Socket Intents: OS Support for Using Multiple Access Networks and its Benefits for Web Browsing

Philipp S. Tiesel, Theresa Enghardt, Mirko Palmer, Anja Feldmann

Abstract-In today's Internet, mobile devices are connected to multiple access networks, e.g., WiFi/DSL and LTE. To take advantage of the networks' diverse paths characteristics (delay, bandwidth, and reliability) and aggregate bandwidth, we need smart strategies for choosing which interface(s) to use for what traffic. In this paper, we present an approach how to tackle this challenge as part of the Operating System (OS): With the concept of Socket Intents, applications can express what they know about their communication pattern and their preferences. Using our Socket Intents Prototype and our modified BSD Socket Interface, this information is used to choose the most appropriate path or path combination on a per message or per connection basis. We evaluate our system based on the use case of Web browsing: Using our prototype and a client-side proxy, we show the feasibility and benefits of our design. Using a flow-based simulator and a full factorial experimental design, we study a broad range of access network combinations (based on typical DSL and LTE scenarios) and real workloads (Alexa Top 100 and Top 1000 Web Sites). Our policies achieve performance benefits in more than 50% of the cases and speedups of more than factor two in 20% of the cases without adding overhead in the other cases.

I. INTRODUCTION

TODAY, mobile devices can usually access the Internet over more than one access network. For example, mobile phones often have the choice between WiFi and cellular networks. By using the paths provided by these networks at the same time, it is possible to aggregate their bandwidth or switch between them to increase overall network availability. Moreover, applications can take advantage of the different characteristics of these paths, e.g., delay, bandwidth, and expected availability, by using the most suitable path(s) according to the application's demands.

To illustrate why these characteristics matter, we take a look at two use-cases: If a user fetches news headlines or stock market quotes, quickly loading small objects ensures responsiveness; so the user prefers a path with low latency. If a user streams a TV series over HTTP, high throughput is most important to provide high video quality. Therefore, if the available paths vary in bandwidth and latency, both usecases benefit from assigning their communications to the path with the most suitable characteristics. Similarly, there is often a choice between multiple destinations, e.g., CDN nodes hosting the same content.

Despite the possible benefits, the available diversity is usually not exploited. While the application knows its demands, selecting the path and destination within the application is impractical most of the time. This stems from the fact that information required for an appropriate selection is often not available, such as detailed characteristics of the available paths or information about cross-traffic from other applications. In addition, implementing path selection strategies is a rather complex task which touches resource management and, thus, belongs to the domain of the operating system (OS). At the OS level, all necessary information including path characteristics and traffic information are available. But when a socket gets connected, the OS is not able to distinguish between different application demands. Consequently, there is no way for the OS to know what to optimize for.

So far, application-aware OS support for multiple paths, the focus of this paper, is still very limited. Most related research [1]–[3] or commercial systems [4], [5] focus on WiFi off- or onloading at the network layer or application layer for well-defined use cases. Multi-Path aware transport protocols, including SCTP [6] and Multipath TCP (MPTCP) [7], [8] allow to aggregate bandwidth across multiple access networks and provide fallback in case of network failures, but are agnostic to applications' needs.

To this end, we introduced the concept of *Socket Intents* [9]. Socket Intents allow applications to share their knowledge about their communication pattern and express performance preferences in a generic and portable way. Thus, an application developer can inform the OS about what the intent of the communication is and what she knows about the communication:

- **Preferences** whether to optimize for bandwidth, latency, or cost
- Characteristics expected packet rates, bitrates or size of the content to be sent or received.
- **Expectations** towards path availability or packet loss
- **Resiliences** whether the application can gracefully handle certain error cases

None of these are hard requirements, e.g., transport protocol guarantees or QoS style reservation. However, they are crucial for the OS to do path and destination selection on behalf of the application.

In this paper, we demonstrate *how Socket Intents can be implemented* on top of vanilla BSD Sockets by designing a system that enables automated path and destination selection within the OS and evaluate the *impact of Socket Intents on Web performance*.

Our prototype implementation extends the BSD Socket API to support path and destination selection for two communication granularities: streams and messages. It demonstrates the feasibility of automated path and destination selection within the OS but also reveals major limitations of the BSD Socket API. The main contributions of this paper are:

- We extend our original prototype [9] to support message granularity communication units, e.g., HTTP requests. Guided by the application needs provided by Socket Intents and path characteristics, we realize connection caching and implicit connection pools in the OS. In addition, we enable our prototype to take control over MPTCP.
- We introduce the *Earliest Arrival Time* (\mathcal{EAF}) policy as informed path selection strategy for Web objects. Based on the information provided by the *Size to be Received Intent*, \mathcal{EAF} assigns each request to the path it is predicted to complete on first.
- We demonstrate applicability of our prototype by implementing a client side HTTP proxy. With only 20 lines of additional code, the proxy takes advantage of connection caching, Socket Intents and the *EAF* policy.
- We evaluate the benefits of using Socket Intents with the *EAF* policy in two ways: A small testbed study using our proxy and an extensive simulator study using a custom flow-based simulator. Our simulation uses a full factorial experimental design and covers the Alexa Top 100 and Top 1000 Web sites over a wide range of network characteristics resembling typical residential broadband and cellular network characteristics.

II. SOCKET INTENTS CONCEPT

To perform path and destination selection within the OS, the OS needs to know what to optimize for - the application demands. Therefore, we introduced the concept of Socket Intents [9]. Socket Intents allow applications to share their knowledge about their communication patterns and express performance preferences in a generic and portable way. Intents are hints for the OS, pieces of information, that allow an application programmer to express what they know about the application's needs or intentions for each communication unit. They indicate what the application wants to achieve, knows, or assumes. In contrast to transport features or QoS-style reservations, they are not requirements but only considered in a best-effort manner, e.g., as input to path and destination selection heuristics within the OS. Possible intents, as shown in Table I, include Traffic Category, Size to be Sent/Received, Timeliness, Duration, or Resilience of connectivity.

Applications have an incentive to specify their intents as accurately as possible to take advantage of the most suitable resources. We expect applications to selfishly specify their preferences. Since the OS knows about the available network resources and the intents of multiple applications, it can balance the different requirements and penalize misbehaving applications.

Socket Intents are independent of the actual Socket API and can be applied to message granularity communications, e.g., UDP messages or HTTP requests, as well as stream granularity communications, e.g., TCP connections. The information provided by the application is structured as key-value-pairs. The key is a simple string representing the type of a Socket Intent. Values can be represented as an *enum*, *int*, *float*, *string*,

TABLE I Socket Intents Types

Intent Type	Data Type	Applicable Message	Granularity Stream
Traffic Category	Enum		\checkmark
Size to be Sent	Int (bytes)	\checkmark	\checkmark
Size to be Received	Int (bytes)	\checkmark	\checkmark
Duration	Int (msec)		\checkmark
Bitrate Sent	Int (bytes/sec)		\checkmark
Bitrate Received	Int (bytes/sec)		\checkmark
Burstiness	Enum		\checkmark
Timeliness	Enum	\checkmark	\checkmark
Disruption Resilience	Enum	\checkmark	\checkmark
Cost Preferences	Enum	\checkmark	\checkmark

or a sequence of the aforementioned data types. Table I gives an overview of Socket Intent types as specified in our recent IETF draft [10]. Despite the variety of Intents we define in this section, the remainder of this paper focuses on how to realize Socket Intents as an extension to the BSD Socket API and the benefits of using the *Size to be Received Intent*.

III. CHALLENGES IMPOSED BY BSD SOCKETS

In Unix-like OSes, BSD Sockets are the standard interface between applications and the network stack. Typically, applications that want to connect to a server first resolve the server's hostname using getaddrinfo(), then create a socket file descriptor using socket() and call connect() to establish the connection. To each of these calls information obtained from getaddrinfo() is passed.

With Vanilla BSD Sockets, taking advantage of multiple paths or choosing among several destinations is complicated. One reason is that the BSD Socket API designers considered multi-homed hosts a corner case. The bind() socket call allows applications to choose the source address of an outgoing communication¹. If the system is configured with a routing policy to send traffic with a specific source address over an associated paths, application can set the source address to implicitly choose the outgoing interface and next-hop and, therefore, large portions of the path.

Besides this hack, Vanilla BSD Sockets do not offer support for multiple access networks: Applications that want to use multiple interfaces usually have to have their own heuristics for selecting paths. Choosing among paths is difficult as the necessary information is often difficult to gather and may require special privileges. Moreover, it differs greatly by Unix flavor.

Another complication occurs when selection a destination: When resolving the hostname to obtain a destination address, applications need to ensure not to mix results for the same hostname resolved via a different interface. For example, CDNs and major Web sites often rely on DNS-based server selection and load balancing. These mechanisms are most useful if the DNS query is sent via the same interface as the actual traffic. If the application sends the traffic over another interface, the chosen server may be suboptimal, which can lead to significant performance degradations. Yet, the resolver

¹ Otherwise, the OS uses the IP address of the interface via which it routes to the given destination.

library of the vanilla BSD Socket API does not allow us to isolate results acquired via multiple paths. For a more detailed discussion, see [11].

Furthermore, the communication units used by vanilla BSD Socket API are implied by the transport protocol and must match the socket type passed to the socket () call. Thus, for stream-based communication protocols like TCP, the application can only choose a path and endpoint for the whole stream. But communication units of actual applications are often not aligned with the communication granularity of the transport protocol. For example, requests in HTTP ---the dominant protocol on the Internet [12], [13]- correspond to a message based communication performed over a stream transport. An HTTP based application can choose for each request to either open a new TCP connection or reuse an existing one. The former allows choosing among multiple interfaces using bind(). The latter saves 2RTTs for the TCP handshake, a few 100 KB for the TLS handshake (if applicable), and time spent in TCP slow-start. Therefore, the abstraction provided by the vanilla BSD Sockets does not assist the application in distributing traffic among multiple paths. Rather, it puts a huge burden on applications that want to do so.

In conclusion, these problem areas demonstrate that the vanilla BSD Socket API is not well suited to enable multiple access connectivity in an easy and portable way. In the next Sections Section IV and Section V, we describe how our Socket Intents Prototype overcomes these limitations and provides path- and destination selection as an extension to the BSD Socket API.

IV. SOCKET INTENTS PROTOTYPE DESIGN

We build a prototype implementation to demonstrate the feasibility of path selection and destination selection as features of the client's OS. Despite the limitations outlined in Section III, our Socket Intents Prototype extends BSD Sockets to use Socket Intents and moves the selection logic into the OS. In this section, we first discuss the overall design choices (Section IV-A) and prototype architecture (Section IV-B). Then, we describe the designs of the individual prototype components.

A. Design Objectives and Choices

The prototype is based on the following **design objectives** (shown in bold) and *design choices* (shown in italics).

As we want **insights about the deployability in today's OSes**, we decide to build our prototype as an *extension of the BSD Socket API*, which is the base of almost all networking APIs used today.

Since different applications have different requirements, the system should allow them to **specify their "intent" for a given communication unit** as hint for the OS what to optimize for. Since there is not always a universally "best" interface, the system enables **choosing an appropriate path** for each communication unit, i.e., for each message or connection, The system also allows **choosing a destination** for each communication unit from the alternatives provided by name resolution.

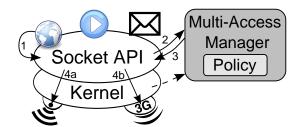


Fig. 1. OS Network Stack with Socket Intents.

To realize these objectives, we provide *three separate API variants* to address the trade-offs imposed by integration with the BSD Socket API.

To improve deployability, we chose to explicitly **not require** or suggest changes to the server or the network. As a consequence, our means of choosing a path are limited to *choosing the source address, the interface, and/or the first hop* for each communication unit.

To evaluate different selection strategies, which we call policies, we realize them as exchangeable modules. These entities decide which access network to use in a given situation based on the available information. To enable joint optimization across all applications that use the Socket Intents Prototype, these policies are hosted in a central place.

We want our prototype to support path and destination selection for the applications' native communication units, e.g., an HTTP request. In today's Internet, these are often not aligned with the communication units of the underlying transport, e.g., TCP streams. An HTTP-based application will typically try to reuse a TCP stream for multiple HTTP requests to reduce overhead and latency. This connection reuse logic has to be implemented by each application individually, although it is, typically, not application-specific. We decided to move this logic into the OS and integrate it with our path and destination selection by providing *implicit connection pools*. For each communication unit, our system needs to **map** communication units to the appropriate connection pool and decides whether to reuse an existing connection or to set up a new one. As we can only choose a path and destination for new connections, we choose to *tightly integrate* the connection reuse logic with the path and destination selection logic.

Finally, to split large communication units and being able to distribute them over an appropriate set of paths, we enable the use of MPTCP and control of MPTCP's path selection.

B. Prototype Architecture

Our Socket Intents Prototype consists of three components, see Fig. 1: Our extended socket API (white), a Multi Access Manager (gray) and a policy modules it hosts (light gray). In a typical use case, an application specifies its Intents through the API (1), then our socket library queries the policy within the Multi Access Manager (2). The policy decides which path(s) to use and communicates this decision back (3), and finally the socket library applies the decision by selecting a path by binding to an appropriate source address (4a/b).

C. APIs

As there is no easy way to integrate application-aware path and destination selection into the BSD Socket API, we provide three different API variants. While maintaining the UNIX file descriptor as abstraction of a connection, each of these variants takes different trade-offs to overcome the challenges described in Section III:

- The *augmented socket-calls* variant, see Section V-A, follows as close as possible to the call sequence of BSD Sockets. It links all socket calls by using an additional context parameter to hold the state needed for path selection beyond the regular parameters. It is meant as a baseline to explore which aspects of automated path and destination selection can be integrated into the vanilla BSD Socket API without changing the application logic (besides providing Socket Intents).
- The *augmented name resolution* variant performs automated path and destination selection as part of the name resolution, see Section V-B. This minimizes the changes to the BSD Socket API, but requires the application logic to change to use the results from path and destination selection.
- The *message granularity* variant, see Section V-C, adds support for access selection at message granularity, e.g., HTTP requests. It moves the connection setup into a single API call. Thus, it completely replaces the usual call sequence of BSD Sockets in order to enable automated connection caching along with implicit connection pooling. This variant is the most powerful one, and therefore used in our evaluation.

For all variants, name resolution is off-loaded to the Multi Access Manager and handled by the policy module to work around limitations of the standard resolver library. See Section III for an extensive discussion about the limitations BSD Sockets impose on implementing automated path and destination selection and their consequences for the implementation of the individual API variants.

D. Policy Design

Socket Intents Policies are entities that decide which access network to use for a given communication unit. They range from simple static configurations up to complex dynamic algorithms that try to take full advantage of the available information.

To enable informed decisions, these policies need to know about the intents of an application as well as interface parameters and statistics, including byte counters and transport protocol state. Within its decision logic, the policy needs to respect the optimization of external communication partners, i.e., it should only rely on DNS replies of the same interface (see Section III).

For the sake of simplicity, we chose not to support perapplication policies, but rely on the information provided by Socket Intents This also enables us to treat communication units of a single application with different communication needs appropriately.

As a first application-aware policy, we introduce the \mathcal{EAF} policy: This policy is based on the idea that downloading objects of different sizes can benefit from different path characteristics, as download time largely depends on the object's file size as well as the RTT and available bandwidth on the path. We use the Size to be Received Intent, which allows an application to hint for the expected size of a communication unit, e.g., allowing an HTTP client to hint about the size of an object to be transferred. Assuming that there are at least two access networks and they vary in RTT and bandwidth, our intuition is that if the communication unit is small, the policy should choose the interface with the shorter RTT. If the communication unit is large it should prefer the interface with the larger available bandwidth. Thus, each unit is scheduled on the interface with the earlier arrival time, resulting in a shorter overall completion time.

V. SOCKET INTENTS PROTOTYPE IMPLEMENTATION

In this Section, we present the implementation and technical challenges of the three components of our Socket Intents Prototype in more detail: the extended socket APIs, the Multi Access Manager, and the policy modules. The source code of all components consists of about 15k lines of C and is available under BSD License².

First, we will explain our extended socket API variants introduced in Section IV-C in more detail. All three variants are implemented within a common wrapper library for the BSD Socket API and are portable across Linux and MacOS. Then, we will dive into the implementation of the Multi Access Manager (Section V-E), and the policy (Section V-F).

A. Augmented Socket-Calls API

With this API variant, we try to stick as close as possible to the call sequence of vanilla BSD Sockets. Support for Socket Intents is provided by adding an INTENT socket option level and thus allowing to specify Socket Intents as socket options. The challenge of implementing this design is that there is no mechanism to pass state needed for path and destination selection between the vanilla BSD Socket API calls, that is not part of their explicit parameters³. To overcome this limitation, we it adds an additional parameter to all socket calls including getaddrinfo().

With this API variant, we can support informed path and destination selection with very few modifications to the application. However, due to our way to pass resolver and selection state, the selection process becomes in-transparent to the application and does not work with applications that want to perform mechanisms like Happy Eyeballs [14].

B. Augmented Name Resolution API

This API variant realizes path selection and endpoint selection within a modified variant of getaddrinfo(). For

²https://github.com/fg-inet/socket-intents/

³ A kernel based implementation could pass information between all socket calls except getaddrinfo() by extending backing struct for file descriptors.

TABLE II Message Granularity Socket Intents API

Call	parameters (excerpt)	in/out
socketrelease socketrelease	<pre>int *socket const char *host size_t hostlen const char *serv size_t servlen struct socketopt *sockopts int domain int type int proto int socket int socket</pre>	in,out in in in in in,out in in in in in in in

all other socket API calls, the application uses the vanilla BSD Socket API. Socket Intents, alongside with other relevant socket options, are passed directly to our modified getaddrinfo() as part of the hints parameter. To do so, we extended the addrinfo struct to include a list of socket options and the source address for the outgoing connection. We also provide a new socketopt struct to pass a list of socket options as part of our extended addrinfo struct. The name resolution implementation of getaddrinfo() is done by the Multi Access Manager, which makes all decisions and returns them in the result parameter as list of endpoints ordered by policy preference. Each endpoint is annotated with the source address the application should bind to and socket options that should be set on the socket. Applications use this information as parameter to the vanilla BSD Socket API call or other APIs. We provide helpers to set all socket options from the result data structure on a given socket.

With this API variant, applications can benefit from informed path- and destination selection provided by the Multi Access Manager while maintaining full control over the connection setup, e.g. to perform Happy Eyeballs [14]. This comes at the cost of having to modify the application to bind to a source address and pass socket options through our API.

C. Message Granularity API

This API variant offers path and destination selection for message granularity communication units, e.g. HTTP requests, using stream transports, e.g. TCP. As stated in Section IV-C, we decided to realize this together with implicit connection pooling to enable connection reuse. As our focus is on supporting simple request/response type protocols, e.g., HTTP/1.1, we presume sequential reuse of connections by the same application. We do not yet support multiple concurrent requests on the same TCP connection as in HTTP/2, since this would also require to implement explicit message extraction like [15] or [16] and does not provide us with insights for path- and destination selection.

To expose the functionality, we extended the BSD Socket API by adding three new calls: socketconnect() to get a new socket or reuse an existing one, socketrelease() to mark a socket as available for reuse, and socketclose() to close a socket (see Table II). Since it realizes the complete connection handling, it moves functionality needed by many applications into the OS.

When an application wants to send a request, it uses socketconnect() to asks our Socket Intents Prototype

for a socket for a specific host, service, and socket options (including Socket Intents) tuple. The in/out parameter for the socket file descriptor allows to explicitly request a new socket or reuse one of a set of sockets. The return value informs the application whether the socket is a new one or an existing one. This allows the application to decide if it needs to add any per-connection actions, e.g., if a new TLS handshake has to be started for a new connection. Future work will remove this complication by integrating TLS into the transport stack. Once the application is done, it can either release or close the file descriptor using the second or third new call. The former enables reuse, the latter does not.

The Socket Intents Prototype manages an implicit pool of active sockets per destination host/service pair. The implementation of socketconnect() checks whether there are currently unused open sockets to the same host and port. It then checks with the Multi Access Manager if any of these existing sockets satisfy the needs of the request according to the Socket Intents⁴. If not, the Multi Access Manager chooses an interface for a new socket to be created. The call then either returns the chosen existing socket or the newly opened one.

Using this API variant is very convenient for writing request/response style applications designed around it, but re-writing applications turned out complicated as it totally changes the way connections are handled in an application.

D. Lessons Learned for the APIs

Adding path and destination selection to BSD Sockets is hard; Its API calls are not designed to defer choices to a moment where all necessary information is available. Our API variants address this problem by choosing different trade-offs:

If limiting path and destination selection to the granularity BSD Sockets typically provide today (TCP connections), the Augmented Name Resolution API variant seems to be a good compromise. Still, it forces applications to change and implement a lot of connection management.

To do path and destination selection at message granularity, we have to make more concessions: We need to add connection caching and pooling to our API. In student projects using this API variant, we encountered a surprising behavior: sockets returned by socketconnect() returned failures on write. This happens if the remote side has already closed the connection while it is still in the pool for re-use. As vanilla BSD Sockets do not provide an explicit mechanism to notify an application about the closure of a socket, we mitigate this behavior by testing the connection before scheduling it for reuse. This API still uses file descriptors as socket abstraction, but otherwise largely diverges from the vanilla BSD Socket API and requires heavily modifying existing applications.

Besides that, the integration of BSD sockets into file I/O system provided us with many implementation challenges and required many shadow state-keeping and other hacks [17]. Retrospectively, it seems easier to move to a new API like the one we design in the IETF Taps Working Group [18].

 $^{^4}$ We assume that the policy takes connection setup time into account for this decision.

E. Multi Access Manager

The Multi Access Manager runs as a user space service that does not require special kernel support. It runs as a service available to all applications on the client and is implemented using *libevent*. Our API uses Unix domain socket to communicate with the Multi Access Manager. After start up, the Multi Access Manager creates a list of all local interfaces with their network prefixes assigned and loads the policy.

The Multi Access Manager does not keep any per request state, but does not prevent the policy to do so. A module implementing a policy consists of a set of callback functions which are triggered by our socket library, the Multi Access Manager, and DNS replies. It can use all functionality provided by *libevent* as well as the *evdns* resolver library that is preconfigured for each interface in the Multi Access Manager.

1) Estimating Path Characteristics: To make decisions, the policy uses various network statistics from the Multi Access Manager. Hereby, the Multi Access Manager periodically queries the OS for smoothed round-trip times (SRTTs) of all current TCP connections to calculate the median and minimum RTT over each available prefix. Also, based on the interface counters it computes the currently used network bandwidth for each interface. The query interval is configurable. Our empirical observations suggest that an interval of 100 ms works well. The Multi Access Manager also gathers additional passive measurements, such as bit error statistics and current channel utilization as reported by the WiFi access point (if available). More data about the current network performance can easily be integrated by adding code to the Multi Access Manager or the policy. All information is stored on a per interface basis within the Multi Access Manager.

2) Resolver Integration: The Multi Access Manager has to guarantee that DNS replies are kept separate on a per-interface basis and, therefore, should only be cached and used for communication on the same interface from which they were acquired. This separation is necessary to avoid interference with DNS-based server selection and load balancing as laid out in [11]. As DNS caching is essential for the performance of applications such as Web browsers, we consider this functionality an integral part of the Multi Access Manager and not of the application.

3) Controlling Multipath TCP with Socket Intents: We use MPTCP to split a TCP stream across multiple paths. This allows bandwidth aggregation for large transfers and thus complements the per-request scheduling. As a result, Socket Intents can choose appropriate interfaces for both small communication units—which the policy can distribute—as well as large ones—which MPTCP can handle. In addition, controlling MPTCP from the Socket Intents Policy avoids opening subflows on already crowded interfaces or on interfaces with a high RTT, which can lead to head-of-line blocking for small objects.

To enable the Socket Intents Policy to control the usage of MPTCP we added an additional path manager to the Linux MPTCP implementation. Our user-space Multi Access Manager uses Netlink [19] sockets to communicate with the kernel-space MPTCP path-manager. If a policy decides to use MPTCP it selects an interface for the initial subflow. If MPTCP is feasible the path-manager notifies the Multi Access Manager, so the policy can choose on which interfaces to add subflows. MPTCP can then distribute the TCP stream over all these interfaces. For more details on our implementation, we refer to [20].

F. Policy

The policy implements the logic for deciding which interfaces, and, thus, which source and destination address pair to use. The actual Socket Intents Policy is implemented as modules for the Multi Access Manager. It is shared by all applications of a host that use our socket interface, as their individual needs are communicated using Socket Intents and are not realized via individual policies. The policy picks a suitable interface for each communication unit. Then, if a set of open sockets is given, the policy tries to reuse one of them by selecting a socket which uses the chosen interface from the given set. If no set is given or if no suitable socket is found, the policy advises the application to open a new connection and suggests an IP address of the chosen interface as source. When the policy has decided which source and destination address pair to use, it instructs the Multi Access Manager to send this information back to our socket library. With MPTCP, the policy may have to keep track of the requests to aid, e.g., the setup of MPTCP subflows.

We implement the following polices in our prototype:

1) Single Interface Policy: This policy always chooses a particular, statically configured interface.

2) *Round Robin Policy:* This policy uses multiple interfaces on a round robin basis.

3) EAF Policy: The EAF policy uses the Size to be Received Intent to predict the completion time for each available interface. It then chooses the one where the communication unit will arrive first. For *EAF* the Socket Intents Prototype uses estimates of the minimum SRTT per prefix and the available bandwidth on the interface. The object size for the Size to be Received Intent is determined via a two-step download that is described in detail in Section VI-B. EAF estimates the available bandwidth by dividing the maximum observed bandwidth on an interface by the number of already scheduled objects on the same interface. We divide the file size by the estimated available bandwidth to approximate the download duration. We add one RTT if a connection can be reused and two RTTs if a new connection has to be established. Finally, the interface with the shortest predicted arrival time is chosen. We do not consider TLS handshakes.

VI. EVALUATION METHODOLOGY

To evaluate the benefits of Socket Intents for Web browsing, we use the following methodology

A. Challenges

When evaluating the Socket Intents Prototype to showing performance implications in a generic manner we face several challenges: First, the evaluations have to be conducted in a realistic environment, including realistic application behavior and network settings. Second, reproducibility of the gained results has to be assured. Third, we have to evaluate a large number of different application and network settings to gain meaningful results. We address these challenges as described in the following. As application scenario for the evaluation we select Web browsing. To achieve a high degree of realism, we use the Firefox Web browser. Due to the high complexity and optimization level of a browser we decided to implement a client side Socket Intent proxy instead of extending the Web browser. As network settings we decided to rely on typical smartphone settings, i.e., two disjoint paths between the client and the server.

Reproducibility of the results is achieved by excluding most uncontrollable influence factors for the chosen Web browsing scenario. We have to cope with problems like changing Web page content over time, content distribution over various Web pages, and varying backend performance. Accordingly, we crawl different Web pages and mirror them in a local testbed. Thus, most uncontrollable influence factors except for varying execution times of the JavaScript are excluded. To eliminate the impact of JavaScript we also rely on synthetic workloads without JavaScript.

To evaluate a large number of different settings we conduct evaluations with around 1000 different Web pages and more than 300 different network configuration. Given the large amount of different settings, we opt for evaluating Socket Intents in two ways.

First, we run a Socket Intents Prototype enabled client-side proxy, which we describe in Section VI-B, in a testbed emulating typical network characteristics, see Section VII-B. Second, we run a simulator, which we describe in Section VI-C, across a wide range of network characteristics and Web pages, see Section VII-C. To ensure consistency of the results, we validate the runs in the testbed against the simulator, see Section VIII-B, and the simulator against the actual download times in the wild, see Section VIII-C. Finally, our results for the proxy are shown in Section VIII-A and for our simulator in Section VIII-D - VIII-F.

B. Web Proxy

To explore to which extent the possible benefits are viable in practice we implemented a Socket Intent enabled HTTP proxy⁵. Our HTTP proxy consists of 2.300 lines of C code. Enabling the Socket Intents Prototype took only 20 lines.

The proxy uses the *Size to be Received Intent* for the objects it downloads, see Section V-F. Since the proxy does not know the size of the objects in advance, we use a two-step download. First, the proxy issues a range request for the first m bytes⁶ to get the initial part as well as the size of the object. If the object is not completely transferred, the remainder is retrieved via a second range request⁷.

C. Web Transfer Simulator

To evaluate the benefits of seamlessly using multiple interfaces and scheduling requests according to our policies at scale across a wider range of network properties and Web pages, we build an event-based data transfer simulator. As evaluation metric we use page load time⁸, which has a high influence on the end-user Quality of Experience [21]. Additional metrics can easily be implemented in the simulator, e.g. [22].

The simulator takes a Web page including all Web objects and their dependencies (represented as a HAR files — see Section VII-D), the Socket Intents Policy, and a list of network interfaces with their path characteristics as input. The simulator replays the Web page download by transferring all Web page objects while respecting their inter-dependencies. It uses the policy to distribute the object transfers across the interfaces and calculates the total page load time.

Since our simulator has global knowledge, it knows all object inter-dependencies a priori. Thus, it can decide when a transfer can be scheduled, i.e, whether all objects that it depends upon have already been loaded. To schedule a transfer we assign it to a connection. This is the job of the policy module which returns either an existing TCP or MPTCP connection, an interface, or a list of interfaces to use to open a new connection, or postpones the transfer if the limit of parallel connections has been reached. A connection is reused if the host name matches and it is either idle or it is expected to become idle before a new connection can be established.

The simulator then determines the next event for this connection, such as the completion of a transfer. When a transfer completes, the simulator records the time, marks all transfers that depend on it as enabled, and schedules them. After the last transfer, the total page load time is reported.

The simulator supports persistent connections with and without pipelining for TCP as well as MPTCP connections across multiple interfaces. It uses a default connection timeout of 30 seconds and limits⁹ the number of parallel connections per server to 6 and the overall number of connections to 17. We simulate TCP slow-start using a configurable initial congestion window size with a default value of 10 segments [23]. Our motivation for simulating slow-start is to get more realistic load times especially for small objects, which are common on Web pages, when they are downloaded on high latency links, which are common in access networks. To simulate slowstart and fair bandwidth sharing, we keep track of the current throughput for each connection. This is updated according to TCP slow-start and capped by the congestion window or the available bandwidth share of that interface to assure TCP fairness¹⁰. Our underlying assumption is that TCP tries to fairly share the available bandwidth between all parallel connections [24]. Rather than fully simulating the congestion avoidance of TCP we assume instantaneous convergence to the

 $^{^{5}}$ In the future, our Socket Intents Prototype can also be implemented in a Web browser.

⁶ Here, m enables a trade-off between RTT and network bandwidth. We see good results for values between 4-8K; values that fit within the initial TCP window of today's Web servers.

⁷The proxy is able to handle various answers including the full object or the remaining part of the object, with and without chunked-encoding.

⁸Here we focus on network time, i.e., the total time to download the objects of the Web page. The complete time to display a Web page also includes times for DNS resolution, page rendering and possibly client-side JS computation.

⁹These values correspond to the defaults of the browser we use to retrieve our workload.

¹⁰In our simulator, a connection leaves slow-start once it reaches the available bandwidth share and never returns to slow-start.

appropriate bandwidth share. The available bandwidth share of each interface is potentially adjusted by each connection event for that interface. If needed the time of the next event is then adjusted accordingly. Note that for MPTCP, our simulator aggregates the bandwidth of the subflows by simulating them as separate TCP flows. We do not implement coupled congestion control because it does not apply to the network scenario we use in our evaluation, see Section VII-A.

D. Simulator Implementation

We implemented our data transfer simulator as a heap-based discrete event simulator. It consists of 3k lines of Python code and is available under a relaxed CRAPL license¹¹. It models the process of loading a Web page by keeping track of the status of the transfers, connections and interfaces.

Each **transfer** corresponds to a Web object which contains the object size, its relationship to other transfers, if the object was transferred via HTTPS, and the server hostname. The **connections** are responsible for estimating and updating the completion times of the transfers which are assigned to them and for simulating (MP)TCP. In case of MPTCP, we maintain a master connection and per-interface subflows. The **interfaces** bundle the connections and are used to calculate the available bandwidth shares.

The **transfer-manager** keeps track of all transfers and informs the policy if a transfer can be scheduled. The **policy** is the main decision-making entity of the simulation. The policy determines which interface(s) to use or which connection to re-use for each transfer by choosing the most appropriate one. The policy then notifies the transfer-manager to schedule the transfer.

To schedule objects in the appropriate order, we derive their interdependencies from the HAR files we gather, see Section VII-D. While identifying all objects of a Web page from the HAR files is straightforward this does not apply to the object dependencies. Some object dependencies are obvious from the base page, the HTML document, and the client-side DOM. However, JavaScript or other Web objects can modify the DOM, by adding or removing Web page objects, at any point during the page load. For example, when a Web site uses JavaScript to dynamically load pictures the simulator should not start downloading these pictures before the JavaScript object has been retrieved. After all, the browser first has to parse the JavaScript before it can download the pictures.

We decided against using sophisticated systems to derive interdependencies, e.g., [25], since their focus is on finding the true dependency tree to speed up future downloads. Thus, using these dependencies often leads to much more optimistic results compared to the capabilities of current browsers. Thus, to ensure compatibility we use a more conservative heuristic. We identify the dependencies from the download times contained in the HAR files. This method is feasible since we use a non-bandwidth limited client to gather the HAR files.

We implement our \mathcal{EAF} policy, see IV-D, for the simulator. Since the simulator tries to provide an upper bound of the benefits it relies on its global knowledge about all currently

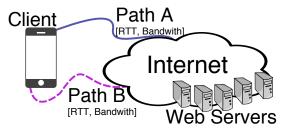


Fig. 2. Simplified Network Scenario.

active transfers. The RTT and maximum interface bandwidth, as well as the size of the objects for the *Size to be Received Intent*, are known a priori. Within the simulation, we add one RTT if a connection can be reused, two RTTs if a new connection has to be established, and two additional RTTs for each TLS handshake.

Since Socket Intents Policies can use transfer predictions, policies can reuse the simulator logic to obtain an estimate of the completion time given the current state and an interface/connection option. This is realized by partially cloning the simulator's state, including all currently active transfers, and simulating the completion time for that transfer.

For MPTCP, the simulator presumes that MPTCP subflows can be opened on all local and remote interfaces. With two network interfaces at the client and one interface at the server, the policy establishes two subflows. The interface for the initial subflow is configurable. We considered two variants: starting the initial MPTCP subflow on the same statically chosen interface (MPTCP if1) or always on a different, randomly chosen interface (MPTCP rnd).

Finally, $\mathcal{EAF}_{\mathcal{MPTCP}}$ combines \mathcal{EAF} with MPTCP. In addition to predicting the arrival time for each interface, it also considers MPTCP for all possible interface combinations. The intuition here is that MPTCP is beneficial for some cases but not all cases. For example, this policy can avoid scheduling small communication units on a high RTT interface. The simulation of $\mathcal{EAF}_{\mathcal{MPTCP}}$ is analogous to the EAF policy, but it includes predictions with MPTCP for all interface combinations, therefore using the interface for the first subflow that is predicted to give the best results. This policy considers neither the Socket Intents nor the current network performance.

We test the basic functionality of the various simulator policies using various traffic patterns that can take advantage of MPTCP, EAF, and EAF_MPTCP , with and without connection reuse. For these cases we manually calculate the expected page load times and check the simulator results against them. In addition, we use extensive assertions and cross checks within the simulator to ensure the consistency of the results.

VII. EVALUATION SCENARIO

To evaluate Socket Intents using the approach presented in Section VI-A, we assume the following network scenario, which serves as the basis for both our testbed setup and our simulator.

A. Network Scenario

The motivation for our network scenario is that, for mobile devices, access networks almost always dominate performance



Fig. 3. Testbed setup used in the emulation.

as the bandwidth bottleneck is likely to be located there and access networks often introduce major delays¹². Thus, our network scenario, see Fig. 2, consists of a client, Web servers, and the paths between them. We presume that all Web servers are reachable via both network paths. Moreover, we choose to neglect the RTT variability introduced by the Internet since queuing delays on Internet core links (\geq 10 Gbit/s bandwidth) are negligible [26]. Therefore, we capture the path characteristics as "interface" RTT and bandwidth. To model connection reuse, we assume a separate server per hostname.

B. Testbed Setup

To evaluate the benefit of Socket Intents in the Web proxy using the Socket Intents Prototype under different access network characteristics we setup a testbed according to our network scenario, see Fig. 3. It consists of three physical machines: Web server, traffic shaper, and Web client. The Web client has two network paths to the Web server via two separate network interfaces. The network characteristics are emulated by the traffic shaper and include three scenarios of access network characteristics. The characteristics range from fully symmetric to asymmetric, see Table III, and are representative of access network characteristics found in literature [27]. On the Web client, we run a Web browser along with the proxy and the Multi Access Manager. Our Multi Access Manager supports \mathcal{EAF} as discussed in Section V-F, which we compare against the use of a single interface.

We automate a browser¹³, which we restart for each measurement to ensure that the cache is cold. We run the DNS server on the Web client to ensure that name resolution does not add delays. Furthermore, since the two-step download does not work with HTTPS, we use HTTP.

The Web server hosts our workload. It consists of handcrafted pages, each with a different number of objects (ranging from 2 to 64) of various sizes (between 1 KB and 1 MB), as well as mirrored versions of several Web pages from the Web workload, see Section VII-D. We select a range of Web pages that represent our workload in terms of numbers and size of objects and hosts.

We use a single Web server since we assume that Web performance is dominated by the different access networks. However, we set up a virtual host per host name to restrict connection reuse appropriately. As we want a lower bound of the performance benefits, we tune the TCP parameters of the

TABLE III
TESTBED SHAPER: NETWORK PARAMETERS.

	Interface 1			Interface 2		
	RTT ms	Down MBit/s	Up MBit/s	RTT ms	Down MBit/s	Up MBit/s
Symmetric	45	10.0	1.0	45	10.0	1.0
Asymmetric	20	6.0	0.768	70	13.0	6.0
Highly Asym.	10	3.0	0.768	100	20.0	5.0

Web server to conservative values¹⁴. On the shaper we emulate a given maximum bitrate using tc and latency with netem and we restrict queue sizes to avoid buffer bloat.

C. Experimental Design for Simulator Evaluation

To evaluate the potential benefits of using Socket Intents across a wide range of parameters, i.e., with different policies under different network scenarios and for different Web pages, we use a full factorial experimental design. Each factor can, in principle, influence the page load time. For each factor, we consider multiple values that cover the possible value ranges. By simulating all combinations, see Table IV, we run 9M simulations.

In our experimental design, the primary factor is the **Policy** used with all of our Socket Intents Policies, see Section VI-D, as levels. The **Web pages** of our workloads, see Section VII-D, are the second factor: Here, the levels are the different Web pages (with their 26 repeated crawls for Alexa Top 100 and one crawl for the Alexa Top 1000).

The remaining four factors describe the network scenario: Since our simplified network scenario as illustrated in Fig. 2 consists of one client using two access networks and various Web servers which are reachable via both interfaces, these factors are: **Interface 1 RTT** and **Bandwidth** as well as **Interface 2 RTT** and **Bandwidth**. The levels for these were chosen to reflect typical interface characteristics: We consider mobile devices that have WiFi as well as cellular connectivity. Interface 1 should resemble the possible characteristics of home broadband connectivity (e.g., DSL or cable) and

TABLE IV Levels of the Factorial Experimental Design.

Factor Levels		
Policy:	Interface 1, Interface 2 Round Robin (starting on if 1), MPTCP starting on Interface 1 (MPTCP if1) or on a random interface (MPTCP md), Earliest Arrival First (EAF), or EAF with MPTCP (EAF_MPTCP).	
Web page:	Alexa Top 100 and Top 1000.	
Interface 1 RTT: Interface 1 Bandwidth: Interface 2 RTT: Interface 2 Bandwidth:	10, 20, 30, or 50 ms. 0.5, 2, 6, 12, 20, 50 Mbit/s. 20, 50, 100, or 200 ms. 0.5, 5, 20, or 50 Mbit/s.	

¹²Internet backbone delays are in the order of a few milliseconds while access delays are typically significantly larger. With increasing access network capacities, the bottleneck might be in the core in some cases. However, capacities are not increasing in all regions of the world.

¹³We use Mozilla Firefox 52.5 Web browser with the Selenium browser automation framework.

¹⁴ We use TCP/Reno with an initial congestion window size of 10 MSS. We disable TCP metrics saving to prevent congestion window caching as well as TCP segmentation off-loading to eliminate interference with the NIC firmware.

Interface 2 should resemble the range of possible 3G/LTE coverages¹⁵. This results in the levels shown in Table IV.

D. Web Workload

To get a wide range of Web pages we crawl the landing pages of the Alexa Top 100 Web sites on 26 consecutive days starting on December 07 2015 and the Alexa Top 1000 Web sites on October 10 2016¹⁶. We focus on the mobile version of the pages by overriding the user-agent of our Firefox browser, impersonating a generic Android mobile device.

As browser we use Firefox version 38.4.0 automated with Selenium and the Firebug 2.0.13 and NetExport 0.9b7 plugins to retrieve the objects and to record the crawled Web pages in the HTTP Archive (HAR) format. Each HAR file contains a summary of all objects of the page as well as their sizes, types, origins (remote sites), and timings. We use a single vantage point with a high available network bandwidth, a virtual machine within a university network.

While most of the pages comprise between 1 and 50 objects there are some with more than 100 objects or even up to 260 objects. Moreover, many Web pages have a low median object size. Furthermore, the number of hosts that have to be contacted ranges from a single one to more than 20 with a median of 7. The total size of the Web pages is between 23.1 KB (5th quantile) and 1.8MB (95th quantile) with a large fraction of pages below 300 KB. These results are in line with previous work [28], [29].

For comparison with less complex Web workload, we add handcrafted Web pages to our workload. These pages consist of different number of objects (ranging from 2 to 64) of various sizes (1 KB to 1 MB), and a mix of these objects.

VIII. EVALUATION

To explore the benefits of seamlessly combining multiple access networks for speeding up Web page load time, we evaluate Socket Intents in two ways, see Section VI.

A. Socket Intent Benefits in Testbed

We setup our testbed as described in VII-B. For each of our Web pages we repeatedly download each page 7 times, using a single interface as well as using \mathcal{EAF} , see Section V-F. We compute the load time of the individual objects, and aggregate the times during which objects were downloaded to compute the total page load time¹⁷. The resulting page load times are shown in Figure 4 using a logarithmic y-axis. It includes all three policies: *Interface 1, Interface 2,* and \mathcal{EAF} . The mixed handcrafted workload shown here consists of 16 objects of 1KB, 8 objects of 10KB, and 4 objects of 100KB.

We see that *Interface 1* is the better choice if the Web objects are small and the network scenario is asymmetric. *Interface 2* is the better choice if the objects are larger or if there are more

¹⁶ http://www.alexa.com/topsites

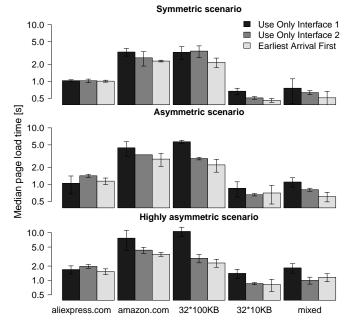


Fig. 4. Proxy: Page load times.

objects. Using both interfaces is, in particular, beneficial for the symmetric scenario. While there is still a benefit of using both interfaces it gets smaller for more asymmetric scenarios.

The \mathcal{EAF} policy takes advantage of the multiple access networks seamlessly. It either uses both interfaces or the better one of the two with only a slight increase in page load time variability. For pages with many objects such as the handcrafted workload of 32 objects of 100 KB, our \mathcal{EAF} policy outperforms the better of the two interfaces with speedups from 25% to 50%. For some of the actual Web pages that we mirror on our testbed, including amazon.com, we get speedups in the range of 20–45%. For other Web pages such as aliexpress.com, we only get a speedup of up to 10%.

The reason for the "decreased" benefits compared to the handcrafted pages are that the mirrored Web pages fetch content from different servers, which limits connection reuse. Furthermore, even for mirrored versions of the same Web page, load times vary based on optimizations in the contained Javascripts, as the Alexa 100 pages are heavily optimized.

Nevertheless, our results highlight the potential of Socket Intents: Even with a proxy the page load times improve. Including Socket Intents within the browser rather than a proxy is likely to yield even better performance.

B. Validation of Proxy and Simulator

We validate our simulation results against the proxy by measuring the Web page load time of our workload in the testbed with similar shaper settings as the interface parameters we use in the simulator. In Figure 5 we compare the simulated and the actual load times for the handcrafted workloads of different sizes, showing the median load time and the 95% confidence intervals. The mixed workload consists of 32 objects of 1KB, 16 objects of 10 KB, 2 objects of 100 KB and 2 objects of 200 KB. Using a single interface with an RTT of 50 ms and a bandwidth of 6 Mbit/s, see Figure 5a, we see slightly higher load times on the testbed both with

¹⁵Costs or restrictions of the data plan are beyond the scope of this paper, but could easily be taken into account by an elaborate policy.

¹⁷The total display time includes page rendering and client-side JavaScript computation, which we exclude here.

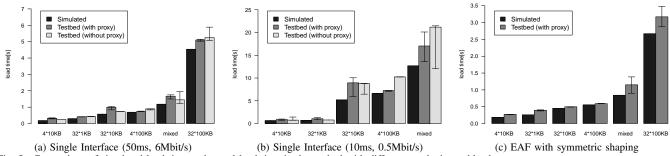


Fig. 5. Comparison of simulated load time and actual load time in the testbed with different synthetic workloads.

and without the proxy, especially for large workloads. Using a single interface with only 0.5 Mbit/s, see Figure 5b, we do not get a page load time for the workload with 32 objects of 100 KB because the browser times out after 10-20 seconds, so we do not show it in this plot. Using our \mathcal{EAF} policy with symmetric shaping (50 ms and 6 Mbit/s on one interface, 50ms and 5 Mbit/s on the other), we cannot test the case without proxy, as we cannot use \mathcal{EAF} without the proxy. Both our simulator and the proxy in the testbed show speedups, see Figure 5c. Note the differences between the y axes, which reflect the speedups observed in Section VIII-A. We get similar results for RTTs up to 200 ms and bandwidths up to 50 Mbit/s.

We get similar load times with and without the proxy. This shows that the two step download in our proxy does not have a major influence on the load time. Overall the simulator is more optimistic than the testbed. However, the differences are quite small. The differences to the simulator can be explained by the following observations: First, the gzip transfer encoding conflicts with range requests: Sometimes the server sends the whole object even though only the initial part is requested. Moreover, disabling compression for the initial request is not feasible as it eliminates compression also for the second request since the content-range refers to the range after compression. Second, the simulator presumes that all independent transfers start immediately, which is not always the case in practice. This can skew timings, in particular for small workloads. These effects are independent from the use of our Socket Intents Prototype. Accordingly, we can use the simulator to conduct a realistic comparison between scenarios with and without Socket Intents.

C. Simulator vs. Actual Page Load Time

We compare the actual page load time to the simulated one for all Web pages of our workload. Given that our crawl uses a machine with a single interface we also use a single interface with the policy "Single Interface". To determine the interface parameters we estimate the available bandwidth as well as the RTT to the servers from the actual download. To estimate the available bandwidth we use all objects larger than a minimum size of 50 KB. Hereby, we take into account that several of these can occur in parallel. Using the median of the estimated bandwidth results in a typically used bandwidth of 67.13 Mbit/s – this suggests that none of the transfers were actually bandwidth bound. To estimate the RTT the simulator issues a series of pings for each Web page. The median RTT

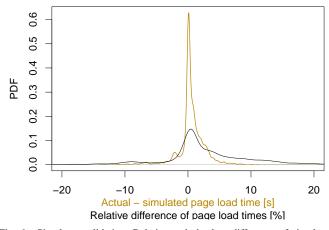


Fig. 6. Simulator validation: Relative and absolute difference of simulated vs. actual page load time.

of all servers of that Web page is then used as an estimator for the interface for the validation run for that Web page.

The simulator as well as the validation uses several simplifications. First, the simulator assumes that all Web objects share a single network bottleneck and that the RTT is the same for all servers. In reality, some embedded objects of Web pages are fetched from hosts with different network bottlenecks and RTTs. We use ICMP ping rather than TCP ping and the pings are not executed while the HAR files are gathered.

Figure 6 shows the absolute as well as the relative differences of the simulated vs. the actual page load times for all Alexa Top 100 Web pages from Section VII-D. The main mass of both distributions is around zero indicating that the simulated page load times are very close to the actual ones. This is confirmed by the median value which is 0.3548/1.5% for the absolute/relative differences. This highlights that the simplifying assumptions of the simulator still enable us to approximate the actual page load times and that we capture most of the intra Web page dependencies.

There are some differences for some Web pages. We manually checked them and find a majority is caused by differences in the estimated bandwidth, server delays, and name resolution overhead. These are, e.g., related to Web back-office interactions [30]. Overall, the results are rather close and show that our simulations result in reasonable approximations of the actual Web page load time.

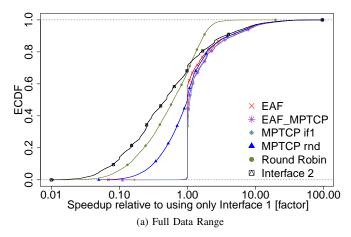


Fig. 7. CDF of Speedups vs. Interface 1 for the Alexa Top 100 workload.

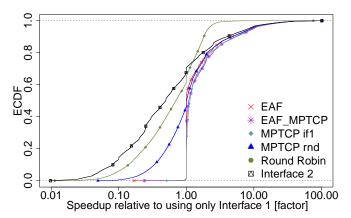
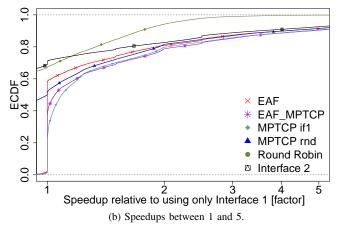


Fig. 8. CDF of Speedups vs. Interface 1 for the Alexa Top 1000 workload.

D. Benefits of Combining Multiple Access Networks

To explore the benefits of combining multiple access networks by using Socket Intents, we compare the speedups of the page load times against the baseline policy *Interface 1*. The baseline policy *Interface 1* resembles what most current mobile OSes do: Use only WiFi and, therefore, the home broadband if available.

Fig. 7a shows the empirical cumulative distribution functions (ECDF) of the speedups achieved using a simulated Socket Intents Policy relative to only using Interface 1, all other parameters being equal. Thus the ECDF shows speedups across all network scenarios outlined in Table IV based on the Alexa Top 100 Web pages and categorized by the Socket Intents Policy used. We see that in more than 42% of the cases for EAF and 63% of the cases for EAF_MPTCP these policies provide a speedup of more than 1, which means that loading a Web page using these policies is faster than using *Interface 1* in the same scenario. In the remaining cases, they almost always provide a speedup of 1, which means that they neither gain nor lose from using multiple interfaces. In these cases, the page load was not bandwidth limited and simply loading the page over Interface 1 was the fastest option. Thus using the other interface in addition did not provide any speedup. Therefore, EAF and EAF_MPTCP simply choose to use Interface 1 in these cases. We also see that in about 1.5% of cases \mathcal{EAF} and EAF_MPTCP is slower than Interface 1 which turned out to be



a limitation of the simulator¹⁸. Overall these results show that using \mathcal{EAF} and $\mathcal{EAF}_{\mathcal{MPTCP}}$ is a good choice in any case.

The speedups of both MPTCP policies are very dissimilar: When establishing the first subflow over Interface 1 (MPTCP*if1*), it shows a speedup greater than 1 in 78% of the cases and neither improvement nor penalty in the other cases. In contrast, if starting the first subflow for MPTCP over a randomly chosen interface (MPTCP rnd), MPTCP performs worse than Interface 1 in 48% of the cases and can be up to 10x slower. We take a closer look at these effects in Section VIII-E.

The other baseline policies, *Interface 2* and *Round Robin*, unsurprisingly show a penalty in about 70% of cases as in most network scenarios Interface 2 has a much higher RTT than Interface 1.

Figure 7b shows the speedups between 1 and 5 from Figure 7a in more detail. From our data, we find that \mathcal{EAF} was up to 2x faster than *Interface 1* in about 23% of the cases and from 2 to 5x faster in about 11% of the cases. We even see speedups of more than 5x in 8.5% of the cases. $\mathcal{EAF}_{\mathcal{MPTCP}}$ and \mathcal{MPTCP} *if1* shows negligibly higher speedups than \mathcal{EAF} . Overall, all three policies perform similarly and can take serious advantage of combining multiple access networks. Even with similar benefits, \mathcal{EAF} has the advantage over MPTCP that it does not need to be supported by the server and that it cannot be blocked by middleboxes.

Finally, Fig. 8 shows the ECDF of the speedups against *Interface 1* for the Alexa 1000. These look similar to the ones for Alexa 100 in 7a. This gives us confidence that our benefits are stable for a wide variety of different Web pages.

E. Benefits of Using the Socket Intents Prototype with MPTCP

As described in Section VIII-D, for our dataset MPTCP if1 and MPTCP rnd behave very differently. While both show gains in almost all cases, MPTCP rnd is at a disadvantage in 48% of the cases while MPTCP if1 almost never imposes a penalty.

In Fig. 9, we compare speedups of our policies for all scenarios and Web pages against MPTCP if1. The curves for

¹⁸ In these cases, the simulator fetches a single huge object via the less suitable interface while the connection limit prevents starting a new connection on the more suitable one.

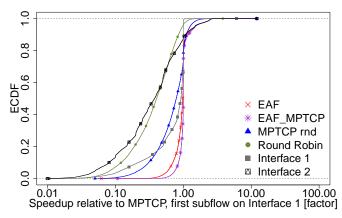


Fig. 9. CDF of Speedups vs. MPTCP if1 for the Alexa Top 100 workload.

 \mathcal{EAF} and \mathcal{EAF}_{MPTCP} are close to 1, which means that the page load times are similar to MPTCP in most cases and never considerably worse. In contrast, if establishing the first subflow for MPTCP over a randomly chosen interface (MPTCP rnd), MPTCP performs worse and can be up to 10x slower than using Interface 1 and about 30x slower than MPTCP if1. The reason for that is that Interface 1 has a shorter RTT in most network scenarios. As many Web page downloads in our workloads were short and not bandwidth bound, MPTCP will often perform most of the download over the initial subflow. Thus, not picking the most suitable one in 50% of the cases bears a considerable performance penalty. EAF MPTCP can always choose the most suitable interface for the first subflow and, therefore, can improve over MPTCP if1 in cases where Interface 1 is not the most suitable interface for the first subflow. Note that EAF shows a similar performance as MPTCP if1. The cases where EAF and EAF_MPTCP perform slightly worse than MPTCP if 1^{19} seem negligible to us given the benefits.

F. Explaining Page Load Time Speedups

To understand how the factors of the scenario and Web page affect the speedups of our policies, we take a closer look at the cases when \mathcal{EAF} is slower, similar to, or faster than *Interface 1*.

In Fig. 10 we bin the simulation results of \mathcal{EAF} into six categories of benefits and show how these distribute among the total Web page sizes and Interface 1 bandwidths. Note that these categories contain different numbers of observations, i.e., *EAF is slower* for just 1.5% of all cases while *EAF is equal* to Interface 1 for 56.6% of all cases.

The CDF in Figure 10a shows the frequency of the speedup categories over the different levels of Interface 1 bandwidths from Table IV. In cases when \mathcal{EAF} was slower or equal to *Interface 1*, higher values for the Interface 1 bandwidth are more prevalent, while high speedups mostly occur when the Interface 1 bandwidth is low. Similarly, we tend to see high speedups for higher levels of Interface 2 bandwidth and for lower levels of Interface 2 RTT (plots omitted).

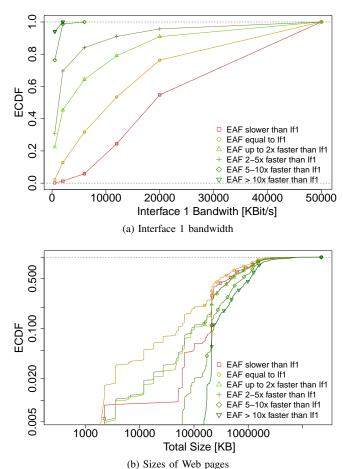


Fig. 10. Levels of factors for which we see a certain level of speedup for Alexa Top 100

To explore what kind of Web pages can benefit from our \mathcal{EAF} policy, we plot the CDF of the speedup categories over the total Web page size in Fig. 10b. As high speedups occur much more frequently for large Web pages, we conclude that unsurprisingly these take most advantage of using multiple access networks. For the median object size and the number of objects in a Web page we see similar results, with high speedups occurring more frequently in cases with high median object sizes.

Both analyses show that our multi-access policies are most useful when Web page download is bandwidth limited.

IX. RELATED WORK

We next review related work regarding multipath support in general and on the end host's Operating System (OS) in particular. We then focus on how application needs are taken into account. Finally, we discuss the benefits of using multiple access networks in the context of WiFi offloading and MPTCP.

1) Multipath: For a comprehensive survey of network layer multipath solutions see Qadir et al. [31]. They present a detailed analysis of the design choices of how to compute and select routes as well as how to split the flow across the chosen paths.

A survey of multipath approaches in some current OSes [32] points out several problems that we also discuss in Section III. Many OSes support mechanisms for source and destination address selection for IPv6 multihoming [33], [34] and there are

¹⁹These cases occur because \mathcal{EAF} and $\mathcal{EAF}_{\mathcal{MPTCP}}$ do not take future transfers into account. They cannot change their decision whether to use MPTCP, while always using MPTCP allows to rebalance traffic between subflows later.

proposed Socket API extensions that enable applications to set preferences [33]. However, these address selection algorithms focus on reachability, while we consider bandwidth aggregation and performance improvement. Some OSes implement a central connection manager to choose the appropriate access network, as is also proposed in current research such as by Kiefer et al. [35]. The latter uses policies controlled by the application and the user and relies on observations of the current network performance, which is similar to our Multi Access Manager. However, it only works on a per-flow basis and not per communication unit. Also, their application policies specify flow prioritizations and constraints, but not different characteristics of the traffic.

2) Application Needs: Previous work where an application can specify its requirements and needs often focuses on QoS, e.g., QSockets [36]. We use the best-effort approach of Socket Intents. The term *Intents* has its origin in Intentional Networking [37], an attempt to explore mobile network diversity by letting applications specify traffic characteristics via an extended Socket API. However, they use a per-packet approach, which introduces complications and overhead, while we use a per-socket/per-flow or per-request approach. Moreover, they imply guarantees while we suggest best-effort. Other approaches include ideas from machine learning to guide application choices, e.g., Deng et al [38].

3) Socket APIs: Alternative socket APIs move parts of the application logic to the socket API, e.g., by requesting a service rather than a protocol, port, and address in the protocolindependent transport API [39] or by exposing all protocols and auxiliary information of the application in a tree-like structure [15]. In contrast, our Socket Intents Prototype takes transport protocols as given. Thus, the idea of Socket Intents is complementary to the before-mentioned work.

4) Offloading: Multi-access connectivity enables one to balance traffic, e.g., from the mobile network to the WiFioffloading-or from WiFi to the mobile network-onloading. Recently, both variants have gotten a lot of attention in the research community, e.g., [1]–[3], [40], as well as in industry, e.g., [41], [42]. For a survey on offloading, we refer to, e.g., Aijaz et al. [1]. For a summary of multi-access connectivity, we refer to, e.g., Schmidt et al. [4]. Examples of recent work on offloading include the work by Lee et al. [40], who demonstrate via a quantitative study the performance benefit of offloading 3G mobile data to WiFi networks, and Balasubramanian et al. [2], who propose Wiffler to augment mobile 3G capacity with WiFi. For onloading, we, e.g., point to Vallina et al. [3]. For an analysis of the economics of offloading see Lee et al. [41]. Offloading typically implements support for using multiple access networks within the network or on the application layer, while we provide support for it within the end-host OS.

5) MPTCP: There have been many studies exploring how an end host can benefit from multiple paths using MPTCP. Chen et al. [43] evaluate MPTCP performance in the wild by comparing its use over a home WiFi network and several different cellular providers to the use of a single path. They find that for small files, using a single path over WiFi is best, while larger files benefit from MPTCP's aggregated bandwidth. This observation is shared by Raiciu et al. [44] and Deng et al. [45], who emphasize that the choice of the interface to establish the first subflow is important, which is in line with our observations.

Han et al. [46] evaluate page load times of HTTP and SPDY over WiFi and LTE using a proxy-based setup. They find that SPDY over MPTCP is always beneficial. This is in contrast to HTTP over MPTCP which in some cases performs even worse than plain TCP. Similarly, Nikravesh et al. [47] observe a performance penalty and energy consumption overhead for apps with small flows in the wild. As a solution, they propose a proxy with persistent connections over multiple paths.

A similar approach for controlling MPTCP (see Section V-E3) has been proposed by Hartung and Milind [48] to use MPTCP for LTE bandwidth management.

6) *SCTP*: Dreibholz et al. propose an advanced stream scheduling policy for SCTP [49] and achieve performance benefits in asymmetric path scenarios using a simulation. It would be possible to integrate such an advanced scheduler into our policies in future work.

X. CONCLUSION

Our Socket Intents Prototype is a system to provide seamless OS support for multiple access networks. It achieves this goal by allowing applications to express their intents and expectations for each communication unit, e.g., HTTP request or TCP connection, toward the OS. The OS then can match the applications' needs and the diverse access networks available in a best effort way.

We evaluate Socket Intents in a testbed and confirm that our Socket Intents Prototype can speed up page load time by up to 50% in typical scenarios. We then use a simulator to explore speedups across a wide range of Web pages and network characteristics. Our simulations demonstrate that Socket Intents with the \mathcal{EAF} policy can improve Web page load time in about 50% of the cases without incurring penalties. Therefore, without kernel modification, we can achieve about the same speedups possible by using MPTCP. In cases where the access networks characteristics are very different, our $\mathcal{EAF}_{-}MPTCP$ helps MPTCP pick the right interface for the initial subflow and therefore prevents performance penalties that can occur from using the "'wrong"' interface.

Socket Intents are a first step for sharing information between applications, the OS, and the network. This idea is not limited to the use case of multiple access networks, but can also be beneficial to automatically choose among transport protocols and can give valuable input to traffic management systems for datacenter networks. Therefore, we are trying to contribute the key ideas of Socket Intents to the IETF *TAPS working group* [50] as means to address the complexity arising from today's transport layer diversity.

ACKNOWLEDGEMENTS

This work has been supported by Leibniz Prize project funds of DFG - German Research Foundation: Gottfried Wilhelm Leibniz-Preis 2011 (FKZ FE 570/4-1).

Thanks to Thomas Zinner for his feedback and support during the revision of this paper.

REFERENCES

- A. Aijaz, H. Aghvami, and M. Amani, "A survey on mobile data offloading: technical and business perspectives," *Wireless Communications*, *IEEE Transactions on*, vol. 20, no. 2, pp. 104–112, 2013.
- [2] A. Balasubramanian, R. Mahajan, and A. Venkataramani, "Augmenting mobile 3g using wifi," in ACM MobiSys, 2010, pp. 209–222.
- [3] N. Vallina-Rodriguez, V. Erramilli, Y. Grunenberger, L. Gyarmati, N. Laoutaris, R. Stanojevic, and K. Papagiannaki, "When david helps goliath: the case for 3g onloading," in *SIGCOMM HotNets*. ACM, 2012, pp. 85–90.
- [4] P. S. Schmidt, R. Merz, and A. Feldmann, "A first look at multi-access connectivity for mobile networking," in ACM workshop on Capacity sharing. ACM, 2012, pp. 9–14.
- [5] F. Rebecchi, M. D. de Amorim, V. Conan, A. Passarella, R. Bruno, and M. Conti, "Data offloading techniques in cellular networks: A survey," *IEEE Communications Surveys Tutorials*, vol. 17, no. 2, pp. 580–603, 2015.
- [6] R. Stewart, "Stream Control Transmission Protocol," RFC 4960 (Proposed Standard), Sep 2007, updated by RFCs 6096, 6335. [Online]. Available: http://www.ietf.org/rfc/rfc4960.txt
- [7] C. Paasch and O. Bonaventure, "Multipath TCP," *Queue*, vol. 12, no. 2, pp. 40:40–40:51, 2014.
- [8] A. Ford, C. Raiciu, M. Handley, S. Barre, and J. Iyengar, "Architectural Guidelines for Multipath TCP Development," RFC 6182 (Informational), Mar 2011. [Online]. Available: http://www.ietf.org/rfc/rfc6182.txt
- [9] P. S. Schmidt, T. Enghardt, R. Khalili, and A. Feldmann, "Socket intents: Leveraging application awareness for multi-access connectivity," in ACM CoNEXT. ACM, 2013, pp. 295–300. [Online]. Available: http://doi.acm.org/10.1145/2535372.2535405
- [10] P. Tiesel, T. Enghardt, and A. Feldmann, "Socket intents," (work in progress), IETF Secretariat, Internet-Draft drafttiesel-taps-socketintents-01, 10 2017. [Online]. Available: https: //www.ietf.org/archive/id/draft-tiesel-taps-socketintents-01.txt
- [11] P. S. Tiesel, B. May, and A. Feldmann, "Multi-homed on a single link: Using multiple ipv6 access networks," in *Proceedings of the 2016 Applied Networking Research Workshop*, ser. ANRW '16. ACM, 2016, pp. 16–18. [Online]. Available: http://doi.acm.org/10.1145/2959424. 2959434
- [12] L. Popa, A. Ghodsi, and I. Stoica, "Http as the narrow waist of the future internet," in SIGCOMM HotNets. ACM, 2010, pp. 6:1–6:6.
- [13] P. Richter, N. Chatzis, G. Smaragdakis, A. Feldmann, and W. Willinger, "Distilling the internet's application mix from packet-sampled traffic," in *Passive and Active Measurement*, ser. Lecture Notes in Computer Science, J. Mirkovic and Y. Liu, Eds. Springer International Publishing, 2015, vol. 8995, pp. 179–192.
- [14] D. Wing and A. Yourtchenko, "Happy Eyeballs: Success with Dual-Stack Hosts," RFC 6555 (Proposed Standard), Apr 2012. [Online]. Available: http://www.ietf.org/rfc/rfc6555.txt
- [15] B. Trammell, C. Perkins, and M. Kühlewind, "Post sockets: Towards an evolvable network transport interface," in *Workshop on Future of Internet Transport (FIT 2017)*, 2017.
- [16] J. Corbet. (2015, 09) The [linux] kernel connection multiplexer. [Online]. Available: https://lwn.net/Articles/657999/
- [17] P. Tiesel and T. Enghardt, "A socket intents prototype for the bsd socket api - experiences, lessons learned and considerations," (work in progress), IETF Secretariat, Internet-Draft draft-tiesel-tapssocketintents-bsdsockets-01, 03 2018. [Online]. Available: https://www. ietf.org/archive/id/draft-tiesel-taps-socketintents-bsdsockets-01.txt
- [18] B. Trammell, M. Welzl, T. Enghardt, G. Fairhurst, M. Kuehlewind, C. Perkins, P. Tiesel, and C. Wood, "An abstract application layer interface to transport services," (work in progress), IETF Secretariat, Internet-Draft draft-trammell-taps-interface-00, March 2018. [Online]. Available: https://datatracker.ietf.org/doc/draft-trammell-taps-interface/
- [19] Linux Foundation, Netlink(7) Linux Programmer's Manual. [Online]. Available: http://man7.org/linux/man-pages/man7/netlink.7.html
- [20] M. Palmer, "Implementation and evaluation of multi-access policies for MPTCP path management in user-space," Master's thesis, TU Berlin, 2015.
- [21] S. Egger, T. Hossfeld, R. Schatz, and M. Fiedler, "Waiting times in quality of experience for web based services," in *Quality of Multimedia Experience (QoMEX), 2012 Fourth International Workshop on.* IEEE, July 2012, pp. 86–96.
- [22] C. Kelton, J. Ryoo, A. Balasubramanian, and S. R. Das, "Improving user perceived page load times using gaze," in USENIX NSDI, vol. 17. Usenix, 2017, pp. 545–559.

- [23] J. Chu, N. Dukkipati, Y. Cheng, and M. Mathis, "Increasing TCP's Initial Window," RFC 6928 (Experimental), Apr 2013. [Online]. Available: http://www.ietf.org/rfc/rfc6928.txt
- [24] J. Wallerich, H. Dreger, A. Feldmann, B. Krishnamurthy, and W. Willinger, "A methodology for studying persistency aspects of internet flows," ACM CCR, pp. 23–36, 2005.
- [25] R. Netravali, A. Goyal, J. Mickens, and H. Balakrishnan, "Polaris: Faster page loads using fine-grained dependency tracking," in USENIX NSDI. Usenix, Mar 2016.
- [26] S. Sundaresan, W. De Donato, N. Feamster, R. Teixeira, S. Crawford, and A. Pescapè, "Broadband internet performance: a view from the gateway," in ACM CCR, vol. 41, no. 4. ACM, 2011, pp. 134–145.
- [27] J. Sommers and P. Barford, "Cell vs. wifi: On the performance of metro area mobile connections," in ACM IMC. ACM, 2012.
- [28] S. Ihm and V. S. Pai, "Towards understanding modern web traffic," in ACM IMC. ACM, 2011, pp. 295–312.
- [29] M. Butkiewicz, H. V. Madhyastha, and V. Sekar, "Understanding website complexity: measurements, metrics, and implications," in ACM IMC. ACM, 11 2011, pp. 313–328.
- [30] E. Pujol, P. Richter, B. Chandrasekaran, G. Smaragdakis, A. Feldmann, B. M. Maggs, and K.-C. Ng, "Back-office web traffic on the internet," in ACM IMC. ACM, 2014, pp. 257–270.
- [31] J. Qadir, A. Ali, K.-L. A. Yau, A. Sathiaseelan, and J. Crowcroft, "Exploiting the power of multiplicity: a holistic survey of network-layer multipath," *CoRR*, vol. abs/1502.02111, 2015. [Online]. Available: http://arxiv.org/abs/1502.02111
- [32] M. Wasserman and P. Seite, "Current Practices for Multiple-Interface Hosts," RFC 6419 (Informational), Nov 2011. [Online]. Available: http://www.ietf.org/rfc/rfc6419.txt
- [33] E. Nordmark, S. Chakrabarti, and J. Laganier, "IPv6 Socket API for Source Address Selection," RFC 5014 (Informational), Sep 2007. [Online]. Available: http://www.ietf.org/rfc/rfc5014.txt
- [34] D. Thaler, R. Draves, A. Matsumoto, and T. Chown, "Default Address Selection for Internet Protocol Version 6 (IPv6)," RFC 6724 (Proposed Standard), Sep 2012. [Online]. Available: http: //www.ietf.org/rfc/rfc6724.txt
- [35] R. Kiefer, E. Nordström, and M. J. Freedman, "From feast to famine: managing mobile network resources across environments and preferences," in *Proceedings of the 2014 International Conference on Timely Results in Operating Systems*. Usenix, 2014, pp. 7–7.
- [36] H. Abbasi, C. Poellabauer, K. Schwan, G. Losik, and Richard, "A quality-of-service enhanced socket api in gnu/linux," in *Real-Time Linux Workshop*, 2002.
- [37] B. D. Higgins, A. Reda, T. Alperovich, J. Flinn, T. J. G. uli, B. Noble, and D. Watson, "Intentional networking: opportunistic exploitation of mobile network diversity," in ACM MobiCom. ACM, 2010, pp. 73–84.
- [38] S. Deng, A. Sivaraman, and H. Balakrishnan, "All your network are belong to us: A transport framework for mobile network selection," in *ACM HotMobile*. ACM, 2014, pp. 19:1–19:6.
- [39] M. Welzl, S. Jorer, and S. Gjessing, "Towards a protocol-independent internet transport api," in *ICC*, 2011, pp. 1 –6.
- [40] K. Lee, J. Lee, Y. Yi, I. Rhee, and S. Chong, "Mobile data offloading: how much can wifi deliver?" in ACM CoNEXT, 2010.
- [41] J. Lee, Y. Yi, S. Chong, and Y. Jin, "Economics of wifi offloading: Trading delay for cellular capacity," *Wireless Communications, IEEE Transactions on*, vol. 13, no. 3, pp. 1540–1554, 2014.
- [42] Cisco Systems, Inc., "Architecture for mobile data offload over wi-fi access networks (whitepaper)," 2012. [Online]. Available: http://www.cisco.com/en/US/solutions/collateral/ns341/ ns524/ns673/white_paper_c11-701018.html
- [43] Y.-C. Chen, Y.-s. Lim, R. J. Gibbens, E. M. Nahum, R. Khalili, and D. Towsley, "A measurement-based study of multipath tcp performance over wireless networks," in ACM IMC. ACM, 2013, pp. 455–468.
- [44] C. Raiciu, C. Paasch, S. Barre, A. Ford, M. Honda, F. Duchene, O. Bonaventure, and M. Handley, "How hard can it be? Designing and implementing a deployable multipath TCP," in USENIX NSDI. Usenix, 2012, pp. 29–29.
- [45] S. Deng, R. Netravali, A. Sivaraman, and H. Balakrishnan, "Wifi, lte, or both?: measuring multi-homed wireless internet performance," in ACM IMC. ACM, 2014, pp. 181–194.
- [46] B. Han, F. Qian, S. Hao, L. Ji, and N. Bedminster, "An anatomy of mobile web performance over multipath tcp," in ACM CoNEXT, 2015.
- [47] A. Nikravesh, Y. Guo, F. Qian, Z. M. Mao, and S. Sen, "An in-depth understanding of multipath tcp on mobile devices: Measurement and system design," in ACM MobiCom. ACM, 2016.

- [48] L. Hartung and M. Milind, "Policy driven multi-band spectrum aggregation for ultra-broadband wireless networks," in *Dynamic Spectrum Access Networks (DySPAN)*. IEEE, 2015, pp. 82–93.
 [49] T. Dreibholz, R. Seggelmann, M. Tüxen, and E. P. Rathgeb, "Trans-
- [49] T. Dreibholz, R. Seggelmann, M. Tüxen, and E. P. Rathgeb, "Transmission scheduling optimizations for concurrent multipath transfer," in *Proceedings of the 8th International Workshop on Protocols for Future, Large-Scale and Diverse Network Transports (PFLDNeT)*, vol. 8, 2010.
- [50] S. Dawkins and others, "Taps working group charter," IETF Working Group Charter, IETF, IETF Working Group Charter charter-ietf-taps-01, September 2014.