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Multihomed Communication with SCTP

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Preface

SCTP (Stream Control Transmission Protocol) is a relatively new standards-track transport layer protocol in the IETF [RFC4960]. While SCTP was originally intended for telephony signalling over IP, it was recognized and designed as a general-purpose transport protocol for the Internet. Like TCP, SCTP offers a reliable, full-duplex connection with mechanisms for flow and congestion control. Unlike both TCP and UDP, SCTP offers new delivery options and several other features and services. This book focuses on SCTP multihoming: an innovative feature that allows a transport layer association to span multiple IP addresses at each endpoint. SCTP multihoming allows an endpoint to simultaneously maintain and use multiple points of connectivity to the network: thus, fixed and mobile users could connect to the Internet via multiple service providers and/or last hop technologies, and could use one or potentially all of those connections.

5

Low Delay Communication and Multimedia Applications

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This chapter focuses on low delay communication methods that SCTP can provide for real-time multimedia applications. The basic strategy to select a path with lower delay for transmission is described. Most works use SCTP internal variable SRTT (smoothed round-trip time) as estimation of the current delay on each path. It is calculated based on acknowledgments received (data and heartbeat). Other parameters such a delay threshold, time guard, losses are also employed to prevent unnecessary route changes. Quality improvements under specific scenarios are evaluated and several examples are shown. Asymmetric communication aspects are described and commented. Discussions and future directions finishes the chapter.

This is a Book Title **Name of the Author/Editor**

5.1 Introduction

5.1.1 Motivation: Need for Low Delay, Low Jitter for Real-Time Multimedia Applications

One important measure that can be optimized in a multihomed scenario is end-to-end packet delay. This is particularly relevant for real-time multimedia applications. Multimedia data are information conveyed in form of video, audio, image or a combination of them. Multimedia applications can be divided in three classes regarding the users' need for interaction. First class includes files that are transmitted to be used later. There is no need to immediate interaction. Second class includes application like video and audio on demand. There is a limited need for interaction in terms of pause, rewind, skip, etc. A fairly large amount latency are accepted for those actions. They can take a few seconds to take place. The third class of multimedia applications are related to interaction that must occur in real-time, like voice conversation or video-conferencing. There is need for a sub-second synchronization between the two parts involved in the conversation. The exact amount of accepted delay varies according to users expectation. ITU recommends value of 100 ms for maximum network delay for realtime high-interaction applications (VoIP, Videoconferencing) (ITU-T 2006).

There is an increasing demand for all types of multimedia communication. From the communication network point of view the class of interactive multimedia applications presents a greater challenge because of the stringent time limits requirements and large delays variations that data networks tend to exhibit. Buffers can be employed to accommodate delay variations but they are more suited for on-demand video and audio applications since extra delay is introduced.

Although other mechanisms of SCTP presented on previous chapters are still applicable to all multimedia transmission in a general form they are not specifically suited for real-time multimedia communication where the tight time constraints plays an important role on perceived quality by the end user.

Concurrent Multi-Transfer (CMT) is a good approach for bulk transfers but it does not necessarily provide the low delay and low jitter requirements for realtime applications. Voice transmission does not require high throughput (rate is between 5 to 80 kbps) but packet must arrive in timely fashion. Video conferencing would require some more bandwidth depending on the quality employed but low packet delay and jitter are the prominent desired feature. It is possible to imagine some scenarios where CMT would not be a good solution for delay sensitive applications. When one of the available paths have intrinsically higher delay than the other (e.g. satellite link or a path traversing many intermediate routers) it is not beneficial to use it for transmission concurrently with other faster path. Some packet would experience long delays unnecessarily and jitter would be increased significantly as well.

The failover mechanism of standard SCTP is certainly an important method to provide resilience for an established data communication session. But the time it takes for this standard approach to effectively switch transmission to alternate path is very high. It depends on the protocol parameter Path Maximum Retransmissions (PMR) which indicates the number of retransmission that can be carried out on a path. Thus PMR+1 retransmissions will denote a path failure and cause further transmission to use an alternate idle path. Protocol recommended PMR value of 5 results in a time to failure of more than one minute. Lower PMR values can be used (0 to 4) and failure detection time can be reduced significantly, but the possibility of spurious failovers increases accordingly. Caro Jr. et al. (2004) have investigated performance of various PMR values (0 to 5) concluded that PMR=0 performs well for simulated bulk transfers . Different policies for retransmission have been proposed. Three of them were evaluated in another work by Caro Jr. et al. (2006): All Retransmissions to Same path (AllRtxSame), All Retransmissions to Alternate path (AllRtxAlt) and Fast Retransmissions to Same path, Timeouts to Alternate path (FrSameRtoAlt), which attempts to balance the tradeoffs between the other two. The results obtained with network simulator for file transfer scenarios indicated policies perform differently depending on the network conditions. FrSameRtoAlt presented a good tradeoff balance.

Realtime multimedia applications need a very fast handover time. Ideally, the handover on a ongoing session should be such that the receiver application would receive all packets in due time during this process. An analytical estimation of the failover time in SCTP multihoming scenarios is provided by Budzisz et al. (2007).

5.1.2 Audio and Video Coding

The initial stage for digital transmission of video and audio is the coding and decoding process. This is accomplished by the coder and decoder (Codec) block. The analog information of video and audio needs to be converted to a digital representation and coded using the least number of bits to save transmission bandwidth. The process of coding can be simple and fast usually yielding a higher bit rate or complex and slow allowing smaller rates. Compression algorithms usually rely on psycho-visual or psycho-acoustic model of human perception to balance the number of bits where the received information will be more or less accurately perceived. Some speech coders employ other approach by performing a parametric coding of the voice source based on some predetermined mathematical model of voice formation. Small bit rates can be achieved (around 2 kbps) at the expense of loss of natural voice sounding. Speech Codecs can be divided into 3 classes: waveform, source and hybrid coders. The latter attempt to fill the gap between waveform and source coder. They allow better speech quality with a smaller increase in bit rate (to around 10 kbps) compared to source coders. A large number of Codecs are in use nowadays. Some of them are proprietary. ITU has many standards published for audio and video codecs. Recommendation H.263 ITU-T (2005) defines protocols to provide audio/visual communication over packet-based networks. It relies on a series of other recommendations that specifies codecs and call control procedures. A brief description of popular Codecs is given below.

1. Voice

   G.711 ITU-T (1988) is largely used in conventional telephony and also still used in some voice over IP (VoIP) applications due to its simplicity. It is a waveform coder where signal is sampled at 8 kHz and each sample is logarithmic compressed to be efficiently represented by 8 bits. This pulse-coded modulation (PCM) stream has a constant bit rate of 64 kbps. Other examples of waveform coders are G.722 and G.726 that uses adaptive differential PCM (ADPCM) to reduce the bit rate to 32 or even 16 kbps with a small decrease in speech quality. Examples of hybrid codecs are specified in G.729, G.729a and G.722.2. They employ Algebraic Code Excited Linear Prediction (ACELP) to yield good speech quality at low bit rate (6 to 23 kbps). These codecs are typically used in mobile or VoIP communication. G.729a is compatible with G.729 but requires less computation. G.722.2 implements Adaptive Multi-Rate Wideband (AMR-WB) algorithm and provides wider voice bandwidth 50–7000 Hz when compared to other codecs designed for conventional telephony (300–3400 Hz). Those speech codecs usually operate at constant bit rate but some codecs can use silence suppression to save bits or even choose to code the voice signal with varying number of bits yielding a variable bit rate (VBR). This means that lower bit rate can be achieved for the same speech quality or conversely more quality can be obtained for the same mean bit rate.

2. Video/Audio

   Video codecs which usually operates with variable bit rate. ITU provides a family of video coding standards. H.261 ITU-T (1993) is an old standard and designed to support two frame sizes: CIF (common interchange format – 352x288) and QCIF (quarter CIF – 176 x 144). It could operate at video bit rates from 40 kbps to 2 Mbps. H.263 ITU-T (2005) was an evolution from H.261 and was largely deployed in videoconferencing systems. The most recent Standard is H.264 ITU-T (2009) called Advanced video coding for generic audiovisual services. It is intended to cover a broad range of applications such as videoconferencing, digital storage media, television broadcasting and Internet streaming. It was designed in close collaboration with ISO/IEC Moving Picture Experts Group (MPEG) where it is named MPEG-4 Part 10 (ISO/IEC 2008). The standards are jointly maintained, they have identical technical content. MPEG-4 was first released in 1998 and absorbed many features of previous standards MPEG-2 and MPEG-1. They specify a collection of methods for compressing both video and audio. The compression of the audio part of the movie was initially standardized by MPEG-1 which provided three layers with increased compression level (MP1, MP2 and MP3 – the latter gained wide acceptance for audio storage and transfer becoming the popular coding method for portable players). MP3 was further enhance in MPEG-2 specification which also introduced new coding method called Advanced Audio Coding (AAC) sought to be the successor of MP3. MPEG-4 uses AAC.

5.1.3 Assessing the Quality: MOS, E-Model (ITU) and Others

Media degradation that occurs during transmission due to packet loss can be determined by comparing the received signal with its original version. There several ways to express this error. Mean squared error (MSE), Root Mean Squared Error (RMSE) and Peak Signal to Noise Rate (PSNR) are some examples. This objective measure of quality does not always reflects the true degree of perceived quality.

Perceived quality of the received media is a subjective matter but it is considered statistically in terms of the average of individual opinions. Mean opinion Score (MOS) provides a numerical indication of the perceived quality. Table I shows the scale that goes from 1 (bad) to 5 (excellent) regarding the quality or 1 (very annoying) to 5 (imperceptible) regarding the impairment (ITU-T 1996). The assessment of quality via MOS for several medias is standardized by ITU recommendations as described in Table 8.1.

[Table 2 about here.]

The assessment of quality via MOS for several media is standardized by ITU recommendations as described in Table 8.2.

[Table 3 about here.]

MOS assessment is a time-consuming and expensive procedure. There are however objective calculations that consider some forms of psycho-visual-acoustic elements of human perception. Those are called perceptual evaluation methods. Many methods have been proposed and this theme is still a matter of discussion. Some methods currently in use are:

PESQ – Perceptual evaluation of speech quality (ITU-T 1998).

PEVQ – Perceptual evaluation of Video Quality (ITU-T 2008c).

PEAQ – Perceptual evaluation of Audio Quality (ITU-R 2001).

A simple but not so accurate way to assess speech quality is obtained with E-model (ITU-T 2008b). This is a computation model to help transmission planners to build systems ensuring users will be satisfied with end-to-end transmission performance. It has been adapted from conventional telephony systems to VoIP transmissions. E-model relates the impairments due to several factors such as noise, codecs, network delay and jitter to provide one figure of merit named R rating according to Equation 5.1.

$$R = R_0 - I_s - I_d - I_e + A \tag{5.1}$$

where R_0 represents the transmission impairment based on the signal-to-noise ratio, I_s is the effect of impairments to the voice signal, I_d is the effect of impairments due to delay, I_e is the degradation of quality caused by low bit rate codecs and A is a compensation factor based on user expectation. A simple calculation tool and tutorial is provided by ITU-T (2008a). R rating can be converted to the MOS scale using the Equation 5.2.

$$R = \begin{cases} 1, & R < 0 \\ 0.035R + R(R-60)(100-R) \cdot 7 \cdot 10, & 0 \leq R \leq 100 \\ 4.5, & R > 100 \end{cases} \tag{5.2}$$

Table 8.3 describes range of values for R, its meaning, and equivalent MOS.

[Table 4 about here.]

Although E-Model does not takes into account the type of delay distribution experienced by the packets nor differentiate burst losses from uniform losses which might provide more accurate estimation it is simple, straightforward and fast to calculate. It is a useful tool for comparing VoIP quality in different simulated scenarios (Santos et al. 2007).

5.2 Delay-Centric Strategy: Switch Transmission Path for Low Latency

After the publication of SCTP standard (RFC2600) in 2000 many works began considering this new protocol as a good alternative to improve the tranport of multimedia traffic (Caro Jr. et al. 2001; Kashihara et al. 2003). A simple idea to use path delay to perform a handover with SCTP was proposed by researchers of Performance Engineering Lab at University College Dublin (Kelly et al. 2004). They considered a WLAN scenario where path latency may degrade due to several factors including traffic congestion. An estimation of the active path delay is obtained by using SCTP internal variable SRTT which is a low pass filtered version of the instantaneous RTT. When an ACK chunk is received, SCTP updates SRTT value for the active path according to Equation 5.3:

$$\text{SRTT} = (1 - \alpha)\text{SRTT} + \alpha\text{RTT} \tag{5.3}$$

where $\alpha = 0.125$. The inactive paths are probed less frequently by HB chunks. The interval between two probes is given by: $H_i = RTO_i + HB.interval(1 + \delta)$ where RTO_i is the latest RTT time-out value for destination i, and δ is a random value between -0.5 and 0.5. The standard parameter HB.interval is 30 s. RTO_i usually has a small value and δ serves to introduce some variability to the probe times. The RTT measurement on the secondary paths is considered merely a guide to the expected RTT if data traffic were to be carried on that path. In order to demonstrate the proposed scheme a WLAN scenario showed on Figure 1 was setup (Kelly et al. 2004).

[Figure 47 about here.]

Two multihomed SCTP hosts running Linux operating system were used as the endpoints of the association. They modified standard SCTP linux implementation available in (Stewart and Xie 2001) for delay-centric handover to one of the hosts only. No changes needed to be made to the second host providing a form of backward compatibility to standard SCTP. Host A can communicate to host B through two separate wireless networks with different IP address ranges. Once the association is established traffic flows on the primary path (0) to host B.

After some seconds several wireless stations start transmitting UDP packets to increase background traffic. This caused congestion in the cell and led to an increase in path delay. When they stop transmitting congestion ceases and delay on primary path returns to baseline level. This example illustrated the possibility of simple delay-centric strategy to perform a vertical handover on a congested network to a less congested one and back again to the first network when congestion ended.

Other work by the same group (Noonan et al. 2004) presented controlled simulations to further demonstrate the technique they also called Delay Sensitive SCTP (DS-SCTP). A network Simulator (NS2) was used to investigate some wireless scenarios. Figure 2 shows a representation of the employed topology. Wireless nodes 0 and 3 are multihomed. They can communicate through interfaces 1-4 over network A or interfaces 2-5 over network B. Scenario 1 represents a badly performing wireless LAN where Network A has 8 interfering sources, while network B has only 2. A voice application transmitting from node 0 to 3 over network A is started at 5 seconds, and at about 9 seconds, it is shown that DS-SCTP hands over to the least loaded network in order to improve performance.

[Figure 48 about here.]

Another scenario simulates a condition where both networks are lightly loaded with only 2 interfering sources. The load is divided quite evenly (5:4 ratio) between the two networks.

The experiment is run several times for 120 seconds while a hysteresis parameter is varied. A handover is programmed to occurs only when average delay on the idle path plus the hysteresis is greater than the average delay on the current path.

Figure 3 shows the network used for transmission each time for a 20 millisecond hysteresis. The superimposed square wave indicates the network used for transmission.

[Figure 49 about here.]

As summarized in Table 8.4, hysteresis reduces the number of handovers and does not seem to affect the ability to change paths when needed.

[Table 5 about here.]

The last scenario investigates a changing condition. Network A has 3 sources of traffic throughout the experiment. The load on network B starts with one source every 15 s has another source added. Figure 6 shows that initially network B, the less congested, is used but once it becomes congested network A is mostly selected.

The authors remarks that this scheme could be used to allow a user to select between a number of available networks, depending on which was able to offer the best level of service. Multimedia applications are often more sensitive to delay variations and this could be significant when networks use different types of technology. They plan to study the influence of delay and jitter in future work.

Another Path selection method (PSM) proposed by Kashihara et al. (2003) uses a different approach. Their selection algorithm considers the bottleneck bandwidth (BBW) which is obtained by packet pair measurement. The mobile host sends two HB packets consecutively, and BBW is calculated from the difference between the arrival times of the two packets (ΔT), as given by

$$BBW = \frac{\text{HB packet size}}{\Delta T}$$

They show the result of a simulation on NS2 where a mobile host roam within an overlap area between networks. A VoIP transmission at 64 kbps using SCTP switches to backup path when there is a decrease in the main path bandwidth. One problem with this approach is that bandwidth estimation based on packet pair is notably inaccurate (Prasad et al. 2003). This may lead to erroneous path change or prevent a path change when it is necessary. The idea of on-the-fly bandwidth estimation is very interesting and more investigation on dynamic scenarios may prove it robust and advantageous.

A more accurate estimation of path bandwidth is obtained by monitoring the path throughput. TCP Westwood (Mascolo et al. 2001) employ this method to adjust its transmission window. The same principle was applied to SCTP for a smart load balancing among the available paths (Fiore and Casetti 2005). At the beginning of the association round-robin is used to uniformly distribute the chunks among the paths, until a bandwidth estimation is obtained for every path. Then the chunks to be transmitted are distributed among the paths concurrently but in proportion to path bandwidth. The authors show some simulations on NS2 where this strategy performs better or equal than a simple concurrent multipath transmission. The idea is to avoid sending data over a channel as soon as some room in its congestion window is freed to prevent reordering delays at the receiver. They remark that packets sent over slower channels can arrive at the receiver much later than those sent over faster channels. This yields poor quality in sound and image display as well as large duplicate SACKs transmissions.

A similar approach that estimates path bandwidth but select only one path for transmission is proposed by Fracchia et al. (2007). Bandwidth on primary path is estimated by the ratio between the amount of transmitted data and RTT. On secondary paths the HB packet is replaced by a train of 6 packets (2 small, 2 large, 2 small) and their dispersion is used to estimate bottleneck bandwidth. Because only one path is in use at a time eventual head-of-line blocking that may occurs in CMT is avoided. The proposed technique also uses throughput on the primary path to differentiate losses due interferences in the wireless transmission or due to packet drop on router congested queue.

One can speculated approaches based on bandwidth will not work well for cases where path with higher bandwidth has intrinsic high delay (e.g. satellite links) compared to low latency paths with much smaller but sufficient bandwidth. The proposed method is though a good solution for bulk transfers and may perform well for realtime multimedia communication in many scenarios.

The absolute delay experienced by packets may have different consequences to the perceived quality depending on the type of Codec in use. Fitzpatrick et al. (2006) propose another metric to decide the handover: An estimation of user perceived MOS which is continuously calculated using E-Model. The method has the advantage of taking into account not only packet delays but also packet losses. They perform some simulations with an 802.11b WLAN scenario to show that an online estimation of MOS is similar to its offline calculation. The original heartbeat mechanism was modified to send multiple packets to imitate a VoIP traffic of a G.711 codec. A train of 25 heartbeat packets is sent to each endpoint in the association every T seconds. The chunks were set to have size of 80 bytes transmitted every 10 ms. If packets were excessively delayed they were considered lost. The loss rate was calculated considering a delay threshold of 300 to 350 ms (twice the maximum one way delay with the addition of encoding and decoding delays).

The estimated MOS and the MOS calculated for a CBR traffic were compared. The results are displayed in Figure 4 for different numbers of VoIP calls that represents the load of the network. It was verified a good agreement of both values for all three wi-fi rates. G.729 codec also also evaluated and yielding good MOS estimation.

[Figure 50 about here.]

This proposed scheme has a good potential of assessing more accurately the perceived user quality. But there is the overhead of transmitting a train of packets which needs to be weighted. Nonetheless, the HBs need to be sent only to the alternate paths because the primary path may have the MOS estimated from its own VoIP packets acknowledgments. This idea were further developed in later works where the authors called it ECHO: Quality of Service based Endpoint Centric Handover scheme for VoIP (Fitzpatrick et al. 2009, 2008).

A relevant question that needs to be answer for wide deployment of such delay-centric selection method is about the overall system stability when a large number of users would be using the same strategy to select minimum-delay path. Would the overall system utilization converge to a stable, low-delay situation for all users? The end systems may detect and select an alternate path with low delay but if they all change to this path it would be quickly congested. Another path may be select by all of them and the end systems would keep oscillating between congested paths causing overall communication to exhibit high latency. An initial study has shown that this may happen under some circumstances (Gavriloff 2009).

A simulated scenario on NS2 considered SCTP sources transmitting VoIP traffic of G.711 Codec. Two hosts are dual-homed to two ISP represented by two routers. There are two distinct paths between these two routers. Each SCTP agent estimates path latency through its internal SRTT variable. This update by every acknowledged received on the active path with standard parameter $\alpha = 0.125$. On the alternate path heartbeats (HB) are sent every second and HB-ACK updates this path SRTT accordingly. All agents are initiated on first path and will switch path if they detect that second path has smaller latency. They are randomly started over the first 5 seconds of simulation to avoid synchronous operation. Link capacities on each path (C) needs to be equal or greater than half of total aggregate traffic bandwidth (B). The C/B ratio was varied from 1 to 2 by adjusting the capacity for a given traffic bandwidth. Simulations considered 6, 12, 24 and 48 SCTP agents.

Three types of behavior were observed depending on simulation parameters. Figure 5 shows the number of agents on the first path during the 250 s of simulation. The number of agents on the second path is simply the complement to 6. First plot (a) shows the case where the agents keep switching paths during the whole simulation and the system never stabilizes. This occurs when there is no or just a small slack on the capacity. Second plot (b) shows the case where agents switch paths for a long time but they stabilize evenly after a while. Third plot (c) shows the case where agents switch paths and quickly stabilizes to an even distribution between the two paths. This is the case when the link capacities were not so tight compared to traffic bandwidth.

[Figure 51 about here.]

Table 8.5 shows whether path switching stabilizes for several C/B ratios and different numbers of SCTP agents. These results obtained by visual inspection suggests that a slack of around 10% is necessary to prevent unstable behavior for 6 SCTP agents. For greater number of competing SCTP sources stabilization occurs with smaller C/B ratios.

[Table 6 about here.]

In order to help overall system stability and prevent that the agents keep changing paths the delay-hysteresis parameter was considered. An agent switches path only if secondary path SRTT is smaller than current path SRTT less the hysteresis value. This approach did not displayed significant differences.

Another mechanism was proposed and investigated. Time guard is a period an agent must wait to switch path. If anytime during this interval the current path presents lower latency the switch operation is canceled. To further prevent syncronous operation among the competing agents a new random value is considered as the time guard for each agent when it detect lower delay on the alternate path. A uniform time distribution between 0 and 3 s was used. This mechanism showed some small improvement toward stability as can be verified in Table 8.6.

[Table 7 about here.]

Another measure of quality that was investigated was the estimated perceived user quality mapped to MOS (Mean Opinion Score) calculated by E-model. An unstable behavior induces higher latency for the VoIP traffic reducing final voice quality. Figure 6 shows the average calculated MOS for 10 simulations as a function of C/B ratio.

[Figure 52 about here.]

Low quality indicated by small MOS values occurs for tight capacity values. This is an indication that path oscillations were responsible for quality degradation. When the number of SCTP sources is increased better MOS is obtained. This is because less oscillations were observed but also because higher path capacity imply smaller transmission delays for each packet. Those results suggest that stability should not be a big problem for most scenarios but it may happen in limited circumstances witch involve high utilization of path capacity by small number of competing SCTP streams.

The same work (Gavriloff 2009) has also showed some results about the perceived user quality of one SCTP VoIP flow over stochastic background traffic. A simple scenario with two dual homed hosts was considered as displayed in Figure 8.

A poison process where packets inter-arrival time is give by exponential distribution is considered. In this mathematically well know model (M/D/1 in Kendall notation) packet delay in the system is given by

$$\bar{d} = \bar{s} + \frac{\rho \bar{s}}{2(1-\rho)}$$

where ρ is the link utilization (ration between traffic average bandwidth and link capacity). Because the link transport both this background and SCTP (CBR) traffic, actual utilization is described by: $\rho = \rho_B + \rho_S$ VoIP transmition simulated G.711 codec. Figure 7 shows the average delay as a function of link utilization. Aggregated traffic composed by CBR transmission and background traffic displays smaller delays when compared to pure M/D/1 system. Link capacity of 500 kbps was considered. Confidence intervals of 95% were obtained after the results from 30 simulations.

[Figure 53 about here.]

[Figure 54 about here.]

The same averaged delay was specified for both paths. One experiment with standard SCTP transmitting on only one path was run for comparison. Figure 9 shows the computed MOS for both cases single-path and multi-path.

[Figure 55 about here.]

Even though both paths on the long run have the same mean delay they are uncorrelated and for short periods of time one of the paths has smaller delay than the other. Path selection algorithm based on SRTT was able to select the lowest delays instants of each path keeping SCTP overall latency low. The resulting MOS was always high while single path transmission quality degrades for increased mean delay. It can be remarked that this is an specific result for Poisson Traffic (M/D/1) which may not be representative of real background traffic pattern. Real traffic tend to be more variable and bursty exhibiting fractal characteristics. Long tail distributions are frequently used to model this behavior. Nevertheless it is expected that delay-centric selection algorithm would perform well or even better for traffics with higher variability.

5.3 Asymmetric Round Trip Path Approach for One-Way Delay Optimization

The strategy to select the lowest delay path can be further refined if one-way delay is considered in the place of round-trip delay. Historically, most delays estimations are based on the simplifying assumption that forward and return delays are the same and equal to half the round-trip time (RTT) of a route. This is rarely true because of the asymmetry present on most traffic profiles of internet applications, e.g. web surfing, file download and video streaming. Access technology like ADSL (Asymmetric Digital Subscriber Line) try to accommodate such imbalance by providing higher bit rate in the downstream direction.

Different delays on forward and return paths have been a concern for TCP congestion control (Barakat et al. 2000). TCP delay estimation for each side of is based on RTT measurements and cannot distinguish if an increase in delay is due to the forward or reverse path, possibly resulting in under-utilization of the available bandwidth. This problem has been addressed by TCP extensions (Fu and Liew 2003) that change the way congestion window is calculated.

Multi-homing introduces a whole new perspective in the sense that not only one-way delay over the available paths can be compared but an action can be taken to select the most appropriate path for each direction, independently. The first effect of this strategy is to increase the number of paths combinations that can be utilized and the chances of a lower delay communication. A collateral effect is that by using the least congested paths in each direction this asymmetric communication may be helping to 'fill-in' the bandwidth gaps induced by other applications and contributing to balance the overall delay on both directions. This could be considered beneficial for TCP and other applications that rely on symmetric RTT that might be running over the same paths.

The approach to used asymmetric paths for low delay communication has been proposed recently (Ribeiro and Leung 2005, 2006). It is shown how one-way delays can be compared in order to select the lowest-delay path for transmission in each direction. An example listing forward and reverse path names is displayed in Figure 10 where hosts H1 and H2 are multihomed to 3 and 2 ISPs respectively. Circles 1, 2 and 3 represent the network interfaces of H1, while 4 and 5 represent the network interfaces of H2. It is assumed that packets can be transmitted freely between any pair of interfaces (nodes) of the two hosts. Each forward path (relative to H1) is designated f_i and the corresponding reverse path is designated r_i, where $i = 1, \ldots, P$ ($P = 6$ in Figure 10) spans the one-way paths existing in each direction. Transmission delay over the corresponding one-way path is d_{f_i} (or d_{r_i}).

Latency on the round-trip paths could be estimated by sending special probing packets that are returned by the receiver via a specific return path (e.g, via heartbeat chunks (HB), possibly extended to specify the return paths for the replies).

To determine relative delays for the forward paths, not all P^2 RTT combinations are required, but a set of P probes along different forward paths returned via the same reverse path are sufficient. The difference in RTTs between round-trip paths $f_i r_k$ and $f_j r_k$ yields the delay difference between forward paths f_i and f_j:

$$\begin{aligned} \mathrm{RTT}(f_i r_k) - \mathrm{RTT}(f_j r_k) &= d_{f_i} + d_{r_k} - (d_{f_j} + d_{r_k}) \\ &= d_{f_i} - d_{f_j}, \qquad i \neq j; \quad i, j = 1, \ldots, 6 \end{aligned} \tag{5.4}$$

To determine relative delays for the forward paths, a set of P probes along different forward paths returned via the same reverse path is sufficient. The difference in RTTs between round-trip paths $f_i r_k$ and $f_j r_k$ yields the delay difference between forward paths f_i and f_j:

Forward path with lowest latency is determined by comparing their relative latencies. The same strategy can be applied by the other host (H2) to determine its forward path with lowest latency.

[Figure 56 about here.]

Table 8.7 displays an example of hypothetical one-way delays in milliseconds for each forward and reverse path. Each cell is the sum of forward and reverse delays representing the path round trip time. Forward path delays vary from 10 (f_6) to 800 ms (f_1) while reverse path delays vary from 50 (r_1) to 1400 ms (r_6). Shaded diagonal cells in the main diagonal give the RTTs of the symmetric two-way paths. A simple delay-centric path selection method that measures only symmetric RTTs will choose the lowest among these shaded values and pick path $f_2 r_2$ which has a RTT of 400 ms. However, packets would experience a 300 ms delay over the forward path, which may exceed the delay requirement of some real-time multimedia application. The proposed asymmetric algorithm will make host H1 to select forward path f_6 which has the lowest delay value (10 ms) among the forward paths and host H2 to select r_1 which has the lowest delay (50 ms) among the reverse paths. Resulting RTT will be 60 ms.

[Table 8 about here.]

Another example of the advantage of considering asymmetric paths is illustrated by figures 11–13 (Ribeiro and Leung 2005). This scenario considers two hosts multihomed through two access networks. Forward paths are designated by lowercase letters a,b,c & d while reverse paths are designated by the corresponding uppercase letters A, B, C & D. Time evolution of forward and reverse path delays are displayed on Figures **??** and **??**. They are simply linearly changing values with added Gaussian random noise over a 60 s time interval sampled every second.

[Figure 57 about here.]

[Figure 58 about here.]

[Figure 59 about here.]

Each side tries to use its lowest delay forward path. It is supposed that the host H1 samples its forward path latency every other 2 s, while host H2 do it with 4 s interval starting at 2 s. Figure 12(a) shows what would be the selected forward paths (circles) by H1 which calculates it every 2 seconds and the smallest delay path for every second (dots). Transmission starts with path d, then at t=9 s changes to path c, and finally at t=32 s changes to path b. Notice that the smallest delay path (dot) is not selected all the times because the H1 does not sample the path every second. On the same figure it can be verified that the experienced delay closely follows the minimum available delay. Figure 12(b) shows the same type of plot for the reverse path, or the forward path with respect to the H2. Figure 13 compares delays for the all symmetric round-trip paths with asymmetric path. Lower delays can be obtained during most part of the transmission.

This example illustrated how SCTP can benefit from asymmetric paths communication when delays on forward and reverse directions are not the same. Another consequence is that the number of round trip paths to choose from are higher increasing the chances to find a not congested path.

The number of asymmetric round-trip paths that can be used for communication deserves some discussion. Consider the example in Figure 10 where host H1 is multihomed with $M = 3$ interfaces and host H2 is multihomed with $N = 2$ interfaces. There are $P = M \times N = 3 \times 2 = 6$ possible one-way paths in each direction.

Although all P paths are virtually possible, only a subset of that can actually be used in standard SCTP/IP implementation. When SCTP in host H1 wants to send a package to host H2 it hands over the package to layer-3 for delivery. Because routing is based on destination address exclusively, the IP layer can only send the package to one of the two possible addresses (A4 or A5). The output interface and hence source address to be used is determined by its routing table which is fixed and previously set. Let us suppose that the output source addresses were, respectively, A1 for destination address A4, and A2 for destination address A5. On the other side, when host H2 wants to send a package to host H1 it would have 3 possible destination addresses to send the package. If the paths to A1 and A2 are simultaneously down for some reason, communications between H1 and H2 will break because the routing table in host H1 does not know how to send a package to host H2 through its third interface IP3 as remarked by Stewart and Xie (2001). This example suggests that the number of different one-way paths that actually could be used is $\min(M, N)$. The number of symmetric round-trip paths is the same.

SCTP alone should be easily modifiable to work with asymmetric paths without any change to layer-3. The number of possible symmetric and asymmetric round-trip path combinations would be $\min(M, N)^2$. In this example there are $\min(3, 2)^2 = 2^2 = 4$ round trip paths that could be used considering a standard destination routing IP layer. They could be f_1r_1, f_6r_6, f_1r_6 and f_6r_1 for a particular routing table setup.

If a cross-layer optimization on SCTP/IP stack that allow layer-4 to tell layer-3 which outgoing interface to use then all possible one way paths $(M \times N)$ in each direction could be used. This is a total of $(M \times N)^2$ possible round-trip path combinations or $(3 \times 2)^2 = 36$ (!) in this example.

The basic idea of exploring asymmetric round trip communication is to have more options to choose from to find a less congested one-way path. This could be accomplished with standard IP implementation, which gives $\min(M, N)^2$ total round-trip paths, or with cross-layer optimized stack, which gives $(MN)^2$ options. Table 8.8 summarizes the relative gain of round-trip path options when asymmetry is also considered for both cases. For simplicity $M \geq N$ is assumed.

[Table 9 about here.]

In a mixed scenario where only one end system has the cross-layer optimization it is easy to verify that the number of possible round-trip path combinations is M^2N when only the stack in host H1 is optimized or MN^2 when only the stack in host H2 is optimized.

The RTT probes could be obtained in different ways. 1) A control package can be transmitted and the elapsed time to receive the corresponding acknowledgment gives the RTT. 2) During normal operation of SCTP, SRTT are constantly being updated on the primary communication path and less frequently on alternate paths when heartbeats are ACKed. 3) The Heartbeat (HB) mechanism could be slightly changed to suit the needs of this method. The modification required is to respond to all the arriving HBs by returning ACKs over a common return path (e.g. primary path) and not to the incoming address. It is important to note that the main function of HBs is not to probe for path delays originally but to test if a destination address is active (its default interval is 30 s). A delay probe nonetheless, can estimate whether a given address is inactive if it times out before receiving a reply. Although both functions are very similar in nature, they could should share a common implementation or a separate control chunk may be used for delay probes.

5.3.1 Simulations

Version 2.29 of the network simulator (NS2) (*Network Simulator 2* n.d.) was used to test the asymmetric path selection method. Standard SCTP implementation (Caro and Iyengar 2005) has been modified to support asymmetric communications. The heartbeat mechanism has been modified to allow for probing all the possible forward paths to the destination. Instead of replying to the source IP address that originated the HB, all HB-ACKs go from current interface to the primary destination IP address of the correspondent node. This way all the packages take the same return path and allow RTT comparisons at the sender.

The topology used on simulations is illustrated on Figure 14. Both sender and receiver are multi-homed with two interfaces. The routers are interconnected in a way that all cross-combinations of source-destination addresses between the hosts are possible.

[Figure 60 about here.]

The symmetric round-trip paths are: f_1r_1 (nodes 6-8), f_2r_2 (nodes 7-9), f_3r_3 (nodes 6-9) and f_4r_4 (nodes 7-8). Other asymmetric round-trip may be selected as well. All links have a capacity of 2 Mbps and a fixed delay of 1 ms. Drop-tail queues are used and the length of each queue is set to 750 kB thus allowing enough queue space to prevent packages from being dropped.

The simulated scenario involved only CBR cross-traffic at different rates to establish variable queue occupancy and hence to provide dissimilar delays along the available paths. Figure 15 shows the delays experienced by cross-traffic UDP packets when traversing the intermediate routers. This is a baseline scenario for comparison where only cross-traffic is present and SCTP transmission is not active.

[Figure 61 about here.]

Paths f_1, f_2 and f_3 starts with their queue not empty and background traffic at this path has a deficient CBR rate that causes their queue occupancies to decrease as time progresses. Forward path f_4 starts with lowest queue occupancy corresponding to a delay of 20ms at t=0, but background CBR traffic progressively fills up the queue to 200ms at t=50s. Note that at around t=16ms, f_3 becomes the lowest delay forward path as can be seen in Figure 15(a). Reverse path r_1 begins with its queue almost empty (10 ms delay at t=0) and has a growing delay up to 100 ms delay at t=50 s. Path r_4 has an initial delay of 300ms but its queue occupancy decreased with time ending up with 10ms delay at the end of the simulation. At around t=38 path r_4 becomes the lowest delay reverse path, as can be seen on Figure 15(b). SCTP packet size was 100B and packet interval was 0.4 s giving a data rate of 2 kbps. SCTP association started with nodes 1 and 4 set as primary addresses thus using path f_1. HBs to probe for path delays were sent every 2s and instantaneous RTT comparisons were used to select the forward path.

A comparison with standard SCTP is illustrated by Figure 16: (1) standard SCTP would transmit only to the primary destination and would not change its destination since there were no path failures. Data packages would experience a delay from 200 ms increasing to 230 ms represented by solid squares. (2) Symmetric delay-centric SCTP would have the same behavior because in this particular case primary round-trip path $f_1 r_1$ has the lowest RTT. (3)

Asymmetric-path SCTP experiences lower transmission delays of 25 to 60 ms for most part of the communications as shown by the circles. After the first HB, at t=2.2, path f_4 is detected as having the smallest one-way delay and hence selected for transmission. Around t=12 s path f_3 becomes the lowest delay path. As soon as this is detected by the next HB probe around t=14 s it is selected for transmission.

[Figure 62 about here.]

The proposed minimum-delay asymmetric SCTP is aimed at real-time applications which requires latency minimization. Although no considerations have been given to bandwidth availability, this method may show some benefits even when there is some demand for bandwidth. The variable delays that packages experience during transmission are due to queue occupancy on intermediate routers either by packages from cross-traffic or by previous packages from the same flow. This latter case indicates a pressure for more bandwidth by the current flow. Depending on the traffic characteristics of the flow, the lowest-delay path selection method may also be able to deal with the increase in latency due to additional bandwidth requirements of the flow itself. This would be related to how frequent the HB probes can successfully detect the increase in path latency. Figure 17 displays such situation. The SCTP flow has its data rate doubled to 4 kbps. At times t=39, 41, 46 and 48 s the selected path keeps changing between f_3 and f_2. The momentary increase in delay at each of these paths is due to self traffic because otherwise they would exhibit the same background traffic delay pattern previously obtained.

[Figure 63 about here.]

RTT probe interval can be adjusted as a compromise between responsiveness to delay changes on the paths and the overhead traffic generated by the probes themselves. Some optimizations can be done when implementing this method. The current path in use do not need to have HB probes sent on it because SRTT is already being estimated by normal protocol operation based on SACKs received. The alternate paths also have their SRTT updated when an HB-ACK is received. The required modification is to have the HB-ACK also sent by the receiver to its primary destination. Delay monitoring is to be done for every SCTP association. Some implementation may choose not to repeat measurement to same destination address and share the SRTT information among associations with same endpoint (layer 3 address).

5.4 Discussions

Multihoming capability of SCTP has established an important framework that is being explored to improve communication performance. Realtime multimedia applications can greatly benefit from such approach. Wireless devices in heterogeneous networks are transforming the classic network scenario. Networks with diverse characteristics in terms of signal propagation, bandwidth, delay and packet loss offer different opportunity for multimedia traffic distribution. The dynamic nature of such environments where those characteristics may change in a matter of seconds bring up the challenge to have a good strategy for selecting the most adequate network for use at a given time.

The simple idea to select the path which has the current lowest end-to-end estimated delay is very promising. It has been demonstrated through simulation and experimental tests that this method provides seamless handover when necessary granting low overall latency for the realtime communication. Simulations with controlled background traffic have shown how users' perceived quality in terms of MOS calculated by E-model behave as a function of the delay in the network for a VoIP CBR transmission. A simplistic scenario where background traffic was generated by a Poisson process with the same mean packet interval on both paths presented an interesting result. Delay-centric handover method can maintain an elevated quality even for high mean delay path values where a single path transmission would have its perceived quality significantly reduced. This is a good demonstration of the ability of the method to take advantage of the short periods of less traffic on each path. It is expected a similar behavior for real traffic which exhibit higher variability and fractal properties and are often modeled by long tail distributions.

The problem of greedy selection of the paths when many simultaneous SCTP sessions employ the same delay-centric method was analyzed. Simulations have shown that oscillations may occur causing a reduction in voice quality (MOS as calculated by E-model) compared to an ideal distribution of the sessions between the two paths. Nevertheless, this unstable behavior only occurred when the utilization factor due to VoIP CBR transmissions was very high (close to unity). It can be argued though that this is not an expected situation because one do not foresee to operate an aggregate of CBR-only traffic close to this limit. On the other hand, conventional single-path transmission would probably not reach an ideal balance among the paths. Any unbalance would cause a significant degradation of the MOS for the path with more VoIP sessions (which would have increasing delays on its queues). Anyway the delay-centric method was not robust enough to deal with this boundary situation gracefully. There is room for some improvement in this sense.

Asymmetric round-trip path method for selecting one-way path with lowest delay is a very interesting approach to further improve the low-delay communication requirement. There is a increase in the number of possible round-trip paths that can be used. There are M^2N more options for a $N \times M$ multihomed system when compared to symmetric round-trip paths. The lowest one-way path can be used for the multimedia transmission in each direction. Examples illustrated the case when a forward path has the lowest latency but its corresponding return path has a high latency. A symmetric round-trip path method does not select this path for transmission. Asymmetric method resolve this problem as it can compares one-way delays.

Simulations on NS2 demonstrated the operation of the asymmetric method in a scenario where a variable number of TCP background traffic induces delay on both one-way paths differently. The modified SCTP implementation was able to compare SRTTs for each asymmetric round-trip path and determine the lowest one-way path.

For a successful wide deployment of low delay SCTP communication further investigations should be carried on. How the loss rate would interfere with path selection mechanism is certainly an important topic specially for wireless networks. The interaction between delay-centric selection and SCTP failover mechanism is another point that needs further exploration. For a mobile device it will be useful to combine the path selection method with other layer-2 information like RSS (Received Signal Strength). So far many works have addressed those situations separately but few works have studied them together. Also the fine-tune of important parameters like HB rate, delay histeresis and SRTT calculation should be further examined.

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Network Simulator 2

Network Simulator 2 n.d. http://www.isi.edu/nsnam/ns.

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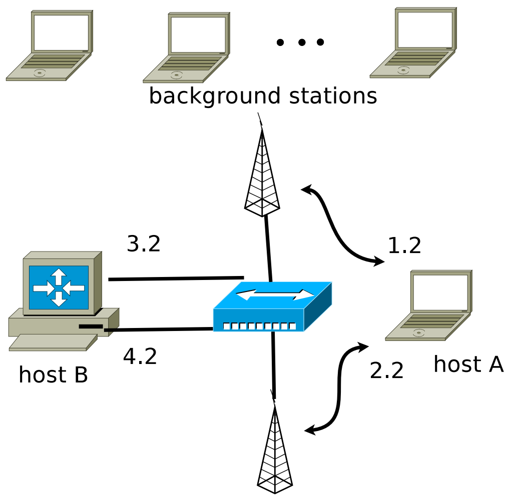


Figure 1 Equipment Setup

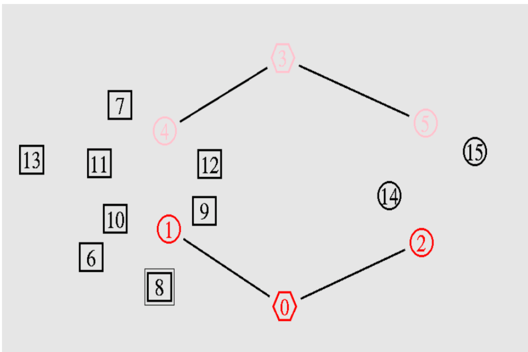


Figure 2 Wireless topology of the experiment

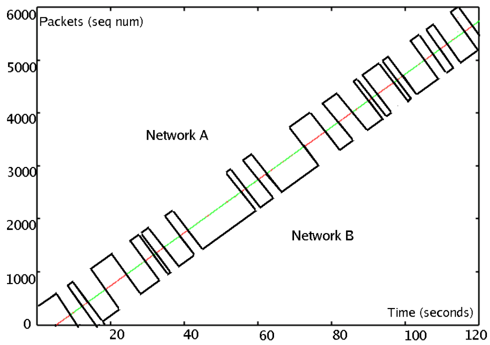


Figure 3 Network selected for transmission

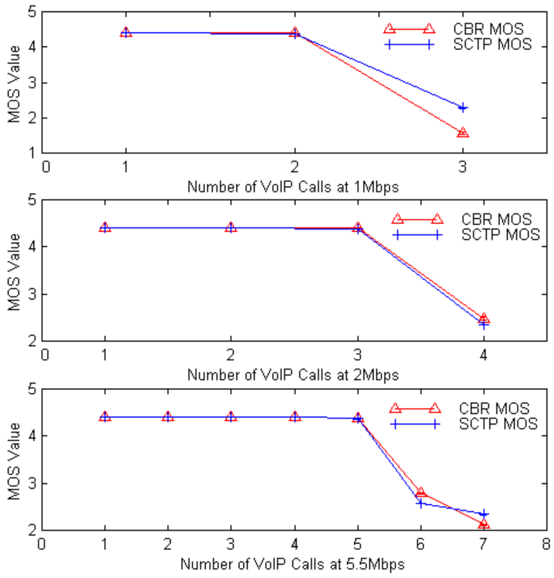
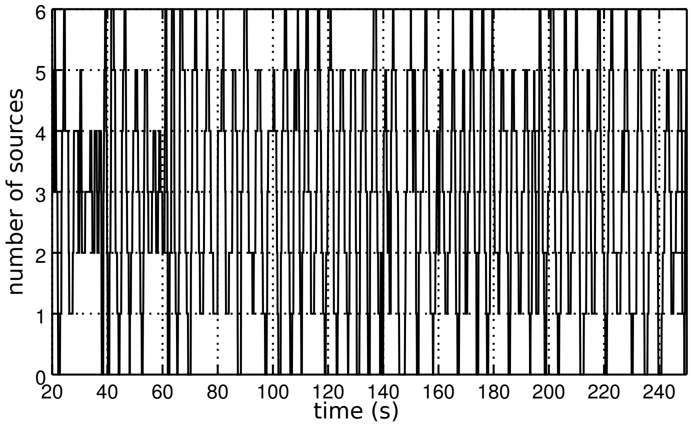
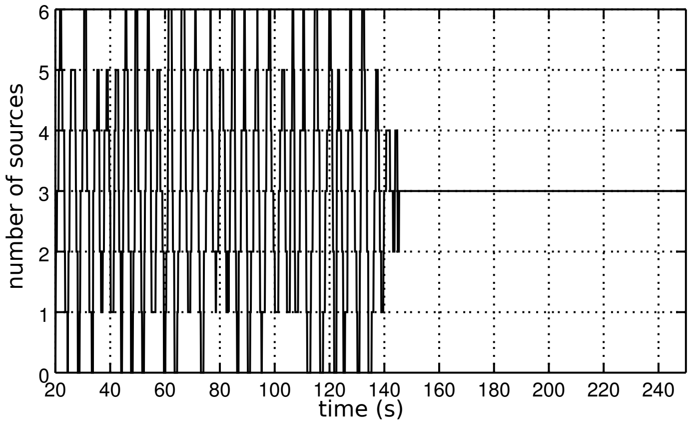


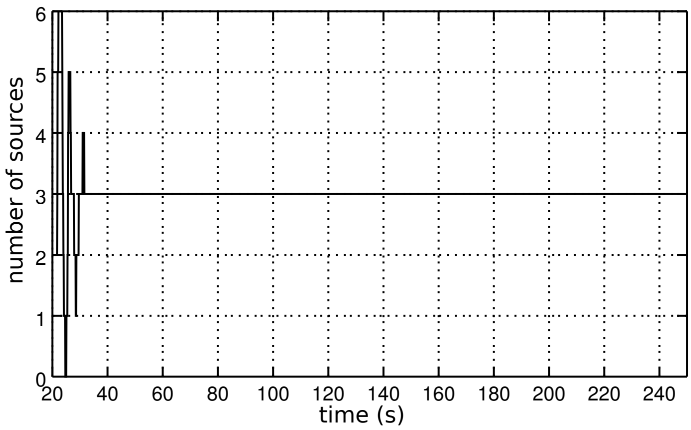
Figure 4 CBR and SCTP MOS for G.711 at different data rates



(a) Unstable



(b) Partially stable



(c) Stable

Figure 5 Path selection oscillations

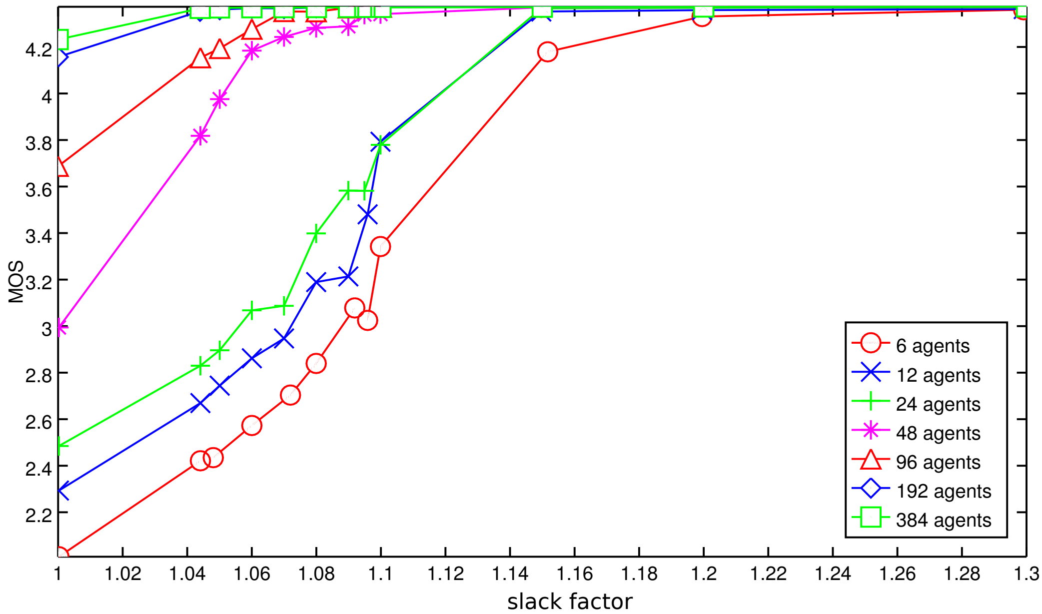


Figure 6 Calculated MOS for different number of competing SCTP transmissions versus slack factor

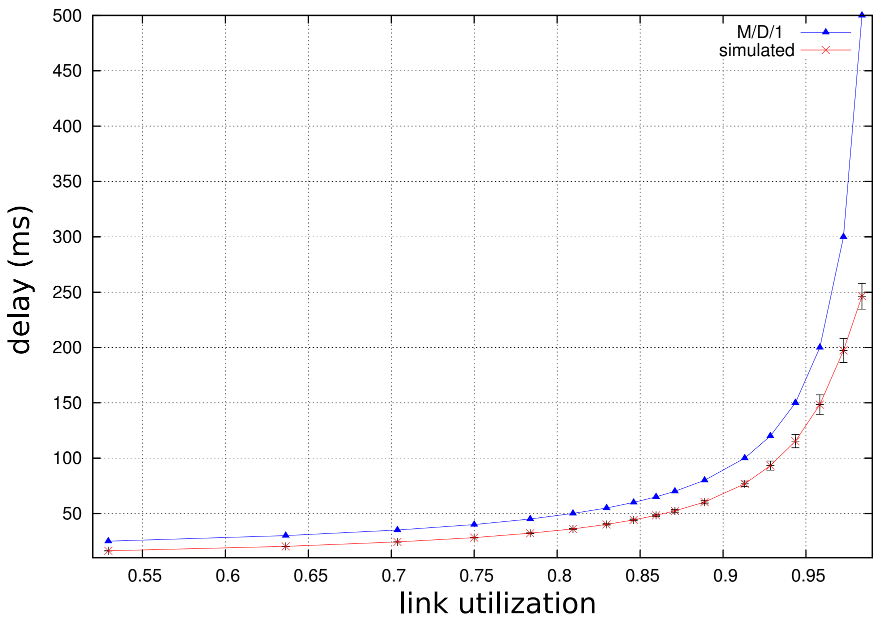


Figure 7 Averaged delay versus system utilization

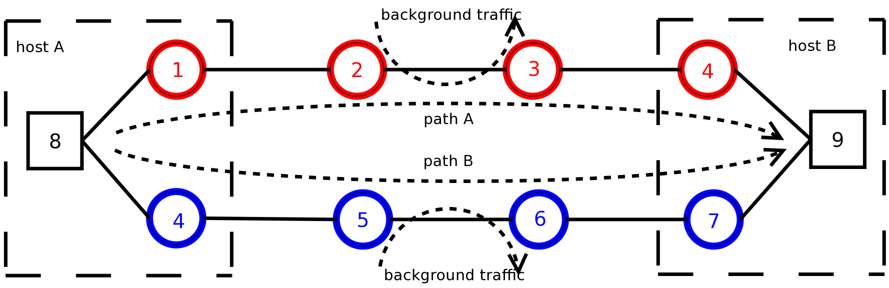
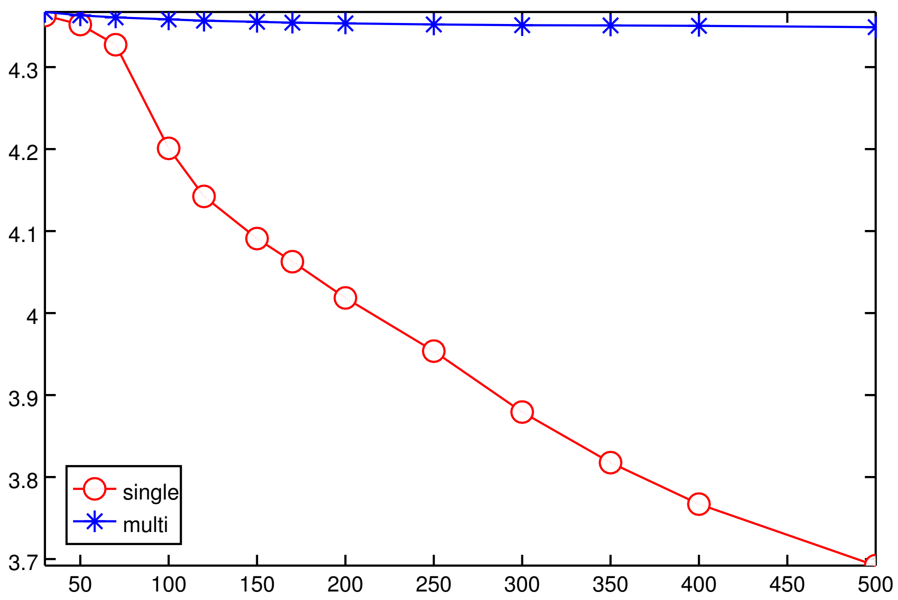


Figure 8 Topology used on simulation



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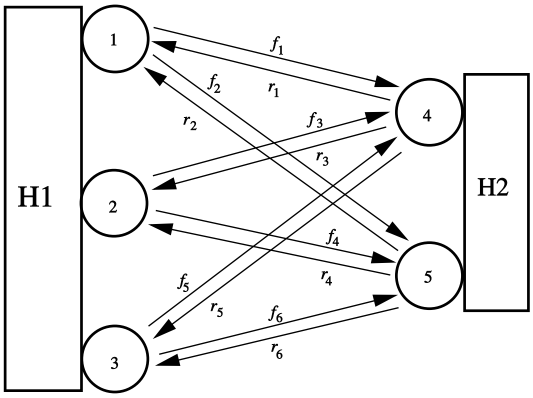
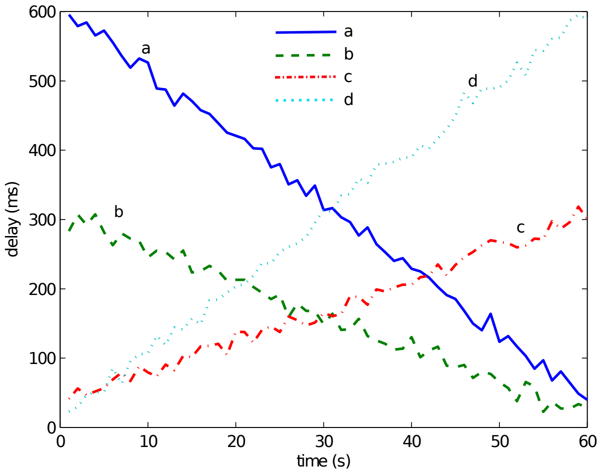
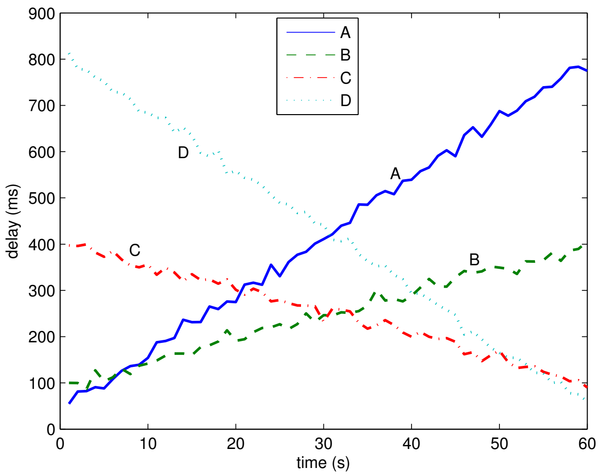


Figure 10 Multi-homing with asymmetric paths. Forward and reverse paths are identified.

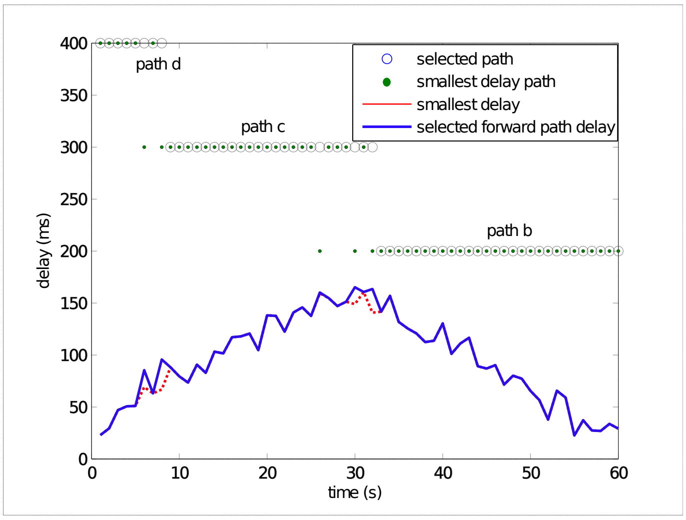


(a) Forward paths delays

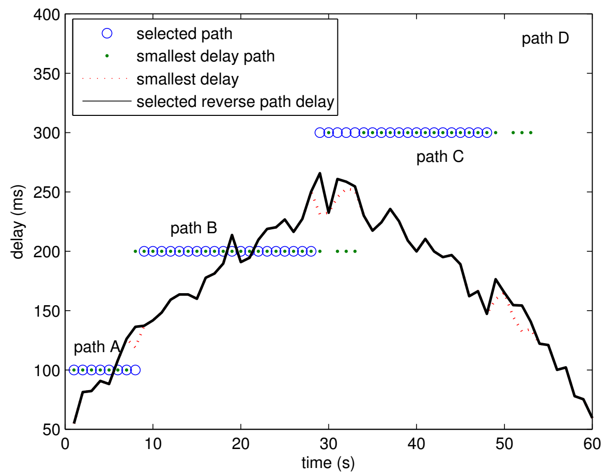


(b) Reverse paths delays

Figure 11 Forward and Reverse paths delays over time



(a) Selected forward paths



(b) Selected reverse paths

Figure 12 Paths delays and selected paths

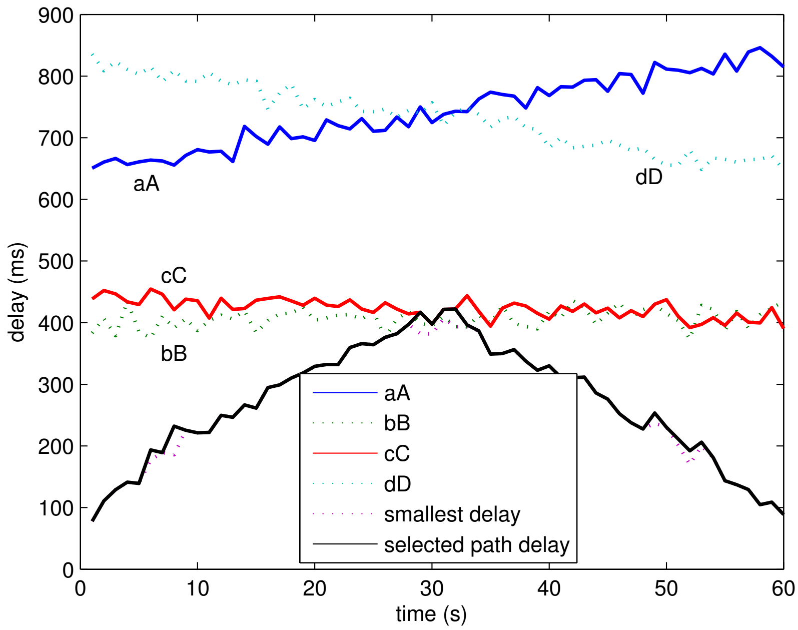


Figure 13 Selected Round-trip paths and delays

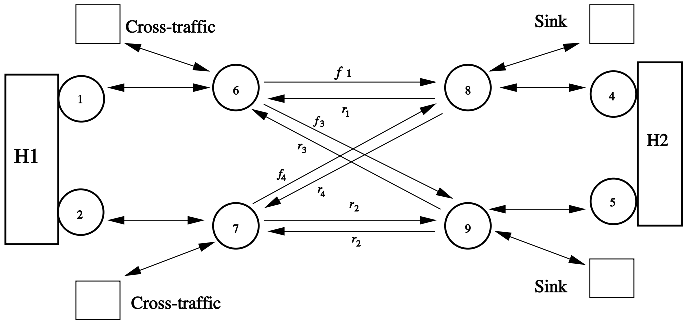


Figure 14 Network topology in simulations - SCTP and cross-traffic

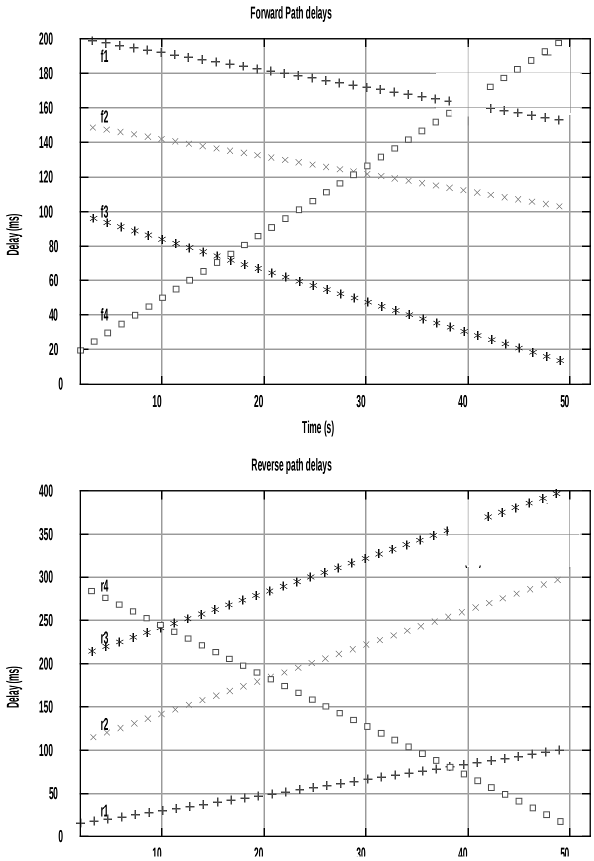


Figure 15 *UDP packet delays - CBR traffic only. (a) forward paths. (b) reverse paths.*

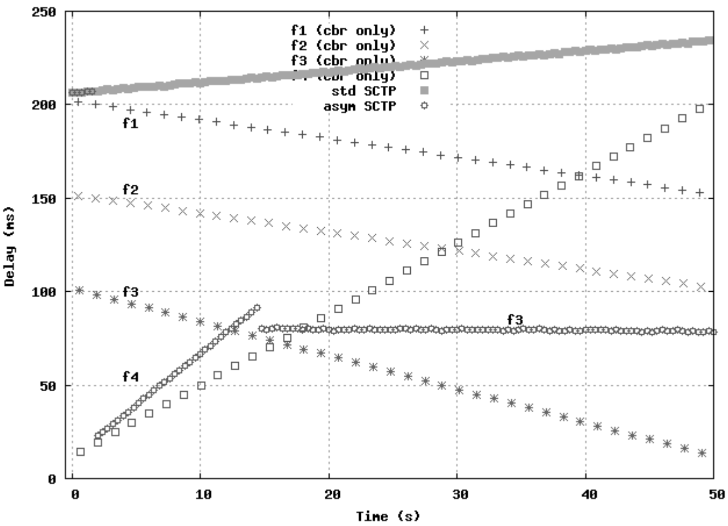


Figure 16 *Comparison between: (1) standard SCTP (primary path only: f_1); (2) Symmetric delay-centric SCTP (primary round-trip path ($f_1 r_1$) has the lowest RTT; (3) Asymmetric SCTP (paths f_1, f_4, f_3); (4) Delay of CBR background cross traffic (with no SCTP traffic) was also plotted for reference.*

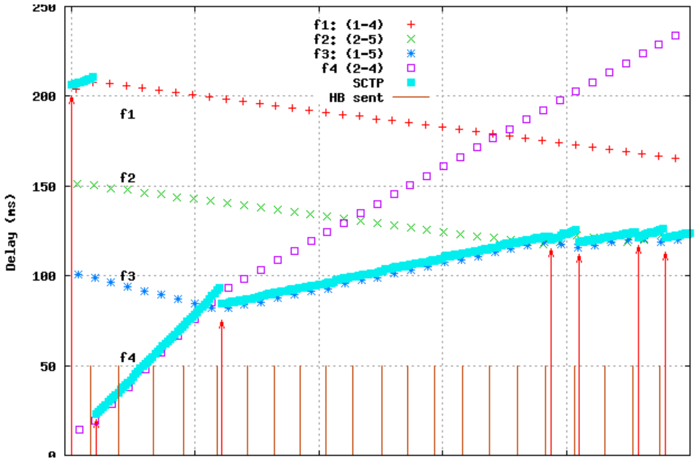


Figure 17 *Delays on forward path with SCTP bandwidth increase (packet size=100, interval=0.2 s). Vertical lines at the bottom indicate when HB probes were sent. Vertical arrows indicate when transmission is switched to a new path with lower delay. At t=38 s SCTP switches back and forth between paths f_2 and f_4 due to self traffic loading on current selected path.*

List of Tables

Table 8.1 MOS: quality and impairment scales

Quality			Impairment
5	Excellent	5	Imperceptible
4	Good	4	Perceptible, but not annoying
3	Fair	3	Slightly annoying
2	Poor	2	Annoying
1	Bad	1	Very annoying

Table 8.2 Recommendations for subjective assessment of quality

Category	Recommendation
Voice	ITU-T P.800 Methods for objective and subjective assessment of quality
Video	ITU-R BT.500 Methodology for the subjective assessment of the quality of television pictures ITU-T P.910 Subjective video quality assessment methods for multimedia applications
Audio	ITU-R BS.1284 General methods for the subjective assessment of sound quality (audio alone) ITU-R BS.1286 Methods for the subjective assessment of audio systems with accompanying picture ITU-T P.911 Subjective audiovisual quality assessment methods for multimedia applications

Table 8.3 Speech Transmission User Satisfaction

E-Model Rating	quality	category	MOS
$90 \leq R < 100$	Best	Very satisfied	$4.3 \leq MOS < 100$
$80 \leq R < 90$	High	Satisfied	$4.0 \leq MOS < 4.3$
$70 \leq R < 80$	Medium	Some users dissatisfied	$3.6 \leq MOS < 4.0$
$60 \leq R < 70$	Low	Many users dissatisfied	$3.1 \leq MOS < 2.6$
$50 \leq R < 60$	Poor	Nearly all users dissatisfied	$1.0 \leq MOS < 2.6$

Table 8.4 Increasing Hysteresis with lightly loaded networks

Hysteresis (ms)	# Handovers	Ratio A:B
0	69	1 :1.25
4	56	1 :0.88
10	43	1 :1.42
20	28	1 :0.96
40	14	1:114

Table 8.5 Stabilization of path switching regime for different number of SCTP competing agents and path capacity / bandwidth ratio

Agents	C/B ratio							
	1.000	1.050	1.070	1.080	1.090	1.095	1.100	2.000
6	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\simeq$	=
12	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\simeq$	=
24	$\approx$	$\approx$	$\approx$	$\approx$	$\simeq$	$\approx$	=	=
48	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	=	$\simeq$	=

Table 8.6 Stabilization of path switching regime with time guard mechanism

Agents	C/B ratio							
	1.000	1.050	1.070	1.080	1.090	1.095	1.100	2.000
6	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\approx$	$\simeq$	=
12	$\approx$	$\approx$	=	=	=	=	=	=
24	$\approx$	$\approx$	$\approx$	$\approx$	=	=	=	=
48	$\approx$	=	=	=	=	=	=	=

Table 8.7 One-way delays and all RTT combinations in ms

			reverse path					
			r_1	r_2	r_3	r_4	r_5	r_6
		one-way delay	**50**	**100**	**400**	**800**	**1200**	**1400**
forward path	f_1	**800**	850	900	1200	1600	2000	2200
	f_2	**300**	350	400	700	1100	1500	1700
	f_3	**100**	150	200	500	900	1300	1500
	f_4	**40**	90	140	440	840	1240	1440
	f_5	**20**	70	120	420	820	1220	1420
	f_6	**10**	60	110	410	810	1210	1410

Table 8.8 Relative gain of round-trip options compared to standard SCTP

	standard stack	selectable output
Symmetric round-trip paths	1	M
Symmetric and Asymmetric round-trip paths	N	M^2N