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Abstract:

This document collects validation plans and results of WP4. The document describes for each validation topic the objective of the validation, the selected test scenarios, the applied tools and test environment, the expected results, the results, and the involved partners.

Keywords:

WP4 validation plans, test scenarios, validation tools and environments, expected results

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Executive Summary

This document describes the validation plans and results of traffic management solutions proposed in MEVICO. For each plan there is a description of objectives, validation scenarios, validation tools, expected results, results, applicability of the results and partners involved. The sections on validation objectives collect the claimed challenges resolved by the particular technology, and the key performance indicators (KPIs). Validation scenarios describe the test cases where the technologies show their benefits, furthermore any important issues on the planned test cases. The sections on validation tools describe the validation environments.

The proposed solutions cover three traffic management areas, and one network planning area:

- microscopic traffic management
    - traffic classification with deep-packet inspection and bulk traffic analysis
    - traffic modeling for better demand estimation: traffic variability analysis
    - passive, network-side QoE estimation of video traffic
    - capacity aggregation and load balancing for TCP traffic using proxy-based MTCP
- macroscopic traffic management
    - Load balancing using network-based IFOM
    - ALTO assisted Connection Management in EPS

    - evaluation of novel multi-criteria cell selection methods in hierachical cellular network with extended criteria set
    - selection of HO candidates in Wi-Fi hotspots for better offload
    - enhanced Gateway selection for better load distribution in the network
- Improved resource selection
    - performance evaluation of mobile P4P assisted P2P traffic
- Network planning issues
    - Evaluation of different GW placement strategies

List of acronyms and abbreviations

ANDSF	Access Network Decision and Selection Function
ALTO-COMEPS	ALTO assisted Connection Management in EPS
ALTO	Application Layer Traffic Optimization
BR	Backbone Router
BID	Binding Identifier
CoA	Care-of-address
CM	Connection Manager
CDN	Content Delivery Network
CN	Core Network
DPI	Deep Packet Inspection
DSL	Digital Subscriber Line
DSMIP	Dual-stack Mobile IP
EP	endpoint
EPC	Evolved Packet Core
FID	Flow identifier
HoA	Home-address
IP	Internet Protocol
ISP	Internet Service Provider
IFOM	IP Flow Mobility
KPI	Key Performance Indicator
LRU	Least Recently Used
LAN	Local Area Network
LTE	Long Term Evaluation
M2M	Machine-to-Machine
MSISDN	Mobile Subscriber ISDN number
MME	Mobility Management Entity
MCCS	Multi-Criteria Cell Selection
NB-IFOM	Network-based IFOM
PGW	Packet Data Network Gateway
P2P	Peer-to-Peer
PCC	Policy and Charging Control
PCRF	Policy and Charging Control Function
PMIP	Proxy-Mobile IP
QoE	Quality of Experience
QoS	Quality of Service
SLA	Service Level Agreement
SGW	Serving Gateway
SINR	Signal to Interference and Noise Ratio
TEHO	Traffic Engineered Handover
TE	Traffic Engineering
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UE	User Equipment
VoIP	Voice over IP

1. Introduction

This document describes the validation plans of WP4. The goal of validations is to demonstrate with simulations and/or proof-of-concept systems the feasibility and the significance of the achieved results. The document describes for each validation topic the objective of the validation, the selected test scenarios, the applied tools and test environment, the expected results, and the involved partners.

Work Package 4 focuses on the traffic engineering (TE) architecture of future EPC architecture. The main objective of traffic management is to improve quality of experience for the user and to enable efficient usage of infrastructure and IT resources. Document "IR4.1.1b Traffic description" contains the specification of reasonable traffic metrics and traffic classes as well as the estimation of the traffic demand for the targeted study period 2008 - 2020. This is used as basis for the validation of the performance of traffic management mechanisms.

The main building blocks of traffic engineering have been identified in "D4.2.1 Traffic management building blocks in next generation mobile telecommunication systems". From the five basic blocks three require support from the core network side, i.e., micro-, macroscopic traffic management and improved resource selection and caching. The research contributions and validation plans cover mechanisms under these categories.

The main challenges in microscopic TE are to design well scalable traffic classification algorithms that will help to understand traffic demands and usage patterns without the need of deep packet inspection. The network must guarantee appropriate end-to-end QoE for users through different mappings between the virtual EPS bearer QoS class identifiers and the real QoS classes implemented in the LTE and the transport network technologies. Furthermore, multipath communication must be supported to improve QoE of the use, but fairness between different flows on the network must also be kept.

The most important requirement in macroscopic TE is that traffic steering along alternative paths must take into account transport resources on the whole routing path between the UE and PDN GW and not only radio resources.

The main challenge under improved resource selection and caching is that the EPC architecture should support Application Layer Traffic Optimization (ALTO) functions, detect and recover from unfavourable resource/cache selection.

1.1. Microscopic traffic management

1.1.1. Performance evaluation and functional validation of traffic identification and classification using deep packet inspection techniques

1.1.1.1. Objectives

The objectives of this validation topic are to evaluate the performance and functional aspects of traffic identification and classification using DPI techniques. Traffic identification and classification is a required component in many traffic management and monitoring tasks. The validation consists in measuring the accuracy and correctness of the traffic identification techniques of network flows. More precisely, this can be done by measuring and analyzing the following functional KPIs:

Completeness of application classification is the ratio of the application detection count over the expected detection count. It may be more than 100%. Low detection completeness indicates many false negatives. A false negative is the inability to classify a flow of application.

Accuracy of application classification measures how correct the detection technique is. It is the ratio of the number of correct detections over the detection count. It may not be more than 100%. The lack of accuracy leads to false positives; that is, the classification of application B as being application A. The higher the false positives are, the lower the accuracy is.

Traffic identification and classification is resource consuming; the performance of the classification is critical when we consider high speed data links. Several techniques can be used to increase the performance of the classification. The following metrics can be used for the performance evaluation:

Speed of application classification measures the number of packets/data volume required before successfully classifying a traffic flow.

1.1.1.2. Validation scenarios

In the validations, we will first consider trace files of self generated traffic for a controlled evaluation. In a second evaluation phase, if possible we will use trace files from a real live network environment. The collected trace files will contain a diverse application mix in order to widen the scope of the application classification mechanisms. Most importantly, Video applications, P2P applications and Social networking traffic will be considered.

1.1.1.3. Validation tools

The validation tool will be a proof-of-concept DPI probe developed by Montimage. This tool will run under both Linux and Windows operating systems. It will perform traffic analysis, identification and classification. In the first phase, the tool will be applied without any optimization. In the second phase, we will check the performance of the classification with different optimization techniques. Finally, if possible, we will test the tool on a live network.

1.1.1.4. Expected results

The validation is expected to show that traffic classification using DPI has a good level of completeness and accuracy, and can be optimized using simple mechanisms in order to increase its speed and rate. On the other hand, we expect that encrypted and obfuscated traffic are harder to identify. The results of this validation can be seen as complementary with the bulk traffic analysis (see section 1.1.2).

1.1.1.5. Results

The validation consisted on collecting a number of traffic trace files (by Ericsson Turkey) and analysing them using the proof-of-concept DPI probe. The traffic trace files were collected on PC machines connected to the fixed Internet by running the applications of interests (set of P2P applications, Web video, Skype) and recording the traffic activity. Additionally, a number of trace files with mobile traffic generated using smartphones connected through WIFI to the internet where analyzed. The objective was to measure the KPIs defined in section 1.1.1.1.

In order to measure the completeness of the classification, we should know the expected classification results of the trace files. This was possible due to the fact we relatively knew what each trace file contained. Table 4.1 presents an overview of the application classification results; it show that the classification accuracy is relatively high reaching around 96% of the traffic data in terms of volume and number of packets. However, the accuracy in terms of number of flows is lower (82%). This result was expected as the trace files were captured on a local network, where the broadcast signalling (using UDP) is relatively high. Only 0.74% of TCP flows were left unclassified. Among the unclassified flows (TCP and UDP flows) only 5% has more than 10 packets. Table 4.2 provides the distribution of unclassified flows based on the number of packets. It shows that the majority of these flows contain few packets. While analysing the unclassified flows, we noticed that some of them were already initiated when the network sniffing was started. These cold start flows account for 62% of the data volume of unclassified flows though there number accounts for only 0.34% of the total flows number.

Table 1.1.1: Overview of the application classification results

	Flows Number	Packets Count	Bytes Count
Total	30513	2733347	1916196927
Total Classified	25081	2622685	1839604663
Total Classified (%)	82.2%	95.95%	96.00%
Total Unkown	227	99657	75449273
Total Unkown (%)	0.74%	3.64%	3.93%
Total Unclassified (Unknown + UDP)	5432	110662	76592264
Total Unclassified (%) (Unkown + UDP)	17.8%	4.04%	3.99%

Table 1.1.2: Distribution of unclassified flows based on the number of packets

Packets Distribution	Flows Number	Bytes Count	Flows Number (Cold Start)	Bytes Count (Cold Start)
2 <= Packets	4751	0.67 MB	0	0

[3 : 10] Packets	423	0.34 MB	0	0
[11 : 100] Packets	213	0.92 MB	2	5.2 KB
100 + Packets	60	235.2 MB	17	147.7 MB
	5447	237.15 MB	19 (0.34 %)	147.7 MB (62.25 %)

Table 1.1.3: Sample of classified applications

Application Name	Flows Number	Packets Count	Bytes Count
Bittorrent	15459	1371049	1198970278
Skype	1370	617276	251476274
https	301	376751	181093125
http	4646	103548	82374039
Youtube	44	83611	79238629
Unknown	227	99657	75449273
Gnutella	206	16073	12545929
Dailymotion	20	11429	10789797
udp	5205	11005	1142991
Facebook	16	306	171212
Twitter	15	178	74266

As the number of Internet protocols and applications is high, the objective of the classification was not to identify each individual protocol/application; rather, we analyzed the popularity of the protocols/applications, identified the top ranked ones and included them in the classification engine. This explains the difference between the classification of WEB sites/applications like Facebook or Twitter and sites of lower popularity that were identified as HTTP traffic (see Table 4.3).

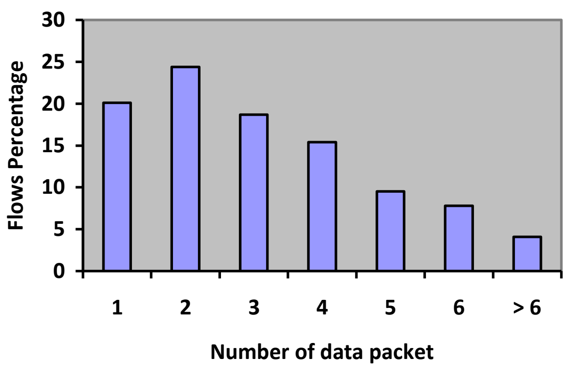


Figure 1: Distribution of the classified flows with the classification speed

Figure 1 illustrates the classification speed of the DPI probe. It considers the number of data packets required for classifying a flow. We can see that for about 80% of the flows, the classification requires a maximum of four packets. Only 4% of the flows require more than 6 data packets to get classified. This result is interesting as if shows that the classification speed is high. We should also mention that the considered trace files where mainly P2P and VoIP applications that use diverse techniques of encryption and obfuscation. For HTTP based traffic that accounts for the lion share of the mobile as well as Internet traffic, the classification speed is rather very high as it requires one or two data packets.

1.1.1.6. Applicability of the results

Application identification and classification can be integrated in EPC element nodes when appropriate (e.g. for nodes performing application based traffic engineering). Additionally, it can be applied on mobile user traffic data in order to obtain deep insights on the traffic and usage trends. Finally, traffic identification and classification is a key block in network and traffic monitoring.

1.1.1.7. Partners involved

Montimage is the only partner involved in this validation. However, due to the complementarities with traffic identification using bulk analysis methods, Ericsson Turkey collaborated on this topic by providing the traffic trace files and through a big number of collaboration workshops.

1.1.2. Performance evaluation and functional validation of traffic identification and classification using bulk traffic analysis

1.1.2.1. Objectives

The objective of bulk traffic analysis is to generate a map of the network data utilization.

Deep Packet Inspection tools in network operators investigate the payload and can determine the exact application type such as Skype, Youtube, Dailymotion, Google+, etc. Please see **Figure 2**. This may be required due to many reasons such as the operator's pricing strategy, campaigns or regulations. That is, these tools are mainly utilized for policy enforcement in real-time. However trying to map all the network traffic via real-time DPI would be extremely costly and many times unnecessary.

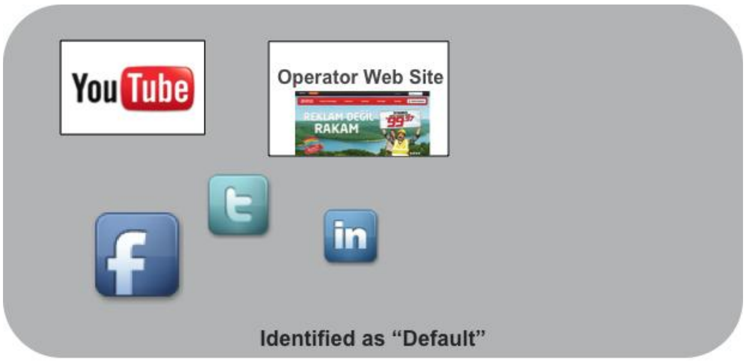


Figure 2: ***Illustration of how DPI classifies network traffic***

For example, in case there is no specific restriction required, or any pricing strategy, there is no reason to identify whether a VoIP application is Skype or Google talk, however in order to satisfy QoS/QoE requirements and manage network efficiently, determining the family of application – that the application is VoIP- is very beneficial..
Ericsson Turkey has worked on a statistical method of classification. This method aims to draw a map of the network traffic, that is, to classify the total usage according to varying time into several classes. The initial idea of classes were:

- Videostreaming
- VoIP
- Instant Messaging
- P2P filesharing
- Web surfing
- Gaming, and
- M2M

This has been modified and finalized as:

- Video streaming
- P2P
- Conversational (contains IM, VoIP and video chat)
- Web
- Gaming, and
- M2M

The main reason for grouping IM, VoIP and Video Chat in one class as conversational is that, the success of the statistical method in distinguishing the three was not acceptable. Merging these can be acceptable since the amount of packet flow is not as much as Video streaming and these three are similar in requirements and sufficiently different from the other classes.

This idea of mapping is illustrated in **Figure 3**.

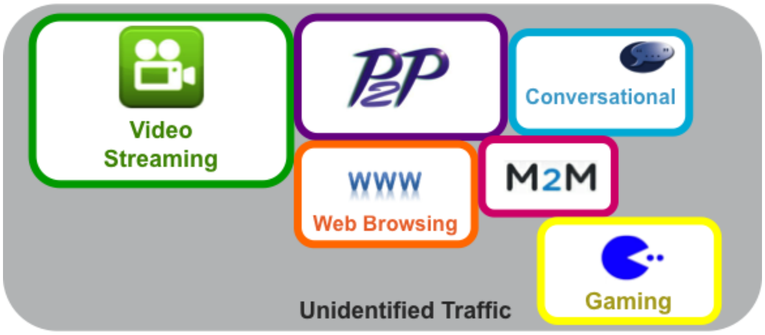


Figure 3: ***Bulk data analysis aims to bring out the distributions of applications in the MNO.***

The initial aim is to get a mirror copy of data traffic going over the network for a period of time like one hour in a set pattern such as:

- Weekday work hour
- Weekday after work hour
- Weekday night
- Weekend daytime
- Weekend evening
- Weekend night

and get a map of the network as in **Figure 4**. In case there are enough many such intervals, variation of usage during the week can be observed and the distribution of traffic into different applications can be observed.

This can be utilized to set QoS parameters according to distribution and the prioritization of the network operator. A very important concern which can be addressed by such a statistical solution is the subscriber privacy in the network. The amount of information gathered about usage would not disturb the end users.

Another aim could be to classify users according to the applications and times they utilize the network and make proper campaigns to customers in order to make better use of the limited bandwidth.

In case this method can be improved so that it can run in real-time, it can be run on the flows where dpi is not running on and help network utilization in real-time and more importantly it may help network neutrality as well. The current progress does not allow real-time usage.

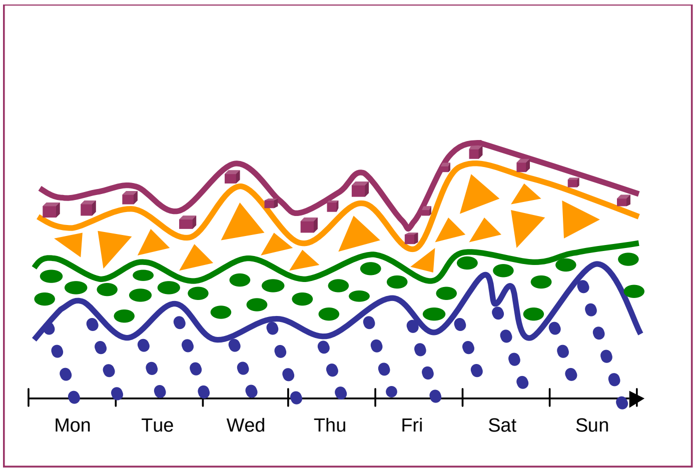


Figure 4: ***Weekly distribution of network data traffic into applications.***

1.1.2.2. Validation scenarios

1.1.2.2.1. Scenario 1

Traffic streams for the following traffic types have been generated utilizing own computers:

1. Video streaming
2. P2P (encrypted and not encrypted)
3. Video chat
4. VoIP
5. Instant messaging
6. Web Surfing

These samples have been fed into Montimage DPI tool and some of the results have been used for training the algorithm and the rest have been used for testing the algorithm.

1.1.2.2.2. Scenario 2

Video Streaming, P2P (encrypted and not encrypted), VoIP, Web Surfing traffic have been generated utilizing a single MSISDN at the AVEA test lab and mirroring has been done using the Gi interface. These application types have been generated consecutively for easy validation purposes.

The samples have been fed to dpi, however the results were not much meaningful. The reason was initially an algorithm with no training was being used. However when this was replaced with an algorithm which requires training, this Scenario ended up having very few samples. Scenario 4 took more effort, however results were more meaningful.

The architecture being used for Scenarios 2, 3 and 4 are as follows. In LTE networks, Bulk Analysis tool can be directly placed in the P-GW with almost no change in functionality.

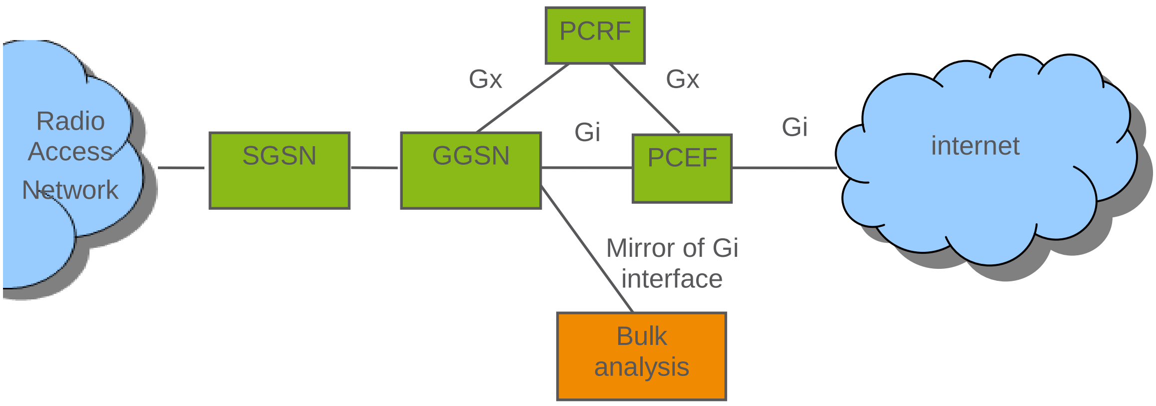


Figure 1 – Architecture for bulk analysis.

1.1.2.2.3. Scenario 3

Video Streaming, P2P (encrypted and not encrypted), VoIP, Web Surfing traffic have been generated utilizing a two MSISDNs at the AVEA test lab and mirroring has been done using the Gi interface. In this test data, the applications have been utilized concurrently.

The samples have been fed to dpi, however the results were not much meaningful. The reason was initially an algorithm with no training was being used. However when this was replaced with an algorithm which requires training, this Scenario ended up having very few samples. Scenario 4 took more effort, however results were more meaningful.

1.1.2.2.4. Scenario 4

Live network data, that is, internet traffic of subscribers has been collected from AVEA and this has been fed to dpi and bulk analysis tool and results have been compared.

M2M traffic is not currently available in networks. Therefore we have not been able to validate the success rate for this type of traffic.

1.1.2.3. Expected results

We expect that the result of the comparison table will be as follows:

Flow ID	Source IP	Destination IP	DPI class	Bulk Analysis Class

1	IP address 1	..	youtube	Video streaming
2	IP address 2	..	skype	VoIP
3	IP address 3	..	bittorrent	P2P
4	..	..	facebook	Web Surfing
5	..	..		
6	..	..		

1.1.2.4. Results

Firstly we would like to mention the results for Scenario 1 in Section 1.1.2.2.1.

The work conducted during the statistical tool was basically working on a method to classify the flows from basic packet statistics. The particular features used were the packet size and packet direction.

Below figures are screen shots of the tool for the statistical classification algorithm.

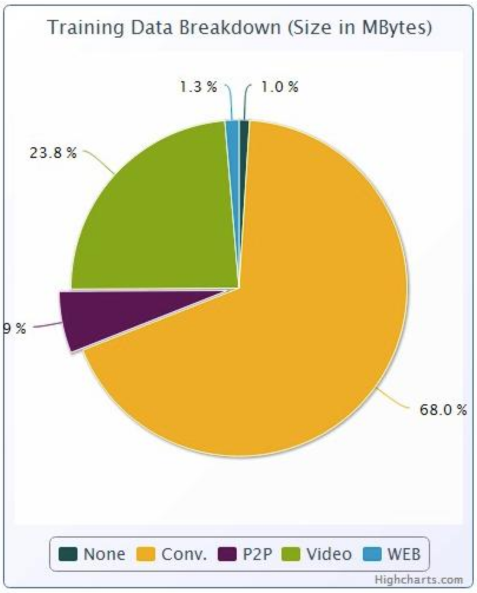
Figure 5 is the display where the information on the input files and the algorithm parameters can be seen.

Figure 5

Figure 6 shows the statistics of the training and test samples. Some application types have numerous flows in terms of percentage of flows, however their percentage in the total size of the files may be less. This is very normal, for example P2P applications generate a lot of small sized flows, however this does not reflect on the byte size. Correct classification of large dimensioned flows is much more important than classification of small sized flows. From the figure, we can see that Conversational traffic held the major portion both in the training data and in the test data. Video traffic holds the second major portion. Since this was generated by our sample captures, this will unlikely represent the distribution in a real network.

A. Training and Test Data for the Experiment

	Training data		Test data	
	Number of Flows	Total Size (MBytes)	Number of Flows	Total Size (MBytes)
NONE	551	19	0	0
P2P	164	110	128	63
VIDEO	42	442	19	138
CONV.	143	1265	49	312
WEB	11	25	4	36
TOTAL	911	1861	200	549



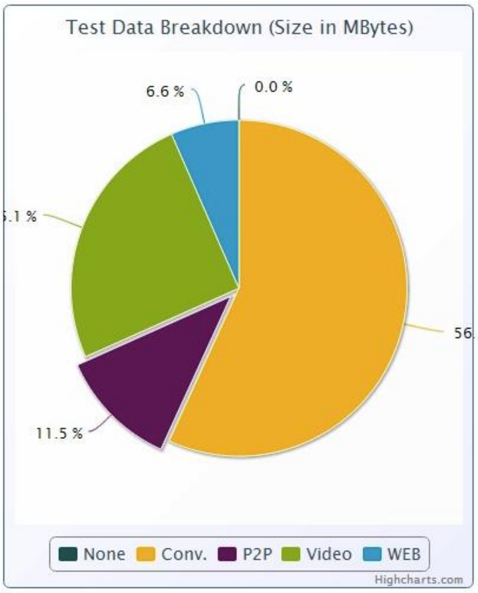


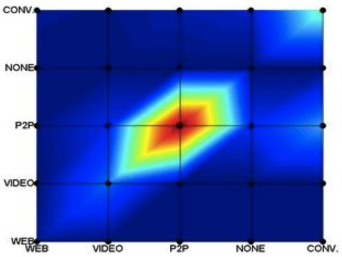
Figure 6

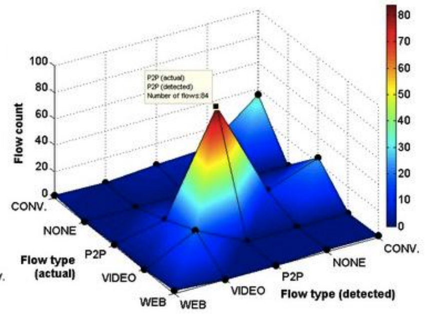
Below figures, Figure 7 and Figure 8 show the classification success rate of the algorithm on the test set after being trained by the training data.

Confusion matrix signifies the distribution of test application types onto classes by the algorithm. The more the diagonals are full, the better the results. Off diagonals represent misclassifications. Results have been provided both flow count based and megabyte based. Video and P2P seem to be highly successful. Even the megabyte based success rate is quite high for conversational. However there is more work to be done so that tests can be done with improving the training set. The training set and test set both had torrent protocol in abundance compared to the other protocols. Similarly Conversational class had more Skype samples than other application types. Improving the training set will improve the results and is an effort that has to be periodically spent so as to keep the algorithm up-to-date.

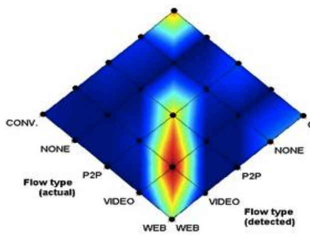
B. Confusion Matrices (Test Results)

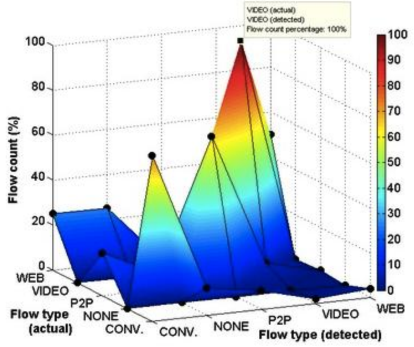
Confusion Matrix (Flow count based)	CONV	NONE	P2P	VIDEO	WEB
CONV	36	6	4	1	2
NONE	0	0	0	0	0
P2P	24	10	84	9	1
VIDEO	0	0	0	19	0
WEB	1	1	0	0	2





Confusion Matrix (Flow count based – Percentage)	CONV	NONE	P2P	VIDEO	WEB
CONV	73.5	12.2	8.2	2.0	4.1
NONE	0	0	0	0	0
P2P	18.8	7.8	65.6	7.0	0.8
VIDEO	0	0	0	100.0	0
WEB	25.0	25.0	0	0	50.0

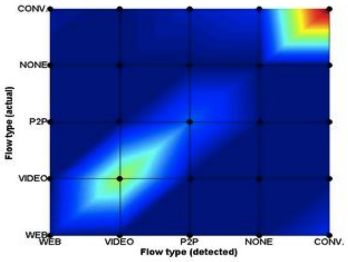


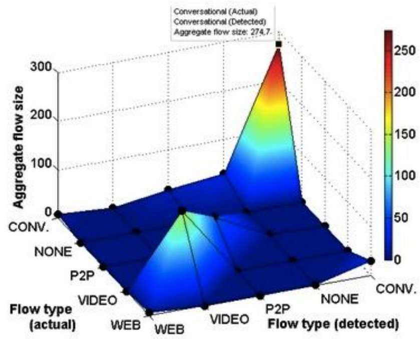


Percentage of identified flows : 70.5%
Best identified class : Video (P_d=100.0%)
Worst identified class : P2P (P_d=71.9%)

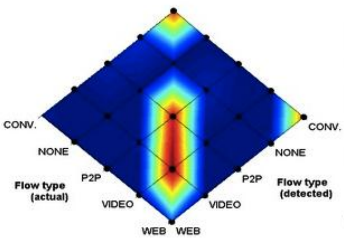
Figure 7

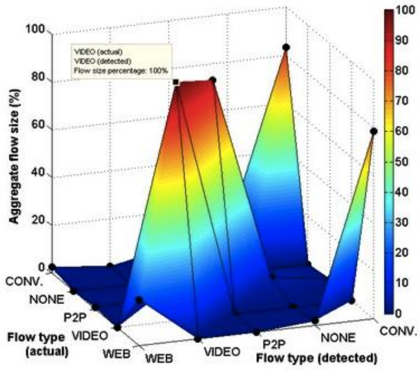
Confusion Matrix (Aggregate size based – MBytes)	CONV	NONE	P2P	VIDEO	WEB
CONV	274.7	29.3	16.9	0.7	8.4
NONE	0	0	0	0	0
P2P	1.0	3.5	60.1	1.3	0.6
VIDEO	0	0	0	145.3	0
WEB	29.9	0.8	0	0	7.4





Confusion Matrix (Aggregate size based – Percentage)	CONV	NONE	P2P	VIDEO	WEB
CONV	83.3	8.9	5.1	0.2	2.5
NONE	0	0	0	0	0
P2P	1.5	5.2	90.5	1.9	0.9
VIDEO	0	0	0	100.0	0
WEB	78.5	2.1	0	0	19.4





Percentage of identified flows : 84.1%
Best identified class : Video (P_d=100.0%)
Worst identified class : Web (P_d=19.4%)

MEVICO Project Camp @ Espoo, FI, Nov. 2012

Figure 8

Scenario 4 Results: Since the data was collected from live network, it is worthwhile to talk about Section 1.1.2.2.4 Scenario 4 results.

In this scenario, our groundtruth labels were not the same labels as in the previous scenario. In the previous scenario, our groundtruth was generated by Montimage DPI and therefore our labels were more QoS requirement based. However in Scenario 4, there was privacy concerns for the end subscriber and therefore IPOQUE open source DPI was used for identification of applications. This changed the label set quite a bit, however general idea is that the algorithm can be trained with any set of labels. Actually we had two algorithms, one of them based on Dirichlet Process Mixture Markov Model (DPM) and the other one is K-Nearest Neighbors (KNN).

In Figure 9, the results of analysis for training set size versus classification accuracy have been provided. We see that the performance indeed sharply increases after observation of a reasonable amount of flows and then levels off. The jitter in the graphs is due to the particular random permutation of the data. To average out the effect of a random permutation we repeated this procedure for 20 different permutations and averaged the results. In Figure 10, we show the results of this experiment for the KNN classifier. The thick lines denote the average performance of 20 different permutations and the thin dotted lines show the 3σ error bars of the performance. We observe that there are indeed some particularly good cases and bad cases of training data. In any event, on automatic DPI labels, a byte classification performance of about 90% seems to be easily achievable. The important point is that, with enough data one may expect to see a better performance from the DPM model as well.

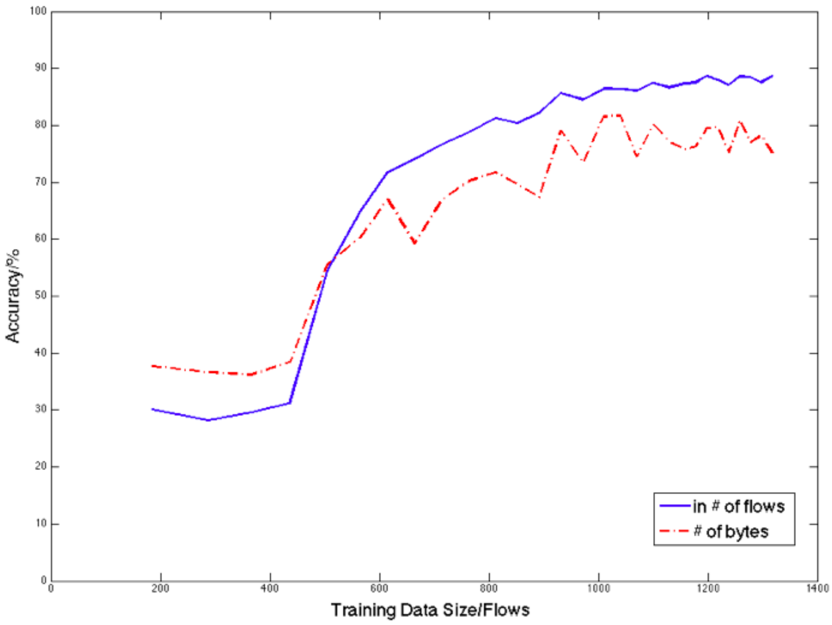


Figure 9 The effect of training set size on test performance (DPM classifier)

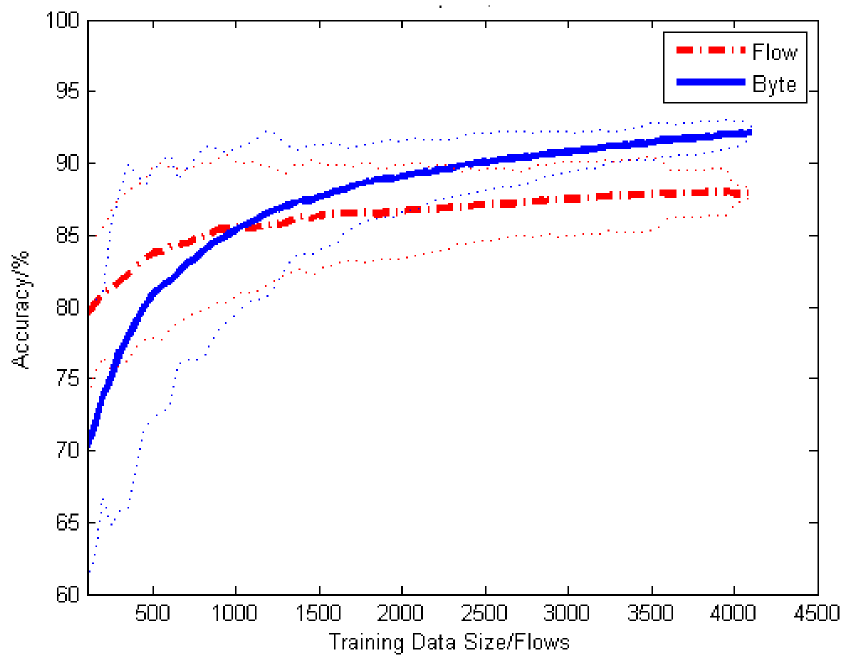


Figure 10 The effect of training set size on test performance (KNN classifier)

1.1.2.5. Applicability of the results

This algorithm can be implemented over P-GW or a separate box for LTE networks. However it needs to be enhanced. Even though there was a various number of application types for each class, this can still be enriched. Currently the algorithm is most successful in correctly classifying the Video class.

1.1.2.5.1. A very useful byproduct

During the course of the research project, due to close collaboration with Montimage, we produced a rather useful byproduct. This is the Broadband Reporting Tool. It is a reporting tool based on network captures and their DPI outputs. Some of the reporting tool outputs can be seen in the following screenshots:

Capture Analysis – Flow Statistics

	P2P	VIDEO STR.	P2P STR.	GAMING	CONV	HTTP	OTHER	TOTAL
Number of Flows (count)	8868	93	15	24	375	2959	9991	22325
Number of Flows (%)	39.72	0.42	0.07	0.11	1.68	13.25	44.75	100.0
Size (MBytes)	637.26	608.12	0.06	123.52	74.04	243.45	428.79	2115.23
Size (%)	30.13	28.75	0.0	5.84	3.5	11.51	20.27	100.0

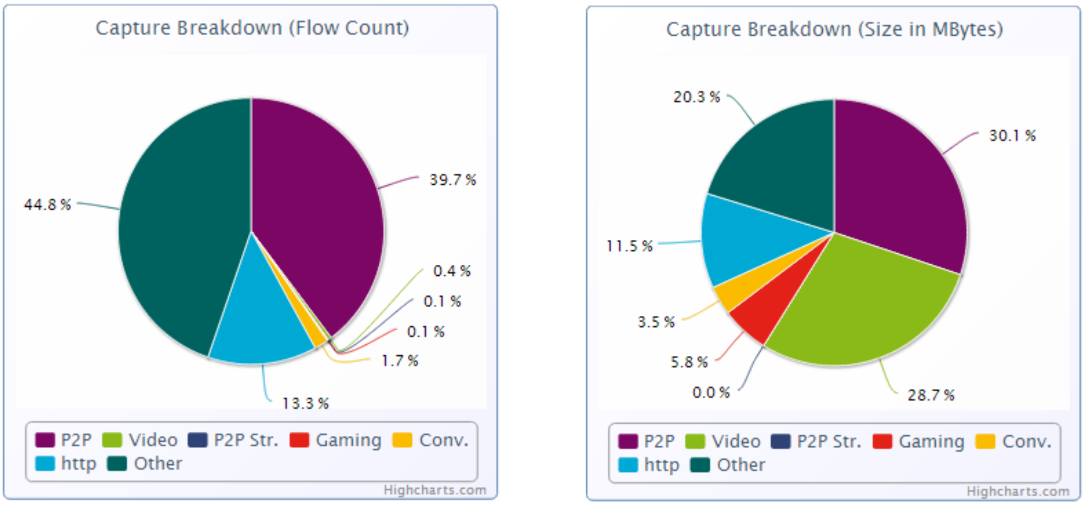


Figure 11

Conversational Traffic Drill-down

	Skype	MSN	GTalk	Yahoo	Jabber	Other	TOTAL
Number of Flows (count)	203	102	124	40	67	88	624
Number of Flows (%)	32.5	16.3	19.9	6.4	10.7	14.1	100
Size (MBytes)	159	66	111	24	54	45	459
Size (%)	34.6	14.4	24.2	5.2	11.8	9.8	100

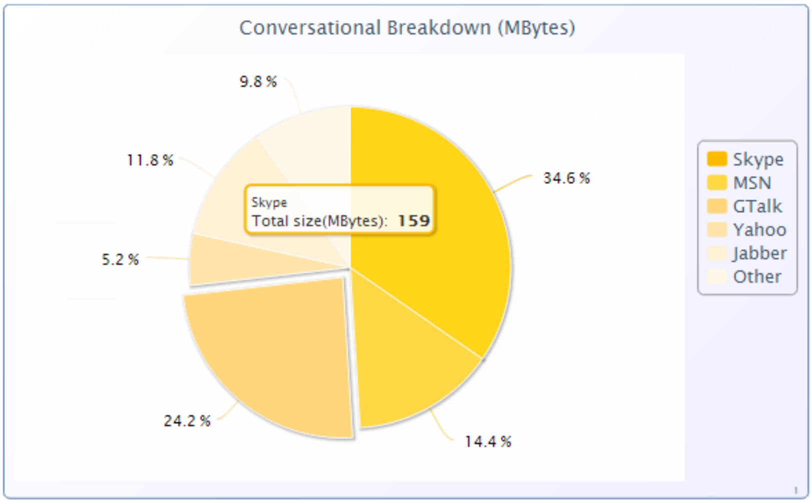


Figure 12

HTTP Usage : Top 5 domains

	google.com	bbc.co.uk/news	msn.com	facebook.com	gmail.com	TOTAL
Number of Hits (count)	36	24	18	12	6	96
Number of Hits (%)	37.5	25.0	18.8	12.5	6.3	100
Size (MBytes)	6.8	5.5	7.9	11.0	10.8	42
Size (%)	16.2	13.1	18.8	26.2	25.7	100

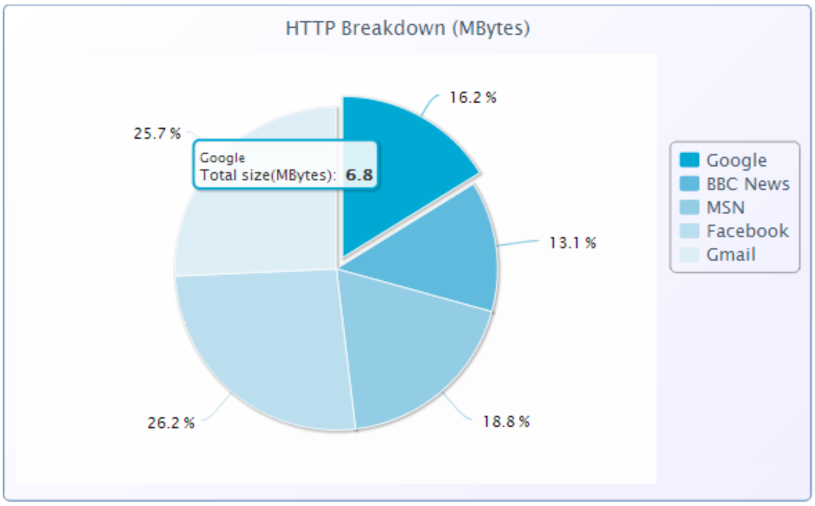


Figure 13

The Broadband Reporting Tool can be readily deployed at network operators for benefiring the marketing departments and network dimensioning people.

1.1.2.6. Partners involved

Ericsson Turkey: Statistical analysis algorithm developed and application based samples collected for algorithm training. Worked on anonymization of Avea data and performed tests at Avea test platform prior to getting live samples from Avea network.

Avea: Deployed server and arranged server for mirroring of live network traffic. Traffic has been mirrored over the Gi interface for a total of ten hours within one week.

Montimage: Provided required output format for the capture samples of various applications by passing through Montimage DPI. Also visited Turkey for holding workshops.

1.1.3. Peformance evaluation of network-based application QoE measurement and application traffic flow manipulation Framework ISAAR

1.1.3.1. Objectives

The objective is to evaluate the network-based application QoE measurement and traffic flow manipulation system called ISAAR (Internet Service quality Assessment and Automatic Reaction). By this Framework, applications get just the transmission parameters required for sufficient QoE. Alternatively low prio traffic might be throtteled, yielding a better throughput for high prio traffic. Also it is possible to change the prio of observed traffic flows.

1.1.3.2. Validation scenarios

The Framework is evaluated by performing experiments with different traffic load and application mixes and checking the resulting application specific QoE by human test users. Both offline as well as online testing is planned.

1.1.3.3. Validation tools

The traffic management testbed of CUT depicted in Fig. 3 is used to carry out the experiments decribed above.

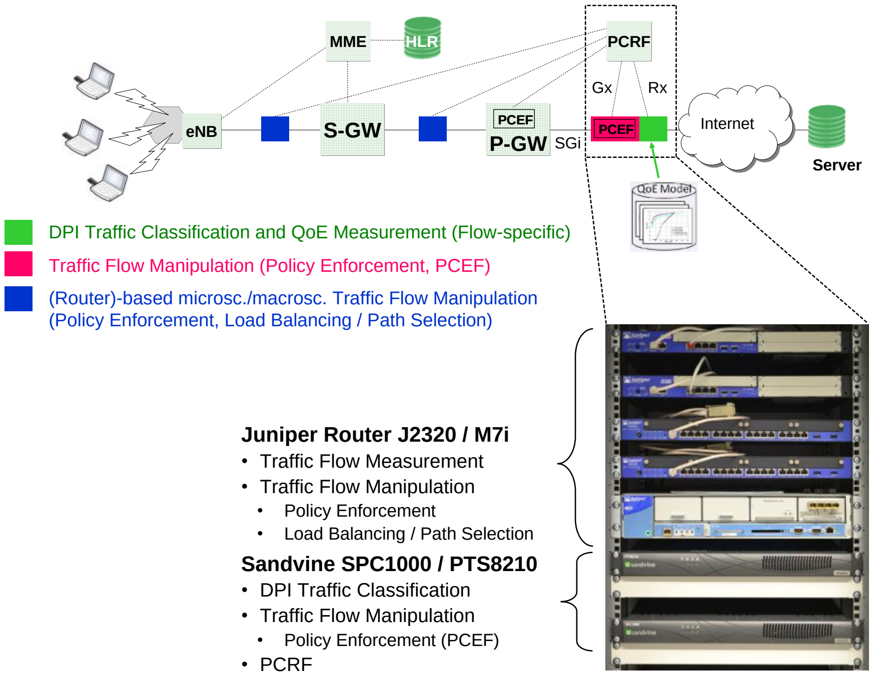


Figure 2 - Traffic Management Testbed (CUT)

1.1.3.4. Results

So far only the offline version of the video QoE monitoring has been evaluated. Therefore, we have shown, that the exact method of nearly reaches the same outcomes as a group of test persons wrote down on their notes. The throughput based estimation method has a higher performance, but the accuracy of the results is decreasing with the gain in processing speed. The combined method shows a higher processing speed as the exact method but a lower than the throughput based method, but there is no decrease in accuracy. The results are given in the following table.

estimation interval stepping	processing time	# re-buffering events	re-buffering time
	Both algorithms - good case video		
human	-	0	0 s
every packet	6 s	0	0 s
10 packets	3 s	0	0 s
50 packets	3 s	0	0 s
100 packets	3 s	0	0 s
150 packets	3 s	0	0 s
250 packets	3 s	0	0 s
	Estimation Method based on Throughput - bad case video		
human	-	10	58 s
every packet	12 s	10	56,6 s
10 packets	6 s	10	56,0 s
50 packets	6 s	10	54,4 s
100 packets	6 s	10	53,7 s
150 packets	6 s	9	51,3 s
250 packets	5 s	6	49,1 s
	Combined Estimation Method - bad case video		
human	-	10	58 s
every packet	12 s	10	56,6 s
10 packets	8 s	10	56,6 s
50 packets	8 s	10	56,6 s
100 packets	8 s	10	56,6 s
150 packets	8 s	10	56,6 s
250 packets	7 s	10	56,6 s

Table 1: Video QoE Monitoring results

1.1.3.5. Applicability of the results

If the results are successful, the network-based ISAAR fraqmework might be implemented as shown in Fig. 3. Different implementation options are possible: separate PCRF controlled traffic flow manipulation by means of automated router configuration, or full integration of traffic classification, QoE measurement and flow manipulation into a PCC solution.

1.1.3.6. Partners involved

ISAAR is solely developed and validated by Chemnitz University of Technology (CUT)

1.1.4. Improving Multipath TCP Performance by Means of Transparent Solutions to TCP End-Points

1.1.4.1. Objectives

The main objective of using more than one path to distribute TCP traffic over multiple paths is the aggregation of the capacities of the used paths. In addition to aggregation of the path capacities, in conjunction with an appropriate congestion controller, TCP data packet distribution may also provide means to balance the Internet congestion in a stable way.

The second objective is the easy deployment of the solution. Despite the potential benefits and a large body of work, multipath TCP solutions could not be successfully deployed. It is because of the requirements from the TCP end-hosts. Operating system vendors have to include the solution within their TCP/IP implementations to benefit from it, which is (if not possible) hard to realize. Thus, our second objective is to develop multipath TCP solutions that don't require change in the TCP/IP protocol implementations of the end-hosts.

The covered challenge is *C.Tm. 4: Improve traffic load distribution*. The TCP throughput/goodput is the major KPI parameter to show benefits of splitting TCP trafic over multiple paths.

1.1.4.2. Validation scenarios

We will test our solution within network scenarios, in which TCP packets may be distributed via multiple paths. Figure 14 shows a possible network which may be used in validation of the developed solution.

The network consists of two local networks (LANs) with clients and local routers (LRs). Two LANs are connected to each other via a core network (CN). LAN clients may have wired or wireless (e.g., Femtocells or LTE) interfaces. LANs are connected to the CN via multiple gateway (GW) links (e.g., for redundancy or since GW links have low capacities).

CN consists of a mesh of backbone routers (BRs). The BRs and LRs may be connected to each other via wired as well as wireless links (e.g., a wireless mesh network backbone). In addition, LAN GW links may be connected to separate CNs (e.g., different ISP networks).

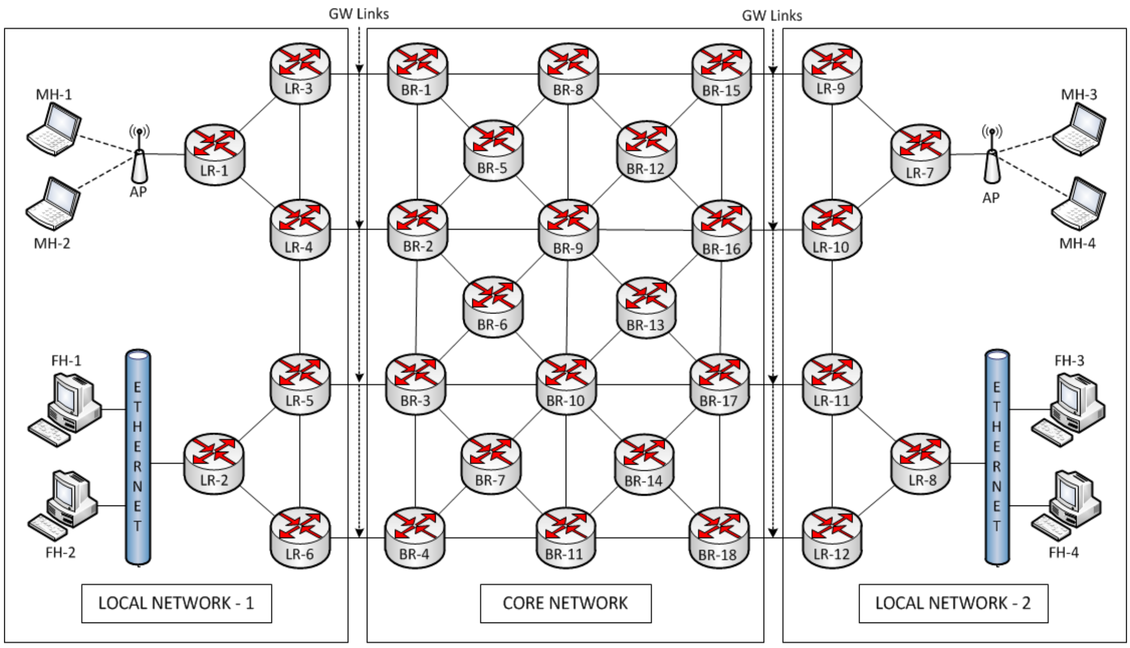


Figure 14. Network with Multiple Gateways (GWs)

Figure 15 shows two split flows in this network: data transfer from FH-2 to FH-4 and from MH-3 to MH-1. In the flow from FH-2 to FH-4, LR-2 detects the TCP connection and distributes data packets to its neighbors (i.e., LR-5 and LR-6). If GW links have low capacity (e.g., 1.544 Mbps T1 links or DSL lines) then the TCP connection may benefit from the aggregated capacities of both GW links. In addition, since split point is outside of the end-hosts, the deployment will be easier. All of these concepts may be applied in the reverse direction, for the connection between MH-3 and MH-1.

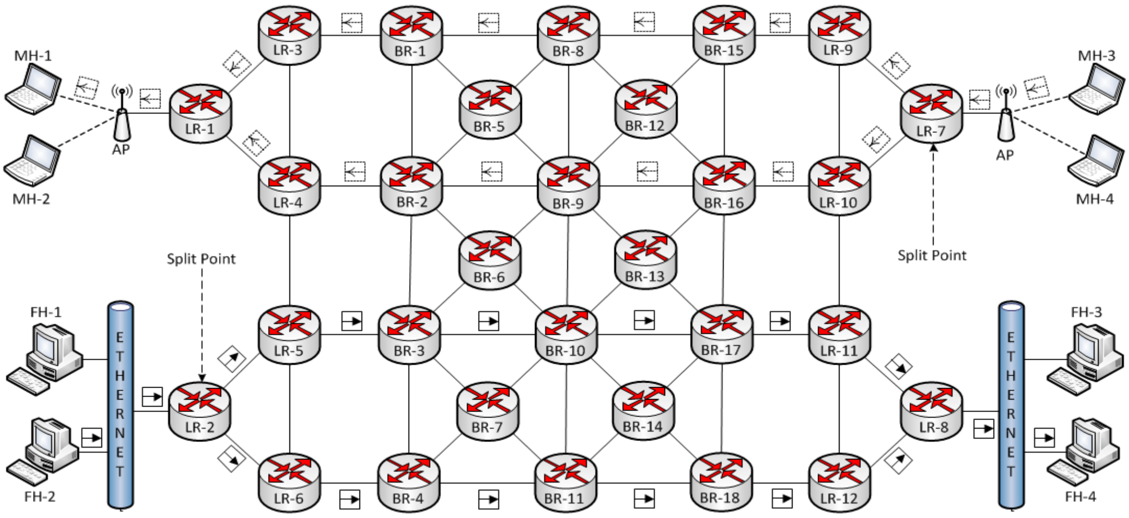


Figure 15. Two Split TCP Flows

1.1.4.3. Validation tools

NCTUns simulation and emulation tool will be used in evaluating our solution. NCTUns simulator will be used to construct scenarios which allow distribution of TCP traffic over multiple paths. NCTUns emulation will be used to test our solution with different operating system implementations of TCP sender.

1.1.4.4. Expected results

Increase in TCP throughput is main expectation from the results. We expect increase in TCP throughput proportional to the number of multiple paths used since each path adds its capacity to the available capacity to the TCP.

1.1.4.5. Results

We have constructed NCTUns emulation scenario shown in Figure 6 based on the architecture shown in Figure 16:

- Our solution (i.e., DUPACK Estimation and Filtering (DEF) algorithm) is transparent to TCP end points. Thus, DEF must effectively work with TCP implementations of various operating systems (OSs). Different OSs may use different variants of TCP by default or support different TCP variants which may be set by the end-users [1]. Thus, we used TCP sender as an external machine and tested DEF performance by using TCP sender as a host that runs one of the following OSs from different OS families: Microsoft Windows XP Professional Version 2002 Service Pack 3 (Windows), Ubuntu 11.04 (Linux), and PC-BSD 8.2 (BSD).
- There are 4 routers within the local backbone (nodes 2 to 5). The links between the routers have 10Mbps capacity and 3ms delay. Node 2 is used as the split point. It distributes TCP data packets to multiple paths based on round-robin (RR) scheduling. The results are collected either by using DEF algorithm or pure RR scheduling on the split point. In packet distribution, either 2 or 3 paths are used.
- Local backbone is connected to the Internet via 3 GW links, each with a capacity of 1.544 Mbps (as in T1) and a delay of 5ms.
- Out-of-order packet arrivals are generated by using Internet paths with different delays. Upper path is the default path with delay of D1 = 10 ms. In two paths experiments, paths with delays D1 and D2 are used. In 3 paths experiments, paths with delays D1, D2, and D3 are used. D1, D2 and D3 are set as shown in Table 2.
- TCP receiver is a simulated node that downloads data from the TCP sender for 600 seconds. TCP goodput is calculated by dividing downloaded amount of data to the download time.

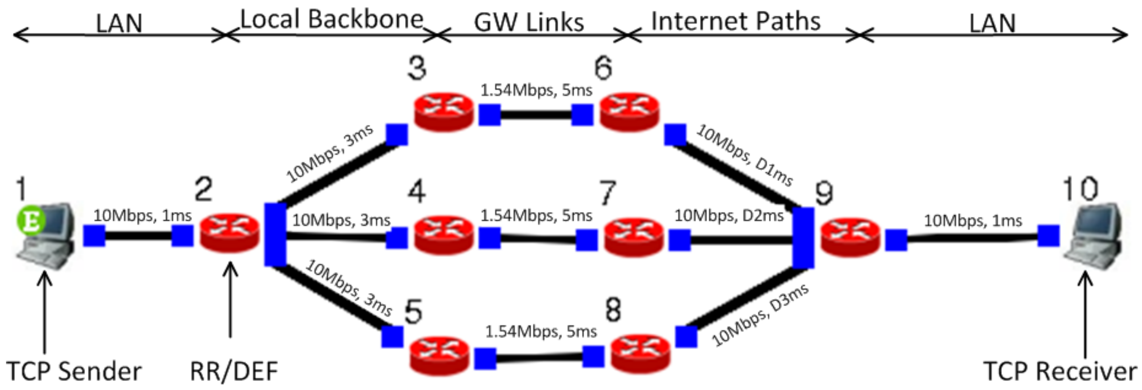


Figure 16. NCTUns Scenario for TCP over Multiple Paths Performance Evaluation

D1/RTT1 (ms)	D2/RTT2 (ms)	D3/RTT3 (ms)	Delay/RTT Difference (ms)
10/40	10/40	10/40	0
10/40	50/80	90/120	40
10/40	90/120	170/200	80
10/40	130/160	250/280	120
10/40	170/200	330/360	160

Table 2. Path Delays and RTTs Used in NCTUns Emulations

Figure 17 and Figure 18 show 2 and 3 paths experiment results, respectively. As expected [2], the Windows XP and PC-BSD TCP goodputs degrades sharply in RR with the increasing path delay difference and it is worse than the single path case when there is even a small delay difference (e.g., 40 ms). The only exception of performance decrease in the RR scheduling is the Linux. RR performed well, since Linux has its own algorithms for out-of order packet reception. Linux TCP implementation dynamically adjusts the DUPACK threshold to trigger fast-retransmit (i.e., the threshold value is not always 3 and adjusted based on the reordering in the network) [3].

On the other hand, DEF gets better TCP goodputs than single path use (i.e., TCP data transfer over one path, without DEF or RR) in all cases. Even though Linux has its own mechanisms to cope with the out-of-order packets, the results show that DEF did not harm the performance of Linux. We observed 75% to 100% increase in the TCP goodput when 2 paths are used and 125% to 200% increase when 3 paths are used, as compared to the single path use.

Because of the network dynamics, we see some fluctuations in DEF performance (not more than 10% when 2 paths are used and not more than 20% when 3 paths are used, between the best and the worst case for a given OS). It is caused by the buffering in the network which influences accuracy of the DEF DUPACK estimations, especially when number of used paths increases and equal-RTT paths are used.

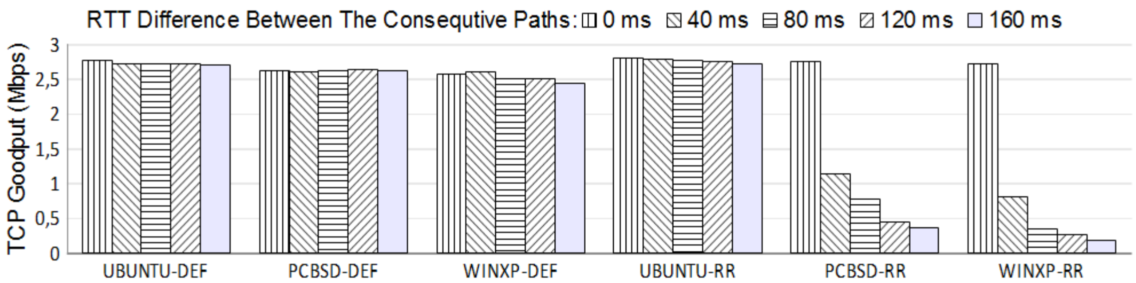


Figure 17. NCTUns Emulation Results when 2 Paths are Used

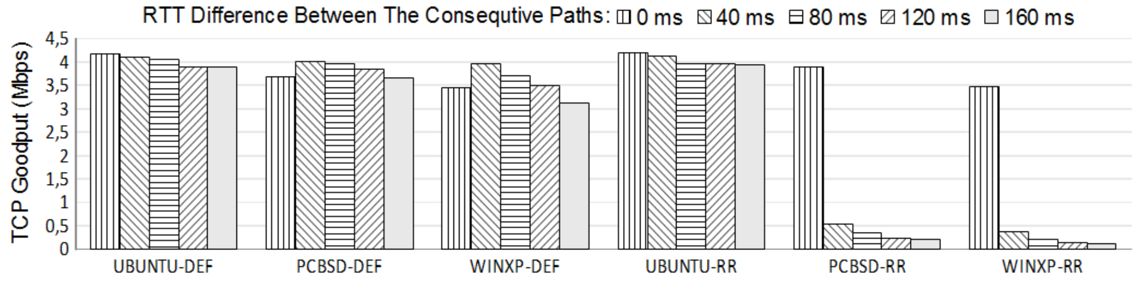


Figure 18. NCTUns Emulation Results when 3 Paths are Used

1.1.4.6. Applicability of the results

Since the main property of the proposed solution is its transparency, it may be used in any scenario in which use of multiple paths may result beneficial and including all reference scenarios from WP1.

1.1.4.7. Partners involved

Technical University of Berlin

1.1.4.8. References

[1] P. Yang, W. Luo, L. Xu, J. Deogun, and Y. Lu, ”TCP Congestion Avoidance Algorithm Identification,” IEEE 31st International Conference on Distributed Computing Systems (ICDCS 2011),
June 2011.
[2] M.Gerla, S. S. Lee, and G. Pau, ”TCP Westwood Performance Over Multiple Paths,” UCLA CSD Technical Report #020009, January 2002.
[3] P. Sarolahti and A. Kuznetsov, ”Congestion Control in Linux TCP,” Proceedings of Usenix 2002/Freenix Track, pp. 49–62, June 2002.

1.1.5. Measurement of IP traffic growth and application mix

We investigate the variability of traffic in different time scales ranging from 1 millisecond up to 1 hour. Standard reporting procedures of routers, which provide the mean load over 5 or 15 minute intervals, do not capture much higher burstiness of traffic on short time scales. Moreover, we distinguish traffic profiles of different applications as components of the aggregated traffic. It is in the main focus

- to estimate the efficiency of buffers in network elements
- to estimate the impact of the load levels on QoS/QoE degradation and
- to set up and confirm dimensioning rules with regard to QoS/QoE.

Moreover, we discuss validation aspects on measurement of global IP traffic trends, which are used to estimate the scalability of future Internet technology.

1.1.5.1. Representative measurement on global trends

The composition and growth of traffic on the Internet is estimated in a number of measurement studies. Deliverable D4.1.1 includes an overview of relevant trends for the MECIVO project. Most well known are periodical reports by Cisco [4] or Sandvine [26], who are also offering measurement equipment and software tools for traffic analysis as vendors. Those measurements show common main trends on traffic composition and growth, but nonetheless there is always some uncertainty about where and when the measurement has been taken. Experience with measurement of IP traffic at DT indicates that measurements can be highly sensitive especially with regard to the daily traffic profile and the upstream-/downstream direction. The four diagrams of transport layer port statistics in Figure 19 have been taken on the same day from the same link in Deutsche Telekom’s aggregation network.

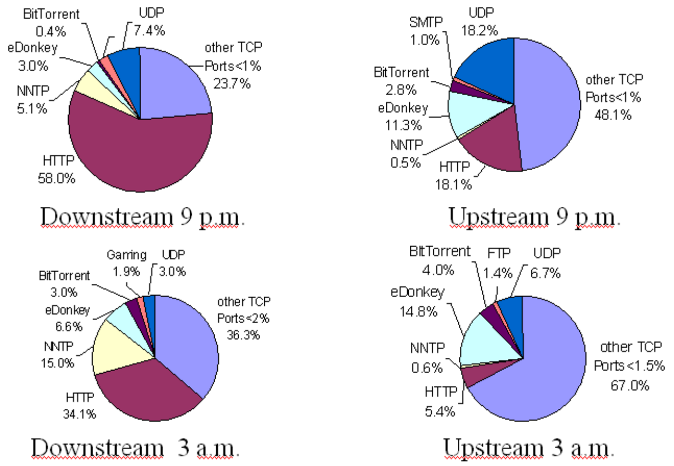


Figure 19: Different IP traffic measurement results in up-/downstream and at day/night time

HTTP is observed to be dominant in downstream direction and at day time, but the picture is completely different in the upstream and at night. Therefore it is important to clearly indicate whether measurement results have been taken

- in upstream or downstream direction or represent mean values over both directions of a bidirectional measurement,
- in specific time intervals (e.g. in the busy hour) or over one or several days / weeks,
- from a single or a set of links, in unique or different technologies/environments (broad-band access, aggregation, core, university, enterprise network, Ethernet, WDM, etc.)

The reports [4][26] partly distinguish upstream / downstream or busy hour measurement. Other main data for verification is often missing. Global results and results for continents do not contain many hints on how many measurement points and samples are included or where and when measurement probes have been taken. Moreover, figures are usually given at the precision of percentiles or per mille, although there is no indication whether the confidence intervals of the results are also in this range or can be much larger. An analysis of the precision and the variance in measurement over time and/or space is often missing.

1.1.5.2. Comparison of trends in global IP traffic measurements

Therefore collecting results from different sources and the comparing their conclusions seems to be the only way to validate measurement statistics on global Internet traffic. Figure 20 shows a step towards such a validation, involving official statistics by administrative bodies, which are available at least for some countries, i.e. Australia [1], Hong Kong [20] and Germany [2]. The main trends are similar, indicating a growth of Internet traffic by a factor of the order 100 over the past decade. However, annual growth rates differ over time and for each region. The variances in each curve and the global trend results are obviously not in the range of a few percent or per mille.

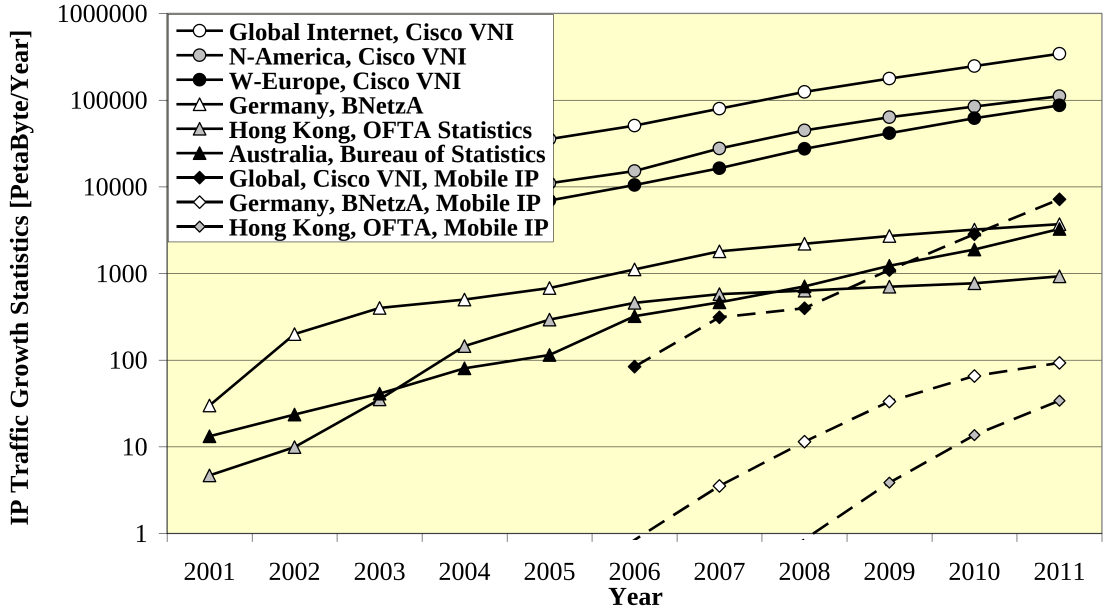


Figure 20: Trends in IP traffic growth reported from different sources

1.1.6. Traffic variability of different applications in LTE/EPC networks

1.1.6.1. Objectives

The variability of traffic is modelled and evaluated on different time scales in the range from 1 millisecond up to 15 minutes. Traffic profiles of different applications are in the focus as well as the aggregated traffic on links and other communication network interfaces. The aim is

- to estimate the efficiency of buffers in network elements
- to estimate the impact of the load levels on QoS/QoE degradation and
- to confirm dimensioning rules.

A regular task for broadband Internet access providers is the dimensioning of transmission links and, in particular, the determination of appropriate load thresholds for link upgrades in order to cope with a substantial growth of IP traffic as illustrated in Figure 20. Such load thresholds depend on the type of the link. Core and backbone networks often have a resilient layout with moderate load thresholds for low loss on partly overprovisioned links, whereas higher thresholds usually apply for expensive links which are often found at network boundaries, transcontinental connections or for air interfaces in mobile networks. The impact of link loads on the quality of service (QoS) in terms of delay and packet loss mainly depends on traffic variability, where each application has its own traffic profile and QoS demands.

For dimensioning, at least the mean superposed traffic rate μ and the variance σ^2 (or standard deviation σ) should be known. As a simple rule of thumb, a bandwidth of $B = \mu + 3\sigma$ is sufficient to obtain a limited loss rate, since for e.g. Gaussian distributed traffic rate this value is exceeded with only 0.13% probability even without buffering. But the loss probability can be much higher for other rate distributions and the variance σ^2 strongly depends on the time scale, i.e. the interval length at which the traffic rate is measured.

Therefore we represent traffic variability in different time scales by the 2nd order statistics, i.e. the variance σ^2_N of the traffic rate in time slots of length $N{\cdot}\Delta$ in the range 1ms - 1000s, i.e. starting with Δ = 1ms and for N = 1 , …, 1 000 000. Self-similar processes are defined based on 2nd order statistics. The autocorrelation function is a common alternative and equivalent representation of σ^2_N [17][30]. In packet-based measurement, the size [byte] and a time stamp for each packet are captured in wire speed [6][9]. A time resolution well below 1ms enables precise byte counts per 1ms intervals, from which σ^2_N is computed.

1.1.6.2. Modeling and verification of traffic variability by 2-state Markov models

In order to support conclusions from measurements of traffic variability on packet loss, the effect on buffering and proper dimensioning can be verified via simulations and partly by analysis. An exact analysis is usually only feasible based on simplifying assumptions whereas simulations can cover a wider range of realistic scenarios. Simulation results are also subject to variability such that verified results need to be expressed in confidence intervals, which usually are becoming tighter in longer simulation runs. However, it is challenging to get tight confidence intervals for simulation of packet loss and other QoS characteristics for Internet traffic, because long simulation runs are required to cope with rare events and long range correlation in the traffic rates over time.

We study 2-state (semi-)Markov processes as a simple modeling approach for traffic variability. Queueing analysis is feasible for a system with constant forwarding capacity and 2-state (semi-) Markov arrivals [8][25], which enables an exact evaluation of the loss probability and delay distribution for a link with bandwidth B loaded with 2-state (semi-)Markov traffic with arbitrary mean rate μ and standard deviation σ.

We give a brief overview of validation aspects and challenges of traffic measurement, simulation, and analysis results.

Measurement

- enables direct evaluation of real scenarios;
- is often available only in limited samples; representativeness of data has to be checked for covering a network over a considered time frame;
- often does not adequately include rare events, which are relevant for network performance evaluation, i.e. packet loss at low loss rates, failure events, etc.

In general, measurement provides only partial information on statistical system behaviour for relevant and critical situations, from which the traffic profiles and performance measures have to be carefully evaluated.

Simulation

- is a flexible option for performance evaluation, which allows modification of parameters and components of a system;
- leads to variability in the results depending on stochastic processes describing the system behaviour, which can be bounded by confidence intervals;
- is expect to improve precision of evaluation through long simulation runs.

Analysis

- is applicable to simple models of system behaviour;
- is tractable only for limited system complexity (limited state space of the model);
- leads to exact results, which usually clearly reveal the influence of parameters,
- includes transient as well as stationary long term system behaviour and performance even when depending on rare events.

In general, analysis methods are less flexible than simulations. The underlying model and the analysis results have to be validated against simulation and/or measurement. Vice versa, the precision of simulation results and confidence intervals can be checked against exact analysis results by testing the simulation with a closest possible, i.e. most realistic analytically tractable model.

In the reminder of this section we investigate the accuracy of the simple type of 2-state (semi-) Markov model adaptations for the 2nd order statistics of measured traffic in comparison to other, especially self-similar models. The parameters for the model are determined in a fitting procedure. When compared to popular self-similar traffic models [17], it turns out that 2-state models achieve an essentially closer fit of the 2nd order statistics and thus are well suited for evaluation and validation of traffic profiles. In this way analytic approaches are confirmed to be useful for checking dimensioning rules.

1.1.6.3. Evaluation of variability and 2nd order statistics of measured IP traffic

Measurement evaluations have been performed in the IP network of Deutsche Telekom AG by means of a hardware tool for quality management which captures headers and timestamps of IP packets being exact to the microsecond. We have analyzed traffic on three parallel 1 Gbit/s links connecting aggregation routers at the boundary of the backbone. The monitored links carry residential user's traffic to the IP network with DSL access bandwidths of up to 16 Mb/s. The following results are based on a typical measurement trace from December, 18th 2011, 7-9 p.m.

Within these two hours, around 600 million IP packets have been counted in the downlink. For the purpose of our statistics, packets are aggregated into flows. A flow is a single TCP connection or UDP application specified by addresses and ports and a QoS marking in the packet headers. This quintuple enables us to evaluate statistics per flow and for aggregates of flows. In order to obtain the 2nd order statistics, we start with the byte count and the corresponding traffic rate of a flow or an aggregate in each 1ms interval of the 2 hour measurement period. Having measured the traffic rates $R_\Delta(t)$ for Δ = 1ms, rates for longer intervals $N\Delta$ are also determined as mean values over several intervals. In particular, we evaluate double length intervals $2^K\Delta$ up to 1000s.

In this way we have examined the total downstream traffic (in the following abbreviated by TDS), and in addition different classes of the traffic, in particular voice traffic (VoIP), and traffic coming from the popular platforms Facebook (FB), YouTube (YT) and RapidShare (RS). The latter is a direct download application with popularity especially in Germany. Traffic has been classified based on IP addresses and ports rather than deep packet inspection. Due to this deliberately simple approach, only a subset of each considered application is identified.

In Figure 21, the burstiness of the aggregated traffic of each class is compared over several time scales. The total traffic has lower burstiness of 0.3 in terms of the ratio of standard deviation to the mean rate even in 1ms intervals due to the statistical multiplexing effect. The dimensioning rule $B = \mu + 3\sigma$ then still leads to $B = \mu + 3 \cdot 0.3 \cdot \mu = 1.9\mu$, i.e. to a required bandwidth of almost twice the mean traffic rate. The burstiness is decreasing in longer time scales, i.e. when the variation of the traffic rate is evaluated over longer intervals. The traffic components that have been filtered for single applications show much higher burstiness because small fractions of a few percent of the traffic can only partly benefit from statistical multiplexing. The decrease of the burstiness over longer time scales is different for those applications, with an almost constant level at 1 for Rapidshare from 0.01s to 10s as one extreme.

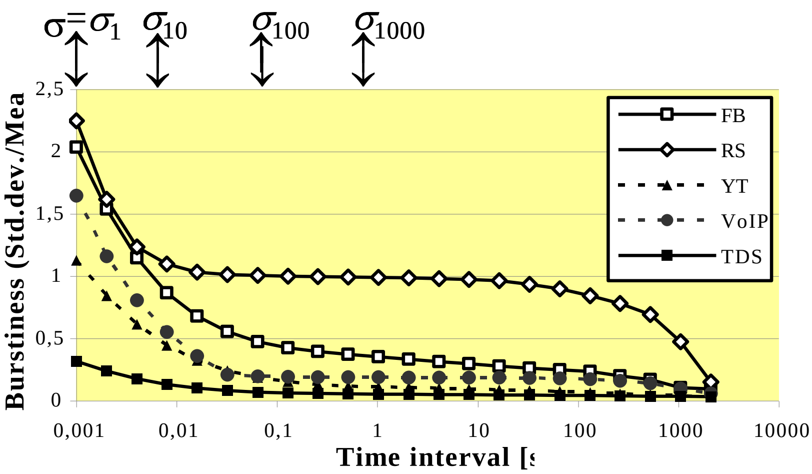


Figure 21: Aggregated traffic burstiness over time scales

1.1.6.4. 2-state Markov models

2-state models were proposed about fifty years ago by Gilbert and Elliott [5][7] for channels with correlated error events. In principle, results on their 2nd order statistics and autocorrelation are covered by results on *M*-state Markov models [19][25][31]. General *M*-state Markov results involve Eigenvalue solutions and computation of the roots of a characteristic polynomial [29][31] to represent the 2nd order statistics.

On the contrary, an explicit formula of the 2nd order statistics of 2-state (semi-)Markov processes can be derived with clear implications on parameter fitting and the suitability for traffic and error profiles. To the authors` knowledge, explicit results are still missing in literature despite many case studies on 2-state processes contained in work on Markovian performance analysis [1][5][7][13][16][18][21][23][28].

Extended 2-state schemes as represented in Figure 1 are shown to provide more flexibility for an essentially closer fit of the 2nd order statistics without adding much complexity in modeling and analysis. 2-state models are often combined in 2^M-state Markov models by

- superposition of M 2-state models [19][25],
- hierarchical refinement of a 2-state model involving 2-state models per state on the next level [16],
- extensions of 2-state models to include memory in an M-th order Markov chain [24].

Structured 2^M-state Markov models of those types facilitate the performance analysis and the parameter adaptation to fit measurement. Nevertheless, the studied approaches again start from different special 2-state formats. Starting from a more flexible 2-state model, new potential can be expected to improve the accuracy or to reduce the state space.

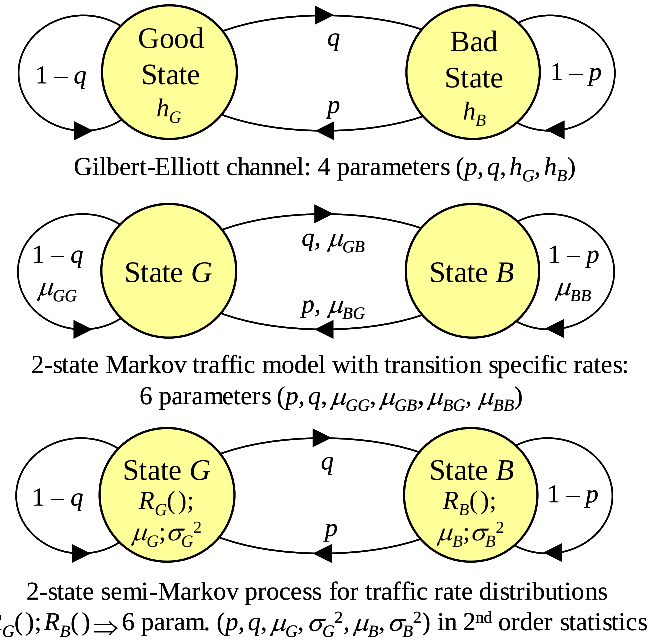


Figure 22: Gilbert-Elliott and general 2-state Markov models

We introduce 2-state semi-Markov processes (SMP(2)) for traffic profiles. As illustrated in Figure 22, they provide a generalized scheme including Gilbert-Elliott channels and other 2-state models studied in literature.

We characterize SMP(2) processes as a series $\{R_t, S_t\}$ with discrete time index t = 0, 1, 2, …, where R_t is a random variable for the traffic rate and S_t for the state at time t. In order to represent the variability of IP traffic, we refer to a discrete time slotted model, such that t refers to a sequence of intervals of constant length Δ. The time index t can also refer to arbitrarily embedded points in time, for example, packet arrival times for the analysis of packet errors.

For the binary state space $S_t \in \{G, B\}$ of a 2-state model we still associate both state symbols with the usual Gilbert-Elliott channel notation for “Good” (G) and “Bad” (B). In the “Bad” state of Gilbert-Elliot channels, there is a high probability for errors to occur. In SMP(2), however, we assign a higher mean traffic rate to the “Bad” state that may cause overload phases.

States can change at every time instant t with probability q from G to B and vice versa with probability p. We assume $0<p,q\leq 1$ as a precondition for an irreducible Markov chain S_t.

$$\Pr\{S_{t+1} = G \mid S_t = G\} = 1-q;\ \Pr\{S_{t+1} = B \mid S_t = G\} = q;\quad \Pr\{S_{t+1} = B \mid S_t = B\} = 1-p;\ \Pr\{S_{t+1} = G \mid S_t = B\} = p. \qquad (1)$$

In the most general representation, the traffic rate R_t is assumed to be transition specific, i.e. depending on S_t and S_{t+1}:

$$R_{ij}(k) = \Pr\{R_t = k \mid S_t = i;\ S_{t+1} = j\} \quad \text{for } i,\ j \in \{G, B\}. \qquad (2)$$

2-state approaches are prevalently state specific with rates depending only on the current state. The transition specific extension has only minor effect on the result of the 2nd order statistics. However, it is indispensible for some use cases, e.g., for the relationship of bit and packet errors in 2-state channels [12][14]. The main 2nd order statistics result of equation (7) depends only on the mean values μ_{ij} and variances σ^2_{ij} rather than the complete distribution function $R_{ij}(k)$

$$\mu_{ij} = \sum_k k \Pr\{R_t = k \mid S_t = i; S_{t+1} = j\}; \quad \sigma_{ij}^2 = \sum_k k^2 \Pr\{R_t = k \mid S_t = i; S_{t+1} = j\} - \mu_{ij}^2. \tag{3}$$

Based on transition specific values we obtain stationary state probabilities $p_G = p/(p+q)$ and $p_B = q/(p+q)$ as well as the steady state mean μ and variance σ^2 of the traffic rate of 2-state Markov models

$$\begin{aligned}\mu &= p\frac{(1-q)\mu_{GG} + q\mu_{GB}}{p+q} + q\frac{(1-p)\mu_{BB} + p\mu_{BG}}{p+q};\\ \sigma^2 &= p\frac{(1-q)(\mu_{GG}^2 + \sigma_{GG}^2) + q(\mu_{GB}^2 + \sigma_{GB}^2)}{p+q} + q\frac{(1-p)(\mu_{BB}^2 - \sigma_{BB}^2) + p(\mu_{BG}^2 + \sigma_{BG}^2)}{p+q} - \mu^2.\end{aligned} \tag{4}$$

1.1.6.5. Defining 2nd order statistics and autocorrelation function

The 2nd order statistics is a basic characteristic of the variability of stochastic processes. Self-similar models are defined via properties of their 2nd order statistics.

Therefore, a covariance stationary stochastic process $R = \{R_t;\ t = 0, 1, 2, \ldots\}$ with mean $\mu = E(R)$ and variance $\sigma^2 = E(R^2) - \mu^2$ is considered. For $N = 1, 2, \ldots$, we denote the averaging process over sequenced blocks of size N in the time series R_t as

$$R^{(N)} = \{R_t^{(N)};\ t = 0, 1, 2, \ldots\} \quad \text{with} \quad R_t^{(N)} = (R_{tN} + R_{tN+1} + \cdots + R_{tN+N-1}) / N. \tag{5}$$

Then, we denote the variance σ_N^2 of $R^{(N)}$ in steady state as the 2nd order statistics of the process including $\sigma_1^2 = \sigma^2$. An alternative and equivalent measure for the correlation in different time scales is the autocorrelation function $\alpha(k) = \mathrm{cov}(R_t, R_{t+k})/\sigma^2 = E((R_t - \mu)(R_{t+k} - \mu))/\sigma^2$. The relationship is due to computation rules for the variance of a sum of N random variables $R_{tN} + R_{tN+1} + \cdots + R_{tN+N-1} = NR_t^{(N)}$ via covariance coefficients $\mathrm{cov}(R_{tN+i}, R_{tN+j})$ [30]:

$$N^2\sigma_N^2 = \sum_{i=0}^{N-1}\sum_{j=0}^{N-1} \mathrm{cov}(R_{tN+i}, R_{tN+j}) = \left(N + 2\sum_{k=1}^{N-1}(N-k)\alpha(k)\right)\sigma^2$$

This equation determines σ_N^2 from $\alpha(1), \ldots, \alpha(N-1)$ and σ^2 and can also be transformed in order to obtain $\alpha(N)$ from $(\sigma^2 =)\ \sigma_1^2, \sigma_2^2, \ldots, \sigma_{N+1}^2$.

2nd order statistics of self-similar processes

A process R is exactly 2nd order self-similar with Hurst parameter H $(0.5 \le H < 1)$ [17][30], if

$$\forall N:\ \sigma_N^2 = \sigma^2(R^{(N)}) = \sigma^2 N^{2H-2}. \tag{6}$$

The same property is also valid for each of the derived processes $R^{(N)}$. In this way the result is transferrable to time scales of coarser granularity, i.e. longer time frames with increasing N. The autocorrelation function is proven to be invariant for all derived process $R^{(N)}$ [17][30].

2nd order statistics for 2-state semi-Markov processes

We extend the result for the 2nd order statistics from Gilbert-Elliot channels as derived in [11][12] to 2-state semi-Markov traffic models under the same steady state assumptions. As a most general 2-state result for transition specific rates we obtain

$$\sigma_N^2 = \frac{1}{N}\left(\sigma^2 + \beta\left[1 - \frac{1-(1-p-q)^N}{N(p+q)}\right]\right) \tag{7}$$

where σ^2 is given in equation (4),

$$\beta = \frac{2pq(E_G - E_B)}{(p+q)^2}\left[\frac{E_G - E_B}{p+q} + \mu_{BG} - \mu_{GB}\right]$$

$$\text{and} \qquad E_G = (1-q)\mu_{GG} + q\mu_{GB}; \quad E_B = (1-p)\mu_{BB} + p\mu_{BG}.$$

The general result preserves the same format as derived in [11][12] for the special case of Gilbert-Elliott channels.

As a special case lying in between Gilbert-Elliott and 2-state semi-Markov processes with transition specific rates, we obtain the 2nd order statistics for state specific rates in equation (8). Therefore we set $R_{GG}(k) = R_{GB}(k) = R_G(k)$ and $R_{BB}(k) = R_{BG}(k) = R_B(k)$:

$$\sigma_N^2 = \frac{1}{N}\left(\sigma^2 + \frac{2pq(1-p-q)(\mu_G-\mu_B)^2}{(p+q)^3}\left[1-\frac{1-(1-p-q)^N}{N(p+q)}\right]\right) \quad (8)$$

$$\text{where } \sigma^2 = \frac{p(\mu_G^2+\sigma_G^2)+q(\mu_B^2+\sigma_B^2)}{p+q} - \mu^2;\ \mu = \frac{p\mu_G + q\mu_B}{p+q}.$$

It is important to notice that for arbitrary rate distributions $R_{ij}(k)$ only the mean values μ_{ij} and variances σ^2_{ij} as well as the transition probabilities p, q have influence on the 2nd order statistics. This fact facilitates the parameter fitting for 2-state models in order to match the correlation of measured traffic profiles and it suggests the result (8) not only to hold for discrete distribution $R_{ij}(k)$, but for continuous ones as well. Obviously, the result is proven to be valid for any distribution function that can be arbitrarily closely approximated by a series of discrete distributions [8][29]. Therefore, the class of exponential and phase type distributions in each state G and B and, for instance, 2-state Markov modulated Poisson processes (MMPP) [25] as another 2-state variant are covered by equation (8).

In the next part, we look at packet measurement of Internet traffic on aggregation and backbone links. Traffic profiles of main applications are evaluated regarding the variability and 2nd order statistics per flow. This allows us to check how realistic traffic models appear when adapted to IP traffic traces.

1.1.6.6. Parameter fitting for self-similar and 2-state semi-Markov models

We compare the 2nd order statistics of different traffic classes of our measurement as shown in Figure 21 with by self-similar and 2-state Markov adaptations. For this purpose, we evaluate time scales from 1ms - 8.192s, covering the most relevant range for the efficiency of buffering, since longer burst tend to produce buffer overflow. In general, the basic time scale Δ can be flexibly chosen as most appropriate for modeling.

In self-similar modeling, the 2nd order statistic is given by $\sigma_N^2 = \sigma^2 N^{2H-2}$ [17][30]. Based on two parameters σ and H we match the variance of the smallest (Δ= 1ms) and the largest considered time frame ($2^{13}\Delta$ = 8.192s) in order to approximate the 2nd order statistics of measurement results. We set $\sigma^2 = \sigma_1^2 = \sigma^2_{\text{Mesmt.}}(\Delta)$ and compute the Hurst parameter H:

$$\sigma^2_{2^{13}} = \sigma^2 \cdot 2^{13(2H-2)} = \sigma^2_{\text{Mesmt.}}(2^{13}\cdot\Delta) \Rightarrow \quad (9)$$
$$H = 1 - \log_2[\sigma^2_{\text{Mesm.}}(1\text{ms}) / \sigma^2_{\text{Mesm.}}(8.192\text{s})]/26.$$

Figure 25 - Figure 28 compare the 2nd order statistics of the measurements with self-similar and 2-state semi-Markov approximations.

Next, we show how 2-state models with state specific rate distributions as illustrated at the bottom of Figure 22 with 6 parameters p, q, μ_G, μ_B, σ_G^2, σ_B^2 can be adapted with regard to the 2nd order statistics given in eq. (8).

The state specific mean values μ_G, μ_B and variances σ_G^2, σ_B^2 can be arbitrarily chosen as non-negative real numbers for adaptation. In general, knowledge on the traffic rate distribution function can be matched by 2-state semi-Markov processes as rate distribution in one of the states or as steady state rate distribution of the complete model. The rate distribution function and the correlation over time can be fitted independent of each other [19][25][29]. Naturally, including the rate distribution function enables more precise queueing and buffering analysis. Nonetheless, we stay focused on the rate variance and its correlation as the other main characteristics in traffic profiles.

We consider two constraints in order to match the mean μ and the variance $\sigma^2 = \sigma_1^2$ of the traffic rate distribution. σ^2 marks the left side of the 2nd order statistics in Figure 25 - Figure 28.

The linear constraint $\mu = (p\mu_G + q\mu_B)/(p+q)$ on mean rates is exploited in order to replace μ_G and μ_B by a single parameter m:

$$\mu_G = m\,\mu \quad \Rightarrow \quad \mu_B = (1+(1-m)p/q)\mu;$$
$$\text{where } 0 \le m < 1 \Leftrightarrow 0 \le \mu_G < 1;\ 1 < \mu_B \le (1+p/q)\mu. \quad (10)$$

Then the factors σ_1^2 and β of the representation (8) of σ_N^2 are expressed based on m:

$$\sigma_1^2 = \frac{p\mu_G^2 + q\mu_B^2}{(p+q)} - \mu^2 + \frac{p\sigma_G^2 + q\sigma_B^2}{(p+q)} = (1-m)^2\,\frac{p}{q} + \frac{p\sigma_G^2 + q\sigma_B^2}{(p+q)};$$

$$\beta = \frac{2pq(1-p-q)(\mu_G-\mu_B)^2}{(p+q)^3} = \frac{2(1-p-q)(1-m)^2\mu^2 p/q}{p+q}. \quad (11)$$

Both terms of equation (11) implicate a separation of the parameter impact, such that the variances σ_G^2, σ_B^2 only have influence on σ_1^2, whereas β and the reminder of equation (8) depend only on *m, p* and *q*. For large σ_1^2 we have freedom of choice in σ_G^2, σ_B^2, whereas small σ_1^2 leads to the restriction $\sigma_1^2/\mu^2 \geq (1-m)^2 p/q$.

A closer look on equation (8) reveals β to depend on only two parametric terms, namely, *p* + *q* and $(1-m)^2 p/q$. Consequently, parameter fitting can be reduced to studying the impact of both terms on β and thus on σ_N^2.

We start the parameter fitting from e.g. *m* = *p* = *q* = 0.5 and focus on matching σ_N^2 first and matching the sum $\sum_{k=1}^{N_{\max}} \sigma_k^2$ ($N_{\max}$ = 8192) in a second step. Both steps are facilitated by monotonic dependencies, as expressed in the statements 1., 2. without enough room for proofs:

1. When *p*+*q* is decreasing, all coefficients σ_N^2 for $N \geq 2$ are increasing, if $(1-m)^2 p/q$ is kept constant.

So we can adapt $\sigma_{8192}^2 = \sigma_{\text{Mesmt.}}^2(8.192\text{s})$ in each example of Figure 25 - Figure 28 by modifying *p* + *q* in interval halving steps in the range 0 < *p* + *q* < 2, while *m* and *p*/*q* are kept constant. Long range dependency leads to large σ_N^2 and thus small *p* + *q*. The monotony is illustrated in Figure 23 by four curves for varying *p* + *q* with constant ratio *p*/*q* = 5 and constant parameters listed in Table 1 for the Facebook traffic example.

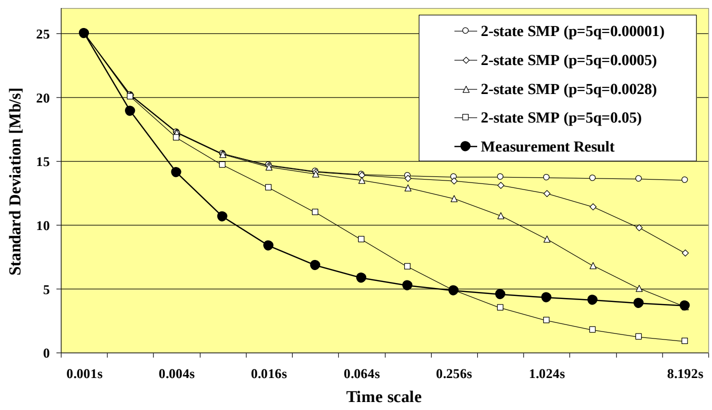


Figure 23: Fitting of σ_{8192}^2 by varying *p*+*q* (Step 1)

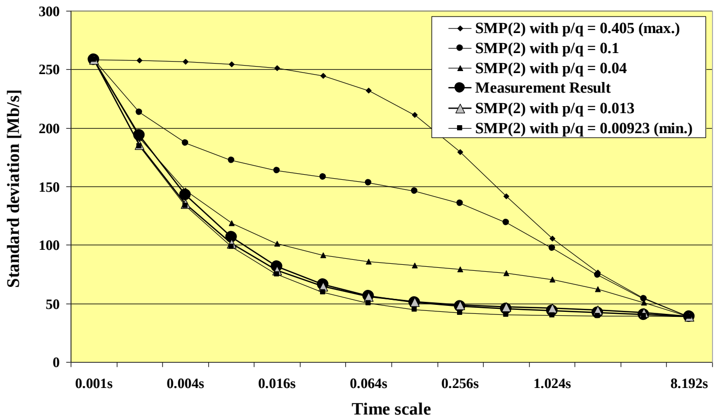


Figure 24: Fitting of $\sum_{k=1}^{8192} \sigma_k^2$ (& σ_{8192}^2) by varying *p*/*q* (Step 2)

2. When *p*+*q* has been chosen in order to match a coefficient $\sigma_{N\max}^2$ ($N_{\max} \geq 3$, e.g. $N_{\max} = 8192$) then the sum $\sum_{k=1}^{N_{\max}} \sigma_k^2$ is monotonously increasing with $(1-m)^2 p/q$ until a maximum is reached at $(1-m)^2 p/q = \sigma_1^2/\mu^2$.

We use this property in order to match $\sum_{k=1}^{N_{\max}} \sigma_k^2$ in addition to $\sigma_{N\max}^2$ in another interval halving procedure. Naturally, not every predefined value of the sum can be achieved by 2-state modeling, but the maximum and minimum for 2-state models can be determined to obtain a closest possible fit. Figure 24 illustrates the procedure, showing a set of 2-state semi-Markov (SMP(2)) 2nd order statistics curves. They are tuned for an exact match of both end points σ_1^2 and $\sigma_{8192}^2 = \sigma_{\text{Mesmt.}}^2(8.192\text{s})$, but they are varying in the term $(1 – m)^2\, p/q$, i.e. in p/q for constant m = 0.5. A match of $\sum_{k=1}^{8192} \sigma_k^2$ with the measurement curve is found for p/q = 0.013. As the boundaries we obtain $0.00923 < p/q \le 0.405 = \sigma_1^2 / (\mu^2 (1-m)^2)$.

The proposed fitting with regard to $\mu, \sigma_1^2, \sigma_{N\max}^2$ and $\sum_{k=1}^{N_{\max}} \sigma_k^2$ is experienced to produce good approximations for various traffic measurement traces in our tests.

A final optimization step can be added with regard to a minimum squared deviation criterion. This criterion does not imply monotonic properties and is implemented by a standard optimization approach for which we can recommend the method proposed by Powell [21][22]. This step is not included in the adaptation results of the examples. As an improvement, this shifts approximations more to the center of the measurement curve, but has a similar effect on all models and does not influence the conclusions of their comparison.

Figure 25 - Figure 28 compare 2-state semi-Markov and self-similar fitting curves for the 2nd order statistic with measurement. The parameters of the examples are listed in Table 1.

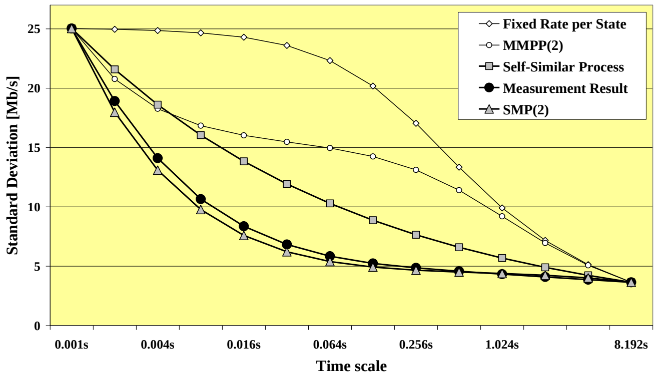


Figure 25: Fitting of 2nd order statistics for Facebook traffic

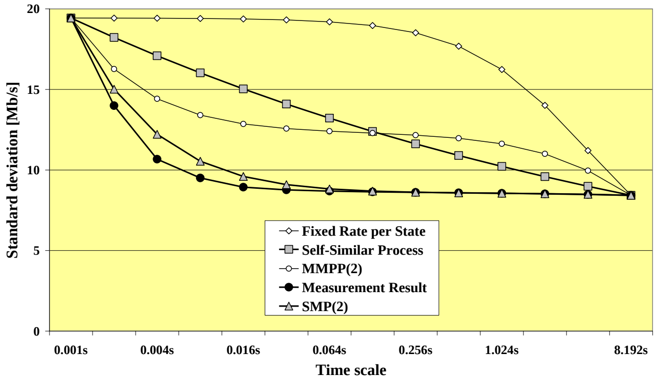


Figure 26: Fitting of 2nd order statistics for RapidShare traffic

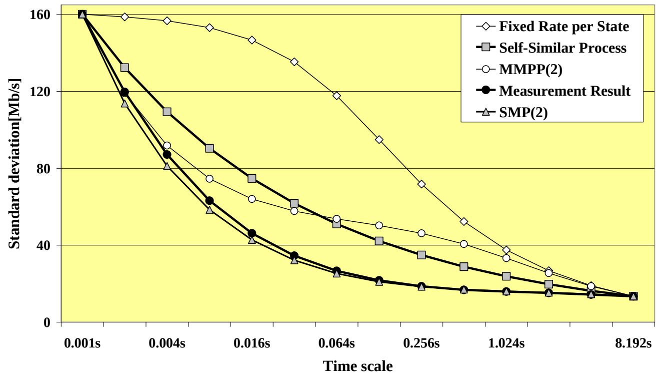


Figure 27: Fitting of 2nd order statistics for YouTube traffic

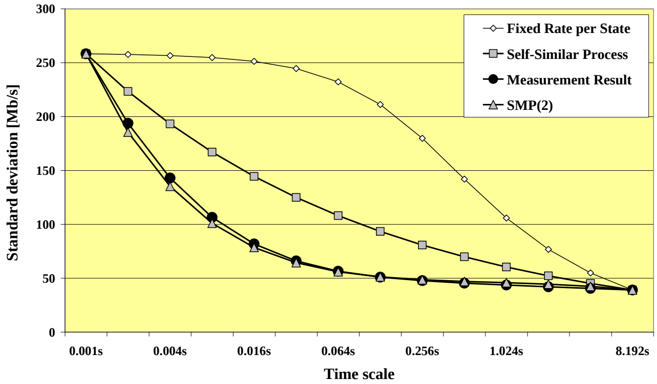


Figure 28: Fitting of 2nd order statistics for the total traffic

Table 3: Parameters of self-similar and 2-state semi-Markov processes fitted to the 2nd order statistics of measured traffic

Model Parameter Traffic Type	μ [Mb/s]	Hurst-P. H	Parameters of the 2-State Semi-Markov Process q	p	μ_G	σ_G	μ_B	σ_B
YouTube	142.1	0.7250	1.182E–4	5.673E–6	71.050	79.197	145.510	162.20
Rapid Share	8.64	0.9073	2.215E–6	8.664E–6	4.322	6.203	25.549	36.674
Facebook	12.29	0.7861	1.068E–4	5.551E–5	6.145	11.587	15.485	29.200
Complete Traffic	811.0	0.7903	1.345E–4	1.743E–6	405.500	126.865	816.254	255.38

1.1.6.7. Restricted fitting flexibility for constant rates per state and MMPP(2)

The Gilbert-Elliott channel has to be extended or modified for applications to traffic profiles. One way is to replace the error probabilities h_G and h_B in each state G and B with fixed traffic rates μ_G and μ_B, respectively.

The mean rate μ and variance σ^2 for this model is given by

$$\sigma^2 = E(X^2) - \mu^2 = \frac{p\mu_G^2 + q\mu_B^2}{p+q} - \mu^2; \quad \mu = \frac{p\mu_G + q\mu_B}{p+q}.$$

With a closer look on σ^2, we can simplify the 2nd order statistics of equation (8) for the special case of fixed traffic rates:

$$\sigma^2 = \frac{pq(\mu_G - \mu_B)^2}{(p+q)^2}; \sigma_N^2 = \frac{\sigma^2}{N}\left(1 + \frac{2(1-p-q)}{p+q}\left[1 - \frac{1-(1-p-q)^N}{N(p+q)}\right]\right).$$

The evaluation for small N yields:

$$\sigma_2^2 = \sigma^2(2-p-q)/2;$$
$$\sigma_3^2 = \sigma^2[9-8(p+q)+2(p+q)^2]/9;$$
$$\sigma_4^2 = \sigma^2[16-20(p+q)+10(p+q)^2-2(p+q)^3]/16;\ \text{etc.}$$

The shape of the 2nd order statistics of the constant rate model only depends on the single term *p+q*. The demand for an exact match of σ_{8192}^2 in step 1 of the previous section determines *p+q* together with the complete 2nd order statistics.

As a consequence, the constant rate model leaves most of the 2-state fitting capabilities unused and is far from measurement results in the examples of Figure 25 - Figure 28.

When we extend the model to constant rates per transition as shown in the 6 parameter Markov model in the middle of Figure 22 then we again obtain a 2-dimensional fitting space with capabilities similar to the SMP(2) model.

Two-state Markov modulated Poisson processes (MMPP(2)) are also restricted to a single parameter in the 2nd order statistics. The MMPP model for traffic suggests exponentially distributed rates with different rates μ_G, μ_B in each state. A similar traffic modeling scheme based on a superposed MMPP(2) variant with geometrical rate distribution is worked out by [19][25] including an elaborate adaptation scheme.

State specific variances for exponential rates are $\sigma_G{}^2 = \mu_G{}^2$, $\sigma_B{}^2 = \mu_B{}^2$. We obtain the MMPP(2) 2nd order statistics ($\sigma^2 \geq \mu^2$):

$$\sigma_N^2 = \frac{1}{N}\left(\sigma^2 + \frac{1-p-q}{p+q}(\sigma^2-\mu^2)\left[1-\frac{1-(1-p-q)^N}{N(p+q)}\right]\right).$$

In our evaluation examples, MMPP(2) models are partly better than the constant rate model, but cannot cope with fully exploited (semi-)Markov fitting results. MMPP(2) fails for the total traffic, since the variance of the total traffic is lower than the minimum MMPP(2) variance $\sigma^2 \geq \mu^2$.

1.1.6.8. Applications of 2-state Model in the Literature

In telecommunication, the two main stream applications of (semi-)Markov models with two or more states are

- traffic profiles for dimensioning and QoS aspects and
- error characteristics for evaluation of coding for reliable transport over error prone and fading channels.

One part of modeling studies deals with measurement and evaluation of Internet traffic profiles with focus on variability and quality of service/experience (QoS/QoE) demands of applications [15]. Such work considers impairments caused by packet loss, delay and other failure cases [6][9][10][17][18] as well as parameter fitting procedures [19][25]. Traffic models are a basic input for queueing analysis of buffered router and switching systems. Factorizations and matrix-analytic analysis [8][29][30] as well as simulations [27] reveal the distribution function of the traffic rate to be relevant in a detailed view beyond the variance. For both, variability measures of the correlation over a range of time scales have to be considered.

The error characteristics of channels is prevalently important in wireless transmission with fading channels [1][18]**Error! Reference source not found.**[24], on the WLAN MAC layer [16] and in cellular networks, e.g. for defining QoS standards in UMTS networks [21]. The performance of error detecting and correcting codes is evaluated for specific coding schemes [12]**Error! Reference source not found.** and in a generalized Markov modeling framework [31]. When digital channels are extended from bit errors to packet and block errors of higher layer protocols, the Gilbert-Elliott model needs to be extended to transition specific error probabilities [11][12][14] as depicted in the middle of Figure 22, because packet errors depend on the state of the bit channel before and the state after a considered packet.

2-state Markov approaches are popular in many more disciplines including

- economics, e.g., for volatility in markets,
- nuclear physics [13],
- statistics in medicine [23],
- or document and image analysis [28] etc.

1.1.6.9. Conclusions and outlook

We compare the 2[nd] order statistics of measured traffic of Internet applications with basic 2-state (semi-)Markov processes, whose 2[nd] order statistics is determined as an explicit formula.

The results show that 2-state (semi-)Markov models in general provide a two-dimensional fitting space for the autocorrelation in different time scales. They enable essentially closer fitting of traffic profiles when compared to self-similar processes including a tractable exact analysis of delay and loss distributions for link dimensioning, which is more complex for self-similar models. Many analytical studies in the literature restrict to special 2-state cases e.g., 2-state Markov with fixed rate per state or MMPP(2), providing only a single parameter and thus a restricted flexibility in the correlation function. In this way, the potential of 2-state models for verification of measurements with conclusions on packet loss and delay distributions may often be under-estimated.

We recommend the SMP(2) semi-Markov model and the 2-state Markov model with transition specific rates in order to fully utilize 2-state fitting capabilities. Both models have 6 parameters as illustrated in Figure 22. A fitting scheme for measurement statistics of correlated traffic or error rates is outlined in the procedure proposed in section **Error! Reference source not found.**. The combined application of simulation and 2-state analytical traffic modeling for verification of link dimensioning with guaranteed QoS properties is for further study.

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1.2. Macroscopic traffic management

1.2.1. Performance evaluation of NB-IFOM, evaluation of decision strategies for flow allocation

1.2.1.1. Objectives

IFOM (IP Flow Mobility) is a solution to allow dynamic management of IP flow routing policies issued by the operator. The architecture builds on top of an IETF proposal which introduces Home Agent initiated flow bindings into Mobile IPv6 signalling. The multiple care-of address registration extension of the Mobile IPv6 protocol makes it possible to use multiple egress interfaces and operate policy based routing using these interfaces by the flow binding mechanism specified in RFC6089. The document describes that in order to initiate a flow binding operation a valid Mobile IPv6 binding is required. Similarly to that technique a HA initiated flow binding operates via the Flow Binding Indication (FBI) and the Flow Binding Acknowledgement (FBA) messages, where the latter is used for the acknowledgement of the FBI message. By relying on the basic concepts introduced by the soon to be RFCd HA initiated flow binding proposal we extended the functionality of the model by defining monitoring points for traffic state and analysis overall the network and adding policy servers to manage the monitoring points and enforce policies based on the processed data.

Despite the fact that IFOM already solves the problem of simultaneous attachment to overlapping radio coverages and allows fine granularity of IP flow mobility between access networks, it has its drawbacks.

When an operator wants to initiate a change in the flow routing policies (i.e. network initiated flow mobility is on the line), the current solution relies on the Access Network Discovery and Selection Function (ANDSF) according to 3GPP TS 23.402:

- The UE registers with an ANDSF server to receive access network information and operator preferences with regard to the selection of an access network.
- The ANDSF service will notify individual UEs about updated flow routing policies.
- UE sends a DSMIPv6 Binding Update (HoA, CoA, Lifetime, BID, FID, flow description) message to the PDN-GW (HA) together with the requested routing rules via the FID mobility option with both the routing filters and the BID.
- The PDN GW sends an IP-CAN session modification request to the PCRF providing the updated routing rules to the PCRF. The PCRF stores the updated mapping between routing addresses and SDFs.
- The PCRF sends an acknowledgement to the PDN GW, including updated PCC rules if appropriate.
- The HA sends a Binding Acknowledgment (Lifetime, HoA, BID, FID) to indicate which routing rules requested by the UE are accepted.
- The PCRF ensures the relevant QoS rules and/or releases resources that were moved away

This solution is very UE centric (the operator firstly delivers the routing policies to the UE, and then the UE must provide these policies to the PDN Gateway). The ANDSF has no interface to the PCC system, therefore requires other ways to get informed about the updated flow routing policy for a particular UE.

The advantages of network-based IP flow mobility (NB-IFOM) enabled operators to enforce IP flow routing policies without involving the UE first, enabling the PCRD (the central policy control entity) to decide on the flow routing policy based on e.g., the available resources in the network, before signaling the policies to the UE.

The solution is more efficient than the ones that rely on the UE to perform policy acquisition and enforcement. According to the current standard it is possible that the network context and resource availability may have changes by the time the UE provides the routing policies to the network; therefore the PCRF will not be able to authorize the new flow policies anymore.

In this validation we consider scenarios from the following groups:

- Simple network-based IFOM use cases, where the decision triggers are delivered by simple node-centric or other mobility events (e.g., UE attaches to the network, UE performs handover, node failure in the network, etc.).
- Advanced use-cases where flows are routed based on the overall status of the network. In these scenarios the central policy manager entity recognizes the problem by periodically receiving reports from the network and sets a goal to remedy the situation. The manager would then change policies for existing and new flows until the situation changes. These emergency policies are revoked from the system when the network performance is again within operational threshold.

This validation is about the performance evaluation of Home Agent initiated flow binding providing Selective IP traffic offload, and evaluation of decision strategies for flow allocation using this flow mapping technique. The technology to be evaluated does not consider the need of elimination of anchor points as it leaves the Home Agent intact. However the design of the decision algorithms and the overall scheme may be applied also in a distributed environment, if the P-GWs are distributed. Test scenarios include mostly demonstrating and evaluating traffic offload capabilities of this technology. High number of users will be emulated. Flow characteristics may not be measured but they will be known in advance in the validation, and given as input to the decision algorithm. It is possible to integrate the validation testbed with DPI solutions from Montimage to measure different network and/or flow characteristics, such as QoS parameters of flows (bandwidth, user QoE related indicators) grouped by location in the backhaul; indicators of usage level/load of different nodes, access networks and backhaul segments.

1.2.1.2. Validation scenarios

Our validation scenarios consider the operator's perspective where in general the network throughput should be maximized but end-user SLAs should also be respected. One possible policy to target that load on the backhaul networks of different access segments should be balanced, however other global objective functions will be studied as well. Decision algorithm and the technology have influence on the overall network performance. The following main validation scenarios will be considered:

- Provision of default flow binding: this scenario means the proper enforcement of Service Level Agreements (SLAs) to users. Three main priorities are to be defined for user profiles (e.g., 3GPP access takes priority over WiFi when providing VoIP service; WiFi access takes priority when providing IPTV service; P2P traffic is only allowed through WiFi connection). Users' default profile can be downloaded through the Home Agent to the UE when registering to the network. This scenario can be particularly beneficial for downstream flows, which can then be directly routed to the pre-configured access (i.e. desired interface). KPI: does the flow binding provision work?
- Flow handover due to node failure: this scenario emulates a hardware failure or maintenance event which can cause an outage one of one or more network nodes. The operator wants to guarantee the fair a continuous maintenance all the user flows, and applies network-based IFOM for these purposes. Two sub cases will be studied. 1) the node failure shuts down the non-3GPP Access Gateway; this triggers a network decision to start routing all the user flows through the 3GPP access. This use case is basically a full network-initiated handover from the non-3GPP access towards the

available connection. 2) a non crucial node goes down (like a redundant router providing multipath communication); this only affects some flows and require reallocation of them. E.g., a network node affected by actual youtube video sessions goes down and the policy server moves flows to working network segments. KPI: successfully moved flows/total affected flows of affected UEs.

- Traffic offload: this is an umbrella scenario considering all kinds of network-based flow handover management events executed due to congested/exhausted links inside the operator’s network (e.g., reallocation of traffic flows from the 3G access to WiFi/WiMAX/etc. due to increasing traffic load in the 3G cellular segment). In this scenario we rely on aggregated measurement results of user traffic and load on different backhaul segments (e.g., amount of youtube video on 3G/WiFi, load on backhaul segments of 3G/WiFi). Intervention: based on static/pre-defined or dynamic optimization functions (e.g., load on the 3G backhaul must stay below threshold) some of the already established sessions (e.g., the youtube flows) are moved by the Home Agent (an updated flow descriptor is to be sent to the UE). Dynamic feedback of the intervention could trigger 1) modifications on running operations 2) initiation of further operations. KPI: backhaul load/throughput; KPI: how much time do we need from recognizing the problem till reaching the goal (transaction time).
- Flow Binding Revocation: this scenario considers network-centric administrative problems (e.g., load of an anchor node reaches a given threshold) which could result in deletion of one or more flow bindings. Mobile nodes should be informed that they are no longer able to use IP mobility service for the affected flow(s). E.g., policy server collects the aggregated eNodeB load data. If the load exceeds a given threshold, flows of mobiles handled by the affected eNodeB are started to be moved to non-3GPP access networks. Optionally pre-defined affinities (like VoIP is preferred to remain on 3GPP access) can also be taken into consideration. KPI: how much time does the affected node above the given threshold/how much flow are affected.

1.2.1.3. Validation tools

The main tool within this validation task is the testbed designed and implemented for network-based IFOM performance evaluation.

The proposed architecture (Figure 5) consists of a central Policy Server which knows the overall state of the policies. It receives measurements and flow descriptions from various parts of the network. The Policy Server periodically checks the predefined QoS parameters and makes decisions based on the received reports in order to move UEs among access networks.

The Policy Server sends a *polCmd* command to the worker agent running on a Home Agent to deliver its decision to the network. The HA will act on behalf on the Policy Server, performing the *FlowBindingInit* and *FlowBindingAck* signalling with the UE. The HA will report the status of the command back to the Policy Server with the *policyAck* message.

It is important to mention that since the Policy Server is not part of the IPv6 mobility framework, it has to be informed of every event related to mobility, such as new client connection, handover event, disconnects. The HA informs the Policy Server with the *policyReq* message to keep the database of the Policy Server synchronized. The Policy Server may reply with a *polCmd* message to perform actions based on the changed state in mobility.

Since the implementation is to be performed on Linux, several open-source tools can be used tools for validation. For traffic generation and basic functional tests we use the ping, netperf and traceroute utilities. To dump packets and verify routing we use the tcpdump utility.

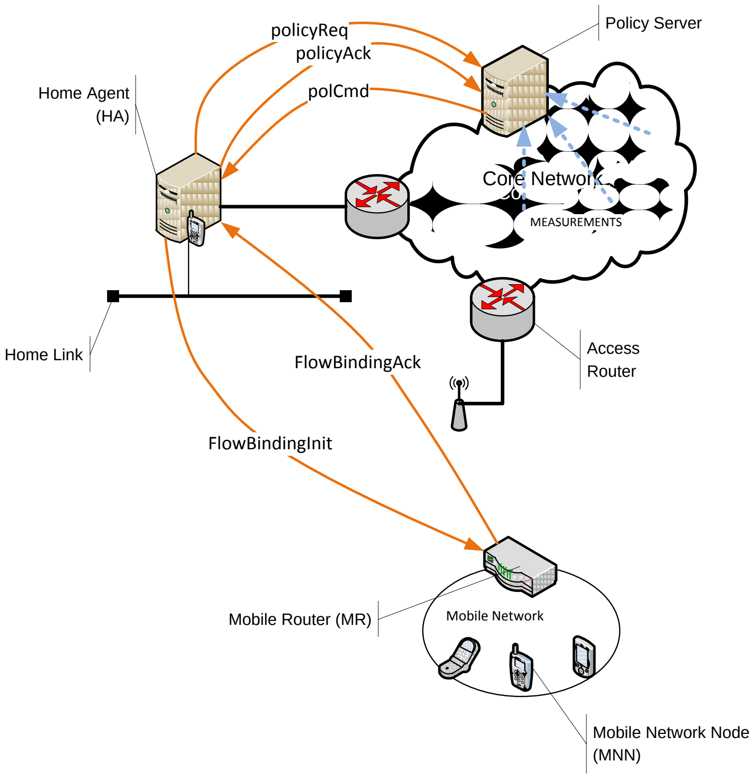


Figure 5 - Validation architecture and main elements for network-based IFOM.

Due to the limited capacity of the real-life test bed (i.e., only a couple UEs, lack of network resources) we could only simulate multiple users by filling up the Aggregated Binding Cache with a simulator and setting the network performance parameters with the same method. Therefore the hybrid simulator-implementation test bed consisted of the following modules on the Policy Server node (see Figure 6 for the implemented setup).

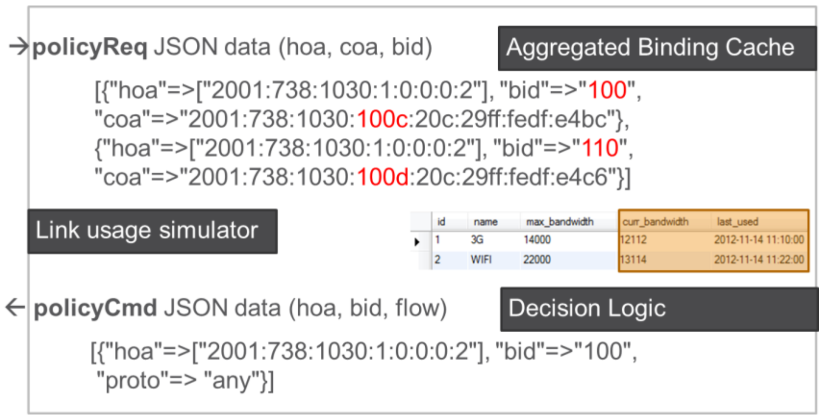


Figure 6 – Hybrid, implemented/emulated Policy Server architecture

1. Aggregated Binding Cache:
   - Contains the Mobile IPv6 bindings of all egress interfaces of a MN
   - Supports multiple Home Agents
   - Key: (HA, HoA, CoA, BID)
   - Always up-to-date (maintained by policeReq)

2. Decision logic:
   Selects default interface for a new user based on a distribution algorithm.

3. Link usage simulator:

   Changes the current bandwidth of the available backhaul links (3G, WLAN) by continuously updating the field in the database with a pre-recorded series (with some entropy). We used a basic input from real ISP statistics and extrapolated the data with random fuzz to simulate more measurement points. The dataset had two daily peaks and linear hourly bursts of users. To better demonstrate the different load on the backhauls we shifted the two datasets to increase the entropy of the system (Figure 7). Aside from current usage, the measured incoming user bursts and the time each user spend on the network have to be simulated as well. Depending on the scenario we chose a variety of distribution setups to successfully demonstrate every aspect of the system.

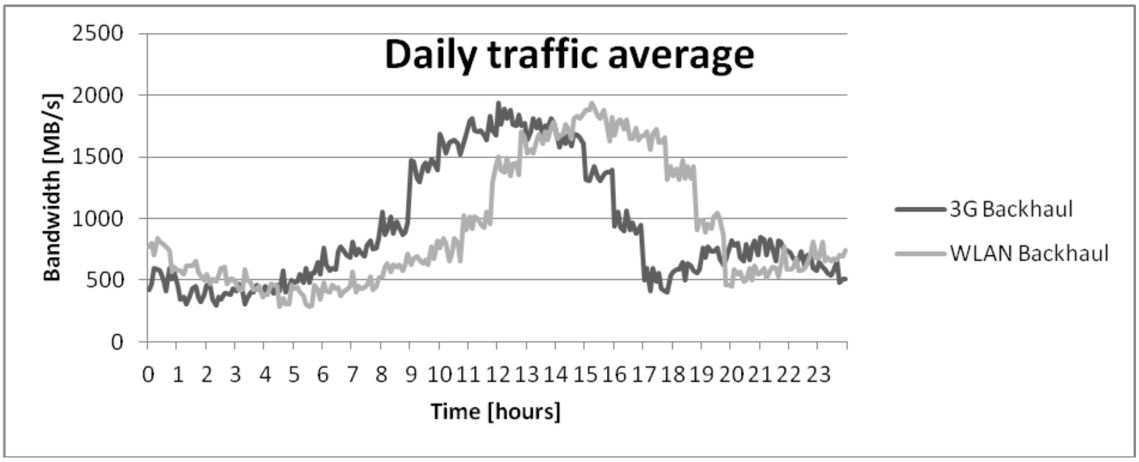


Figure 7 – Link usage simulation

1.2.1.4. Expected results

As functional validation itself is not enough to verify the implemented solution in reference to the overall performance to the network, a number of performance indicators were defined (see Section 1.2.1.2). Using these KPIs in each scenario we expect to see the numerical improvements due to the application of the network-based IFOM scheme and the decision algorithms. It also means that performance drops are expected in cases where the implementation is not available. Performance degradation caused by the implementation (i.e., significant overhead or latency growth) is not acceptable.

1.2.1.5. Results

Three main use cases were examined. Default Flow Binding Provisioning (1) is used for example in an environment where a central entity wants to force Service Level Agreements (SLA) to a customer, e.g., forcing P2P traffic through WiFi while allowing UMTS access for HTTP traffic. The Traffic Offloading use case (2) makes it possible to move certain data flows from one interface to another, e.g., in case of increasing traffic load in 3G segment move video streams to the Wi-Fi segment. Policies can be much complex based on the fact that the core network entities know about their actual traffic conditions. Flow Binding Revocation (3) is useful when due to an administrative decision a certain flow binding is no longer valid for the MN.

All the above use-cases can be broken down to the following tasks:

1. Load-balancing of newly connected users by the distribution of default flow-policies
2. Moving individual flows as the network performance parameters change (e.g., backhaul traffic volume of 3GPP / non-3GPP accesses)
3. Selecting and executing decision algorithms

The NB-IFOM system depends on the successful establishment of Mobile IPv6 bindings, and depending on the policy enforcement mode we can establish the following two concepts:

- Strict provisioning: The HA waits until the flow-policies are installed then registers the binding. Useful if the operator wants to ensure that all data packets are sent on the policy-designated link. It introduces delays in the first MIPv6 binding process.

- Loose provisioning: The HA acknowledges the binding; policy distribution is a parallel task. It is possible that until the new policies take effect the packet flow will use an asymmetric suboptimal path.

All flows belonging to a newly connected user must belong to a default policy, which binds all flows associated with a UE to a predefined egress interface. (e.g., all traffic must go through the WLAN access link).

At this time of the session we don't have per-user flow statistics as we cannot predict the flow usage of a UE in advance, therefore every flow is treated equally.

The triggering scheme of the process (Figure 8):

1. The Home Agents send a policyReq message to the Policy Server
2. The Policy Server enforces the default flow policy selection algorithm on new bindings
3. The PS sends a policyCmd command to the HA
4. The HA initiates flow binding with the MR

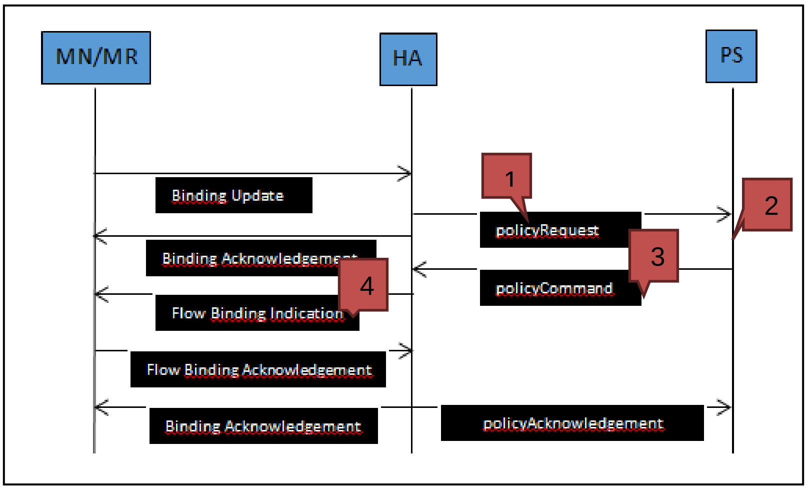


Figure 8 – Initial flow distribution in NB-IFOM

The distribution policy in the Policy Server for delivering default flow policies is static. (i.e., set in a configuration file).

Table 4 summarizes the implemented load-balancing techniques with respect to the overall stability of the network. The objective function of the methodology is to prevent the use of policies (Note: Policies will only be applied when the traffic parameters and QoS inside the backhaul and the core networks are suboptimal), by keeping the load distribution stable. This KPI does not consider rapid changes in existing data flows.

Table 4. The implemented decision algorithms

Distribution alg.	Input argument	Remarks
Round-robin	-	Flows are distributed among available uplinks evenly. Good for Best Effort QoS. Uneven flow bandwidth distribution may result in policy reallocation.
Least Used	curr. bandwidth	Dynamic selection based on the recently used link
Lowest latency	curr. latency	Dynamic selection based on the actual latencies on the links
Overflow	curr. bandwidth	The algorithm waits until one of the links become full, allowing network policy events to be triggered before utilizing all available media.

The Round-robin algorithm requires no input arguments as it always produces the same behavior regardless of the actual network status. The statistics calculation was done in Excel based on the datasets generated by the simulator scripts. The actual per-hour user numbers are represented on the right y-axis, while actual throughput is depicted on the left y-axis. Each user spend exactly 1 hour in the network. If users would continue to use the network throughout the day, the distribution would be an incremental value and it would not demonstrate the properties of the algorithm. Figure 9 shows that the RR algorithm evenly distributes the incoming bursts and keep the backhauls from reaching the maximum capacity and getting overloaded.

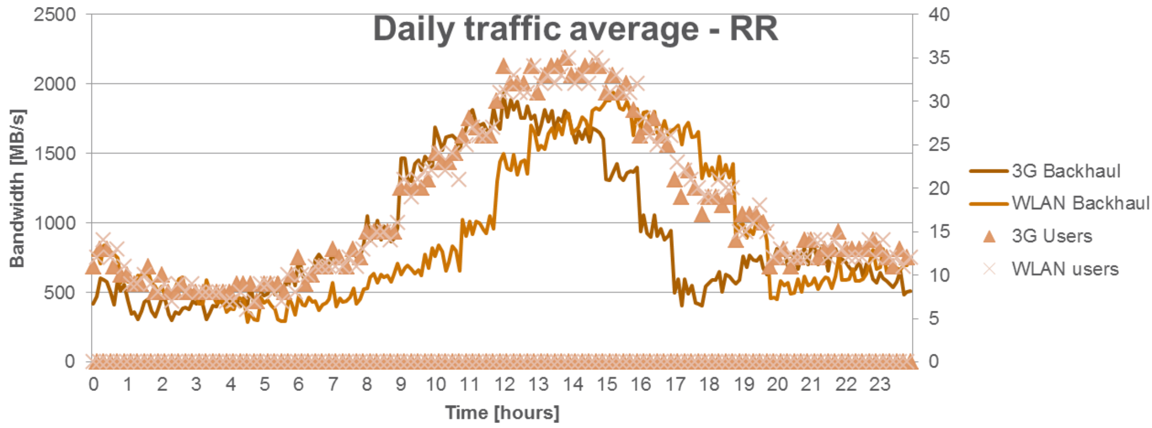


Figure 9 – Performance of the Round-robin decision algorithm

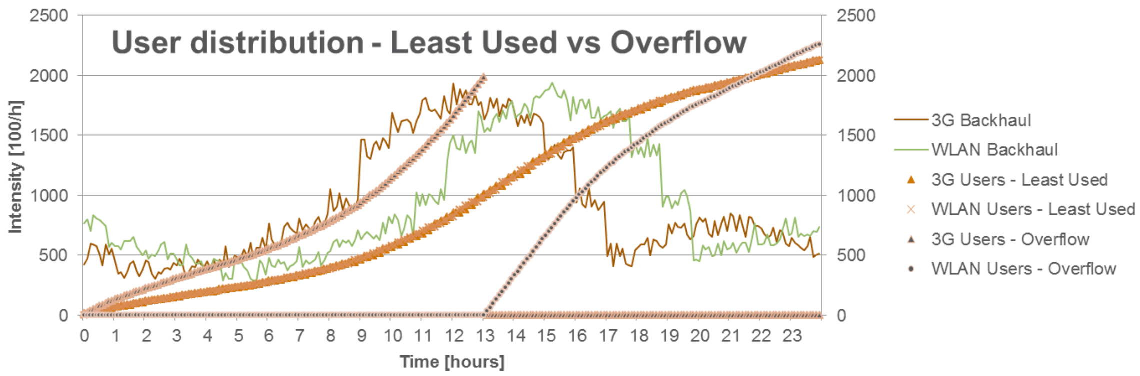


Figure 10 – Performance of the Least used and Overflow algorithms

Figure 10 Figure 10 – Performance of the Least used and Overflow algorithmsrepresents the simulated state where the decision-making algorithm considers additional input parameters such as actual bandwidth, user count of the network. In this scenario the Least Used and Overflow algorithms were applied, where the Least Used solution pushes new UE traffic to the backhaul link where the actual bandwidth is the least, and Overflow selects one backhaul and only diverts flows to a different backhaul when the resources of the current one are exhausted. The following assumptions are valid in this scenario: each user stays connected to the network in order to have accumulating traffic, all users generate a traffic data-flow and previously used bandwidth distribution may be used as incoming intensity for the number of newly connected users per hour. One of the greatest advantages of the Least Used algorithm is when each user consumes the same amount of bandwidth and the distribution is proportional to the incoming intensity; thereby the system stays stable throughout the analyzed timeframe. The Overflow algorithm, however, highly depends on the configuration of the system: when the handoff parameter of the algorithm is set to 100%, the system waits until one of the links reaches 100% load, and only after then it starts to relocate data flows. Therefore we conclude that the latter algorithm is suboptimal due to static input parameters.

Relocating individual data flows represent another challenge in the performance of the system. The performance indicator of the process is not the decision making process, rather the core handover technique used by the IPv6 mobility implementation. The implemented system merely exploits some of the benefits of MCoA and multihoming. One of the biggest disadvantage of general Mobile IPv6 and NEMO protocols is that handover preparation takes place on the active link, disrupting communication while Layer-2 reconfiguration and connection establishment are taking place. Using the Layer-3 routing feature that comes with the MCoA protocol extension, it is possible to route data-flows on established Layer-2 links, significantly reducing the time needed to bind flows to a specific egress interface. The execution of the handover process is completed when both the MR and the HA changes the binding of a data flow to the same interface. This is done with the HA initiated flow binding process implemented by the SSH RPC command described in D4.3.2 deliverable.

By measuring the time of the execution of both local and remote commands we can measure the performance of the HA initiated handover, therefore we can approximate the overall performance of the system by adding round-trip-times measured on the control channels of the architecture (where signaling among nodes take place inside the core network).

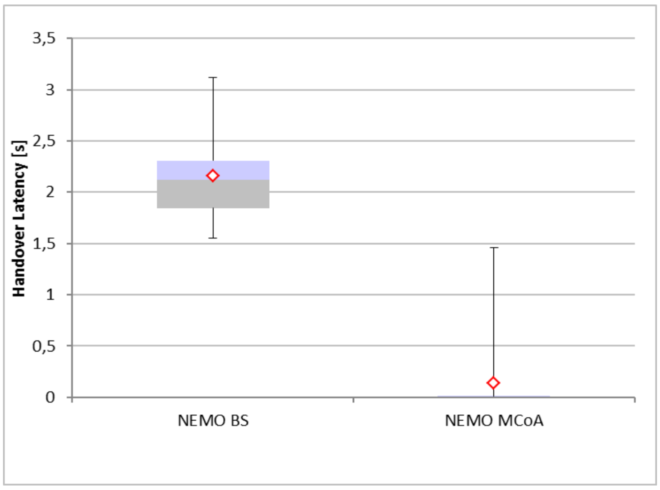


Figure 11 – Handover latency

The results are presented in the box-and-whisker diagrams of Figure 11 are displaying the collected numerical data groups in a compact way. The depicted statistical information are as follows: the lowest sample value (lower line), the lower quartile called Q1 (the lower edge of the box), the median called Q2 (the delimiter of the two distinctive colors of the box), the upper quartile called Q3 (the upper edge of the box), the largest sample value (the upper line), and the mean of the collected data (red lined rhombus). In our graphs the Q1-Q2 interval is indicated by grey color and the Q2-Q3 interval is colored with light blue. The length of boxes (i.e., the interquartile range) represents the middle fifty percent of the measured data. Diamonds show the mean (average) value of the measurements, the solid line in the background depicts the range of measured data.

Time stamped log messages and kernel events provide the measured latency in seconds passed between the handover decision and the availability of the data flow on the selected egress interface. As a result of MCoA handover the average handover time was around 130 msec which is mostly due to the latency of the last-mile medium (WLAN or 3G) at the UE. Therefore actual signaling latency in the core and decision-making latency was negligible in our scenario.

1.2.1.6. Applicability of the results

Several network elements should be extended in order to implement the proposed technology. These extensions are not seriously affecting the applicability of the scheme, as mainly the already standardized Mobile IP (DSMIP or PMIP) and PCRF functions should be slightly modified. However, a sub-optimal usage of DPI and monitoring functions could cause serious scalability issues. It means that NB-IFOM decision algorithms must not rely on high frequent measurement data of many nodes: the definition of the appropriate level of aggregation is crucial.

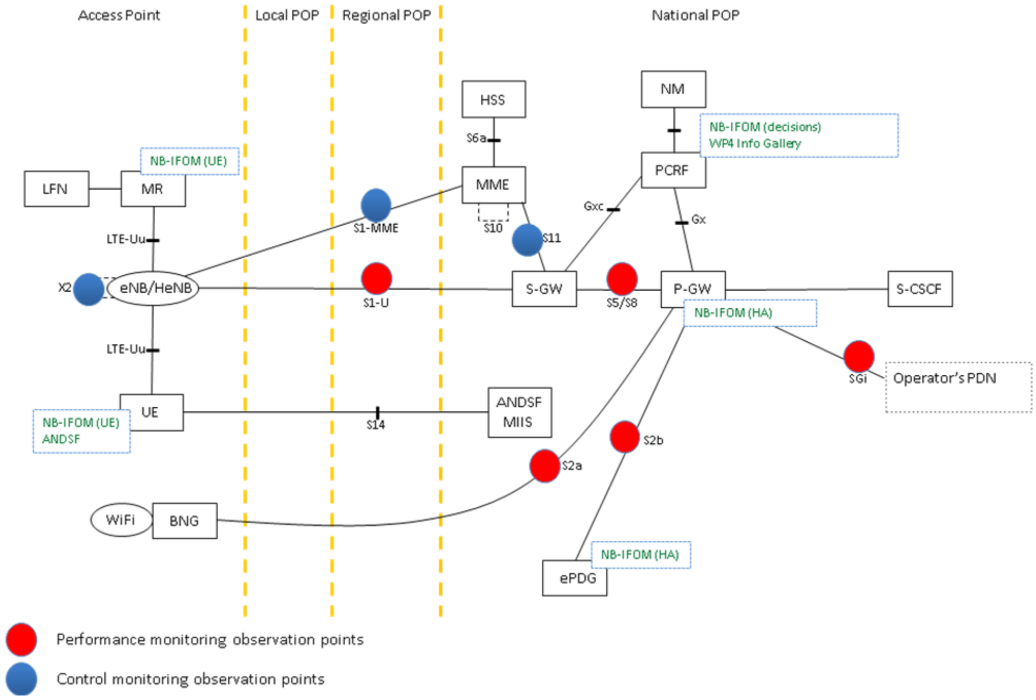


Figure 12 - Affected elements by NB-IFOM in the 3GPP Release 10 reference architecture.

1.2.1.7. Partners involved

BME-MIK

1.2.2. Performance evaluation of TEHO with ALTO upon IP Mobility.

1.2.2.1. Objectives

The objective is to validate the concept of ALTO assisted connection management in Evolved Packet Systems (EPS) (ALTO-COMEPS). The basic idea of this concept is to improve the broadband service continuity by guarantying the service quality by connecting an ALTO Client to the Connection Manager (CM) and using the ALTO protocol in 2 possible ways:

- In a proactive way: when the UE is MAPCON capable and has the possibility to distribute its flows among several P-GWs via their associated interfaces. In that case, the CM hands the set of candidate CNs to the ALTO Client that requests information from the ALTO server in order to rank them w.r.t. their path cost to the UE. The UE is then appropriately connected to the selected CNs.
- In an adaptive **way**: when IP Flow Mobility (IFOM) performed on a flow leads to a change of Serving Gateway (SGW). The IFOM has been driven by the ANDSF that considers the conditions of the network at the EPS scope and is completely agnostic to the QoE needs of the application sessions running on the UE. After the IFOM, the path between the UE and the PGW may have changed and therefore its costs. This change may affect the overall cost between the UE and the CN and therefore the CM notifies to its co-located ALTO Client that it must update its cost values to the candidate CNs.

The validation focuses on the adaptive way and uses IP Mobility rather than IFOM, to show the applicability of the concept to the Layer 3 as well.

1.2.2.2. Validation scenarios

The objective is to demonstrate that after mobility in LTE that may itself improve the QoE, an ALTO transaction, checking the cost of a connection to the candidate application Endpoints may further improve this QoE or even restore a high level of QoE that has been previously lost. IP Mobility has been used instead of specific mechanisms such as IFOM and MAPCON, to show that Adaptive COMEPS is valid for all these options and whether the architecture is distributed or not.

Figure 29 illustrates the scenario used for the functional demonstration of the advantages of ALTO-COMEPS, with an IP Mobility-capable UE having one IP address. The scenario includes the following steps:

1. IP Mobility causes a change from SGW1 to SGW2 and subsequently a change of the associated path cost from the user equipment (UE) to endpoints (EPs) in the packet data network.
2. once the HO is performed, the connection manager (CM) requests the ALTO Client to update its path costs to the current CNs, say the EPs. Suppose that the cost of the end to end path (from UE to EP) is calculated as MAX[P(EP, PGW), P(PGW, UE)], and is to be minimized. In this example, the path cost from UE to PGW evolves from 7.5 to 5. With SGW2, the least cost EP becomes EP2 with C=5, where as the cost with EP1 equals 7.5, so EP2 is preferable.

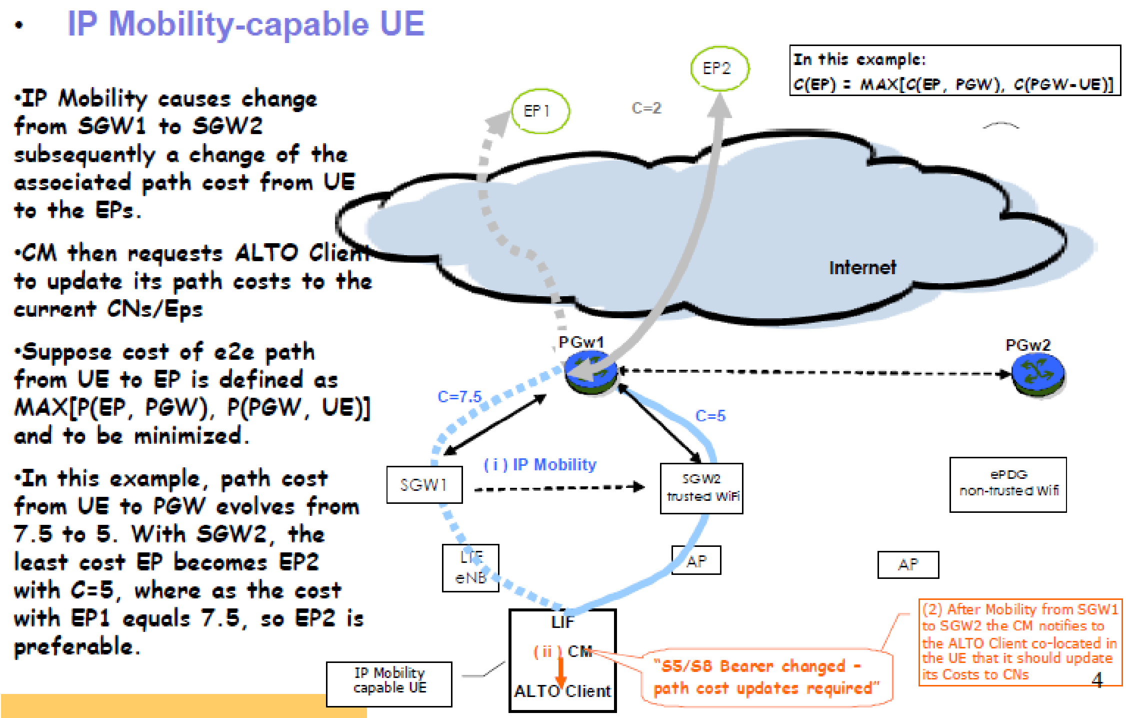


Figure 29 ALTO used upon IP mobility to update path costs between UEs and Enpoints and triggering connection to a better candidate application Endpoint (EP2)

1.2.2.3. Validation tools and scenario

The concept of ALTO-COMEPS has been evaluated through functional validation. The demonstrated prototype includes:

- IP Mobility functionalities,
- an ALTO Client developed by ALBLF and an ALTO Server developed by Bell Labs US.
- a video download application (VLC) client where the video content location is chosen with guidance provided by the ALTO Service.
    - The content location ID is reflected in the VLC playlist.
- A transport network composed of the following elements:
    - 2 involved Mobility Access Routers (MAR) playing also the SGW function in LTE
    - One router beyond the MARs playing also the role of a PGW
    - Beyond this router, 2 video Servers and an ALTO Server
    - A UE equipped with an ALTO Client, a VLC Client and accessing the network via WiFi/ePDG

The demonstration scenario includes 4 steps, illustrated in Figure 30, Figure 31, Figure 32 and Figure 33:

1. The UE is connected via MAR3 on a low bandwidth path. ALTO selects **S2** a server that provides low quality but that has a low ALTO Routing Cost w.r.t. the path quality between UE and S2. The video at **S2** is well adapted to a low quality link.
2. With IP mobility, EU moves to MAR2 and gets a better path quality.
3. Another ALTO request selects **S1,** because its ALTO routing cost is better w.r.t. the path quality from UE to **S1**.

4. The high quality video of **S1** is now well adapted to the high bandwidth path. The user QoE is improved.

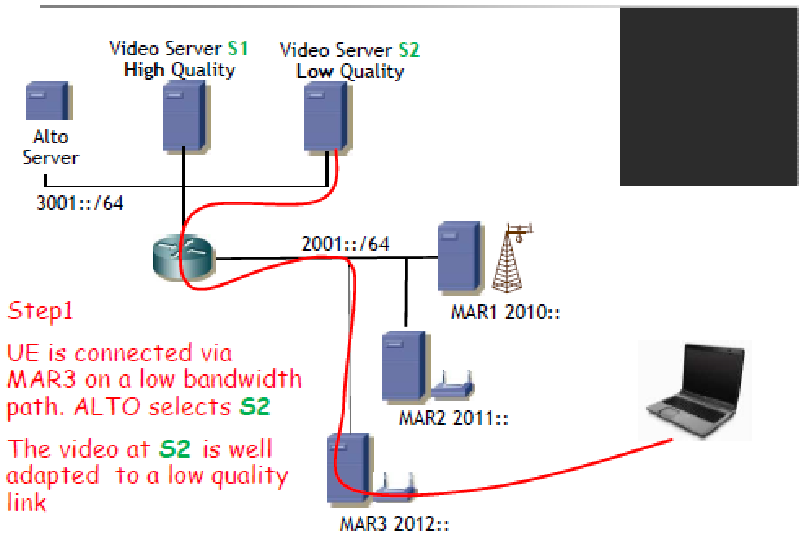


Figure 30: Demonstrated ALTO upon IP Mobility scenario – Step 1/4

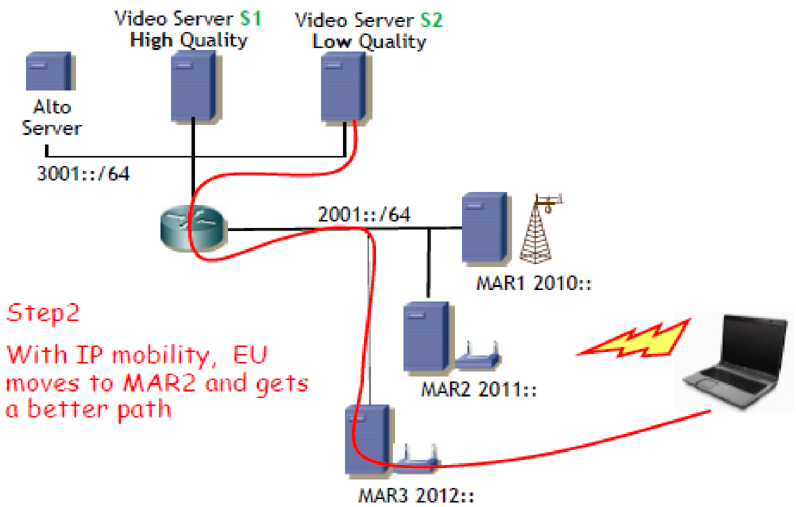


Figure 31: Demonstrated ALTO upon IP Mobility scenario – Step 2/4

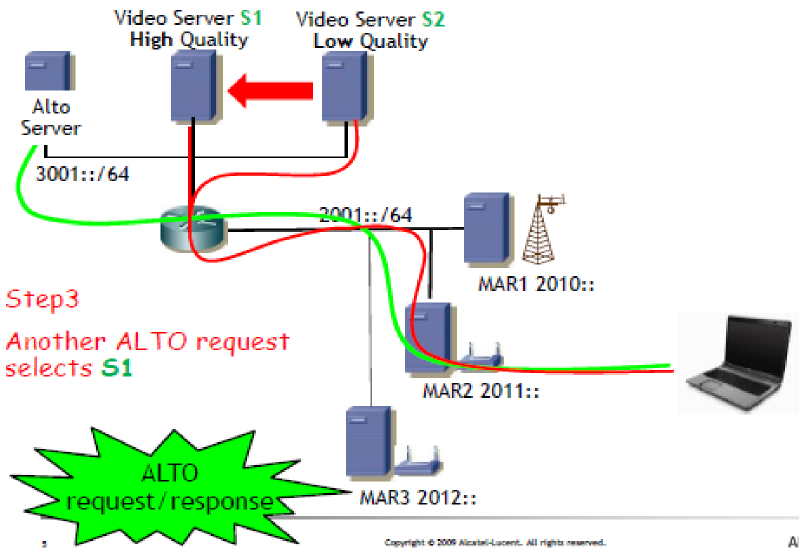


Figure 32: Demonstrated ALTO upon IP Mobility scenario – Step 3/4

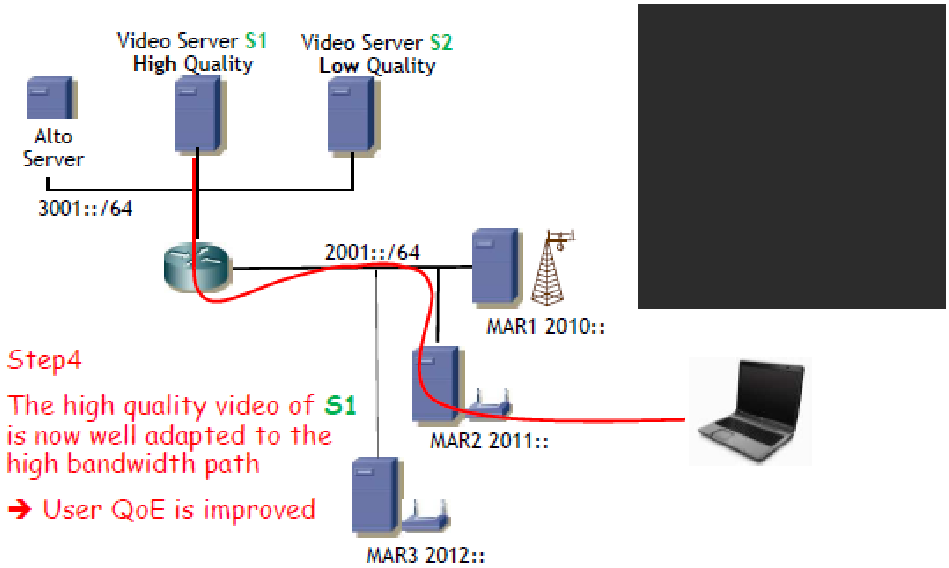


Figure 33: Demonstrated ALTO upon IP Mobility scenario – Step 4/4

1.2.2.4. Expected results

The expected result is that as EP2 provides a better QoS than EP1, the QoE improvement is visible through a better video quality at the UE.

The improvement is measured visually by looking at a video downloaded from different endpoints selected by the ALTO protocol and seeing that the visual quality is better.

1.2.2.5. Results

The results have been demonstrated at the final MEVICO Project review, illustrated in the above figures and the video of the demonstration is available upon request to the MEVICO consortium for readers outside the project scope.

1.2.2.6. Applicability of the results

The results can be used in the case of an application where a UE uses resources, be it computation cycles or content that are available from different possible Endpoints. It is assumed that:

- Cost values related to the access and usage of these Endpoints are available via requests to one or several ALTO servers.
- ALTO information is available at the granularity of the EPS. For instance it includes the costs of accessing resources from the mobile network and other access networks in the EPS.
- The CM is linked to the ALTO Client so that it can notify it of a change in the S5/S8 bearer.

1.2.2.7. Partners involved

This demonstration has involved partner ALBLF.

1.2.3. Evaluation of multi-criteria cell selection algorithms

1.2.3.1. Objectives

The objective of the Multi-criteria cell selection (MCCS) is to investigate cell selection criterion for User Equipment (UE) in the wireless networks consisting of femto, pico and macro cells. The cell selection methods SINR based, Distance Based, Biased SINR based and Traffic Load Based (proposed) are studied in LTE/EPC architecture. Local cell selection algorithms which ignore the system load information and only consider the channel quality will be replaced by global cell selection algorithms which converge to the optimum solutions in selecting the cells since they use both channel quality information and system's load information and lead to better load balancing by taking into account the number of handovers and/or delay constraints. Traffic Load Based method is an example of global cell selection algorithms.

The performance of MCCS algorithms are measured considering the user throughput and utilization of cells. The performance of the previous cell selection algorithms are illustrated for comparison through simulation and/or numerical calculations.

The covered challenge is C.Tm.13 *Multiple Cellular Coverage*. The load variance, throughput and QoS parameters are the major KPI parameters to illustrate the benefit of the MCCS algorrihtm.

1.2.3.2. Validation scenarios

In order to validate multicriteria cell selection algorithms in wireless networks, a three tier heterogeneous network where femtocells and picocells are deployed in the coverage of the macrocells is considered as illustrated in Figure 11.

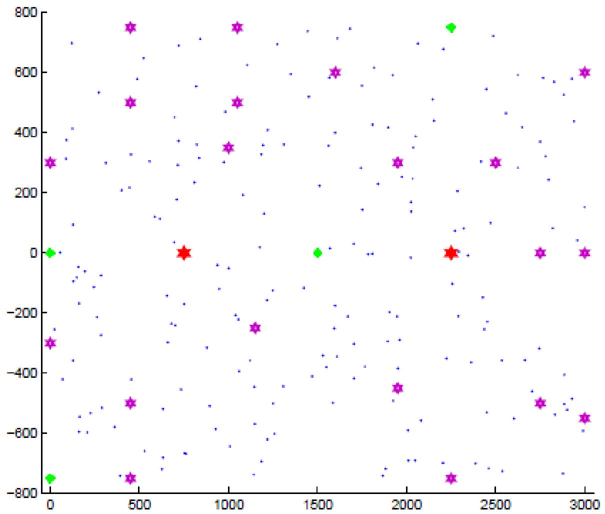


Figure 34: The locations of BSs and the mobile users. Macro BSs are illustrated in red, Pico BSs are green, Femto BSs are purple and the users are shown in blue

As a performance metrics the overall network rate and load are obtained to compare cell selection algorithms such as SINR, distance and biased SINR based and proposed methods.

For the system model all the BSs are considered as fixed located. Picocells are placed at the cell edges of the macrocell and femtocells are uniformly distributed in the network. All the femtocells are assumed as open access. There are 2 picocells and 10 femtocells per macrocell and there are 2 macrocells in the network.

The system parameters are chosen from LTE system with the channel model Extended Pedestrian A model (EPA) and these parameters are listed in Table 5.

Table 5: System Paremeters

Parameter Name Parameter	Value
Macrocell Transmit Power	46 dBm
Picocell Transmit Power	35 dBm
Femtocell Transmit Power	20 dBm
Bandwidth	20 MHz
Carrier Frequency	2.1 GHz
Noise Power	-134 dBm/Hz
Lognormal Shadowing	10 dB
Macrocell Radius	750m
Distance between Macro and Pico	750m
Pathloss (for Macro and Pico)	$Lp = 128.1 + 37.6\log 10(R)\, dB; \quad R \text{ in } km$
Pathloss (for Femto)	$Lp = 140.7 + 36.7\log 10(R)\, dB; \quad R \text{ in } km$
Number of Antennas	(1, 1)
Antennas	Omnidirectional

The traffic in the system is determined according to the user demands as voice service, data service and video service. These demands are thought as a load in the system and for each mentioned service, some load coefficients, wk, are assigned to each user. These coefficients are chosen for voice service as 1, for data service as 5, and for video service as 25.

1.2.3.3. Validation tools

We will use MATLAB and/or NS2/NS3 to validate system performances. In order to characterize mobile radio channel, we will use statistical channel models derived by using MATLAB and path loss and channel coefficients produced by Wireless Insite.

1.2.3.4. Expected results

The load variance among the cells, throughput and QoS will be obtained to illustrate the benefit of the proposed MCCS algorithm. We will also focus on low signal to interference ratio (SINR) to consider cell edge users. Load variance is expected to decrease since the proposed algorithms target the load balancing. Overall throughput is expected to increase based on better utilization of the cells. QoS is expected to incease owing to better utilization of the resources.

1.2.3.5. Results

In the following, performance results are given of all the mentioned cell selection schemes in the above sections. It is seen that the results for biased SINR cell selection (CRE method) and the proposed traffic load based cell selection method are very close to each other with the biasing factors *{1, 4, 11.9}* for macrocells, picocells and femtocells, respectively. These biasing factor values are found heuristically and it is stated that these values are dependent on the transmit powers of the BSs rather than their densities.

The performance comparisons of the cell selection algorithms are implemented for two different traffic load densities, one of them is for a normal traffic in the network and other one is for the network with a denser traffic.

For the illustration of a normal traffic density, the percentages of the load coefficient distributions to the users are *{50%, 25%, 25%}*, respectively.

In order to compare the cell selection methods the cumulative distribution function (CDF) of the average rate is shown in the Figure 35: The CDFs of overall rate in the heterogeneous network with its zoomed version. One can see that the biased SINR based and the proposed traffic load based cell selection schemes performs better than the SINR based and distance based cell selection algorithms especially at low rate. The distance based cell selection method can catch the proposed traffic load based schemes at a rate of 0.38 bits/s/Hz, SINR based cell selection method catches up at a rate of 0.21 bits/s/Hz and the CRE method catches up at a rate of 0.27 bits/s/Hz.

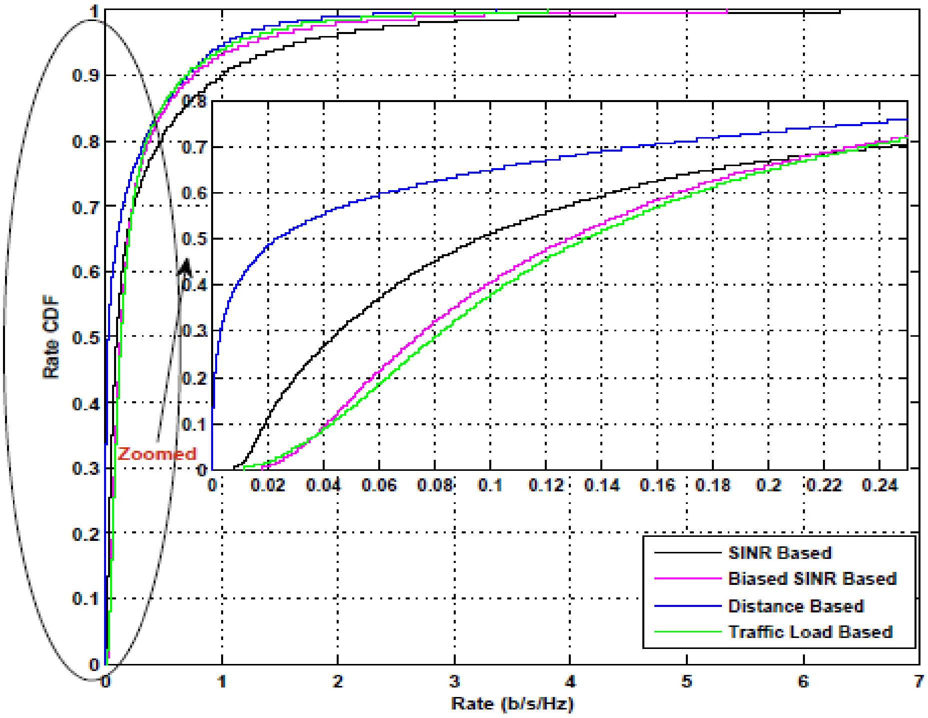


Figure 35: The CDFs of overall rate in the heterogeneous network

For the illustration of a denser traffic in the network the amount of data and video service demands are increased since their weight coefficients are higher and the amount of their distributions to the users are changed as *{20%, 40%, 40% }* for voice, data and video services respectively. When the traffic is increased in the network, the difference between the reached throughput is increased between the biased SINR cell selection scheme and the proposed traffic load based cell selection scheme. This result can be seen in Figure 36 with its zoomed version in order to see the performance difference between all methods in detail. In this figure the distance based cell selection method can catch the proposed traffic load based method at a rate of 0.39 bits/s/Hz, SINR based cell selection method catches up at a rate of 0.23 bits/s/Hz and the CRE method catches up at a rate of 0.29 bits/s/Hz. The amount of improvement for dense traffic network can be easily seen from the Figure 35 and Figure 36, as the proposed scheme is 0.07 bits/s/Hz better on average than the biased SINR scheme in the dense network while there are a 0.03 bits/s/Hz change on average between these two methods in the network with less traffic.

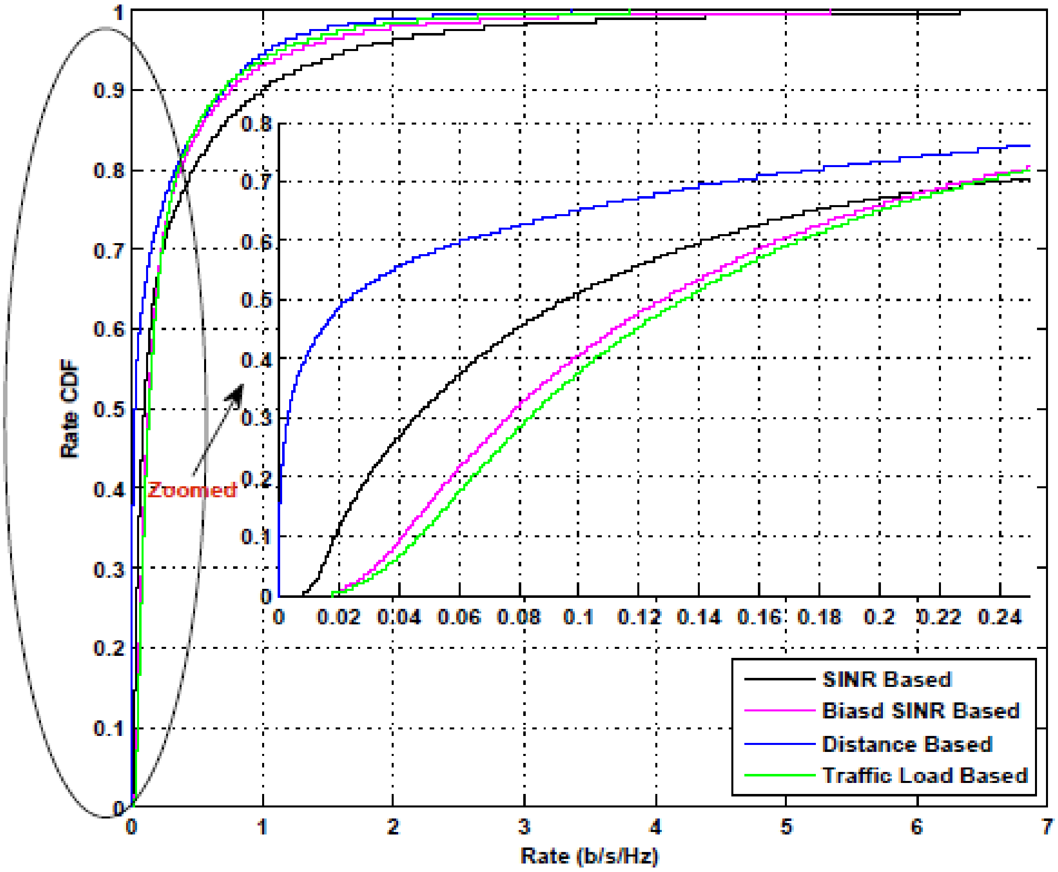


Figure 36: The CDFs of overall rate in the heterogeneous network with a denser traffic

Load distributions over the BSs for all cell selection schemes are shown in Figure 37. SINR based cell selection method distributes the load over the BSs in a very unbalanced way, such as macrocells are overloaded while small BSs are very lightly loaded. On the other hand the load is shifted to the lightly loaded small cells in the other cell selection methods. It is distributed to BSs in a more fair way in the proposed traffic load based cell selection approach than the biased SINR based cell selection method. Lastly the distance based approach gives the big portion of the load to the small BSs while macro BSs serves to a very few part of the users; however the throughput is the worst among all the cell selection schemes.

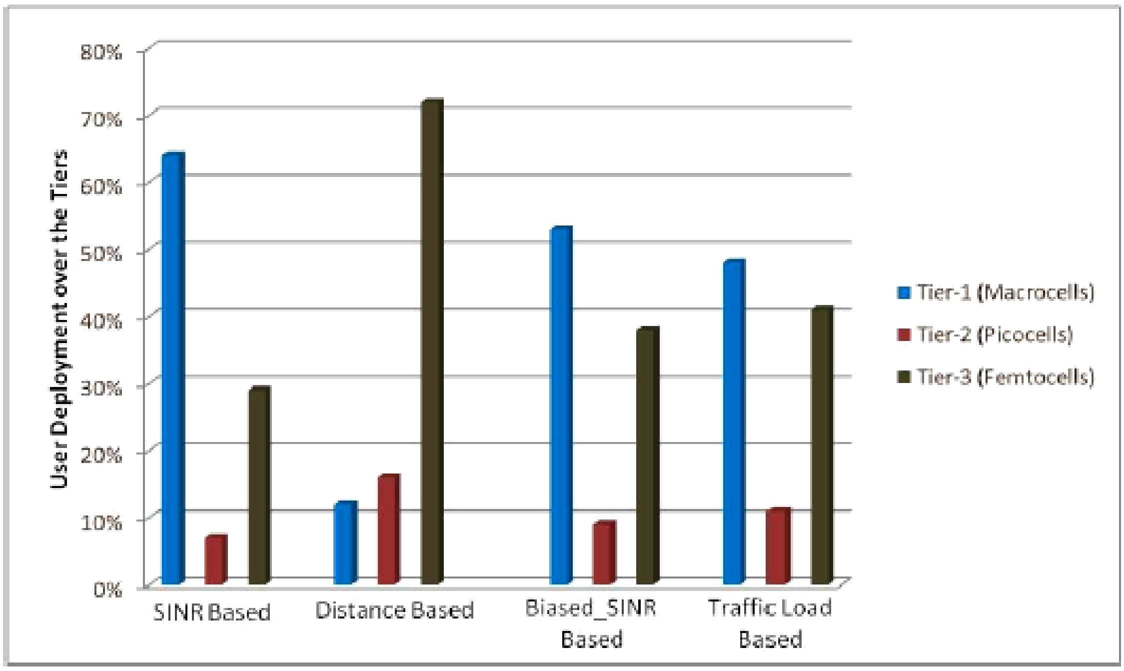


Figure 37: Distributions of average number of users per tier according to different

Numerical results show that our proposed approach gives better performance according to the user throughputs among all these schemes. It can be noticed that the performances of the CRE approach and the proposed approach are close to each other when compared to the other SINR and distance based cell selection methods. However the determination of the optimal biasing factors is not an easy work and it is achieved heuristically. In addition the proposed approach performs much better in a dense network. Another observation can be drawn from the point of BSs' view as the load is distributed more fairly with the proposed method. The congestion over the macrocell is shifted to the small cells while the overall rate is increased.

1.2.3.6. Applicability of the results

We will consider MCCS algorithms and their impacts on LTE-Advanced EPC architecture. The proposed MCCS algorithms can be implemented to eNodeB and MME.

1.2.3.7. Partners involved

Turk Telekom is the only partner in this validation.

1.2.4. Performance evaluation of selection schemes for offloading traffic to WLAN hotspots

1.2.4.1. Objectives

Offloading data traffic to IEEE 802.11 hotspots is one of the key techniques with which mobile operators aim at dealing with the increasing traffic demand of their users in cellular networks. Usually, these offloads are simply conducted once an end-device has WLAN connectivity. The objective of this work is to identify gains if the decisions for accommodating certain traffic flows in WLAN consider the characteristics of this technology, i.e., the suitability of traffic being offloaded in terms of occupation of the channel and MAC overhead as result of contention, interference, and fluctuating channels. The contribution of this work is twofold: A performance evaluation compares the novel scheme designed within the Mevico project with common received-signal-strength-based decisions as well as with simpler randomized choices for offloading particular traffic flows. Our scheme shows substantial improvements over a broad range of factors. Lastly, the results identify operational points of the WLAN cell allowing operators to dynamically choose the mix of the offloaded traffic according to their requirements.

1.2.4.2. Validation Scenarios

We investigate three scenarios with different traffic mixes: pure VoIP, VoIP with FTP downloads, and VoIP with FTP uploads. For all three scenarios, we vary the size of the quadratic area with edge lengths of 50, 75, 100, 125, and 200 meters; the AP resides at a corner of this area. Larger edge lengths lead to greater sizes of the areas such that the data rates applied for 802.11 transmissions further and further reduce. The cumulative distribution function (CDF) of the data rates for the different edge lengths is shown in Fig. 1.

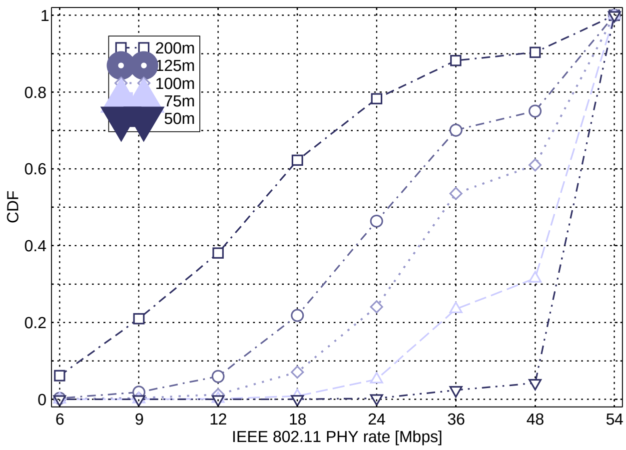


Fig. 1. Distribution of IEEE 802.11g PHY rates for different edge lengths

1.2.4.3. Validation Tools

For our performance evaluatuion, we use the network simulator ns-2. The WLAN hotspot is represented by an IEEE 802.11g Access Point (AP) that is 11e-capable by providing EDCA functionality. All WLAN devices apply 802.11g Extended Rate Physicals (ERPs) with OFDM modulation—from 6 up to 54 Mbps. The 802.11e/g parameters were chosen according to [9]. To take into account that radio signals are not only affected due to path loss but also due to multipath propagation, we use a Log-Distance Path Loss, a Ricean Fading and a SINR model. Models' details and parameters of the applied network simulator ns-2 are described in [7].

Further, in this work, we consider VoIP traffic as well as data transfers via FTP. For VoIP, we use the same model and parameterization as in [7], i.e., an exponential on/off model generating packets according to the ITU-T codec G.711 (160 Byte audio packets each 20 ms during ON periods). Similar to [7], we define the QoS limit for VoIP in terms of losses, consisting of lost and late packets. If five or more percent of the VoIP packets are lost, the quality is assumed to be lousy.

Data traffic is generated by FTP clients, which either down- or upload a large file of infinite size via TCP. TCP/IP segments have a size of 1500 Bytes and the TCP-SACK option is used. For FTP clients, we set a hard QoS limit of 128 kbps.

1.2.4.4. Results

We compare the performance of our four selected decision schemes regarding the number of VoIP flows and the volume of data traffic that can be accommodated by a WLAN cell thus unloading CT. To trade off both traffic types against each other, we additionally consider the question how much FTP traffic one can transport at the costs of a reduced number of VoIP clients. For this, we apply a two-stage process: we firstly determine the capacity of the WLAN cell in terms of VoIP users which can be simultaneously served without violation of QoS constraints. In the second stage, we consider VoIP and FTP traffic mixes in the WLAN cell.

For our performance evaluation we apply the following metrics:

1. *Number of Clients:* For an understanding about the number of supported clients from different traffic types, we consider the number of VoIP and FTP clients, which reside within the WLAN hotspot having satisfied QoS constraints.

2. *Goodput/VoIP-Reduction Ratio (GVR):* To allow a com- parison of the decision schemes in the scenarios with traffic mixes of VoIP with elastic traffic such as FTP, we

introduce the Goodput/VoIP-Reduction metric. It is a revenue/cost ratio metric, where the costs are the number of VoIP streams that were removed from the WLAN cell. Contrary, the revenue is the aggregated MAC goodput of the FTP traffic ($G_{FTP, MAC}$) that is accommodated by the WLAN cell instead. In other words, since we consider the capacity of a WLAN network in terms of VoIP calls, we define a measure that gives us the revenue for the case that we replace certain VoIPs by FTP clients in the WLAN network:

$$\mathrm{GVR} = \frac{G_{\mathrm{FTP.MAC}}}{N_{\mathrm{VoIPmax}} - N_{\mathrm{VoIPcur}}}$$

There, NVoIPmax is the capacity of the network in terms of VoIP users (from the VoIP only case), while NVoIPcur current number of served VoIP clients by WLAN.
For the GVR metric, where the denominator is not just a constant, a proper computation of confidence limits suffers from the fact that ratios of (sample) means do not obey a normal distribution anymore. As a solution, we apply Fieller's method [11] to obtain the 90-percent confidence intervals. It utilizes the property of the difference between two normally distributed random variables x and y, which obeys a normal distribution again. As result, with a ratio $\rho = y_m/x_m$ of the mean values, also the difference $y_m - x_m\rho$ is normally distributed. By normalizing this difference with the joint standard deviation of $ym - xm\rho$ to a standard normal random variable and solving the equation $t^2_{1-\alpha/2} = (y_m - x_m \rho)^2 / \sigma^2$, Fieller was able to determine bounded confidence limits for ρ, iff $x_m^2/\sigma^2 > t^2_{1-\alpha/2}$. Regarding the case-by-case analysis of Fieller's method, the reader is referred to Luxburg et al. [12].

We consider the results regarding the VoIP capacity, the operational points of the network with traffic mixes, and finally compare the different strategies by means of the GVR metric.

1. *VoIP capacity:* Fig. 2 shows the results for the first simulation setup. With increasing sizes of the area, the number of VoIP clients reduces for all decision schemes. For RSSI and both cost-function schemes, the VoIP capacity behaves similar. This is a result of the homogeneous traffic, for which only the surcharge value of the inefficiency metric may have an impact, as the overhead factor is just a constant for all traffic flows. This confirms previous results [7], where the surcharge value increases with the distance between AP and STA due to the increasing probability for low rate transmissions and higher number of retransmissions. Only the VoIP capacity with the random selection drops significantly faster, which is a result of the technology-agnostic decisions.

   The range of the overall number of simultaneous VoIP calls for RSSI and cost-function decisions corresponds with results of previous work [13] that showed 105 VoIP calls for 802.11g- only-scenarios. We gain a little higher capacity values for scenario sizes up to 100 meters, as we apply the 802.11e EDCA MAC protocol, which has smaller CW_{min} and CW_{max} values. Thus the AP stays on average less time in the (post-)backoff process enabling some more calls to be served.

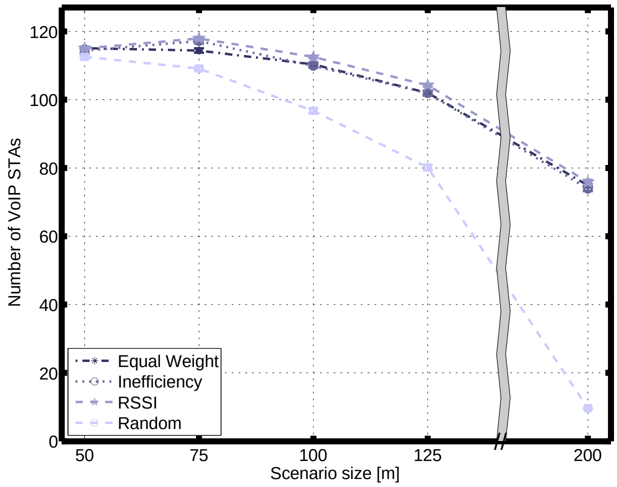
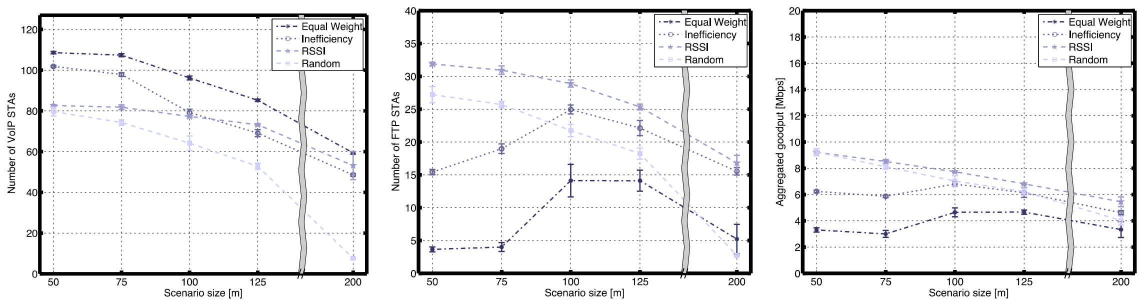


Fig. 2. VoIP capacity with the different decision schemes

2. *VoIP-FTP Traffic Mixes:* Results for VoIP plus FTP download and upload traffic are shown in Fig. 3 and 4. First, let's consider the number of VoIP terminals in Fig. 3(a) and 4(a). In both, the Equal Weight (EW) decision scheme comes up with the highest number of VoIP users, followed by the Inefficiency, RSSI and Random Decision scheme.

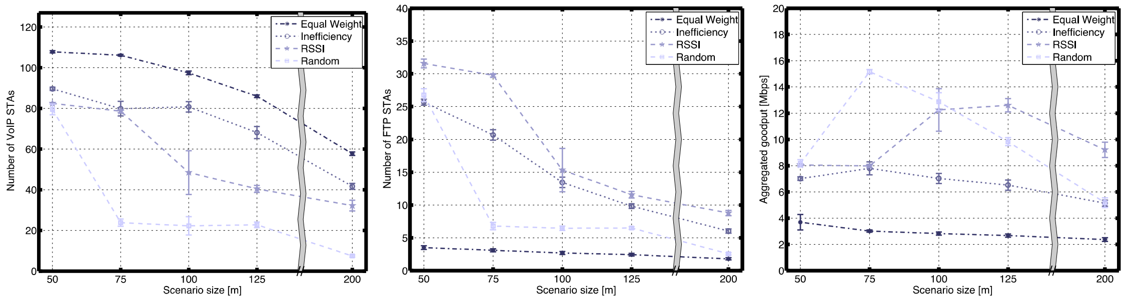
   Consequently, the number of FTP users, shown in Fig- ure 3(b) and 4(b), are the smallest for the EW scheme. With the RSSI scheme, the highest number of FTP users are gained, while Random and Inefficiency decisions range in between.

   To complete the overall picture, Figs 3(c) and 4(c) show the aggregated MAC level goodput for all accommodated FTP STAs. For FTP downloads, the aggregated MAC goodput curves follow the shape and the relations of the number of FTP STAs pretty closely. Contrary, for the FTP uploads, Random and RSSI decisions have high peaks in the goodput curves at 75 and 100 to 125 meters, which corresponds with the great reduction regarding the number of VoIP STAs at these points. Finally, we can conclude that the differences in the FTP MAC goodput lead to different operational points of the network. As one can see from Fig. 3(a) and 4(a), also the VoIP curves are affected. The difference in FTP goodput is a result of the MAC scheme which ensures fairness on a per station basis. In hotspots, where the AP serves all STAs and has to contend with the uplink traffic, this naturally results in higher queuing delays and drops for FTP data traffic in the downlink. As this lowers the FTP throughput, it leaves more capacity for VoIPs. Contrary, FTP uploads have higher throughputs, leading to a smaller number of VoIPs as their QoS constraints are violated earlier such that handovers are conducted earlier. In the IEEE 802.11 area, the phenomenon of higher delays and drops in the downlink is known as asymmetry problem and has been analyzed in the literature for VoIP [13] and TCP traffic [14].



(a) VoIP STAs (b) FTP STAs (c) Aggregated MAC goodput for FTP STAs

Fig. 3. FTP downloads together with VoIP traffic



(a) VoIP STAs (b) FTP STAs (c) Aggregated MAC goodput for FTP STAs

Fig. 4. FTP uploads together with VoIP traffic

3. *Goodput/VoIP-Reduction Ratios (GVR):* Lastly, we consider GVR for FTP up- and downloads in Fig. 5. Note that the relatively large confidence intervals are a result of the denominator, which is $\Delta N_{VoIP} = N_{VoIPmax} - N_{VoIPcur}$. Although the numbers of VoIP STAs itself have small variances and confidence intervals (cf. Fig. 2, 3(a), and 4(a)), these small variances lead to higher impact on GVR as they reside in the denominator. In other words, the high confidence intervals for EW and Inefficiency schemes stem from small ΔN_{VoIP}. In scenarios where ΔN_{VoIP} becomes larger, the confidence intervals for GVR behave similar to the other schemes.

   Now, for the FTP downloads, both Equal Weight and Inefficiency decisions outperform RSSI and Random for the first two scenario sizes. While Equal Weight is still better also at 100 meters, Inefficiency and RSSI are not significantly different anymore. For the remaining scenarios, Equal Weight, Inefficiency, and RSSI perform similar. Despite the smallest and largest scenario, Random decisions clearly lead to the worst results, which is not surprising as the number of VoIP clients was stronger reduced than with the other schemes.

   For FTP uploads, EW outperforms again for the first two scenarios, while the Inefficiency strategy is only slightly better than RSSI for the smallest scenario. Contrary, RSSI decisions are the best for large-sized scenarios, where all other schemes behave similar but worse than RSSI.

   Overall, it is surprising that the well accepted RSSI-metric for handover decisions is better only for FTP upload traffic in very large, rather unlikely scenarios. For the more realistic cases, the results show (partially great) improvements for the novel decision schemes presented in this work.

1.2.4.5. Applicability of results

Within the MEVICO project, we have designed a novel selection scheme for traffic to be offloaded from cellular networks to IEEE 802.11 hotspots. The scheme incorporates means to identify the suitability of traffic flows for WLAN cells. In our performance evaluation, we compared the gains of our approach with the classical RSSI as well as random decisions. Overall, our scheme outperforms all others for the dense settings in which 802.11g STAs transmit at medium to high PHY rates. In sparse settings, where the AP covers a large area, all decision schemes performs similar. Additionally, our results show a tendency for operational points that is of interest for mobile operators: if only few flows with elastic data traffic should be served, the Equal Weight strategy is the right choice, whereby the Inefficiency scheme may be used for more data flows but fewer VoIPs.

1.2.4.6. Partners involved

TU Berlin

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1.2.5. Performance evaluation of traffic management via enhanced Gateway Selection algorithms

1.2.5.1. Objectives

The objective is to validate the capability of enhanced Gateway selection algorithms for macroscopic traffic management. The target function of enhanced Gateway selection algorithms might include multiple (weighted) criteria (e.g. GW load, transport link load, expected subscriber mobility behavior, etc.). It has to be evaluated whether the algorithms fulfill the predefined targets in realistic traffic load situations.

1.2.5.2. Validation scenarios

The considered validation scenarios comprise different network architecture scenarios (centralized, distributed, flat, see D1.1) with different GW deployment options as well as different traffic load test traces.

1.2.5.3. Validation tools

Validation will be carried out by simulation. The GW selection optimization algorithm is implemented in MATLAB or AMPL/CPLEX.

1.2.5.4. Expected results

It is expected to show the improved performance of the enhanced GW selection algorithms (based on multicriteria optimization) compared to the simple GW selection algorithms that are used today.

1.2.5.5. Results

Since the German project part of MEVICO is still ongoing validation results could be not provided here

1.2.5.6. Applicability of the results

The enhanced GW selection algorithms might be implemented in the MME (for 3GPP release 8 or later).

1.2.5.7. Partners involved

CUT

1.3. Improved resource selection

1.3.1. Performance evaluation of mobile P4P

1.3.1.1. Objectives

This validation deals with performance evaluation of mobile P4P deployed in LTE/EPC. P4P technology can significantly reduce load on the network caused by traditional P2P traffic.

The evaluation targets the challenges C.Tm.5 *Increasing volume of P2P and multimedia traffic* and C.Tm.7 *Improved resource selection and caching*.

KPIs of the validation are *cross-domain traffic* and *end-to-end delay*.

1.3.1.2. Validation scenarios

In the validations there will be at least two Network Providers both equipped with their own iTracker which keeps the entire network cost map. There are two test scenarios as given in Figure 13:

- In the first test scenario (1), UE will communicate with iTracker and appTracker for any content request.
- In the other scenario (2), iTracker will communicate with the appTracker regularly and provide the cost map, thus the UE will only communicate with the appTracker for any content request.

The results will be compared to regular P2P transmissions.

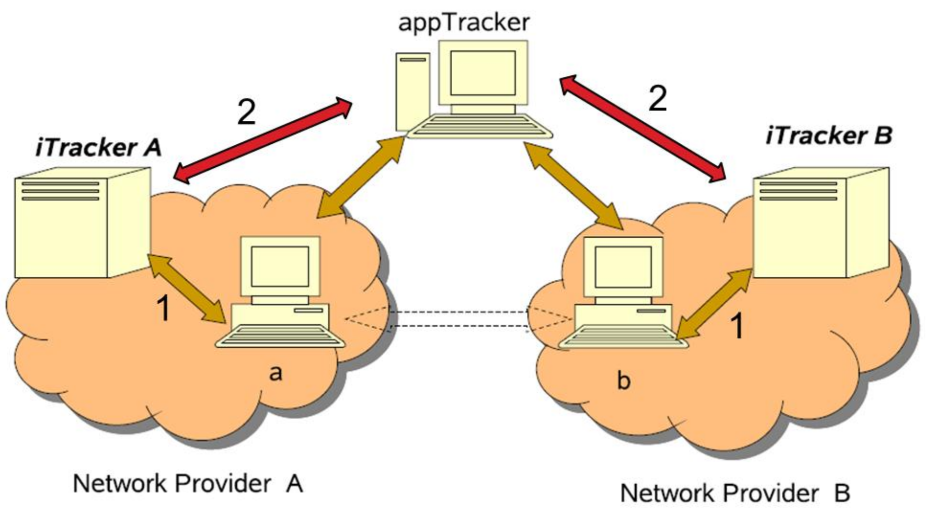

Figure 13 – Test scenarios for the evaluation of mP4P.

In this study, we simulated two different BitTorrent-Like file sharing scenarios with and without ALTO Service, those include 780 peers, one tracker and three ALTO Servers each of them are placed to each ISP. In the first scenario, when a peer wants to join the overlay, it first sends a request to the tracker. While the peer is registering to the overlay, it also sends information about which chunks of the file it has or has not as well. Then, when the tracker receives the request from the corresponding peer, it registers the peer and sends a randomly created peer list that includes 20 peers' IP addresses. The peer that receives the peer list start to establish TCP connections with the peers those are in peer list. Besides, every peer in the overlay sends updated information about which chunks it has to the tracker with a 10 seconds period of time. In the second scenario, three ALTO Servers are made active in each ISP to provide Network Map and Cost Map information. Therefore, before creating and sending a peer list, tracker obtains the Cost Map information from the corresponding ALTO Server, as shown in **Error! Reference source not found.**. Once getting the related cost values, tracker creates a ranked peer list and sends it to the requester peer for better performance. Also upload and download bandwidth rates are distributed to the peers according to Table 6. It is important to note that every ISP has a direct link to other two ISPs as shown in Figure 38. Besides, the internal physical topology of an ISP is shown in Figure 34 , as well. In that topology model we see that, for simplicity the connections between routers are made according to tree based model all in three ISPs. In other words, in overall topology, there is only one path from any peer to any peer. The size of the content is set 16 Mbytes and the simulation is run for 6 minutes.

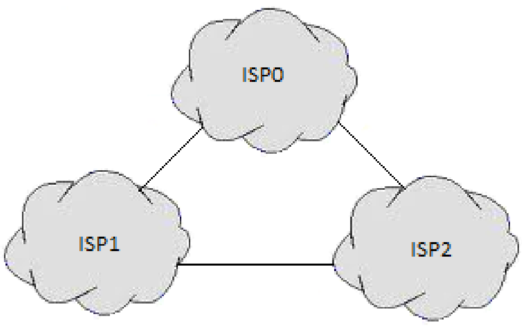

Figure 38: Proposed Network Model

Table 6 : Bandwidth Distributions of Peers

Percentage(%)	Upload Bandwidth(Mbps)	Download Bandwidth(Mbps)
56	0.5	0.25
21	3	0.4
9	1.5	0.9
3	20	2
11	20	5

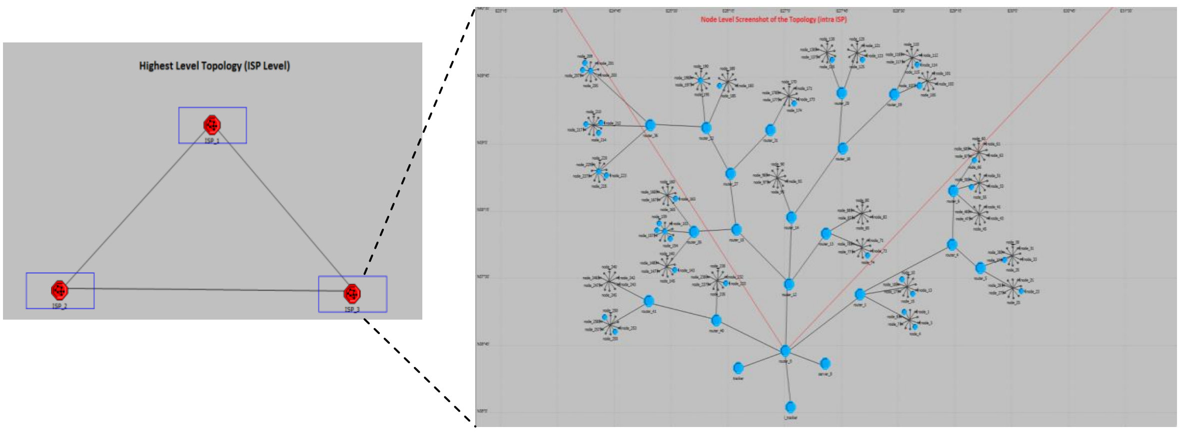

Figure 39: ISP Topology

1.3.1.3. Validation tools

The validation tool is chosen as OPNET.

1.3.1.4. Expected results

Cross domain traffic and end-to-end delay are the two main KPIs to be investigated in this validation. Cross-domain traffic is expected to decrease since the cost map provided by the iTracker will enforce the closer UEs to take part in P4P content sharing. Similarly, the end-to-end delay is expected to decrease since cross-domain traffic is expected to be reduced.

1.3.1.5. Results

During simulations in both of the scenarios, initially 30 peers in the overlay are assigned as seed (who have entire content) and they are equally distributed to the three ISPs. In this study we focus on two results that are Inter-ISP traffic rate and content downloading completion time of the peers. In Figure 40, Figure 41 and Figure 42 we see the average traffic rates owing between ISP1-ISP3, ISP1-ISP2 and ISP2-ISP3 respectively. In all of the graphics we see that, the average cross domain traffic rates are reduced approximately half of them by using ALTO service. As an interesting note, in Figure 40 we see a different behavior than Figure 41 and Figure 42. As we see there is a dramatic decrease at time about 4 min. The reason why such a decrease has occurred in ALTO free scenario is, or in other words the reason why the Inter-ISP traffic is so high in the beginning of this scenario is, random peer selection process causes inefficient peer lists in terms of cross-domain traffic and also since the bandwidth distribution is not the same of the peers. However, after a considerable time Inter-ISP traffic has reached to it stable value as like in Figure 41 and Figure 42. Since all the topology is symmetric we see similar results in all figures.

Similarly, if we look at the content downloading completion time in Table 7, we see a small improvement, as well, since we focused on the minimum delay while creating peer list for any peer. Here, since there is no congestion between any links, except (peers' link) we don't see such a big gap in terms of content downloading completion time.

The reduction in inter-ISP traffic and download times are promising solutions for mobile P4P applications as well.

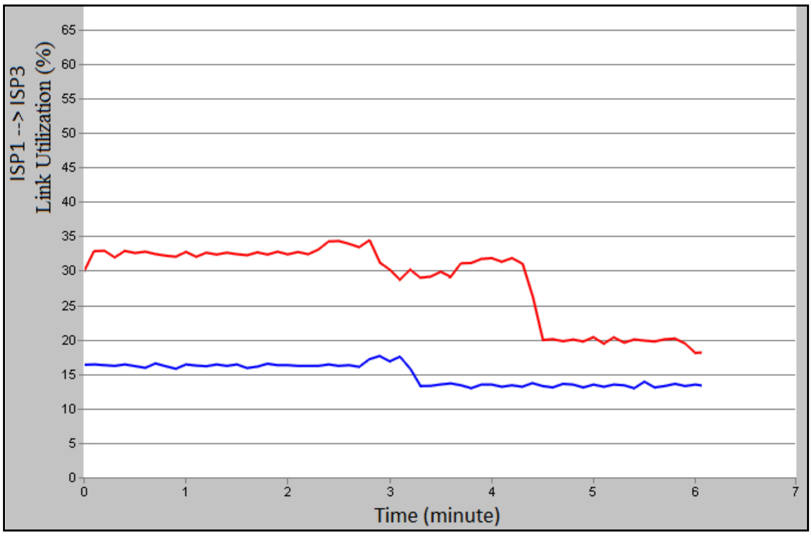

Figure 40: Inter ISP traffic: ISP1 to ISP3

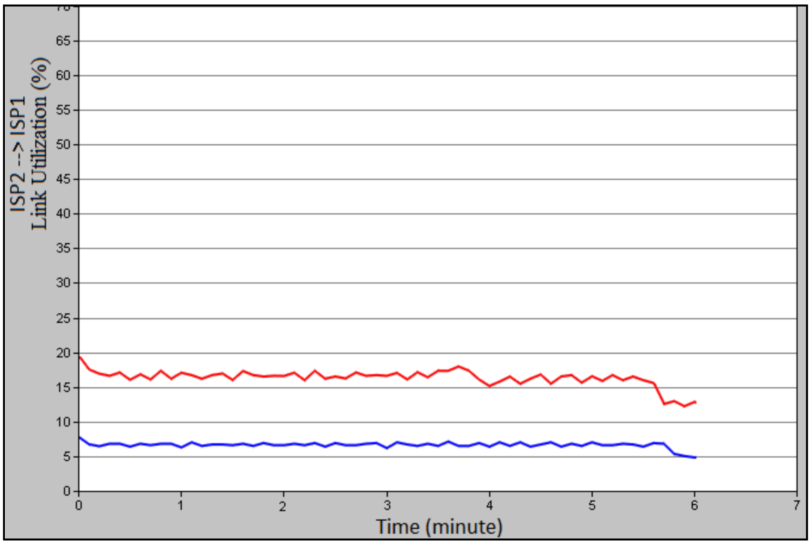

Figure 41: Inter ISP traffic: ISP2 to ISP1

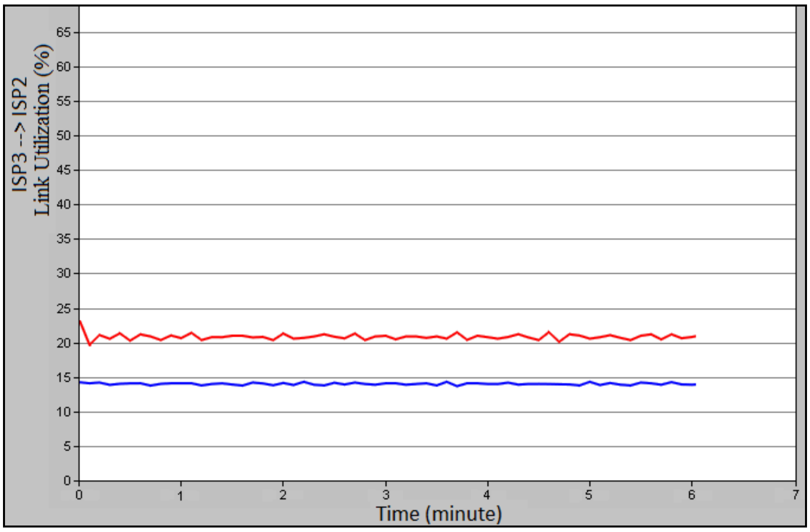

Figure 42: Inter ISP traffic: ISP3 to ISP2

Table 7: Download Completion Time

With ALTO Service (sec)	Without ALTO Service (sec)
375	383

1.3.1.6. Applicability of the results

The system is fairly easy to implement in 3GPP EPC. The iTracker can be implemented as a new entity by the P-GW of each Network Provider. No modifications are necessary in the EPC. The communication between iTracker and UE or appTracker can be done via any standard protocol.

1.3.1.7. Partners involved

Turk Telekom is the only partner in this validation.

1.4. Planning issues of GW and cache placement

The following topics are not about traffic management mechanisms but are closely related, i.e., planning the location of caches and GWs.

1.4.1. Evaluation of GW placement strategies

1.4.1.1. Objectives

The objective is to validate placement strategies for GWs. The GW placement algorithm is formulated as an optimization problem (MILP) and implemented in MATLAB. As a solver CPLEX is used.

1.4.1.2. Validation scenarios

The validation scenarios are realistic transport network scenarios, for example taken from the sndLib (http://sndlib.zib.de). A generic scenario model is provided in Figure 43. It is assumed that the IP transport network is given as well as the locations of the internet exchange points. Furthermore at all router sites an access region (e.g. a group of eNodeB) is attached. The considered traffic model is based on IR4.1.1b.

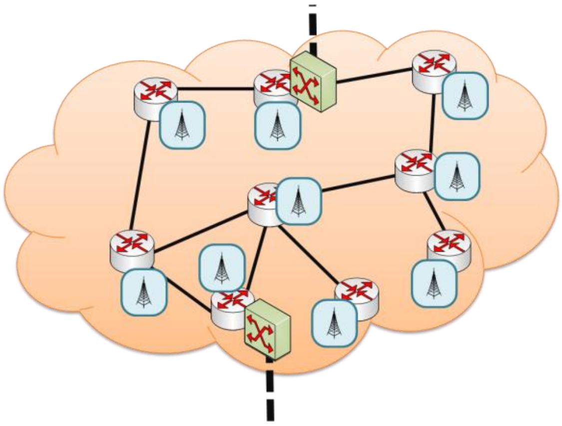

Figure 43: Network Model GW placement algorithms

1.4.1.3. Validation tools

As a validation toll MATLAB is used, to compare different optimization goals (e.g. objective functions) and network topologies, in order to evaluate the performance of the placement algorithm.

1.4.1.4. Expected results

It is expected that the GW placement strategies provide the optimal location of GW for given transport topologies in order to minimize the considered KPIs (network cost, e2e delay, etc.).

1.4.1.5. Results

So far only first results are available, since the German project part of MEVICO is still ongoing. These first results show that for Internet traffic no further path length reduction could be achieved, if more GWs are placed than IXPs are available. But since the GTP protocol introduces an additional overhead (10% pure GTP and 25% with IPsec encapsulation), so from a maximum throughput point of view it could be useful to place the GW close to the access network

1.4.1.6. Applicability of the results

GW placement algorithm could be used by network operators in order to find the best locations of GW during the initial network planning phase or it could be used to find the best location for a new GW within an existing core network.

1.4.1.7. Partners involved

CUT

2. Conclusions

This document has collected the validation plans and results of technology solutions and some traffic modeling methods investigated in the traffic management work package of the MEVICO project.

The main results are summarized below for each technology solution:

Technology	Description of the gains achieved in terms of the KPI
Throughput gain in 3GPP access and backhaul	
The proposed technology will increase the network throughput and will be measured in terms of increase of number of packets and packets size in the access and backhaul. For example, expected throughput based on modest estimation for a large European country is 300 Gbs. The requirements in 3GPP defined for classes of traffic should be considered when measuring this KPI.	
Bulk Traffic Analysis (BTA)	Analysis of network usage will enable the operator to prioritize traffic better and develop campaigns to move some traffic such as P2P to non-busy hours, hence improving usage distribution and in effect capacity of the network.
Multipath TCP Proxy (MPTCP-Pr)	SCAPs are transparent to TCP. This allows their use on multi-homed end systems. By offloading some of the TCP traffic, access network will be prevented from excessive data traffic.
Selective admission control	Selective admission control will help to improve throughput. Work not completed yet. Will be finished in 2013 within the German part of MEVICO.
Cross-layer interference detection	Mitigation of interference will help to improve throughput. Work not completed yet. Will be finished in 2013 within the German part of MEVICO.
caching	Distributed caching at a local level (e.g. BTS) will directly reduce backhaul and transport network load. Project is scheduled till end of 2013. Results were included into the German part of MEVICO.
Application Layer Traffic Optimization (ALTO)	ALTO lead to choose the connection nodes involved in applications so as to minimize metrics including hop count. Throughput gains can be obtained by minimizing the needless use of links. The proposed protocol extension called ALTO Cost Schedule will help applications to schedule their connection at times favorable wrt. network resources usage. Thus application will avoid to needlessly load the access network.
Deep Packet Inspection (DPI)	The DPI by itself will not increase the overall throughput. Rather it is a mechanism that can be used to increase the throughput of selected application classes by prioritizing their traffic (if the legislation allows such prioritization) or reducing the throughput of undesired non vital services.
Multi-Criteria Cell Selection (MCCS)	In the text generation LTE cellular systems, the users have access to more than one cell region. A single user will have access to femtocells, picocells and macrocells at the same time. With MCCS techniques the serving cell can be chosen optimally via a system level selection procedure. The optimal selection of serving cell enables optimal utilization of the system resources. The simulation results have validated that the resources can be utilized better via MCCS techniques and the throughput increases.
QoE estimation & traffic manipulation	The QoE estimation will not increase the overall throughput, but it will allow increasing the throughput of selected services. The QoE of selected service flows is estimated and, if there is a low QoE detected, the priority of this flow will be increased which leads to a higher throughput for the specific service flow. The gain in throughput is achieved by lowering the throughput of a non-vital service.
Backhaul and RAN influence on E-E delay	
Requirements for delays in the backhaul and last mile come from the requirements from the end user applications. Typical delay budget for backhaul part in LTE case is 10 ms, reducing to 1ms for LTE-A and subms for beyond 4G mobile technologies. BS synchronization transfer and new mobile system features like 3GPP rel 11 CoMP may pose stricter delay and delay variation requirements for backhaul connections.	
caching	Distributed caching at a local level (e.g. BTS) will partly eliminate backhaul delay. Project is scheduled till end of 2013. Results were included into the German part of MEVICO.
Mobile P4P (mP4P)	Utilizing P4P in a communication network yields optimal P2P operation. ISPs can use the cost map of the network to control the P2P traffic. Our simulations focused on implementing an ALTO server for enabling P4P video transmission. The results have shown that the end-to-end delay is reduced significantly since inter-ISP traffic is reduced.
Selective admission control	Selective admission control will help to reduce the e2e delay. Work not completed yet. Will be finished in 2013 within the German part of MEVICO.
Cross-layer interference detection	Interference mitigation will reduce the e2e delay. Work not completed yet. Will be finished in 2013 within the German part of MEVICO.

<table>
<tr><td>ALTO</td><td>As said for mP4P ALTO in general aims at reducing inter-domain traffic and by accordingly unloading the network enable better throughput and thus delay for other traffic. The ALTO cost Schedule information allows picking the best time to connect and avoid needless overload as well.</td></tr>
<tr><td>MCCS</td><td>In a multi-cell access environment, with MCCS techniques the serving cell can be chosen optimally via a system level selection procedure. Using MCCS techniques, the optimal selection of cells yields performance improvement on the utilization of system resources, i.e., macro cells do not become overloaded. In accordance with this, simulation results have shown that maximum delay is reduced by 15% when a distance based approach is used instead of SINR based approach.</td></tr>
<tr><td colspan="2">Efficient load distribution in the backhaul and in the core</td></tr>
<tr><td colspan="2">This KPI should show that the application of load balancing mechanism contributes to the non-congested states of the network in case of high traffic demands. The traffic load, the inter-arrival time and transmission delay should be measured either on end points or in the routers/switches if possible. The KPI could also use global packet loss ratio to measure the congestion of the end to end network</td></tr>
<tr><td>Network-based IP Flow mobility (NB-IFOM)</td><td>Non-congested states indicate that none of the trigger conditions are met to fire a policy change event. In such scenarios the default distribution-ratio of policies depends on the load-balancing algorithm. The following table summarizes the implemented load-balancing techniques with respect to the overall stability of the network. The objective function of the methodology is to prevent the use of policies (Note: Policies will only be applied when the traffic parameters and QoS inside the backhaul and the core network are suboptimal), by keeping the load distribution stable. This KPI does not consider rapid changes in existing data flows.
<table>
<tr><th>Distribution alg.</th><th>Input argument</th><th>Remarks</th></tr>
<tr><td>Round-robin</td><td>-</td><td>Flows are distributed evenly among available uplinks.
Good for Best Effort QoS. Uneven flow bandwidth distribution may result in policy reallocation.</td></tr>
<tr><td>Least Used</td><td>current bandwidth</td><td>Better than RR and Overflow</td></tr>
<tr><td>Lowest latency</td><td>current latency</td><td>Better than RR and Overflow</td></tr>
<tr><td>Overflow</td><td>current bandwidth</td><td>The algorithm waits until one of the links become full, allowing network policy events to be triggered before utilizing all available media, making this method the worst case.</td></tr>
</table></td></tr>
<tr><td>Gateway Selection</td><td>The enhanced GW selection algorithm uses multiple input parameters (e.g. transport network load, GW load), with this information the GW is selected in a way that the overall traffic is distributed over the network. If nevertheless a link is congested the selection algorithm avoids selection a GW which utilizes the congested link.</td></tr>
<tr><td>ALTO</td><td>ALTO provides guidance to select application Endpoints wrt. metrics including hopcount and soon available path bandwidth. One of the effects is thus load balancing.</td></tr>
<tr><td colspan="2">Offload gain due to the usage of multi-access capabilities</td></tr>
<tr><td colspan="2">The KPI can measure the user and operator point of view.
• User point of view. The KPI should measure the traffic that goes on each interface of the UE in case of simultaneous use of radio interfaces or it should measure the end to end delay of transmission.
• Operator point of view. The KPI should measure the load on different elements of the network. It could measure the proportion of load in different access i.e. Wi-Fi access versus LTE access.</td></tr>
<tr><td>NB-IFOM</td><td>User point of view: By utilizing the flexibility granted by flow policies, it is possible to create policies to maximize the user experience. For each type of data flow a preferred network path could be selected, however, at the current stage of the system model, there is no feedback (i.e., QoS or QoE measurement results) from the user side.

Operator point of view: Flow mobility allows simultaneous data flows of a single UE to be offloaded to various available media, depending on the decision algorithm and policies set by the network administrator. To demonstrate the advantages of network based flow mobility, a rule set, which tries to avoid congested states on all available links, shows that it can be guaranteed that load proportion will never exceed 90% on a certain network path. Constant monitoring of not only overall network, but flow statistics, offloading only the few affected flows restores the stability of the network, provided they are not conflicting with other policy requirements, such as user experience requirements introduced above. This fine granularity (i.e., flow based operation) makes the scheme really flexible and provides a wide set of optimization possibilities.</td></tr>
</table>

MPTCP-Pr	MPTCP-Pr provides additional flexibility for offloading also for the single-homed scenarios (no multihoming feature is necessary).
Selection of handover candidates in IEEE 802.11 hotspots	Offloading data traffic to IEEE 802.11 hotspots is one of the key techniques with which mobile operators aim at dealing with the increasing traffic demand of their users in cellular networks. Usually, these offloads are simply conducted once an end-device has WLAN connectivity. Proposed algorithm identifies gains if the decisions for accommodating certain traffic flows in WLAN consider the characteristics of this technology, i.e., the suitability of traffic being offloaded in terms of occupation of the channel and MAC overhead as result of contention, interference, and fluctuating channels.
ALTO	ALTO does not influence offload decisions when associated with IFOM. It does when associated to MAPCON as it chooses the endpoint wrt the path and its resources, including the resources in the RAN & backhaul
Capacity aggregation and E2E QoE sustainment	
This KPI should measure the throughput gain due to multipath communication, including goodput. The KPI will also measure QoS packet delay jitter packet loss plus any additional QoE measurements.	
NB-IFOM	One of the greatest advantage of flow mobility is the ability to aggregate traffic of the UE, by distributing flows on multiple interfaces. Preliminary results suggest that distribution of packets belonging to one data flow may cause degradation in QoE sustainment, due to how transport and application protocols react when multiple links with different qualities are used in the transmission. To ensure end-to-end QoE sustainment it is possible to create network policies which always move each flow to satisfy its requirement while not violating network point of views either.
Gateway Selection	The enhanced GW selection algorithm is aware of multiple access technologies, used by the UE. It assigns a GW which provides the best path towards both access technologies and therefore optimizes the delay and jitter for packets send over different access technologies.
MPTCP-Pr	Aggregation of multiple path capacities increase TCP goodput.
ALTO	ALTO has no influence on multi-path communication
DPI	DPI is a core technology to classify traffic based on the application type and to measure the QoS metrics at the application level. It can also provide input for application QoS/QoE measurement mechanisms.
QoE estimation & traffic manipulation	At a certain set of observation points within the ISP network, the QoE of specific services (e.g., YouTube) is estimated.
Service interruption delay due to handover	
This KPI should measure the packet transmission additional delay due to flow mobility/handover. The KPI may measure the service interruption delay and jitter due to HO, as well. Packet delay budgets for guaranteed bitrate real-time services can be considered as hard constraints for induced E-E service interruption delay. Packet delay budget is the delay defined between the UE and the PCEF. TS 23.203 defines the following packet delay budgets for different guaranteed bit-rate (GBR) service types. <50 ms: real-time gaming; <100 ms: conversational voice; <150 ms: Live (interactive) video streaming; <300 ms: Buffered video streaming; For higher than 300 ms interruption seamless service continuity may not be achieved for real-time GBR services. Hence for these application types, and in case of requiring seamless handover, a given technology is not acceptable.	
NB-IFOM	By utilizing the multihoming advantage of MIPv6 Multiple Care-Of Address (MCoA) support, the overall latency of the Layer-2 handover can be reduced to zero. The process is called MCoA handover, and it depends on the presence of at least two connected and configured links. When both the current and future network is connected, the handover skips Layer-2 configuration delays, and the only requirement of a successful handover is a modified flow binding between the Home Agent and the UE. Latency may be measured from the beginning of the flow binding signaling, until the new binding is registered and acknowledged by the other endpoint, ensuring the symmetric routing of data packets of each flow. A good measure of the KPI is the calculated latency, which is the round-trip-time of the protocol signaling. The following table compares DSMIPv6 and NB-IFOM (i.e., MCoA) handover latencies in light of interruption timeouts described above. The percentage shows the proportion of successful handovers within the required timeframe. **Service type** | **DSMIPv6** | **MCoA**

<50ms	0%	30%
<100ms	0%	50%
<150ms	0%	72%
<300ms	0%	100%

Handover related signalling load on the network

The handover procedures together with handover initialization, preparation, completion phases should be analyzed. Show that compared to state-of-the-art handover the new technologies provide reduced signaling load on different parts of the network.
This KPI may measure transmitted data overhead for HO process or the number of HO messages and their size. Also describe which handover types are supported by the technology, e.g. intra-3GPP, inter-technology, single/multi-access devices etc

NB-IFOM

As the table below summarizes, protocol signaling may be as broad at NB-IFOM as it is with IFOM, however, there are fundamental differences in the way these two protocols operate. Regular IFOM blocks UE until policies are distributed and allocated from a central location and the procedure is triggered by a UE event (i.e., registering to the network). Flow enforcement is up to the UE, and therefore, it cannot be trusted. NB-IFOM uses parallel signaling between UE-HA and HA-PS (Policy Server) node pairs, allowing rapid changes in network configuration via the non-blocking binding registration. Policies are only sent over the network, when a change is necessary, hereby reducing the signaling overhead by more than 40% in most cases. Enforcement of a flow policy is moved to the Home Agent as intermediary between the PS and the UE.

No.	**IFOM**	**NB-IFOM**
1.	UE registers with ANDSF server.	UE sends Binding Update to the HA.
2.	ANDSF notifies UE about updated flow policies.	HA sends Binding Acknowdgement to the UE and sets up default policy. **UE flow is blocked until this step completes.**
3.	UE sends a DSMIPv6 Binding Update with the offered flow policies to the HA. **UE BU cannot be trusted.**	HA forwards BU to Policy Server.
4.	The PDN GW sends an IP-CAN session modification request to the PCRF.	Policy Server may send a policy command to HA regarding any policy change.
5.	The PCRF sends an acknowledgement to the PDN GW.	HA sends Flow Binding Indication to UE.
6.	The HA sends a Binding Acknowledgment. **UE flow is blocked until this step completes.**	UE sends a Flow Binding Acknowledgement to the HA. **Policy is enforced from network side.**

E-E delay between UE and content

This KPI may measure RTT (on UE or server). The 3GPP Requirements by application type in terms of E2E delay budget should be considered. This KPI could also measure the path lengths in terms of number of L2/L3 hops.

NB-IFOM	As NB-IFOM is based on the DSMIPv6 standard and the MIPv6 protocol family, without the completion of the standard's Routing Optimization (RO)/Enhanced Routing Optimization (ERO) procedures all the user plane packets will traverse the Home Agent. This could result in suboptimal communication paths between the UE and its Correspondent Node (i.e., the content) and also could raise the E-E delay between the communicating entities. However, executing RO/ERO for every running flow optimal paths of packets between the communicating nodes can be set up. Also the flow-based offloading takes effect on the E-E delay by redirecting certain communication session from e.g., a 3G to a Wi-Fi connection.
Gateway Selection	In order to reduce the E-E delay the GW selection choses the GW, which is on average on the shortest path between the User and the content. Thus an average delay reduction could be achieved.
caching	Distributed caching at a local level (e.g. BTS) will efficiently and directly reduce the E2E delay between UE and content. Project is scheduled till end of 2013. Results were included into the German part of MEVICO.
mP4P	Utilizing P4P in a communication network yields optimal P2P operation. ISPs can use the cost map of the network to control the P2P traffic. Our simulations focused on implementing an ALTO server for enabling P4P video transmission. The results have shown that the end-to-end delay is reduced significantly since inter-ISP traffic is reduced.
ALTO	Minimizing the e2e delay between UE and content is a core objective of ALTO. The can be achieved by using the ALTO Cost Map and Endpoint Cost Services to select the best possible Content Location wrt transport network resources.