AN ALGORITHM TO ENHANCE THE NETWORK SELECTION PERFORMANCE IN IN AN INTEGRATED UMTS/WLAN ENVIRONMENT

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Abstract: Future wireless networks will be multi-access networks, and the users can benefit from the advantages of any radio access technology. Then, network selection can play a critical role in overall network functionalities. Interworking between UMTS and WLAN is considered as a feasible approach toward next generation wireless networks. In this paper, we propose a method that uses statistical parameters of a WLAN cell traffic load to measure its capabilities to ensure QoS for the requested service. A network selection algorithm can use this measurement and user preference to select the best radio access technology. We investigate the performance of this algorithm through simulation of transferring video traffic in WLAN. The results show that the proposed algorithm has an acceptable performance and in about 95 percent of the times, a user that opt WLAN as his access network will have the requested QoS.

1 INTRODUCTION

Wireless networks have emerged, evolved, and experienced an extensive deployment. These networks include cellular networks like GSM, GPRS, UMTS, and CDMA 2000 for voice and data transmission and wireless LAN for broadband wireless Internet access. Each network has its own merits and most likely the next generation wireless networks will be multiple access networks exploiting the advantages of all existing radio access technologies (RAT).

Interworking between UMTS and WLAN is considered as a feasible approach toward next generation wireless networks (Salkintzis, 2005). UMTS can provide data rates from 384 Kbps up to 2 Mbps for wireless Internet access over a wide geographical area. In hot-spots, where user's mobility is limited, we can use WLAN to provide users with higher data rates (up to 54 Mbps) and also benefit from low deployment cost of this network and its license free bandwidth.

Recently 3GPP has developed two UMTS-WLAN interworking architectures for different usage scenarios (3GPP, 2003), (3GPP, 2005a). A common scenario is when a UMTS subscriber can connect to a WLAN to use Internet and other WLAN data services. This architecture, called "WLAN direct access", supports transferring authentication, authorization and accounting (AAA) signaling between two networks. The other architecture which is named "WLAN 3GPP IP access" enables UMTS subscribers to access UMTS packet switching services through WLAN.

In heterogeneous networks, where the user can select his access technology, the network selection algorithm can have an important effect on the overall network performance and the user's experience of received QoS. An efficient algorithm for network selection should consider the network condition like received radio signal strength (RSS) and network traffic load to select the best RAT for the user.

In this paper, we focus on network selection issue in an integrated UMTS and WLAN environment. We offer an algorithm that measures WLAN cells capabilities in satisfying service requirements and user preference. In this method rather than using instant amount of the cell's traffic load, we use statistical parameters of the network traffic load like mean and variance. These parameters are calculated in Access Point (AP) and are broadcast in the WLAN cell. We show that while in some situation instant amount of the traffic load can not be used, our method have an acceptable performance even for highly stochastic traffics like video.

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The rest of the paper is organizes as follows. In section II, we discuss the network selection issue in heterogeneous networks and introduce the network selection and advertisement scenario that 3GPP has proposed for UMTS and WLAN interworking. In section III, we introduce our algorithm. Simulation results come in section IV. Finally, we conclude in section V.

2 NETWORK SELECTION IN HETEROGENEOUS ENVIRONMENTS

In heterogeneous networks, where user can opt for several access technologies, network selection algorithm plays a vital role in insuring the quality of the received service and efficient usage of the network resources. An efficient network selection technique which keeps the user Always Best Connected (ABC) should consider user preference, service requirement, and network condition (Qingyang, 2005). The concept of ABC as defined in (Gustafsson, 2003) is that the user is not only connected but also connected through the best device and access technology. In a network selection procedure, the first step is collecting necessary information about network condition, application requirement, and user preference. Then, this information is used for making decision.

For interworking between UMTS and WLAN, 3GPP considers the generic network advertisement and selection scenario as been depicted in figure 1. In this scenario, a user which is a subscriber of the 3GPP home network is located in an area which is covered with several WLAN access networks. Using information which is advertised by WLAN access networks, the user decides which WLAN and which 3GPP visited network should be used.

The user can use periodic beacon frames sent by WLAN access networks to gather the necessary information. This method is called passive scanning. The user can also use active scanning to collect this information. In active scanning, by sending probe request frame, the station asks the AP to send the necessary information in the probe response frame.

In any WLAN access network, it is possible to have cells with overlapping areas. In these areas the user can opt for possible APs.

Therefore in an integrated UMTS and WLAN environment, the network selection algorithm should determine whether we must connect to UMTS or WLAN access network, which WLAN should be used and which AP in the WLAN should be selected.

Most simple conventional network selection algorithms use RSS for RAT selection. In (Yilmaz, 2005) the authors show that a simple 'WLAN if coverage' strategy would lead to satisfactory results. This is because when the hotspot is not congested, WLAN with its low service cost and high bandwidth can satisfy the user's preference and the service requirement. This simple strategy can also boost the UMTS performance. Because by selecting WLAN for wireless Internet access, UMTS channels become free for voice traffic.

However, in this paper it is shown that when the hotspot is congested, UMTS can offer higher bandwidth. Hence considering only RSS is not sufficient and the network traffic load should be considered as an input parameter for network selection algorithm, as the authors in (Hyun-woo, 2005) have used for AP selection. The authors show that advertising the amount of traffic processed in AP can lead to better network performance and fairness among users.

Authors in (Qingyang, 2005) propose a general decision method to take into account the other users requirements like security, cost, and reliability.

In this paper, we propose a method for evaluating WLAN capabilities in providing QoS for real-time applications such as video and voice. We show that for video services, the traffic load is highly stochastic and instant amount of the traffic load can not be used for network selection, as it is used in (Hyun-woo, 2005) for AP selection. Therefore in our algorithm, the statistical parameters of the WLAN cell traffic load are estimated in the AP and broadcasted in WLAN beacon frames. A UMTS subscriber can use this information to determine if this AP can ensure transferring data frames in their due time or not.

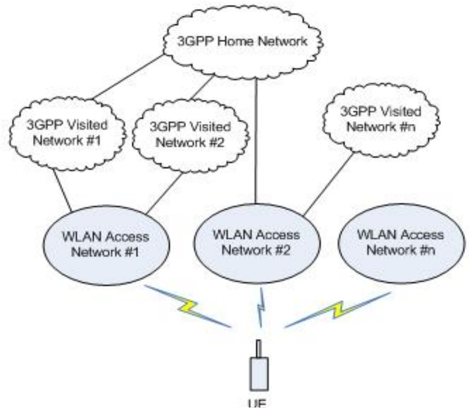

Figure 1: Generic scenario for network advertisement and selection.

3 THE PROPOSED SOLUTION

WLAN stations within a cell can use distributed coordination function (DCF) and optional point coordination function (PCF) for accessing the medium. The former in which users compete with each other to send their frames is not a feasible option to support QoS for real-time application (Rajavelsamy, 2005). Therefore, we have chosen the latter method, PCF, to be used in WLAN cells.

In PCF, the AP periodically announces the inception of a contention free period (CFP) with a specified maximum duration. The period of the CFP and its maximum duration is announced in a standard information element (IE), which is included in the beacon frame. In the CFP, the AP polls any station on a polling list. During this period, stations may transmit only if the AP solicits the transmission with a CF-Poll frame. After elapsing the maximum duration time or reaching the end of the polling list, the CFP ends. After CFP, contention period (CP) begins, in which users can access the medium using DCF.

Depending on the traffic load, it is possible that some station on the polling list can not be polled in the CFP. Now the purpose of our algorithm is to determine if a new user is added to the polling list, can it be polled at least in the x percent of the CFPs or not. The choice of x depends on the service requirement and the user preference. If this AP is not capable of polling the station, we conclude that the WLAN cell is congested and we search for other APs. If there is not any suitable AP, UMTS network is selected. Our algorithm for assessing the WLAN cell capabilities has the following stages: 1) The AP measures the traffic transmission time (TTT) for each CFP. TTT is the time spent for transmitting and receiving data frames to and from polled stations. 2) The AP estimates the mean and the variance of the TTT. 3) These estimated parameters are broadcasted using a non-standard IE which is included in the beacon frame for passive scanning or in probe response frame for active scanning. 4) By assuming normal probability density function (pdf) for TTT, and by considering the service requirements and the user preference, any station is able to estimate the upper bound so that, TTT in x percent of the CFPs is bellow it. 5) By considering CFP maximum duration, the estimated bound, the length of data packets and the transmission rate, the station can determine if this AP is capable to ensure the desired QoS or not.

Before going into detail, we shortly discuss the rationales behind our algorithm. For reasonable handoff decision, we must consider cells traffic load. However in the 802.11 standard, each station can scale back its transmission rate, according to its radio link condition. Considering only cell's traffic load, will lead to ignoring station's transmission rate which has a significant effect on the cell capacity. For this reason, we have used traffic transmission time to consider users rates in our WLAN capability assessment.

Rather than using instant amount of the traffic load, we use statistical parameters, mean and variance. Even though VoIP flows usually have constant bit rate, other real time services like video generate data packets with vastly different packet size. The interval between two successive packets can also be very different. In this condition, the amount of the traffic load in one CFP has little information about successive CFPs. In the next section we will show that the instant amount of the traffic load is not a viable network selection criterion for video flows.

When the number of the users associated with the AP is large enough, the assumption of the normal pdf for traffic transmission time is reasonable. This condition usually occurs within hotspots with high traffic load where the network selection algorithm can play a pivotal role in overall network functionalities.

Now we show how the mean and the variance can be estimated and how the upper bound for TTT is determined. A simple method for estimating mean and variance is the moving average, which is shown in bellow.

$$avr[i] = \alpha \times avr[i-1] + \beta \times T[i]. \quad (1)$$

$$\mathrm{var}[i] = \alpha \times \mathrm{var}[i-1] + \beta \times (T[i] - avr[i])^2. \quad (2)$$

Where: $0 < \alpha < 1$ and $\alpha + \beta = 1$

In these formulas, *T[i]* is the traffic transmission time in the i^{th} CFP and α is called forgetting factor and shows how much we pay attention to the past data. The choice of α is a trade off between algorithm precision and its delay. With greater α, estimation of the mean and the variance would be more prices, but the effect of the entrance of a new user to the WLAN cell is detected with more delay.

The mean and the standard deviation of the TTT can be broadcasted in WLAN cell by using a non-standard IE. Figure 2 shows this IE. In the 802.11 standard each IE has a variable length and is identified by 'Element ID' field. Many of the

Element ID	Length	Standard deviation of CFP duration (msec)	Mean of CFP duration (msec)
2 Byte	2 Byte	1 Byte	1 Byte

Figure 2: Proposed IE for broadcasting traffic information.

Element IDs are reserved for future uses. One of them can be used for this non-standard IE.

If we assume the normal pdf for the TTT, we easily can estimate an upper bound so that TTT in the x percent of the CFPs is less than it.

$$bound = Q^{-1}(\frac{x}{100}) \times \sigma + avr. \quad (3)$$

Where Q^{-1} is the inverse of the standard normal cumulative density function, σ and avr are the standard deviation and the mean of the traffic transmission time. But as a matter of fact, the normal pdf is only an approximation for probability distribution of TTT and the estimation of the mean and variance is not very accurate. Therefore we should use a little higher bound to ensure that the estimated bound is equal or greater than actual bound and our algorithm works well. Our simulation results show that by using only 25% greater standard deviation, the estimated bound has satisfactory performance. So we use (4) to estimate the bound. When the network has high traffic load, this bound is not very different from what is estimated by normal pdf.

$$bound = 1.25 * Q^{-1}(\frac{x}{100}) \times \sigma + avr. \quad (4)$$

By considering this bound, maximum packet size and the transmission rate, if the station selects this AP, it can be sure that x percent of the data packets could be sent in their due time.

The choice of the x is a trade-off between service requirements and user preference on the one hand, and network performance on the other. Some applications like VoIP can tolerate more frame loss. But some applications like video are more sensitive on frame loss. Users also have different preferences. For example, rather than using an expensive UMTS service, a user may prefer to pay less and use a WLAN access network that guarantees to transfer only 95% of the data frames. The network performance is another issue. When transferring of more data frames is required, fewer users can be accepted and WLAN performance in most of the time is bellow its maximum.

4 SIMULATION RESULTS

To study the performance of the proposed algorithm, we have used a simulation scenario, in which 80 UMTS users get into a WLAN cell. The entrance time of each user is randomly chosen from 1 second to 1200 seconds. The entire simulation duration is 3000 seconds.

The AP of the WLAN cell supports 802.11a standard. All UMTS users which want to connect to WLAN should support WLAN radio interface as they do in our simulation scenario. The transmission rate of the users can be different according to their radio link condition. We did not simulate the WLAN physical layer. So we randomly chose 18 Mbps or 24 Mbps as the user's transmission rate to imitate the different radio link condition.

WLAN cell supports PCF for medium access. The CFP maximum duration is 50 msec and CFP announcement interval is 60 msec. Therefore, stations can use DCF for remaining 10 msec. CP was not explicitly simulated. We have used a constant 10 msec period plus a random variable chosen from [0, 1 msec] interval for CP to mimic the real conditions in which CFP inception can be delayed by CP.

All users have a bidirectional (up-link and down-link) real-time video session. Video streams are coded by H.263 standards with target rate of 64 Kbps. H.263, MPEG-4 and H.264 are three codecs which 3GPP recommends for conversational video services (3GPP, 2005b). For generating video traffic, we have used trace files that publicly can be accessed in http://trace.eas.asu.edu/trace /ltvt.html.

Figure 3 shows the traffic transmission time for all CFPs. From this figure it is evident that since the instant values of the TTT are vastly different, it is not a viable criterion for network selection algorithm as it is used in (Hyun-woo, 2005) for AP selection.

Figure 4 shows the cumulative density function of the TTT, when there are 80 users in the WLAN cell. In this figure 95% and 99% of the CFPs have traffic transmission time less than 45 and 49.5 msec respectively. Therefore the actual upper bound for x = 95 and x = 99 is 45 and 49.5 msec respectively. The CFP maximum duration is 50 msec. As a result, if the maximum size of the packets generated by application is 1500 bytes, for sending 99% of the frames, more users can not be accepted. But for sending 95% of the frames in CFP, still more users can be accepted. So we can see that there is a trade off between network performance and service quality.

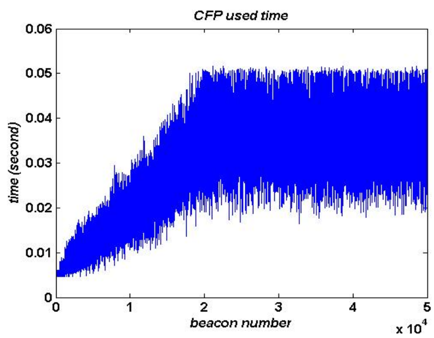

Figure 3: CFP duration for all CFPs.

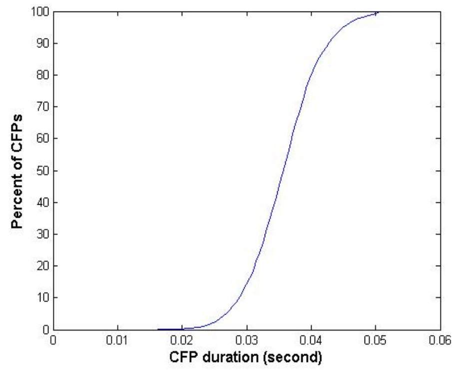

Figure 4: Cumulative density function of CFP durations.

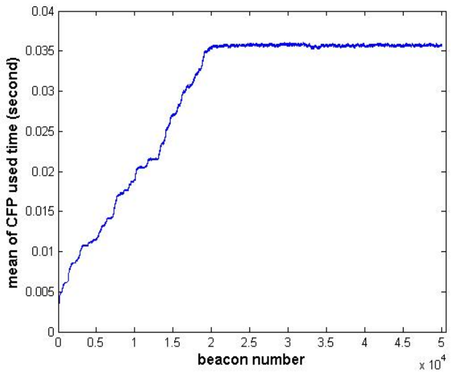

Figure 5: Mean of the CFP duration.

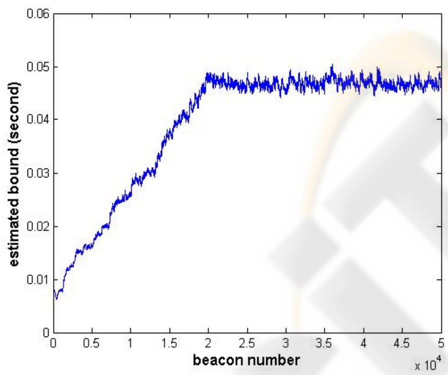

Figure 6: Estimated bound for sending 95% of the frames.

For estimating the mean and the variance of the traffic transmission time we have used (1) and (2) with $\alpha = 0.01$. Figure 5 shows the mean of the traffic transmission time. The mean increases when more users enter the WLAN cell and finally it reaches to about 36 msec. As mentioned previously, to send 99% of the frames, no new user can be accepted. Therefore the performance of the WLAN cell in many times is below its maximum. The source of this deficiency is the stochastic nature of video traffic.

We have used (4) to estimate the upper bound for traffic transmission time. As stated before, this bound is a little greater than what is predicted by normal pdf. In our simulated scenario, the standard deviation is about 4 msec and the mean is 36 msec. therefore the bound estimated by (4) is only 3% greater.

This bound for x = 95 is depicted in figure 6. Our simulation results show the estimated bound by a new user in 98.5% of cases will be equal or greater than its actual value, 45 msec. so in these cases, if the user selects WLAN as its access technology, he can transfer at least 95% of his frames. In only 4% of the cases, the estimated bound is larger than 48.5 msec. In these cases a user with maximum packet size of 1500 and transmission rate of 24 Mbps will not choose WLAN as its access network. In these cases our network selection algorithm has been wrongly too skeptic. But as one can see in this 4% of cases, only a small percent of the cell capacity remains idle. Thus, our network selection algorithm has an acceptable performance.

Figure 7 shows the estimated bound x = 99. Our results show that in 6.3% of the cases, new users wrongly estimate the bound less than its actual value, which is 49.5 msec. This estimated bound can be misleading for new users. For example in 2% of the cases, a new user will estimate that the bound is less than 49 msec. if the transmission rate of this user is 24 Mbps and the maximum packet size is 1000 byte, the network selection algorithm wrongly will choose the AP.

Our results show that the access delay is mainly bellow the CFP announcement period. Access delay is a function of network load, user's place in the polling list and the characteristics of the traffic load generated by the user's application. To demonstrate this issue, in figure 8 we have shown the access delay for three users which are placed in first, 50^{th} and 80^{th} row of the polling list.

From this figure we can see that the access delay of the first user in the polling list is mainly affected by his traffic load characteristics. Due to fact that polling frame for this station is sent after CFP inception. So network load can not affect the access delay. On the other hand, access delay of the last user in the polling list (user 80) is mainly affected by CFP traffic transmission time. By comparing access delay of this user with access delay of the 50^{th} user in the polling list, we can infer that being in the end of the polling list does not mean that the station will have greater access delay.

As a result of being the last one to be polled, the last station has more chance to transfer its frame that has been generated in this CFP interval. Therefore

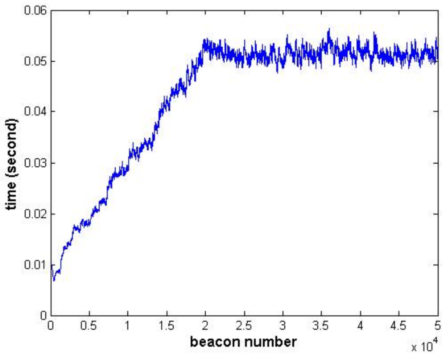

Figure 7: Estimated bound for transferring 99% of frames.

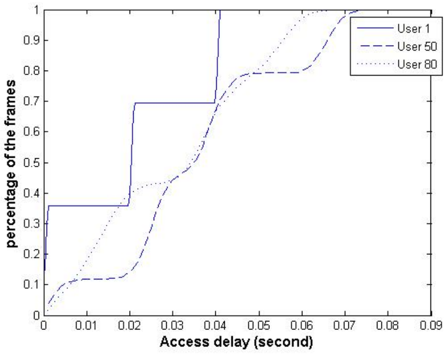

Figure 8: Cumulative density functions of the access delay**.**

the access delay for this station is less than CFP period.

5 CONCLUSION

Next generation wireless networks will be multi-access networks. In these networks, a subscriber can use one of the access technologies to become always best connected (ABC) according to his requested service and network condition. Integration of UMTS and WLAN is a viable solution toward the next generation wireless networks. As we discussed in this paper, the network selection algorithm has a significant effect on the overall network performance. In this paper we proposed a method for measuring WLAN cell's capabilities in such a way that enables the network selection algorithm to consider network condition, service requirement, and user preference to keep the user ABC.

We have used video traffic in order to study the performance of our proposed method. Simulation results show that the proposed method has acceptable performance in measuring WLAN cell capabilities to accept the UMTS users.

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