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Intelligent Selection of Access Nodes for multi-interface Mobile Terminals

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***Abstract*— This paper tackles the problems of intelligent interface selection for mobile terminals connected to Fixed & Mobile Converged (FMC) network and equipped with multiple network interfaces of different access technologies. Selection mechanisms that can be found in today's networks are analysed and their drawbacks are estimated. A new selection policy retaining the VoIP quality above a pre-defined threshold is elaborated. The new policy eliminates the drawbacks of today's algorithms, its easy realisation and integration in networks is possible thanks to the Integrated Mobility and Resource Management Mechanism. The performance of different selection policies is compared using ns2.**

***Key Words*—QoS, network performance, interface selection, VoIP.**

I. INTRODUCTION

FMC access networks become a standard today. IP Multimedia Subsystem (IMS) and Resource & Admission Control Subsystem (RACS) are being considered on the signalling plane for session and resource management respectively.

Currently available mobile terminals with multiple network interfaces of different wireless and/or wired access technologies enable usage of the most suited interface for each session maintained by the user. To be able to select the most suited access interface of the mobile terminal, the basic RACS architecture has been extended with functional elements defined in [1],[2].

A short overview of the introduced Integrated Mobility and Resource Management Mechanism (IMRMM) is given in Fig. 1. The central entity of the IMRMM is the Mobility Management Engine (MME) that is connected with IMS, Mobility Management Database (MMDB) and all access nodes (AN) in the network whereto mobile nodes (MN) are connected. IMS provides to the MME information about the types and requirements of sessions going to be established by MNs. The MMDB stores the information about all MNs like the number of available network interfaces and ANs whereto different interfaces of MNs are connected.

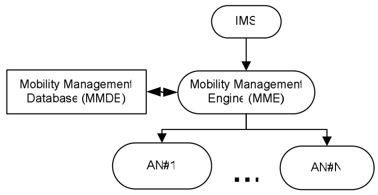


Fig. 1: Integrated Mobility and Resource Management Mechanism realised as extension to IMS in RACS

Upon reception of the new session request, the MME may define which ANs can be used to transfer data of the new session based on the information stored in the MMDB. After that it can exchange any parameters with involved ANs to detect e.g. the modulation technique of the MN. After selection of the MN's interface and appropriate AN, that will be used for the new sessions, the Edge Router (ER) and the MN gets informed about that in order to transmit packets of the new session using the selected AN. Thereafter the MME can install observers on the AN to observe e.g. the modulation of the MN in order to be informed in the case some parameters of the MN or of the AN change. In that case the MME is able to re-initiate selection algorithms to re-define the AN for the established session. If a handover for some sessions must be performed, the ER and involved MNs get informed about that in order to transfer packets of a particular session using the re-selected AN. However, logical algorithms how ANs for particular sessions of MNs will be selected and which impact on the quality of different traffic types the performed selection takes, have not been defined and analysed yet. That issue is tackled in the present paper.

The rest of this paper is organised as follows: Section II gives an overview about the problem of intelligent interface selection, Section III introduces evaluation methodology we have used, Section IV models the today's selections of AN. Based on the evaluation results of today's policies a new policy is elaborated in Section V, Section VI compares the policies and Section VII finalises the paper with a short conclusion.

II. State-of-the-Art and Problem Statement

Simultaneous usage of multiple network interfaces requires intelligent interface selection for different application streams so that optimal load distribution among available ANs is possible. That allows avoiding overload of particular ANs while providing the best possible QoS for user sessions. The general view on the problem is shown in Fig. 2.

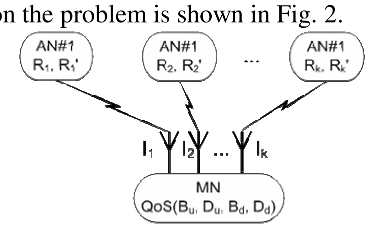


Fig. 2: General problem of optimal interface selection

The MN has k network interfaces $\{I_1,I_2,..I_k\}$, which are connected to the k ANs, which transport resources are $\{R_1,R_2,..R_k\}$, whereby some of them are occupied by other MNs so that transport resources available for the requesting MN are $\{R_1',R_2',..R_k'\}$. The MN is going to establish a new session with QoS requirements $QoS\{B_u, D_u, B_d, D_d\}$, where B is the required bitrate and D is the maximal delay, indexes $[u,d]$ are for uplink and downlink respectively. To optimally distribute MNs' flows among ANs, functions f_i must be defined for k ANs. The f_i defines how many transport resources of the AN are required to fulfil QoS requirements of the MN ($r_i = f_i(QoS)$). One of the solutions for intelligent AN selection can be to maximize the remaining transport resources (L) of an AN after assignment of the required resources for the session:

$$max\{L_i = [R_i' - f_i(QoS)]/R_i > 0\},\ i = 1...k,$$

i.e. the AN with maximal positive L will be selected for the session.

The problems of that general description are that the requirements of TCP based application cannot be presented in such form as well as the definition of f_i is difficult for many access technologies, especially with uncoordinated random access to the media like IEEE 802.11. Furthermore, the behaviour of TCP based application like File Transfer Protocol (FTP) cannot be predicted and mathematically described in a simple way in contrast to e.g. Voice over IP (VoIP) application that is typically Constant Bit Rate (CBR) traffic. That is because FTP application consists of two flows, a flow where data will be transferred with TCP data packets from the source to the destination and a flow in the reverse direction where TCP Acknowledgement packets will be transmitted. The number of packets, inter-packet intervals and therefore the required bitrate of these flows depend highly on each other that make investigation of TCP based applications in a complex multi parameters system almost impossible.

In the following, we use network simulation to detect effects of different selection logics and their influences on different traffic types. We start to investigate selection mechanisms available in today's networks and improve them to the selection logic that will provide the desired QoS in Next Generation Networks.

III. Evaluation Methodology

The performance of different selection algorithms is compared using network simulator ns2 ([3]). An own developed module for IEEE 802.16 PMP access technology has been developed based on the specification [4] while the available module for IEEE 802.11 is used.

A. Network scenario

A typical FMC access network consists of a big number of ANs deploying different access technologies and a huge number of MNs connecting to them. Simulation of such large networks is very time-consuming. To investigate how different selection policies perform in a heterogeneous network environment two ANs of different access technologies are simulated. Two ANs are sufficient for simulations since potential benefits of the proposed IMRMM architecture and used selection policies are investigated. The reference network topology used in all performed simulations is shown in Fig. 2.

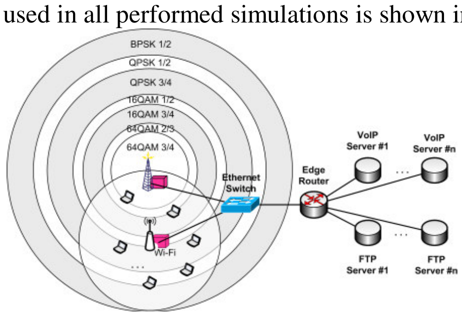


Fig. 3: Reference network architecture

A TDMA IEEE 802.16 based AN in PMP mode and a CSMA/CA IEEE 802.11 based AN are used to enable access to the network for MNs. A smaller coverage area of the Wi-Fi AN is covered by the larger coverage area of the WiMAX AN. MNs are located in the coverage area of the Wi-Fi AN so that they are covered by both ANs. Every MN has both WiMAX and Wi-Fi access interfaces so that it can communicate using a single interface or both. MNs are static, i.e. they do not move during the whole simulation time. Both ANs are connected to an Ethernet Switch that is connected to the Edge Router. The Edge Router is connected to $N_{server}(=N_{MN})$ VoIP and FTP servers that can be used either as sources or destinations for VoIP and FTP streams respectively. All wired links in the network have capacity of 10Gbit/s and propagation delay of 0.5ms. The broadband wired links and the same number of FTP and VoIP servers are selected to reduce possible queuing delays in the wired links. Therewith the bottleneck of the whole network is the capacity of ANs which are used to connect MNs to the wired network. The same basic conditions are assumed in real FMC access networks. The queue lengths of both ANs are limited to 50000 packets. Therewith packet drops due to the queue overfilling are eliminated while the delay of packets waiting for the transmission over air can be very high.

B. *Traffic model*

The main goal of the used traffic model in our simulations is to investigate impact of increasing FTP traffic in the network onto the quality of established VoIP sessions and how the IMRMM performs to retain the VoIP quality at the satisfied level. Two VoIP calls each consisting of two VoIP streams (uplink (UL) and downlink (DL)) are assigned to every MN. First VoIP calls are assigned at the simulation time of 5s, second VoIP calls - at the time of 15s. At the time of 40s an FTP session either in DL or in UL direction will be assigned to a MN every 5s. Session direction is selected randomly *dir=uniform[DL;UL]*. The index of the *i*'th MN idx_i is selected randomly among the complete set of MNs in the network N_{MN} : $idx_i=uniform[1...N_{MN}]$, $idx_i \neq idx_1...idx_{i-1}$, $i \in 1...N_{MN}$. VoIP and FTP sessions are infinite, i.e. after their establishment they run till the simulation finishes. Therewith the load in the network increases with the time and never decreases. The behaviour of different policies how do they handle infinitely increasing FTP load in the network and what is the impact of the increasing FTP load on the established VoIP sessions can be good evaluated using this traffic model.

C. *Evaluation criteria*

In the performed evaluations the ITU-T codec G.711 [5] is used. It is selected as a reference voice codec providing the same quality as in ISDN. To evaluate the quality of VoIP streams well known Mean Opinion Score (MOS) are used. The work of Cole and Rosenbluth in [6] presents a very good summary of ITU-T recommendation [7] that can be used to calculate the R-factor representing a simple measure of voice quality in the range between 0 (worst) and 100 (best). Using the R-factor the related MOS can then easily be calculated using equations from [6]. The equation for the calculation of the R-factor can easily be derived from detailed discussions in [6]. We refer to that and present here only the final (1) that depends on the network delay for VoIP packets d_n and packet loss rate e.

$$\begin{aligned} R = 93.24 - 0.024 d_n + 0.11(d_n - 137.3)H(d_n - 137.3) - \\ - 30\ln(1+15e)H(0.04-e) - 19\ln(1+70e)H(e-0.04) \end{aligned} \quad (1)$$

where d_n is in msecs. and H is a function as defined in (2).

$$H(x) = \begin{cases} 0 & \text{for } x < 0 \\ 1 & \text{for } x \geq 0 \end{cases} \quad (2)$$

Both d_n and e can be estimated based on available simulation traces. The average MOS of all streams in the network is then calculated using (3) and used for comparisons.

$$\overline{MOS} = \frac{1}{N}\sum_{i=1}^{N} MOS_i \quad (3)$$

The ***fairness*** of the policies is assessed based on the standard deviation of MOS (4).

$$MOS_{std.dev} = \sqrt{\frac{1}{N}\sum MOS_i^2 - \overline{MOS}^2} \quad (4)$$

The higher the standard deviation the more difference between particular VoIP streams in the network, that points to an unequal service of various users in the network. According to [6], estimated MOS scores can be rated based on the user satisfaction. During evaluations of our algorithms we consider MOS values corresponding to *Best* and *High* ratings as successful. The lowest threshold for MOS is therefore 4.03 ([6]).

The performance of FTP streams is defined by their goodput. (3) is used to calculate the FTP goodput T_{FTP}.

$$T_{FTP} = \frac{application\_data}{\tau} \quad (3)$$

where *application_data* is the number of useful bytes extracted from all TCP packets received within the time interval τ. These values are also available from trace files after every simulation.

IV. STATIC SELECTION POLICIES

Static selection policies mean that the rules how an AN for a new session is selected will not be adapted to the varying QoS situation in the network. Static selection policies can only be applied for the selection of ANs for new sessions initiated by MNs. Handovers will not be performed.

A. *Random selection*

In today's operating systems the selection of the network interface of the MN for default communication is done based either on the interface priority or based on the static configuration manually performed by the user. Thereby a single default route for IP packets of all applications maintained by the MN is used. It means, only one network interface is effectively used on every MN. Typically, priority of wired interfaces is higher than the priority of wireless interfaces available on the MN. An Ethernet interface will therefore be preferred for the use rather than a Wi-Fi interface if both are connected. Thereby there is no possibility to consider whether the transport resources like possible bitrate available in Ethernet network are really higher than resources available using the Wi-Fi access. Furthermore, there is a number of network interfaces, that can be connected to the MN using USB or the mentioned Ethernet connector. A wired interface can be connected using USB while the Ethernet interface is used to provide the MN a WiMAX connectivity using external WiMAX CPE connected to the MN via Ethernet cable. The MN cannot then recognise that a wired connection is essentially a wireless one. As result, it will be selected as a default network interface while an Ethernet adapter connected via USB has a lower priority. While selecting the default network interface the MN's administrator actually follows the same assumption - a wired interface must perform better, than a wireless one. Since the real type of the network interface does not depend on the interface type recognised by the operating system or by the user, it can be said, the interface selection is performed randomly in that case. It means if the MN has k multiple network interfaces $I_1, I_2, \ldots, I_k$, the interface I_i will be selected according to Rule 1.

$$i = uniform[1 \ldots k],\ if\ I_k = online$$

Rule 1: Random selection of the network interface

Rule 1 means, that an interface will be selected for the session randomly with uniform distribution among all network interfaces, connected to ANs. The selected interface of the MN will be used as long as it remains connected to the AN.

B. *Signal strength based selection*

Analysing the described selection policy it is obviously, that random selection of the AN for the default route is not an optimal solution. MNs can select ANs wherewith they have bad channel conditions, for example. If R is the total amount of transport resources available on an AN and the first MN with bad channel conditions requires r_1 transport resources for a service, e.g. a VoIP stream while the second MN with a better radio channel requires r_2 transport resources, then $r_1 = r_2 n$ and $n > 1$, i.e. $r_1 > r_2$. Assuming all other MNs connected to the same AN have the same channel quality and requires r resources for the same stream, the number of streams k_i that can be provided to these MNs is $(R - r_i)/r$, where $i = 1;2$ means that either the first or the second MN uses the AN. Since $r_2 < r_1$, $k_2 > k_1$ that means that more data streams can be assigned to other MNs if the second MN uses the AN. If $r_2 = \alpha r$, the advantage to assign the MN with a good channel to the AN instead of the MN with a bad channel is $k_2 - k_1 = \alpha(n-1)$. As result, if all MNs connected to the AN have the same priority, connection of a MN with bad channel quality reduces the overall performance that can be achieved using all transport resources of the AN.

Consequently, a logical extension of the random selection policy is the usage of information about link quality of different access interfaces of the MN. Selection of AN based on the signal quality is the typical method used in homogeneous access networks, e.g. in Wi-Fi, GSM or HSPA. Thereby the AN wherewith the MN has the best channel conditions will be selected for the usage. This approach could be extended to work also in heterogeneous networks. However, the main problem in that case is the definition of metrics how radio channels of different access technologies can be compared. A further problem thereby is that the network interface may not be built in the MN like an internal Wi-Fi adapter, but connected to a MN e.g. via an Ethernet cable like an external WiMAX CPE. In that situation the channel quality parameters must be informed to the operating system using additional solutions like SNMP, Telnet or similar. Using the IMRMM, the described selection policy can easily be implemented in the network. The modulation technique of the MN's channel with a WiMAX AN is analysed for the selection. Thereby Rule 2 is used to select the interface for the new session.

$$i = max(mod(WiMAX)) > 16QAM1/2\ if\ type(I_k) = WiMAX$$
$$else\ i = Wi\text{-}Fi$$

Rule 2: Signal strength based selection of the network interface

where $type(i)$ defines the interface type and $mod(i)$ defines the modulation technique used by the i'th interface. If the MN is connected to WiMAX and Wi-Fi ANs and the modulation technique of the current WiMAX radio channel is more efficient, than 16QAM1/2, the WiMAX interface will be selected, otherwise the Wi-Fi interface. The selected interface of the MN will be used as long as it remains connected to the access network.

C. *Performance of static selection policies*

Fig. 4 shows the VoIP quality in DL and UL for the described simulation scenario. The mean number of FTP streams in both DL and UL directions is the same due to the uniform random selection of the stream direction.

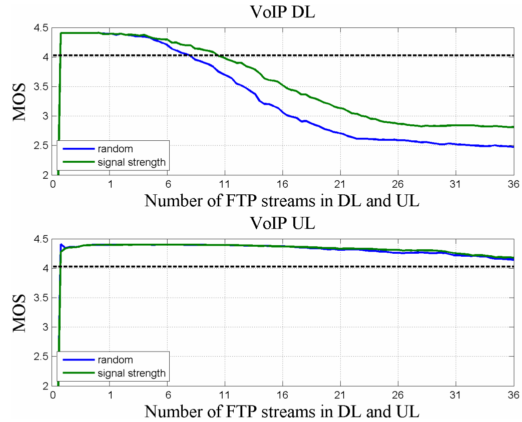


Fig. 4: VoIP performance in DL and UL for static selection policies

It can be seen that, as expected, the VoIP quality in DL decreases faster using random selection policy, than using the signal strength based selection. The difference between two policies is, however, not as large. Using random selection the VoIP quality decreases below the MOS threshold of with ≈8 FTP streams in DL and UL directions (16 streams altogether), while the quality with signal strength based selection comes under the threshold with approx. 10 FTP streams. The interesting result for further investigations is that the VoIP quality does not decrease proportional to the number of FTP streams.

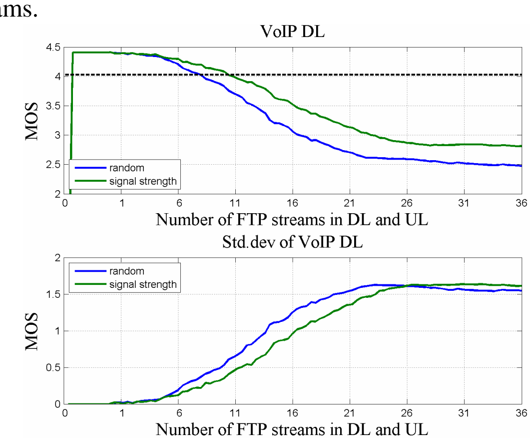


Fig. 5: VoIP mean and std.dev. in DL for static selection policies

Using random selection policy the lowest VoIP quality of ≈2.5 is reached with ≈23 established FTP streams in DL and UL while the lowest VoIP quality of ≈2.8 is reached with ≈26

FTP streams using signal strength based selection. Fig. 5 additionally shows the standard deviation of the VoIP quality in the DL direction. From Fig. 5 it can be seen, that the maximal standard deviation of the VoIP quality is $Q_{VoIP:stddev}\approx 1.6$ and is the same for both selection policies. Considering the mean VoIP quality of $Q_{VoIP}\approx 2.6$ it can be concluded, that there are two groups of VoIP streams in the network. The streams of the first group have good quality of ≈ 4.2 ($Q_{VoIP}+Q_{VoIP.stddev}$), while all streams of another group have extremely bad quality of ≈ 1 ($Q_{VoIP}-Q_{VoIP.stddev}$). To exactly determine which VoIP streams belong to each group, the impacting factors resulting in the bad VoIP quality must be determined and analysed. Since the bottleneck of the packet transmission is the capacity of ANs, the queuing delay of VoIP packets in DL is firstly analysed. Fig. 6 shows the queuing delay of both ANs for VoIP packets in DL for both selection policies.

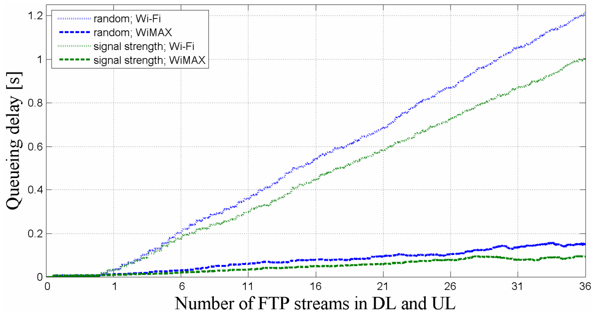


Fig. 6: VoIP queuing delay in DL

From Fig. 6 it can clearly be seen that the VoIP queuing delay of the Wi-Fi AN is much higher, than the queuing delay of the WiMAX AN. Analysing results for the VoIP quality depending on the network delay d_n only (using (1)), it can be concluded, that the VoIP quality goes to 1 when the packet delay is around 0.7-0.8 seconds. Because of broadband wired links, it is assumed that the queuing delay on ANs is the main part of the end-to-end packet delay in the network. Then it can be concluded from Fig. 6 that the VoIP quality of VoIP streams transmitted using Wi-Fi AN goes to 1 if there are ≈ 22 and ≈ 27 FTP streams in DL and UL using random and signal strength based selection policy respectively. Since there are still some additional delay in wired links and possible packet losses, the limits for the number of FTP streams can be seen as the same that have been concluded analysing VoIP quality in Fig. 5. It is clear that the VoIP streams transmitted over the Wi-Fi AN cause reduction of the mean VoIP quality of all VoIP streams in the network. Analysing the VoIP queuing delay of the WiMAX AN it can be concluded, that the quality of VoIP streams transmitted over the WiMAX AN must be around 4.3-4.4 as the queuing delay is around 150ms and 100ms for random and signal strength based selection respectively.

Considering the standard deviation of MOS for static selection policies in Fig. 5, it can be concluded, that both policies are unfair as $MOS_{std.dev}$ for both policies increase up to ≈ 1.6. As discussed above, that shows that MNs using WiMAX for VoIP traffic are preferred since their VoIP quality has *High* or *Best* ranging while the VoIP quality of streams assigned to Wi-Fi AN is unacceptable.

Considering the queuing delay of both ANs is a good approach to approximately predict and/or to assess the performance of real-time VoIP traffic transmitted over both ANs. Queue lengths and average queuing delays of packets of different traffic classes could be used as indicators for the MME to decide whether the AN operates in normal or overloaded mode. When queuing delay of packets of a particular traffic type does not increase and stays at the acceptable level, it means the AN can serve that traffic with good quality. If, instead, the queuing delay permanently grows, it indicates the service quality of the AN decreases for the particular traffic type.

To increase the performance of different traffic types, selection policies must adapt to the varying load in the network in order to prevent overload of particular ANs that can be detected by analysing the queuing delays.

V. Dynamic Selection Policy

Dynamic selection of access nodes means the possibility of the IMRMM to learn about the current QoS situation in the network in order to adapt the selection policy to the varying network load as well as to initiate handovers for load re-distribution if needed.

A. *VoIP retaining selection policy*

Analysing performance results presented in previous section it can be concluded, that in the case of congested wireless access links, different network interfaces must be used for different traffic types in order to retain quality of real-time applications. An obvious selection policy from the presented results could be to use WiMAX for VoIP streams only and Wi-Fi for FTP. That will, however, decrease the FTP performance since most of WiMAX transport resources will not be used if VoIP load in the network is low.

The initial distribution of traffic among ANs can stay the same as using signal quality based selection. However, the MME indicates to all ANs to observe their overload for VoIP traffic. Overload of an AN for the VoIP traffic is defined by queuing delay t_d, $t_d=t_{d.VoIP}+t_{d.stddev.VoIP}$. Analysis of the sum is important to avoid possible high fluctuations of the queuing delay among different MNs with e.g. different qualities of access radio channels.

Threshold values for the VoIP queuing delay in UL, ξ_{UL}, and DL, ξ_{DL}, are assigned by the MME for every AN. The AN permanently observes the VoIP queuing delay in both UL and DL as illustrated in Fig. 7. To avoid reaction to peak impulses of the queuing delay, the reaction time σ is introduced. In this way a kind of a QoS hysteresis has been defined so that a triggering message is generated by an AN and sent to the MME if $t_{dr} > \xi_r$ for the duration $t > \sigma$, where $r=[UL, DL]$. In the message the information about the direction r with decreased VoIP quality is contained.

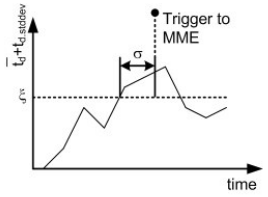


Fig. 7: Observing queuing delay on AN

The duration of σ defines the reaction speed of the MME. The reaction of the MME to the decreased VoIP quality is a rescue handover of a data stream. A rescue handover is always initiated for a one stream. It means, upon $t_{dr}>\xi_r$ for $t>\sigma$, a rescue handover will be initiated for either VoIP, FTP or FTP acknowledgement (FTPa) data stream. The type of data stream that will be handed off by a rescue handover defines the logic of the selection policy deployed in the MME.

Overload of an AN for the VoIP traffic due to the FTP traffic using a single VoIP threshold is defined as $t_{dr}>\xi_r$ while $N_{VoIPr}>0$, $N_{FTPr}>0$ and $N_{FTPar}>0$, $r=[UL, DL]$. Considering performance results for the VoIP traffic in previous section, the limiting distribution of data streams among ANs in the overload case will be done according to Rule 3.

$N_{VoIP}>0$ while $N_{FTP}=0$ and $N_{FTPa}=0$ for WiMAX ANs

$N_{VoIP}=0$ while $N_{FTP}>0$ and $N_{FTPa}>0$ for Wi-Fi ANs

Rule 3: Release of VoIP overload caused by the FTP traffic

The MME can change N_{VoIP}, N_{FTP} and N_{FTPa} by handing VoIP, FTP and FTPa streams off between Wi-Fi and WiMAX ANs.

A session maintained by a MN connected to multiple ANs will be selected for the handover from the overloaded AN applying Rule 4.

if $t_{dr}>\xi_r$ for $t>\sigma$,

VoIP in r at the MN with max(mod(WiMAX)) if $N_{VoIPr}>0$ for Wi-Fi,

FTP in r at the MN with min(mod(WiMAX)) if $N_{FTPr}>0$,

FTPa in r at the MN with min(mod(WiMAX)) if $N_{FTPar}>0$,

VoIP in r at the MN with min(mod(WiMAX)) if $N_{VoIPr}>0$ for WiMAX

Rule 4: Selection of the stream for rescue handover with dynamic policy

Where *max(mod(WiMAX))* and *min(mod(WiMAX))* means the MN having the most efficient and most robust WiMAX modulation techniques respectively. The overall performance of a WiMAX AN depends extremely on the modulation techniques of particular MNs using that AN. Therefore, to increase the overall performance in the WiMAX cell, streams maintained by MNs demanding minimal amount of transport resources will be handed off to the WiMAX AN while streams of MNs demanding maximal amount of transport resources are handed off from the WiMAX AN. The streams to be handed off are firstly FTP data, after that these are FTP acknowledgement streams.

B. Policy performance

There are two basic configuration parameters for the proposed dynamic policy - ξ and σ. The higher the admitted VoIP queuing delay ξ, the worst is the overall VoIP performance since the network delay for VoIP packets is higher. To correctly select ξ values, impact of the network delay d_n on MOS must again be analysed using (1). As it can be calculated, the maximally allowed delay while VoIP quality is still in the successful range (>*High*) is around 220ms. The limit for the *Best* VoIP quality is around 135ms. That is why two thresholds for the queuing delay, ξ_1=0.120s and ξ_2=0.210s have been selected for the investigation. The reaction time for the investigation is set to σ=1.0s since the detailed analysis of its impact on the final results has shown that it is negligible in comparison to the impact of ξ.

Firstly, to see the difference of the dynamic policy to both static policies, queuing delays on both ANs are shown in Fig. 8.

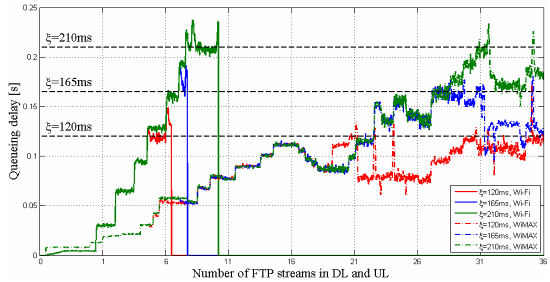


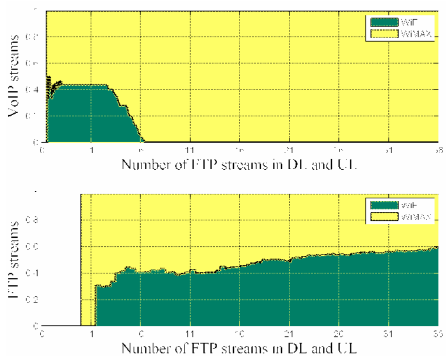
Fig. 8: Queuing delay in DL on both ANs using dynamic selection policy

Comparing queuing delay in Fig. 8 to this with static selection policies in Fig. 6, it can be clear seen, that the maximal VoIP queuing delay does not highly overcome defined thresholds ξ_i. It means, MOS of none of VoIP streams independent of used AN does decrease below the desired quality. The estimated VoIP delay goes to 0 on Wi-Fi AN between 6 and 11 FTP streams in every direction. It is the result of Rule 3 and Rule 4 as streams may be handed off between ANs. To better see performed handovers, Fig. 9 shows the percentage distribution of VoIP and FTP streams among two ANs.

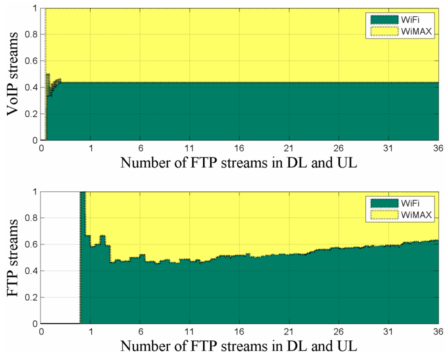
In the beginning of the simulation streams of both types are distributed among ANs almost equally. With increasing load in the network, however, all VoIP streams are handed off to the WiMAX AN while more and more FTP streams are handed off to the Wi-Fi AN. That is required to not to disturb the performance of VoIP streams running over the WiMAX AN.

VI. Static vs. Dynamic Selection Policies

The elaborated VoIP retaining policy is compared with static selection policies comparing VoIP and FTP performance in the network. Fig. 10 presents the VoIP performance for both static and dynamic selection policies in DL. As it was expected, VoIP performance of dynamic policy never decreases below the 4.03 as the MME performs rescue handovers to avoid that. In contrast to that, VoIP using today's random and signal based selection policies decreases with increasing load as it has been discussed in Section IV.C. As expected, the VoIP performance with ξ_1 is better than while using ξ_2 for both, mean values and fairness factor.



(a) DL



(b) UL

Fig. 9: Percentage distribution of streams among ANs in DL and UL

It is because, as mentioned above, the quality of different VoIP streams varies among MNs much more with higher ξ values. The fluctuation is higher since the admitted interval for the VoIP queuing delay is larger so that delays of particular streams may differ very much.

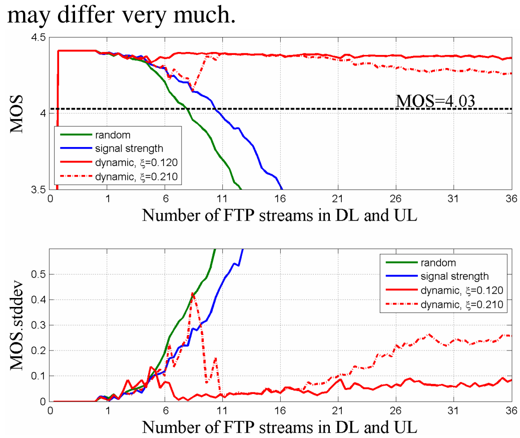


Fig. 10: Static vs. dynamic selection policies, VoIP in DL

Using lower ξ values, the delays of different streams fluctuates in a much shorter range, instead. The main goal of the dynamic policy has been achieved as it is good to see in Fig. 10: the VoIP quality does not decrease below the defined successful threshold of 4.03.

Fig. 11 shows the performance of FTP applications in both directions. As it can be seen from Fig. 11, the FTP performance with dynamic selection policy in DL is approx. the same as when using signal based selection. Instead, the FTP goodput can be increased significantly in UL. A detailed analysis of that issue has shown that this difference is because re-distribution of FTP sessions in UL with increased load where FTP streams handed off to the Wi-Fi AN could achieve better performance than while assigned to the WiMAX AN.

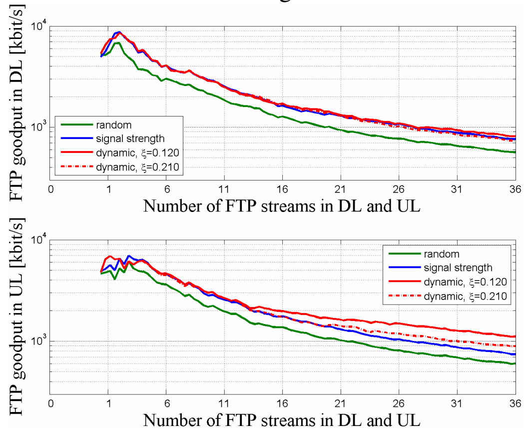


Fig. 11: FTP goodput in UL and DL with dynamic selection policy

That cause leads also to the issue that the FTP goodput is higher using ξ_{l}. More FTP streams will be handed off to the Wi-Fi AN in that case. More transport resources are then offered for these streams because of the particularities of the Wi-Fi access technology. It can be concluded analysing Fig. 9a where it is clearly to see that up to 65% of all FTP streams run over W-Fi AN.

Results for the VoIP quality in UL are not presented since even using static selection policies the VoIP performance does not decrease below MOS=4.03. Mainly it is because of the uncoordinated random access to media in Wi-Fi. All MNs and the AN have the same priority accessing the media that is that every node gets the same transport resources in the case all of them have to transmit the same amount of data. However, the AN has to transmit $N_{MN}D$, where D is the data amount that has to be transferred from/to every MN in DL and UL. That is why UL can still operate in non-overloaded mode while the DL on the AN is overloaded.

VII. Conclusion

In this paper we have modelled the today's available algorithms for selection of ANs for MNs with multiple network interfaces of different access technologies. A new selection algorithm based on the evaluation of the queuing delays on ANs has been proposed. A performed comparison between today's and proposed algorithms shows the ability of the new method to retain the quality of particular applications above the desired level in the network. It provides also a fair service for the selected network applications. As the example of an application which QoS level needs to be kept above a threshold, VoIP application has been considered. It was detected that its performance does not decrease below *High* rating of MOS even with high FTP load in the network that is impossible using today's selection algorithms. Furthermore, the FTP goodput in UL is slightly increased using new dynamic selection policy because of intelligent usage of available transport resources of both ANs.

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