous Points (RP) and are unidirectional in nature. Shared tree traffic always flows from the RP down to the leaf routers. In some types of multicast applications, namely many-to-many type applications, a unidirectional tree may be inefficient.

Other multicast protocols such as CBT and BGMP provide bidirectional shared trees. Bidirectional[40] trees do not have these inefficiencies in many-to-many appli-

cations. In a bidirectional tree, traffic from a source is forwarded directly onto the shared tree at the closest point; the traffic is then forwarded both “up” and “down” the tree to all receivers. This is in contrast to a unidirectional tree when the source packets are sent first to the RP (or root) and then down the tree. Recently, two proposals have been submitted [40] which add bidirectional tree capabilities to PIM-SM.

2.5 BGMP

The Border Gateway Multicast Protocol (BGMP) [33] is a new inter-domain multicast routing protocol which addresses many of the scaling problems of earlier protocols. BGMP attempts to bring together many of the ideas of previous protocols and adds features which make it more service provider friendly. BGMP is designed with being a unified inter-domain multicast protocol much in the same way that BGP is used for unicast routing.

BGMP is designed as an inter-domain protocol in that it adopts particular design features of BGP familiar to providers. Two of these features are that it uses TCP connections for the transfer of routing information and has a state machine (with error notifications) similar to BGP.

In order to accommodate different applications and backwards compatibility, BGMP can build three types of multicast trees, both unidirectional source and shared trees and bidirectional shared trees. Unidirectional trees are useful for single source applications and for backwards compatibility with other multicast routing protocols. Shared trees are useful for many-to-many applications (e.g. multi-player gaming, video-conferencing) and allow multicast forwarding state to scale for these types of applications.

One of the unique properties of BGMP is that its shared trees are rooted at an autonomous system (AS) which is associated with the multicast group address of the tree. Having the root of the tree at the AS which is associated with the address is logical because there are likely members in that domain. Rooting the trees at an AS level also provides stability and inherent fault tolerance.

BGMP requires a way to discover which AS’s “own” which multicast addresses; this can be accomplished through the use of the MASC protocol or through globally assignable multicast addresses (e.g. IPv6 multicast). The MASC protocol allocates temporary assignments from the IPv4 group-D address space; it then distributes these assignments into MBGP so that BGMP will know which AS is associated with which group and therefore where to send join messages.

If globally assignable addresses are available, then BGMP can use any static address architecture for obtaining an AS from a multicast group address. The combination of BGMP and a large multicast address space (e.g. IPv6 address space) provide the best scaling for all types of multicast applications.

3 Transport and Congestion Control: Calling down traffic on a site

Multicast is a multiplier. It gives leverage to senders, but without their knowledge. Multicast (and its application level cousin, the CU-SeeMe reflector)[2] can "attract" more traffic to a site than it can cope with on its Internet access link. A user can do this by inadvertently joining a group for which there is a high-bandwidth sender, and then "going for a cup of tea". This problem will be averted through access control, or through mechanisms such as charging[58] which may result from the deployment of real time traffic support.

The problem is seen as critical by ISPs who have a shared bottleneck in their access technology - this is the case for cable-modem and in some cases for ADSL where a large number of fast lines converge on a slower interface to the backbone. Here, a single user may attract more traffic than this link, without seeing a problem that they cause for other (unicast or other multicast lower capacity separate sessions using the same shared bottleneck). The use of IGMPv3 with authenticated join and configuration management would appear to be a possible solution to these woes. Alternatively, the use of TCP friendly multicast congestion control (as envisaged for reliable multicast, but also as emerging in some realtime RTP[4] applications), would also solve this problem.

3.1 Congestion Control

One of the critical areas to clarify is the role of congestion control in multicast transport protocols[1]. From an early stage, it was established that coexistence with TCP was a critical design goal for protocols that would operate in the wider Internet. Thus systems such as TFMCC[8], PGMcc[53] and receiver driven congestion control[54] all extend the classic work by Jain[15] and Van Jacobson[17] and subsequent evolution[16] on TCP congestion avoidance and control.

Recently, this line of thinking has even been extended back into the unicast world in the application of such control schemes to UDP like flows in the work on the Datagram Congestion Control Protocol[62], suitable for adaptive multimedia flows on RTP for example.

3.2 Reliable Multicast

There is a clear requirement for some sort of analog to TCP, for multicast applications that need a level of reliability. The IRTF's RMRG group [3] has developed a number of prototypical solutions to the problem, which turns out to be quite a large design space (not "one size fits all").

The IETF RMT WG has now been chartered to develop single source reliable multicast transport solutions that meet the current Internet constraints.[1] That group has

[2] CU-SeeMe is a popular MAC and PC based Internet video conferencing package that currently does not directly use IP multicast.

[3] the RMRG site is at `http://ale.east.isi.edu/rm/`

developed a building block approach[12], which is based partly on abstracting components from existing work such as RMTP II[18], RLC[7], MFTP[28], PGM[41] and many other protocols.

Some applications of RMT products are likely to be infrastructural rather than of direct use to the ISPs customers - for example, distributing software to mirror sites seems to be one popular compelling use.

However, reliable multicast is sometimes regarded as something of an oxymoron.

When people talk about "Reliable Multicast" they usually mean a single protocol at a single 'layer' of a protocol stack, typically the transport layer (although we've seen people propose it in the network and even link (ATM!) layers too), that can act as any layered protocol can - to provide common functionality for applications (higher layers) that need it.

So what's wrong with that? Well, possibly 3 things (or more):

Fate sharing Fate sharing in unicast applications means that so long as there is a path that IP can find between two applications, then TCP can hang on to the connection as long as they like. However, if either party fails, the connection certainly fails.

Fate sharing between multicast end points is a more subtle idea. Should 'reliability' extend to supporting the connection for k recipients failing? Clearly this will be application specific (just as timing out on not getting liveliness out of a unicast connection is for TCP - we must permit per recipient Timeouts/Failures).

Performance When a talks to b, the performance is limited by 1 path. Whatever can be done to improve the throughput (or delay bound) is done by IP (e.g. load sharing the traffic over multiple paths). When a talks to b,c,d,e,f, should the throughput or delay be that sustainable by the slowest or average?

Semantics As well as performance and failure modes, N-way reliable protocols can have different service models. We could support reliable one-to-n reliable n-to-one and reliable n-to-m.

Applications such as software distribution are cited as classic one-to-n requirements. Telemetry is given as a n-to-1 reliable protocol. Shared whiteboards are cited as examples of n-to-m.

Now the interesting thing is to look at the reliability functions needed in these. 1-to-n and n-to-1 are effectively *simplex* bulk transfer applications. In other words, the service is one where reliability can be dealt with by "rounding up" the missing bits at the end of the transfer. Since this need not be especially timely, there is no need for this to be other than end to end, and application based. [4]

On the other hand n-m processes such as whiteboards need timely recovery from outages. The implication is that the "service" is best done somewhat like the effect of having

$$n * (m - 1)/2$$

[4] Yes, we know telemetry could be real timeish....but we are trying to illustrate major differences clearly for now.

	Recovery	*Sequency*	*Dalliance*
Network	not in our internet	ditto	int-serv
Transport	one-many	y	adaptive
Application	many-many	operation semantics	adaptive

Table 1: Reliable Multicast Semantics

TCP connections. If used in the WAN, the recovery may best be distributed, since requests for recovery will implode down the very links that are congested or error prone and caused the need for recovery.

Now there are different schemes for creating distributed recovery. If the application semantics are that operations (application data unit packets-worth) are sequenced in a way that the application can index them, then any member of a multicast session can efficiently help any other member to recover (examples of this include mark Handley's Network Text tool[6]. On the other hand, packet-based recovery can be done from data within the queues between network/transport and application, if they are kept at all members in much the same way as a sender in a unicast connection keeps a copy of all un-acknowledged data. The problem with this is that *because* its multicast, we don't have a positive acknowledgement system. Because of that, there is no way to inform *all* end points when they can safely discard the data in the 'retransmit' queue. Only the application really knows this!

Well, this is not to say that there isn't an obvious toolkit for reliable multicast support - it would certainly be good to have RTP style media timestamps (determined by the application, but filled in by the system). It would be good to have easy access to a timestamped based receive queue so applications could use this to do all the above. It might be neat to have virtual token ring, expanding ring search, token tree and other toolkits to support retransmit 'helper' selection....

We illustrate this in table 1 of where we might put functions to provide reliability (retransmit), sequencing and performance (adaptive playout say versus end to end, versus hop by hop delay constraint).

3.3 Router Assist for Reliable Multicast

As mentioned in previous sections, one of the difficulties in end-to-end multicast signaling is the "implosion" of signaling at a source from many receivers. This problem has been addressed in a number of ways, including the use of timers, the use of servers to aggregate signaling, and through the use of router-assisted mechanisms. We now discuss three protocols which make use of router assistance in order to better scale end-to-end multicast protocols.

The Pragmatic General Multicast (PGM) [41] protocol is a NAK based router-assisted reliable multicast protocol. PGM uses routers to aggregate receiver to source signals (e.g. the NAKs) as they flow toward the source. PGM's router support also includes a sub-casting ability whereby repairs will only flow down to receivers who have requested them.

Extending the ideas of router-assist in PGM is the Generic Multicast Transport Service (GMTS) [35]. GMTS provides *generic* fixed simple services for any end-to-end multicast transport protocol. These services include such features as signal aggregation with predicates and sophisticated sub-casting ability. GMTS was used as a basis for Generic Router Assist (GRA) [34], which is similar, IETF standards oriented, and a bit more streamlined.

4 Securing Multicast

Multicast security is more difficult than unicast security in several areas. The key exchange protocols used between unicast hosts do not scale to groups. Rekeying is required more often to maintain confidentiality as group membership changes. And the efficient authentication transforms used between two unicast hosts cannot protect traffic between mutually distrustful members of a group.

These problems are being worked on by the IETF msec and IRTF gsec working groups. Because of the wide range of application requirements in group communication, their work is based upon a building block approach similar to that of the Reliable Multicast Transport group. The blocks being developed are data security transforms, group key management and group security association, and group policy management [49]. An application may use different blocks together to create a protocol that meets its specific requirements.

4.1 Data Security Transforms

A data security transforms block provides confidentiality and authentication services for data being transported between group members. Confidentiality is reasonably easy to provide using standard encryption algorithms. Authentication is more difficult, as the algorithms used in unicast protocols such as IPSEC would not allow a group member to authenticate data as being from another specific group member. This is because the secret used to authenticate the traffic must be shared between all sending and receiving parties. Public-key signatures would solve this problem, but are an order of magnitude slower than symmetric authentication algorithms and hence especially unsuitable for real-time traffic and low-powered communications devices.

Instead, blocks such as TESLA [55] are being developed that trade off small amounts of functionality (such as immediate rather than slightly delayed authentication) to retain the efficiency benefits of symmetric algorithms. TESLA senders use a hash chain of keys $k_{n...1}$ to sign data, where $k_n = hash(k_{n-1})$. They release each key in the chain a short interval after the data the key has signed. As long as other group members received the data during that interval, they can be confident that the signature was made by the sender. If keys are lost during transmission, receivers can recompute any key earlier in the sequence simply by repeatedly applying the hash function used to any later key received. Finally, they can be sure that keys are coming from the sender because the first key in the sequence is digitally signed, while only the sender can know the later keys in the sequence (since by definition, a hash function must not be reversible).

4.2 Group Key Management and Group Security Association

To use data security transforms, group members need to possess the cryptographic keys necessary to encrypt/decrypt and sign/authenticate data. They also need to agree on parameters such as specific encryption algorithms. This building block allows this information to be shared between group members.

The Group Key Management architecture [47] provides a unified model for key management blocks. A central Group Controller/Key Server (GCKS) provides Traffic Encrypting Keys (TEKs) and/or Key Encrypting Keys (KEKs) to new group members after authenticating them with a unicast protocol. The GCKS may also delegate some of its functions to other entities, improving scaleability.

In groups with simple security requirements, this may be the only communication required between a group member and GCKS. But if group changes need to be cryptographically enforced, further TEKs, encrypted using a KEK, may be provided to members by multicast or a more scaleable protocol such as LKH [56] that does not require every rekey message to be sent to every group member. Alternatively non-interactive mechanisms such as hash trees may be used to update keys [48]. Finally, group members may explicitly de-register with the GCKS using a one or two-step message.

Three key management building blocks are being developed. The Group Domain of Interpretation (GDOI) builds on the Internet Security Association & Key Management Protocol [52] to allow the creation and management of security associations for IPSEC and other network or application layer protocols [46]. Multimedia Internet Keying (MIKEY) is targetted at real-time multimedia communications, particularly those using the Secure Real-time Transport Protocol, and can be tunnelled over the Session Initiation Protocol [45]. And a Group Secure Association Key Management Protocol (GSAKMP), along with a GSAKMP-Light profile, have also been developed [51].

4.3 Group Policy Management

The final building block defines policies such as which roles various entities may play in the group; who may hold group information such as cryptographic keys; the cryptographic algorithms used to protect group data; and proof that the creator of a given policy is authorised to do so. A group policy token is used to hold all of this information [50]. All or part of tokens can be made available to users in policy repositories or using other out-of-band mechanisms.

5 Operational Deployment of Multicast

As was mentioned above, multicast seems to be difficult to deploy. One problem is that it has only recently moved from the research community (and typically implemented using tunnels) into the service community (running native IP multicast routing). This means that debugging multicast sessions, applications and routing is a common activity. However, because of the dynamic nature of multicast addresses, and the anonymous nature of the multicast service model, debugging is somewhat harder than for the equivalent unicast case. Luckily, all current native multicast paths are at least computed

from underlying unicast ones, and it is possible to use tools such as `mtrace` and `mrm` to query the underlying router system to try to figure out where things are going on. Of course, the relevant MIBs need to be designed, but mere SNMP access to the variables defined in these may not be enough.

Many multicast sessions are global in scope, and not surprisingly, someone, somewhere, sometime in the session will have a problem. In a way, you only have to look at multicast as a way of sampling large pieces of the Internet at one go to see why its hard to figure. In fact, a research project called MINC[9][57] is using that very observation to build tools of more general use.

5.1 MRM

One recent tool which has been developed to facilitate multicast monitoring and debugging is the Multicast Reachability Monitor (MRM) [32]. MRM consists of two parts; a MRM management station configures test senders and test receivers in multicast networks. A multicast test sender or test receiver is any server or router which supports the MRM protocol and can source or sink multicast traffic. MRM provides the ability to dynamically test particular multicast scenarios; this capability can be used for fault isolation and general monitoring of sessions.

MRM is typically used to configure MRM capable routers as test senders and test receivers from a management station. Routers configured as test senders send multicast packets periodically to a configured multicast group at a configured rate. Routers configured as test receivers monitor traffic to a group and keep statistics which can be reported back via RTCP packets. Test receivers can be configured to send RTCP reports when a given condition has been reached or when polled by a management station. Although the MRM protocol is simple itself, it provides powerful capabilities which can be used by future multicast debugging applications.

6 Research Ideas in Multicast Routing and Addressing

The seeming complexity exhibited by the full panoply of multicast protocols has led some people to develop doubts as to the eventual deployment of multicast. It is far too early to say whether these doubts are well founded. The slow pace of deployment is not just a symptom of this complexity, but also of the underlying complexity of handling growth and evolution of *any* type in such a large system as the Global Internet.

Having said that, it is worth mentioning four of the approaches that have been discussed in the Internet community recently:

AIM Addressable Internet Multicast, by Brian Levine, et al attempts to provide explicit addressing of the multicast tree. The routers run a tree-walking algorithm to label all the branch points uniquely, and then make these labels available to end systems. This allows a number of interesting services or refinement of multicast services to be built. Of some particular interest would be the ability this service gives to end systems to do *sub-casting*, which would be useful for some classes of reliable transport protocols.

Express Explicitly Requested Single-Source , by Hugh Holbrook et al, is aimed at optimising multicast for a single source. The proposal includes additional features such as authentication and counting of receivers, which could be added to many other multicast protocols usefully. It is motivated by a perceived requirement from some ISPs for these additional features. Express makes use of an extended address (channel+group) to provide routing without global agreement on address assignment. A possible source of problem for AIM is the potential for unbounded growth in the size of identifiers for labelling subtree branch points.

RAMA Root Addressed Multicast Architecture, by Radia Perlman et al, is in some senses both a generalisation of Express type addressing, but also requires bi-directional trees (CBT-like, rather than current PIM-SM, although work on bi-directional PIM is underway too). The goal is to offer a single routing protocol for both intra-domain and inter-domain. In fact, RAMA can be implemented by combining the address extensions proposed for Express, and two-level bi-directional PIM as an implementation of BGMP. RAMA and Express (and bi-directional PIM) require a mechanism for carrying additional information in multicast IP data packets. There are two critical problems for carrying this identifier which are hard to solve in general: firstly, it takes new space in the IP packet, and this has to be accessed by both hosts and routers - that represents a deployment problem; secondly, in the general case, the extra field must be examined on the "fast path", in routers which have such a concept, and this takes valuable processing resources which may have to be taken away from some other forwarding task.

CM Connectionless Multicast by Dirk Ooms, et al, is a proposal for small, very sparse groups to be implemented by carrying lists of IP unicast addresses in packets. The scheme is not simply a form of loose source routing, as it would make use of packet replication at appropriate branch points in the network. It may be well suited to IP telephony applications where a user starts with a unicast call, but then adds a third or fourth participant.

DCM EPFL work on Distributed Core Multicast. This aims to address very large numbers of very small groups with mobile users, typically characteristics of mobile IP telephony users making conference/group calls.

ABT MIT have some work on the use of wide area "anycast" addresses for the core/RP. This results in a potential improvement in the availability of trees (and subtrees) for multicast delivery in the even of router or link outage. More importantly it may be possible for a multicast group to survive network partitions (or lack of core reachability) which would make this an invaluable improvement to the service. It depends on the scalability of the wide area anycast solution, which the MIT work shows is at least viable, and certainly worth more attention.

YAM Yet Another Multicast (YAM) routing protocol[30] was devised by Ken Carlberg of SAIC to address the possiblilty of forming different multicast trees based on some QoS metric - the idea is that IGMP is modified to provide a "one-to-many" join, and a receiver sends this with required performance parameters.

Routers receiving the request over links that can provide this service respond. The receiver (sender of the one-to-many IGMP) selects the one to then commit the join to.

QoSMiC QoSMiC is a development from YAM by Faloutsos[29] at Toronto, and slightly modifies the tree building exercise.

MPLS When multicast and MPLS are mentioned together, there is both confusion and surprise. MPLS can be used with multicast in two very different ways. The first method is by building multicast trees over MPLS traffic engineered paths. Some multicast routing protocols already make use of unicast forwarding information for the construction of multicast trees. Using multicast traffic engineered paths is simply an extension of this concept except with one catch. Some multicast routing protocols use Reverse Path Forwarding (RPF) checks on incoming packets to prevent looping; this is accomplished by checking to see if the incoming interface is the "closest" to the source. With MPLS traffic engineering, RPF checks are difficult. A solution has not been presented at this time which addresses this problem. The second method for using multicast with MPLS is through the use of point-to-multipoint virtual circuits much in the same way as ATM point-to-multipoint VC's. In cases where receivers are statically configured to a multicast address or multicast traffic is always to be delivered to a destination then these are useful. Mapping dynamic memberships into a multipoint circuit has proven difficult for example with ATM. There are currently several internet-drafts which propose various solutions for MPLS and multicast [31].

Application Layer Multicast Several groups have been working on end-system only multicast schemes, probably most notably CMU[59].

7 Summary and Conclusions

In this article, we have taken a look at some of the newer ideas in the research and development community in the area of multicast. There is still a lot to be done to close the loop between network services, transport and applications, but we can see that there is plenty going on that will eventually achieve this goal.

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