Dynamic service oriented resource allocation system for interworking broadband networks

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Abstract: Optimising available radio resource efficiently to the diverse traffic categories in a heterogeneous interworking network, is the key issue of radio resource management (RRM). In this paper, an advanced RRM method specified as dynamic application centric resources provisioning algorithm (DAC-RP), to provide users with a dedicated set of suitable channels to real-time (RT) and non-real-time (NRT) services based on bandwidth conditions, to maximise the capacity with satisfied QoS constraints is proposed. The DAC-RP is realised over an ultra mobile broadband (UMB)-worldwide interoperability for microwave access (WiMAX)-wireless local area network (WLAN) hybrid interworking network, linked over novel intelligent internet protocol (IIP) architecture. IIP, an unified architecture, obtained by the merging IMS call session control functions (CSCFs), application services, enhanced IMS and centralised services, under a single layer with a common set of control and routing functions, to converge the heterogeneous protocols, functional entities and applications. The competency of the IIP and DAC-RP is validated by comparing the performance metrics of RT and NRT applications, simulated for the IIP based UMB-WiMAX-WLAN network developed using the OPNET to the scenario with existing IMS and UMTS-WiMAX-WLAN network.

Keywords: radio resource provisioning; quality of service; broadband wireless network; absolute partition; heterogeneous network; call control layer; real-time; non-real-time application; NRT; IP multimedia subsystem; dynamic resource management.

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1 Introduction

The progress of the next generation broadband wireless communication technology necessitates the attention to meet the soaring demands of the increasing multimedia and interactive applications. The prospective of the developing wireless technologies, perceive to reach tremendous growth by employing, interworking of the existing third generation partnership project (3GPP) and non-3GPP networks with a diverse heterogeneous characteristics (Andreev et al., 2014). Each existing active wireless networks has its unique characteristics such as bandwidth, data rate, mobility, coverage, protocols, etc. to support the binding quality of service (QoS) requirements. With wireless local area network (WLAN) offering high bandwidth and less coverage; the worldwide interoperability for microwave access (WiMAX) renders larger coverage region and maintained QoS of broadband services with limited mobility. WiMAX and WLAN, of the current wireless communication industry, are ever growing standards with upgrading features and benefits to meet the requirements of the massive broadband communication requirements. Operating in a wide range of unlicensed and licensed spectrum, they enhance the performance of the future generation communication, through the flexibility and wide interoperability features with any other existing network. To meet the expected data rate of the developing 5G, many research using LTE and LTE-A was being investigated. LTE, with its OFDMA, advanced modulation and coding (AMC) techniques and MIMO antenna technology, strives to meet its theoretical data rate of 300 Mbps, which is practically yet to be achieved. With the present 4G offering a high capacity network with mobile multimedia access, the future 5G will be a set of an interoperating existing and evolving systems designed to meet new requirements, such as virtually, zero latency gigabit experience, to support tactile internet service. Ultra mobile broadband (UMB), an evolution of 3GPP2's EV-DO Rel 8, with similar features of LTE, could also endow with a data rate of 280 Mbps in DL and 68 Mbps in UL. An optimised network functionality to provide flexibility through interoperability of 3GPP and non-3GPP networks is offered by UMB. In addition they could offer a low latency network and improved interference management techniques with a scalable IP based hierarchical architecture and flexible spectrum allocation to support heterogeneous interworking wireless network scenario. To meet the future extreme capacity and performance demands, telecommunication industry is looking ahead for a network that support increased data rate and reduced latency (nearing to zero), for which UMB could be a solution. It supports network users with increased coverage, and spectral flexibility through advanced adaptive modulation, link adaptation, and multiple antenna techniques, akin to the features of active LTE technology. The complementary and requisite characteristics of such networks endow to integrate these heterogeneous systems, to achieve optimisation of the future All-IP broadband wireless technology network functionalities. It intends to offer a flexible and open architecture for a large variety of radio access technologies (RAT), with diverse protocols and applications with specified QoS demands, to provide always a best-connected set of connections (Bellavista et al., 2009). With such prominent notions; it has certain challenges to the network designer in terms of improving system capacity and to meet the growth of wireless connections with user perceived QoS, through more intelligent and flexible network radio resource management (RRM) strategy.

The emerging broadband network is capable of providing rich multimedia and interactive services to make advantage of the vast technical potentials of the multipurpose mobile communication devices (Falowo et al., 2011). Exiting IP multimedia subsystem (IMS) offers the network provider with such utilities, designed to allow the convergence of the existing 3G, 3GPP, and other non-3GPP networks, yet lacks to solve the scalability issues due to the incomplete independency of the underlying layers. To endow with the challenges of the heterogeneous next generation networks, a substantial improvement of the IMS is proposed through a novel intelligent internet protocol (IIP) for UMB-WiMAX-WLAN hybrid interworking network. It offers a complete interaction of the media transport, call session control function (CSCF) and the application server (AS) layers of the IMS, to interact seamlessly over a unified architecture. To resolve the resource utilisation maximisation problem, to increase the capacity and data rate of the heterogeneous interworking network, a dynamic application centric resource provisioning (DAC-RP) scheme is developed, that meets the essential QoS constraint of the divergent real-time (RT) and non-real-time (NRT) applications.

The rest of the paper is organised with the following sections. Section 2 briefs on some existing RRM methods. Section 3; explicate the role of the IIP architecture and its effectiveness in seamless integration of UMB, WiMAX and WLAN network. The proposed DAC-RP for allocating users with appropriate channel to serve the RT and NRT applications, through complete bandwidth partitioning, with dynamic bounds is explained in Section 4. Concert metrics for voice over internet protocol (VoIP), file transfer protocol (FTP), hyper text transfer protocol (HTTP) and e-mail applications are obtained for the interworking architectures simulated using OPNET14.5 is discussed in Section 5. DAC-RP's activeness is validated by comparing the results of IIP based UMB-WiMAX-WLAN network without RRM implementation. The encroachment of the developed IIP architecture's features is validated through network interconnected over existing IMS and with UMTS-WiMAX-WLAN network. The enhancement in the response time and latency metrics obtained for DAC-RP amid IIP architecture is concluded in Section 6.

2 Related work

The need for efficient RRM design is gaining importance in recent research, due to the demand of high data rate for best effort traffic service, dynamic protocols and control functions of a heterogeneous wireless network. Several RRM methods were proposed in literature to satisfy the prescribed QoS needed for multiple applications and traffic services in a heterogeneous wireless networks. A broad classification of this could be made as centralised or distributed methods of managing the available radio resource among the users in an interworking network. Centralised scheme of RRM involves different load balancing and policy based algorithm for managing the spectrum availability. A predictive QoS balancing and load balancing algorithm to achieve improved end user throughput is developed in Ong et al. (2012). Simplified dynamic hierarchy resource management (SDHRM) algorithm composed of prediction-based network-level resource allocation and connection level network selection is proposed in (Juan et al., 2014). The SDHRM algorithm takes advantage of the tidal traffic to exploit the resources efficiently. Connection admission control based on the signal to interference ratio and delay metric of the network are considered in Walingo and Takawira (2015). Capacity and spectrum sharing algorithm for efficient radio resource utilisation for a mobile network were discussed in Panchal et al. (2013). A number of other load balancing and resource provisioning algorithm were presented in literature, but a complete centralised method does not consider the network's QoS policies and increases the overhead by augmenting more decision criteria.

Distributed method of resource allocation mechanism involves, service and user preference based algorithm for efficient utilisation of the available radio resource in a heterogeneous network. A joint allocation algorithm using modified Newton method adopted to maximise the total system capacity by choosing the optimal bandwidth for the services and power allocation to that bandwidth is proposed in Shyam Sundar and Nanda Kumar (2012). A distributed multi-service resource allocation method is given in Ismail and Zhuang (2012) were, each network employs a priority mechanism in order to give a higher priority on its resources to its subscribers on the home network than to the other users. A common resource management used in heterogeneous wireless networks to provide a balance between different levels of data rates, mobility, and traffic is proposed in Skehill et al. (2007). Admission control strategies are essential to the operation of converged heterogeneous networks to select the most appropriate wireless access based on service type and user preference. A cost function dependent resource allocation algorithm for the mobile network with RT and NRT service is given in Bublin et al. (2007). However, it was proved to be efficient for RT than NRT. An absolute distributed RRM algorithm has a disadvantage of increasing the restricted access of available resource, without taking into account the network metrics, but increases the QoS of the user depending on the application profile.

To overcome the demerits of the complete centralised and distributed RRM, this DAC-RP is proposed. It mulls over the network's QoS policies such as the total available channels and call blocking probability, and as well as the user's QoS policies for allocating the radio resource in an interworking network. It is based on partitioning the available channels dynamically, to select the suitable spectrum under the developed hybrid heterogeneous UMB-WiMAX-WLAN RATs integrated over the IIP architecture, to serve all the multimedia and interactive applications.

3 UMB-WiMAX-WLAN integrated over IIP

This section brief on the proposed novel UMB-WiMAX-WLAN-IIP hybrid interworking network, en-route for seamless integration, designed for higher data rate and increased capacity.

3.1 Intelligent internet protocol

IIP is a unified architecture with media transport/connectivity layer, control layer, and application layer servers, as a common core. The intense benefit of the IIP lies in the merged layers of IMS, to provide common application, call control and transport functionality to multiple services, and users across multiple access networks. Without effectively modifying the existing access network and network elements, IIP reduces the cost of deploying new elements. It includes all the basic servers of the existing IMS, and additionally modified Emergency, multimedia, gateway, centralised and application services, over converged IMS layer functionalities. Yet with additional CSCF, core process servers and ASs, to support enhanced functions, the developed IIP architecture slightly tends to increase the signalling and network complexity.

The communication between elements of the interworking network is provided by the CSCF servers (P-CSCF, I-CSCF, and S-CSCF), SIP and the HSS servers of the call control layer. Registration, routing of signalling messages, control of traffic between the layers are the role of these servers. The *Proxy*-CSCF (P-CSCF) is the first point of attachment to the core of the IIP architecture. It is associated with the endowing secure establishment of the session, identification, monitoring the movement of subscribers, maintains SIP signalling procedure, and sustains the QoS level by assigning appropriate bearer resource. I-CSCF; *Interrogating*-CSCF interconnects and routes the signalling messages among the IMS province securely, to retrieve the S-CSCF's address to perform SIP registration. *Session*-CSCF (S-CSCF), is the central component that enables the coordinated interaction of all control layer and signalling entities by inspecting every signalling messages. It carries out routing services to PSTN, accompanied with other call control layer entities such as breakout gateway control function (BGCF), media gateway (MGW), and media gateway control function (MGCF). The home subscriber server (HSS) provides the details of the subscriber database to the other core elements of the home network. These components form the basis (existing IMS) of the IIP core network, and the convergence functionality is enhanced through the additional servers such as emergency CSCF (E-CSCF), BGCF, MGW, MGCF, multimedia resource function controller (MRCF) and media resource function processor (MRFP). E-CSCF resides with the P-CSCF to handle emergency call request during IIP call session. It also helps in reducing the call dropping probability on congestion of a channel. Interworking entities, such a BGCF, MGCF, and MGW are used to route the SIP signalling to the destined subscriber of underlying heterogeneous media/connectivity layer. The main function of BGCF is to offer a connection between IIP packets switched network and non-IIP (PSTN) circuit switched network. The servers, MGCF and MGW, collectively work as a gateway through which BGCF transport the signalling messages, to direct the distribution SIP signal across sessions, and to encodes and decodes the media exchanged between the

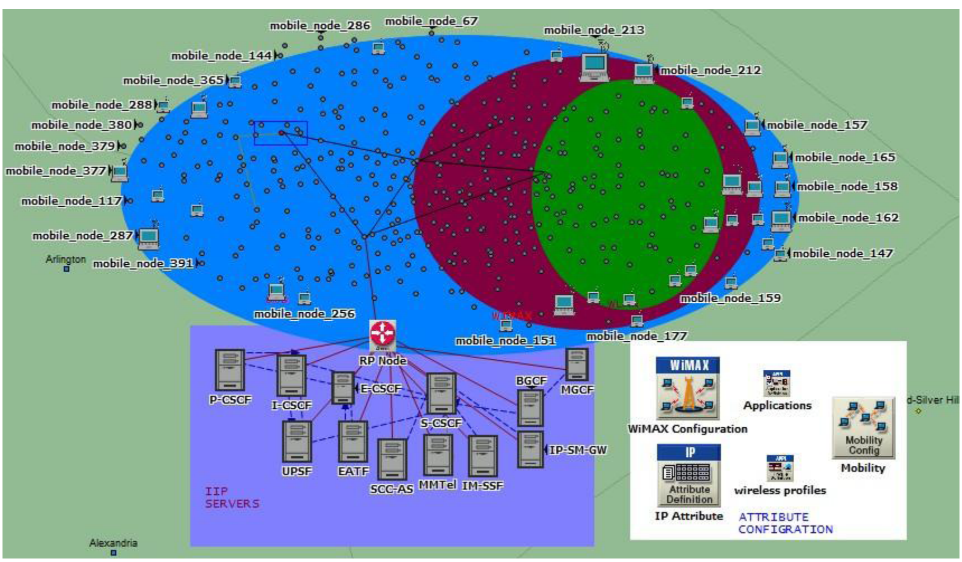
IIP and circuit switched network respectively. MRCF performs media processing required by the AS and controls MRFP to transfer and retrieve the media stream resource such as conferencing, voice mail, recording, voice processing, etc. from the AS and S-CSCF.

The AS layer is the execution platform for one or more ASs that control the end service based on user requirement (Khlifi and Grégoire, 2008). The different servers included in AS are telephony application server (TAS), IP multimedia-service switching function (IM-SSF), service centralisation and continuity application server (SCC-AS), IP-short message gateway (IP-SM-GW), multimedia telephony (MMTel), open service access gateway (OSA-GW), etc. These servers make IIP a reality to support the varied multimedia and interactive services available for digital convergence. A wide variety of servers were added to support both RT and NRT services due to the flexibility provided in the IIP architecture, to form a consistently unified architecture.

3.2 UMB-WiMAX-WLAN hybrid interworking network on IIP

The proposed interworking architecture of UMB-WiMAX-WLAN over IIP developed with OPNET 14.5 simulation tool is shown in Figure 1.

Figure 1 UMB-WiMAX-WLAN interworking network on proposed IIP architecture (see online version for colours)



UMB a part of 3GPP2; is a mothballed technology developed as a successor of CDMA2000 EV-DO-Qualcomm. UMB can subsist as an alternative to 3GPP through its necessary features supporting both RT and best effort traffic with flawless mobility (Juneja et al., 2010). It attains a fast data rate up to 275 Mbps in downlink and 75Mbps of uplink speed, with seamless integration of varied networks. Advanced techniques such as, OFDMA, MIMO, SDMA, and sophisticated control and signalling mechanisms to support inter-technology handover and adaptive reverse link interference management makes it likely to become the primary standard adopted for next generation broadband applications. It is an unripe standard, investigated in this paper to prove its competency to support the future broadband technology. The documentation of its development clogged up at 2008, where further literatures related to it were unavailable. The network is developed in this work with its reference architecture, and with the attributes functioned with respect to the existing LTE network elements.

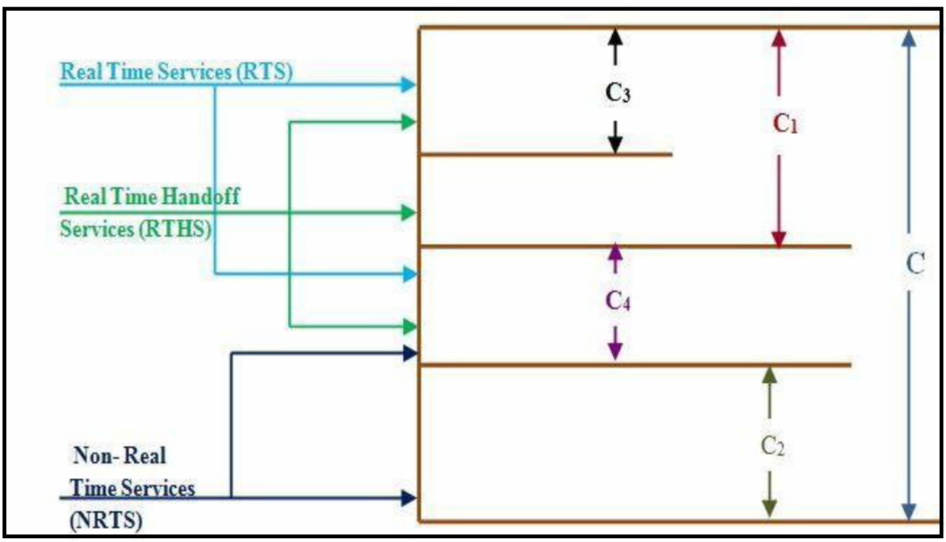
WiMAX, a developed pattern for broadband technology, defines a service flow framework to support multiple QoS classes. It delivers high-speed wireless broadband at a much lower cost (Banerji and Chowdhury, 2013). It operates at 2 to 11 GHz, and below 6 GHz spectrum, for fixed and mobile WiMAX standard respectively. It can provide a theoretical transmission capacity of 75 Mbps in 20 MHz bandwidth with coverage of 40 km, for fixed standard, and up to 30 Mbps per subscriber in a mobile environment. It works with OFDM modulation/demodulation technique at the physical layer and TDMA mechanism at the MAC layer, to provide improved capacity and effective utilisation of the resource, through even distribution of bandwidth among several devices.

WLAN, a standardised technology with various expansions, from IEEE 802.11 ‘a’ to ‘n’, etc. has unique features and functions, with the selection based on the requirements of the user. IEEE 802.11a, in the 5GHz band offers a data rate of 54 Mbps with OFDM physical layer specification. 802.11b can maintain up to 11 Mbps in the 2.4 GHz band. In this interworking architecture of UMB-WiMAX-WLAN, IEEE 802.11 g, standard compatible with both IEEE 802.11 a/b is employed. It offers a data rate of 54 Mbps in 2.4GHz of the band, with the benefit of maximum speed and superior signalling range. The unlicensed frequency spectrum, ease of deployment and data rates had made it the most widely deployed technology through wireless fidelity (Wi-Fi) for wireless internet applications (Banerji and Chowdhury, 2013). The unique characteristics of each network endow the development of the proposed heterogeneous network, to meet the demands of the fast evolving wireless generations. The UMB-WiMAX and the WLAN network are integrated through hybrid coupling, with all the RATs connected together through the core IP and the packets are routed across the RAT with less complex modification of the network architecture. Interconnected with IIP, the service provider is endowed with a horizontal infrastructure, which provides with benefits of services and access independence supporting functions such as authentication, addressing, routing capability negotiation, provisioning, session establishment, etc. The flat architecture provides security and QoS functionality with internet power to the communication world with reduced revenue for all multimedia and interactive applications to make a complete platform to serve next generation network.

4 Dynamic application centric resource provisioning algorithm

The available net bandwidth of the interworking UMB-WiMAX-WLAN network can be efficiently utilised by the DAC-RP algorithm. It is an absolute partition scheme, with the total frequency spectrum of the heterogeneous network is divided into separate pools for each traffic type such as for RT and NRT, and a set of the channel is also explicitly allocated to serve the handoff calls to reduce the dropping probability. In this scheme, the boundary for the partition is designed to be variable and, thus, can effectively deal with the traffic changes in the system. As shown in Figure 2, C is considered to be the total channels available for the interworking architecture. C_1 and C_2, out of C channels are reserved exclusively for new RT and NRT respectively. The remaining number of channels, represented as C_4 ($C-C_1-C_2$) channels are reserved to be shared in a fair manner by both RT and NRT applications, during congestion. As RT ongoing calls have the highest priority, and to reduce the handoff dropping probability, C_3 out of C_1 channels are further restricted, to serve handoff real-time (H-RT) calls. However, the ongoing H-RT and H-NRT calls can also be served by any channel available in C_1 and C_2 respectively or in C_4 region, to decrease the latency further. As the NRT can usually tolerate some degree of service degradation, new NRT calls, and handoff NRT calls are not distinguished in this paper.

Figure 2 Dynamic application centric RRM scheme (see online version for colours)



The dynamic resource allocation scheme can be modelled as a three-dimensional Markov chain. The DAC-RP scheme can be modelled as a three-dimensional Markov chain (López-Benítez and Gozalvez, 2011). Let P_{ijk} be the steady probability with, *i*, *j* and *k* representing the new RTS calls, H-RT and NRT in the interworking system respectively. The steady state property of a call depends on the arrival rate and the rate at which the call is serviced before the call is being dropped. It is assumed that all RT, H-RT, NRT and H-NRT calls in channels experience the same new and handoff call arrival rates. In each cell, the arrivals of new RT calls, new NRT calls, handoff RT calls, and handoff NRT calls are Poisson distributed with arrival rate λ_{rtn}, λ_{nrtn}, λ_{rth}, and λ_{nrth} respectively. Thus, the total RT call arrival rate and NRT call arrival rate are $\lambda_{rt} = \lambda_{rtn} + \lambda_{rth}$ and $\lambda_{nrt} = \lambda_{nrtn} + \lambda_{nrth}$, respectively. Similarly, the service rate of the RT, H-RT, NRT, and H-NRT are defined as μ_{rt}, μ_{hrt}, μ_{nrt}, and μ_{hnrt}, respectively. The steady state probability is defined as the ratio of the arrival rate to the service rate of the call (i.e., $P_{ijk} = \lambda_{ijk}/\mu_{ijk}$). The probabilities for RT, HRT, NRT, and H-NRT are determined from the corresponding arrival rate and the service rate from each application and the overall blocking probability for the RT and NRT service is obtained through the equations (1) and (2) formed below.

The available channels for serving the RT and NRT services are allocated based on the number of RT, NRT and the H-RT calls in the network. The resource provisioning conditions, for allocating the calls to the appropriate available channel is given in Table 1.

Table 1 Channel condition based on resource allocation

Channel condition	*Call acceptance*		
	RTS	*NRTS*	*H-RTS*
$i = j = k = 0$	0	0	0
$0 \le i + j \le C_1$ and $0 \le k< [(C - C_1) / B]$	✓✓	✓✓	✓✓
$0< i + j < C_3$ and $k = [(C - C_1) / B]$	✓✓	×	✓✓
$C_3 \le i + j < C_1$ and $k = [(C - C_1) / B]$	✓	×	✓
$i + j = C_1$ and $k = [(C - C_1) / B]$ or $k < i + j < C - C_2$ and $k = [(C - i - j) / B]$	✓	×	×
If $i + j = C - C_2$ and $k < C_2 / B$	×	✓	×

Note: ✓✓ – call accepted to the allocated channel, ✓ – call accepted (to the shared channel), × – call rejected.

The admission control of DAC-RP Algorithm described as follows. For a new RT request to a network, unoccupied channels in C_3 searched, if there is no available channel there, the free channel in C_4 (shared Channel area) will be searched. Likewise, an NRT call assignment will be first attempted in the C_2 area and if that is not possible, the channels available in C_4 will be examined. When an H-RT call arrives, it searches for available channels in C_1 and in C_4 and, if there is no channel available in both, the call will be dropped. When a new RT call arrives, and if the channel occupancy exceeds the threshold in C_3 and C_4 (i.e., there is no idle channel), the call will be blocked. When an NRT call (new or handoff) arrive, if the number of idle channels in the C_2 or in C_4 area is lesser than B (Bandwidth for NRT), it will be blocked. Based on the conditions explained above in Table 1, the steady state call blocking probabilities, P_{rtb} and P_{nrtb}, for RT and NRT calls are derived respectively as given below

Overall RTS call blocking probability:

$$P_{rtb} = \sum_{C_3 \le i+j \le C_1} P_{ijk} + \sum_{C_1 \le i+j \le C-C_2} P_{ijk} + \sum_{i+j=C-C_2} P_{ijk} \tag{1}$$

Overall NRTS call blocking probability:

$$P_{nrtb} = \sum_{C_3 \le i+j \le C_1} P_{ijk} + \sum_{C_1 \le i+j \le C-C_2} P_{ijk} \tag{2}$$

where

C total number of channels

C_1 channels out of C for new RT and H-RT calls

C_2 channels for NRT calls

C_4 (i.e., $C-C_1-C_2$) for shared channel

C_3 out of C_1 for new RT calls.

5 Simulation performance analysis

The validation of the proposed DAC-RP algorithm and IIP architecture for the interworking UMB-WiMAX-WLAN network, in improving the system capacity by reducing the latency of applications are analysed using the results obtained from OPNET simulation. The enhancement of the IIP architecture could be attained through comparing its results of the interworked network with the existing IMS architecture. Similarly to confirm that UMB could be used as the developing technology to meet the increasing demands of the broadband services; the UMTS-WiMAX-WLAN interworking is also developed with the DAC-RP. Performance metric such as upload and download response time for e-mail, FTP and object and page response time for HTTP are considered for NRT service. And for RT services metrics of voice such as jitter, mean opinion score (MOS), Packet delay variation and packet end to end delay are acquired and compared. The type of service and the number of users considered for each application are specified in Table 2. Based on a number of iterations performed, the suitable number of subscribers served for each network and applications are fixed. The comparisons of following scenarios are discussed in the following part of this section.

1 UMB-WiMAX-WLAN with existing IMS architecture and proposed DAC-RP.

2 UMB-WiMAX-WLAN with proposed IIP architecture and without DAC-RP.

3 UMB-WiMAX-WLAN with proposed IIP and DAC-RP.

4 UMTS-WiMAX-WLAN with proposed DAC-RP.

Table 2 Number of users assigned in each application for simulation

Simulation parameter	*Traffic type*	*Number of users*		
		UMTS-WiMAX-WLAN	*UMB-WiMAX-WLAN-IMS*	*UMB-WiMAX-WLAN-IIP*
VoIP	PCM quality	50	100	150
FTP	Heavy	30	50	90
E-mail	Heavy	30	50	90
HTTP	Heavy	30	50	90
Total number of subscribers		140	250	420
Simulation run time		600 seconds		

5.1 Performance metrics of NRT applications

To verify that the proposed DAC-RP and IIP reduce the latency of the applications, Response time for uploading and downloading an email and File are considered, and the time taken to load the HTTP page is obtained.

The time taken to send the request for an email and receive it from corresponding server, and the time between sending an email to a server and receiving an acknowledgment, defines the email download and upload response time (sec) respectively. The download and upload response time (in seconds) of any application between a client and a server could be expressed respectively as in equations (3) and (4) given below.

$$T-\text{Response}DL_{[s]} = T_{rq[s]} + T_{rr[s]} + T_{sig[s]} + T_{dst[s]} \tag{3}$$

where

T-Response$DL_{[s]}$ is the downlink response time in seconds

$T_{rq[s]}$ is the request delay

$T_{rr[s]}$ is the receive response packet delay

$T_{sig[s]}$ is the signalling delay

$T_{dst[s]}$ is the delay for setup and tear-down.

$$T-\text{Response}UL_{[s]} = T_{sd[s]} + T_{ack[s]} + T_{sig[s]} + T_{dst[s]} \tag{4}$$

where

T_{sd} is the send delay

T_{ack} is the acknowledge delay.

Figure 3 Email application-download response time (sec) (see online version for colours)

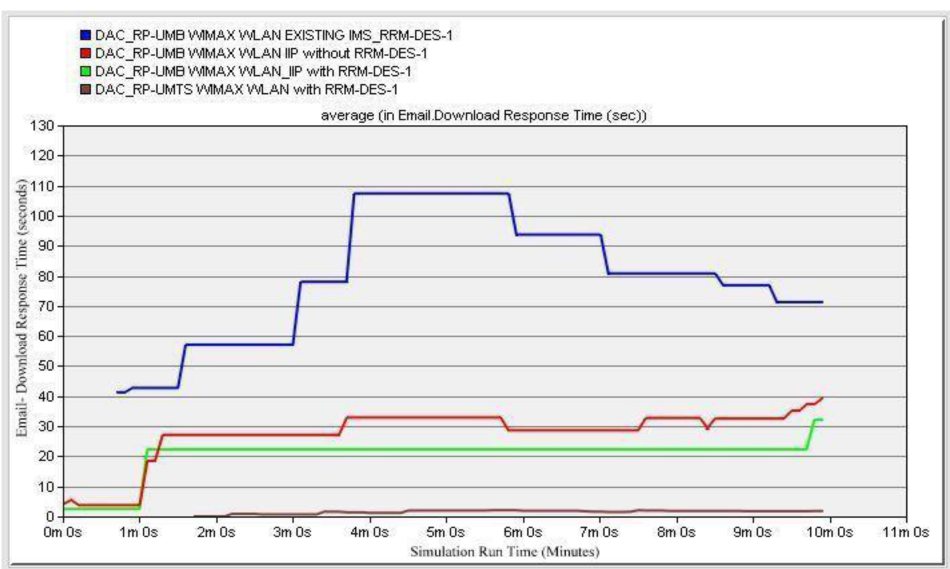
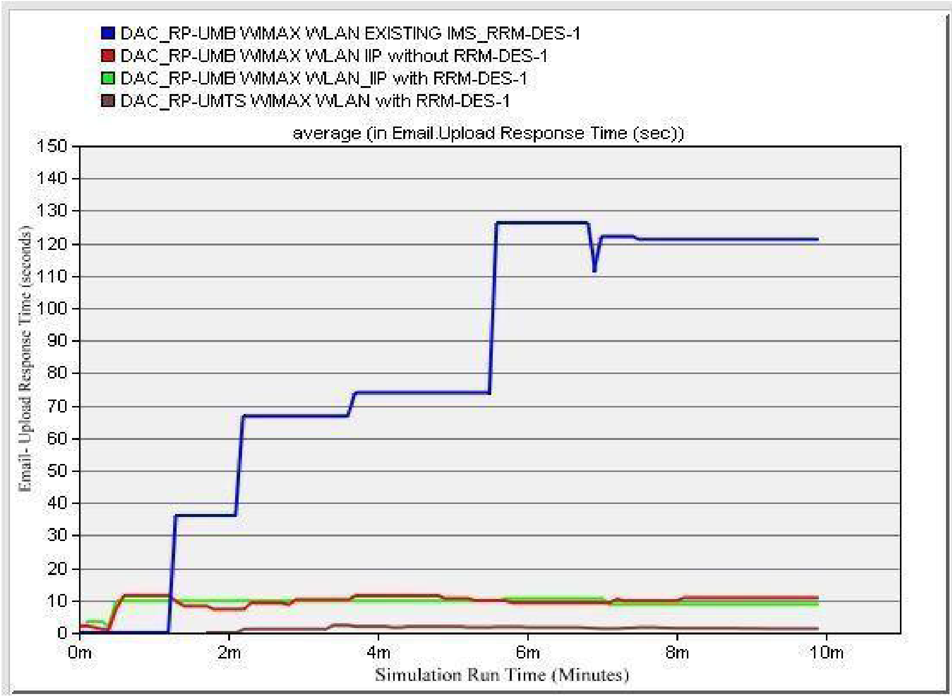


Figure 4 Email application-upload response time (sec) (see online version for colours)



Figures 3 and 4 give the comparison of download and upload response time correspondingly for the scenarios listed above. In both obtained responses, the DAC-RP scheme shows the minimum response time, as the network bandwidth for NRT services utilisation function is dynamically monitored and the network with the maximum capability to serve the application is selected, thus minimising the time involved in processing. Also comparison of network interworking with IMS and IIP, IMS show much higher response time even for 50% lesser Email users, due to increased processing time owing to less interaction between the servers at different layers. The advantage of the IIP and UMB network could be obtained from comparing the interworking network with UMTS and UMB, has the former shows a comparatively increased response time even for 30 users.

FTP download and upload response time for the interworking network is described in Figures 5 and 6 respectively, and they can also possibly be expressed mathematically as in equation (3) and (4). The download response time (sec) is the measured time a user sends a request to the server to the time a response packet is received. The period of time for downloading a file and getting acknowledgment is reduced for the IIP interworking scenario with the DAC-RP scheme is 80 % lesser compared to the network without resource allocation scheme. Between the IMS and IIP architecture, IMS shows a slightly better performance in download response, which depend on the traffic characteristic of the interworking network. As the traffic characteristics of a network erratic with time, the curves will not remain stable, and so the UMTS network shows a larger response time besides of lesser number of FTP services.

Figure 5 FTP application – download response time (sec) (see online version for colours)

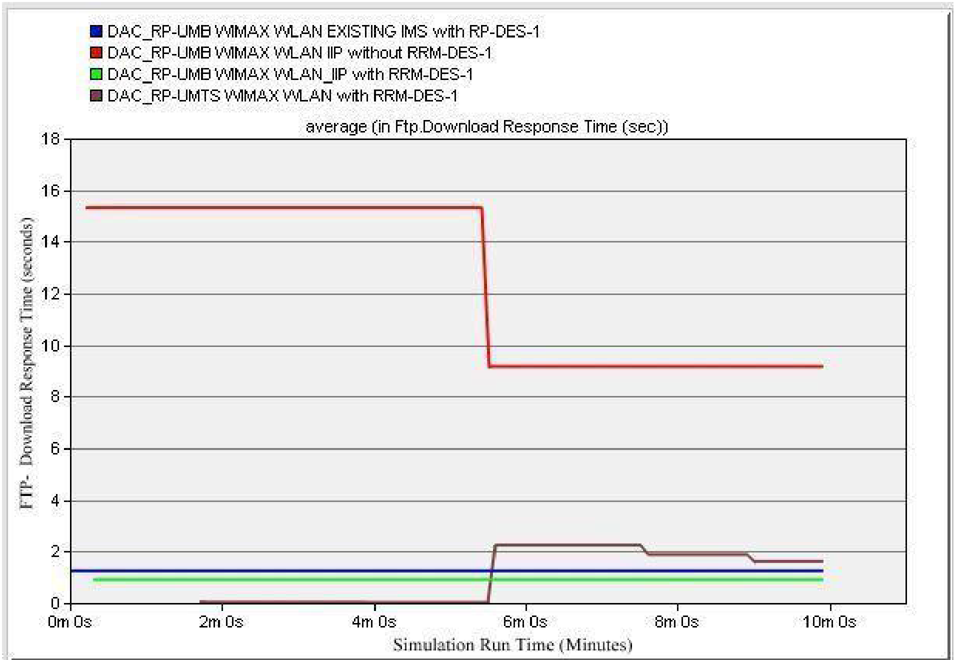
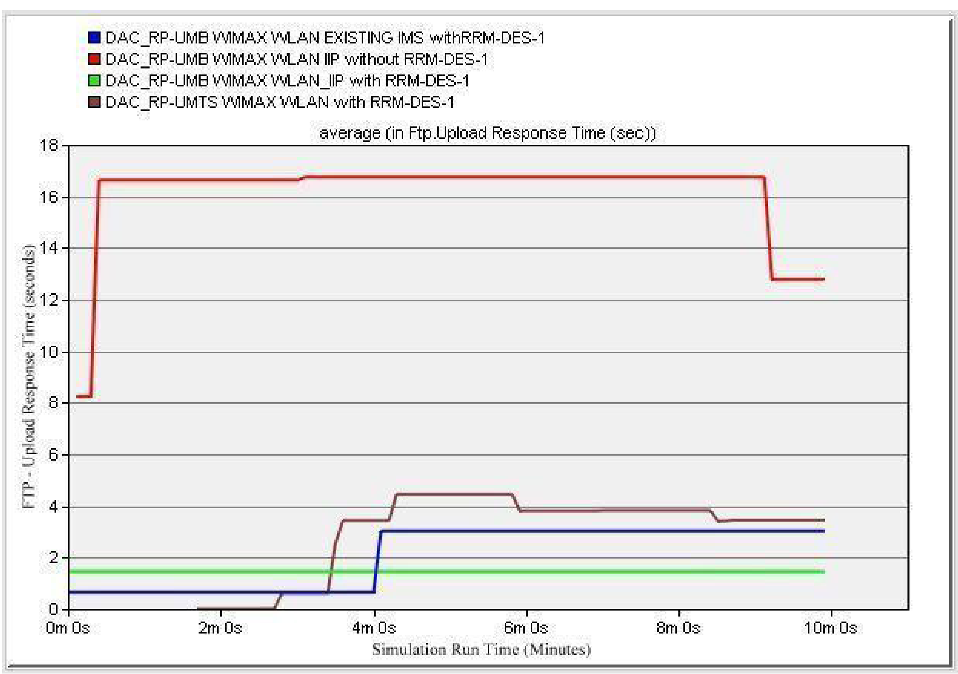


Figure 6 FTP application – upload response time (sec) (see online version for colours)



Likewise, the time taken to uploading a file and getting acknowledgment is the upload response time of FTP. It is lesser for the UMB-WiMAX-WLAN interworking Network, with IIP, compared to UMTS network, due to the flexibility offered by the unified architecture and the interworking operability of UMB network. The HTTP object and page response time defining the time for loading an Inline object of the HTML page and the time for completely retrieve the entire page respectively. The page response time of an HTTP application is the difference between the user request time and the time when the page is retrieved, which can be expressed as in equation (5).

$$T_{pr[s]} = T_{er[s]} - T_{ur[s]} \tag{5}$$

where

$T_{er[s]}$ is the end request instant

$T_{ur[s]}$ is the user request instant.

Figures 7 and 8 represent the above mentioned metrics for the application layer protocol function of HTTP, used to transmit virtually all files and other data on the World Wide Web, in prescribed format. Assigning NRT applications to a separate pool of bandwidth reduce the response time for this stateless application. A number of channels are available for NRT application due the wide flexibility of UMB network and IIP and hence the resource is partitioned among the network hence reduces the time for retrieving the web page, makes the web access faster. Comparing the scenarios, DAC-RP reduces the time for object response and page response with respect to that of the network with UMTS and existing IMS.

Figure 7 HTTP application – object response time (sec) (see online version for colours)

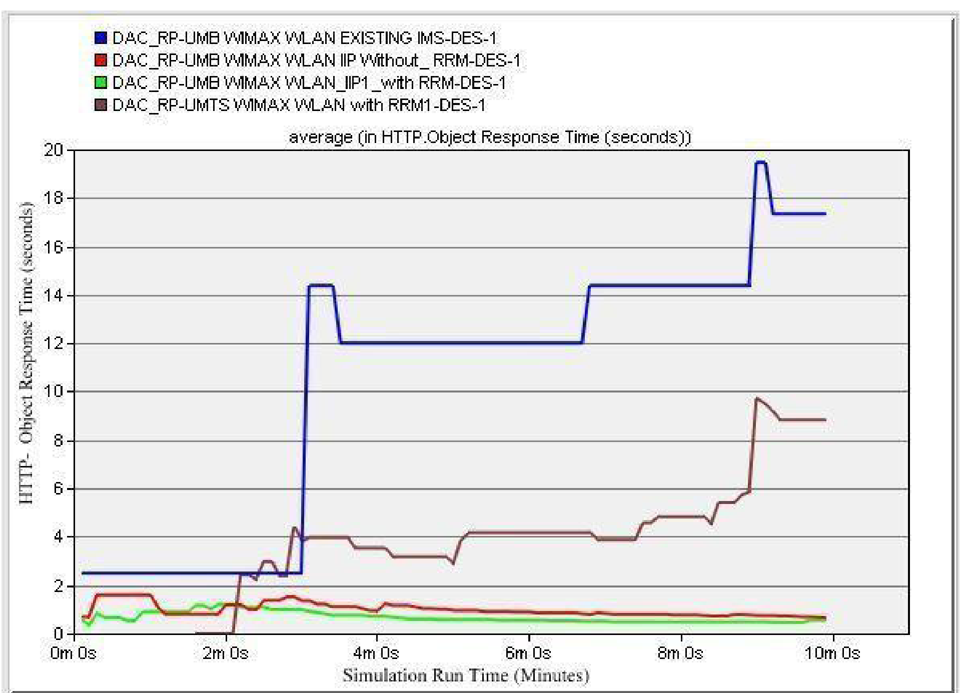
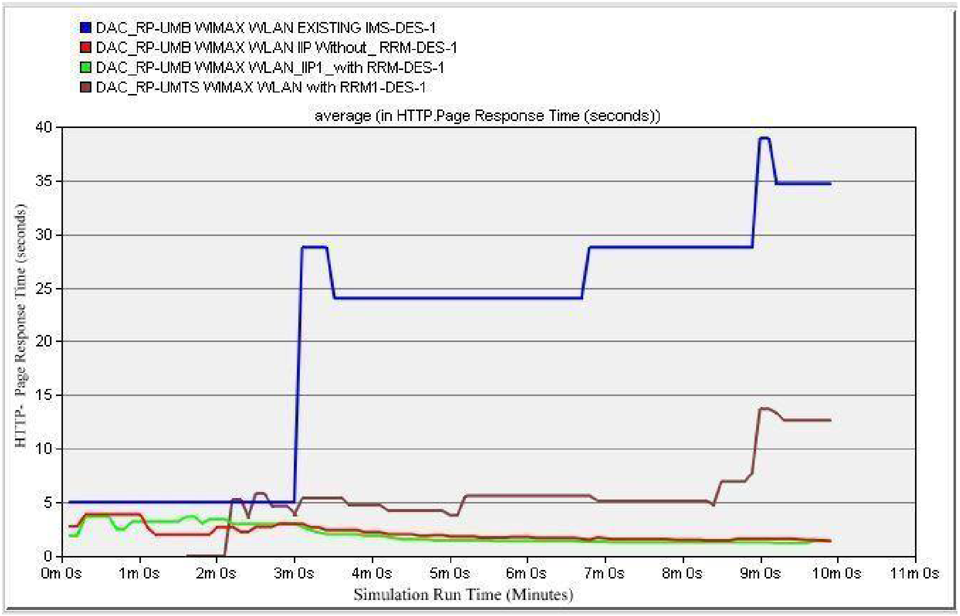


Figure 8 HTTP application – page response time (sec) (see online version for colours)



As a separate set of a dedicated channel are allocated in DAC-RP, an equal radio resource is considered for all applications and it provides an improved performance of the interworking network. And also the unified architecture of IIP, the CSCFs communicate with the ASs, to reduce the latency of data transfer.

5.2 Performance metrics of RT application

Figure 9, is the delayed variation defining the voice jitter. It is the time elapsed between two consecutive packets leaving the source node at different time stamps and the time at which they are played back at the destination. From the results obtained for the existing IMS and interworked network with UMTS, the jitter is much higher compared to the IIP architecture. Owing to the irregular distribution of traffic characteristic of the network, channel allocation criteria, network conditions, the delay in the network can vary, which corresponds to the jitter. A large value of jitter may lead to receiving the packet out of bound of the buffer, which leads to discarding the packets and dropouts. The unified architecture of the IIP with specific AS for supporting the Multimedia applications, and the MMTel, IM-SSF servers designed specifically for media service can handle the continuity of Multimedia packet without time lag, thus reducing the delayed variation time nearly to zero.

Figure 9 Voice application – jitter (sec) (see online version for colours)

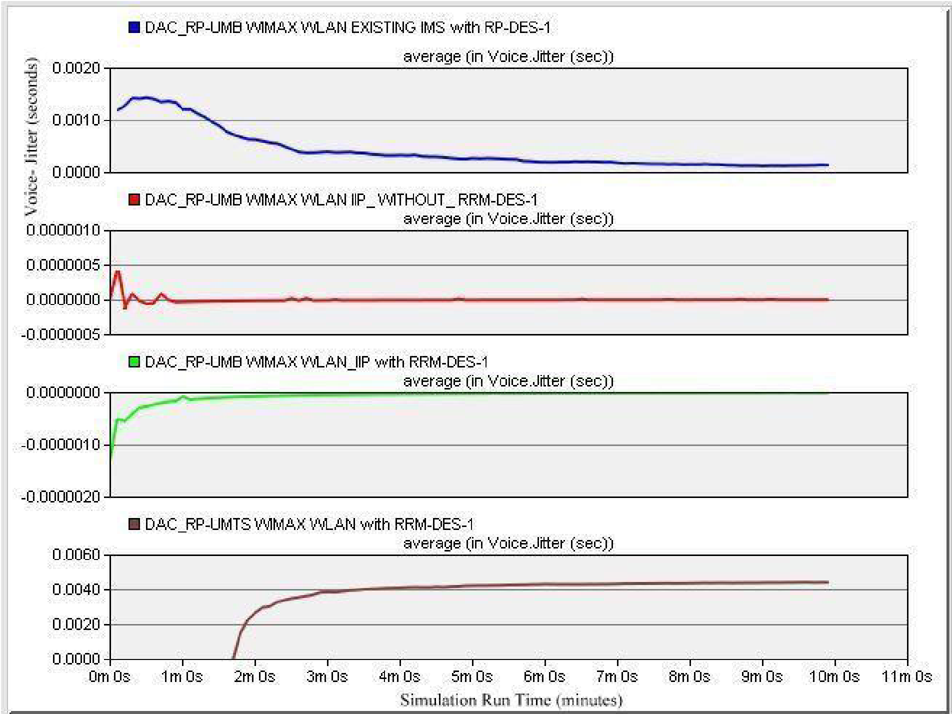
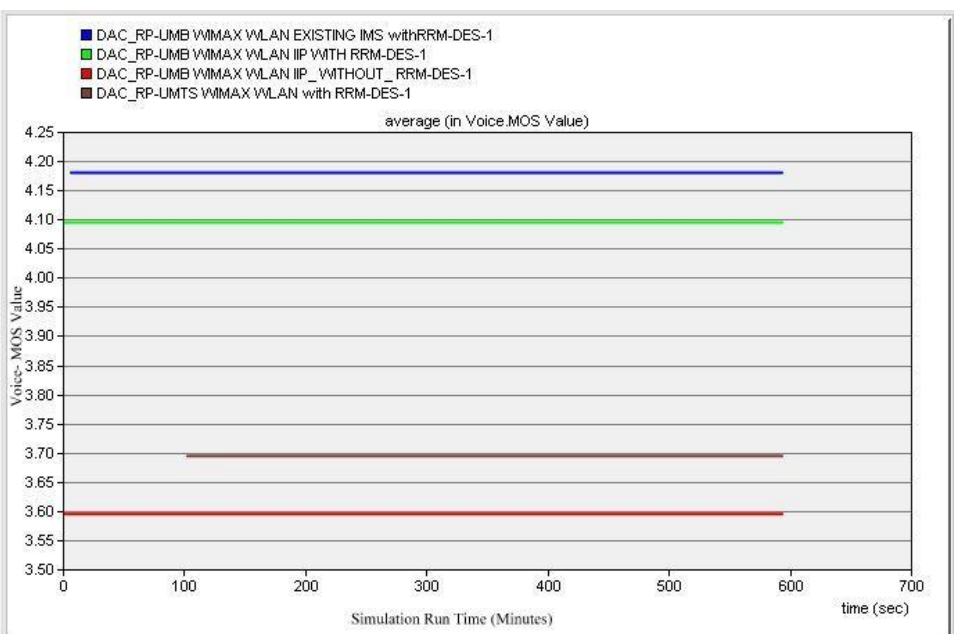


Figure 10 Voice application – MOS value (see online version for colours)



Mean opinion score (MOS) for Voice application; specify the perceived quality of the media received after transmission and eventual processing expressed as in equation (6). The theoretical value of MOS for voice ranges from one to five. Higher the value better is the received media quality. It defines the type of codec used for VoIP signal transmission, the delay and impairment factors of the network. From the Voice quality result obtained in Figure 10, it can be observed that the proposed DAC-RP provide a voice application with better quality. The MOS value for the interworking network with the implementation of RRM for IMS is 4.19 and for IIP, it is 4.1. But for the network without RRM, the quality value is 2.3, representing a medium quality of voice. This is due to that the DAC-RP based on assigning a dedicated channel for RT, thus making it highly prioritised.

$$MOS = B_{SNR} - I_s - I_d - I_{e,eff} + A \tag{6}$$

where

B_{SNR} is the basic signal to noise (SNR) ratio

I_s is the impairments with voice signal

I_d is the impairments due to delay and echo effects

$I_{e,eff}$ is the effective equipment impairment factor

A is the advantage factor (a constant).

Figure 11 Voice application – end to end delay (sec) (see online version for colours)

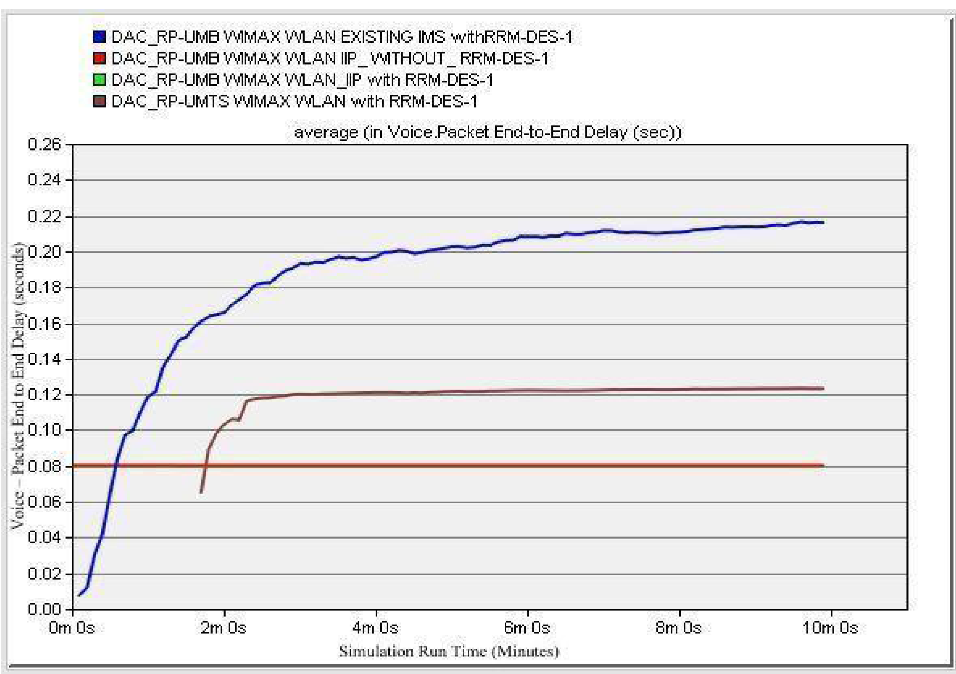
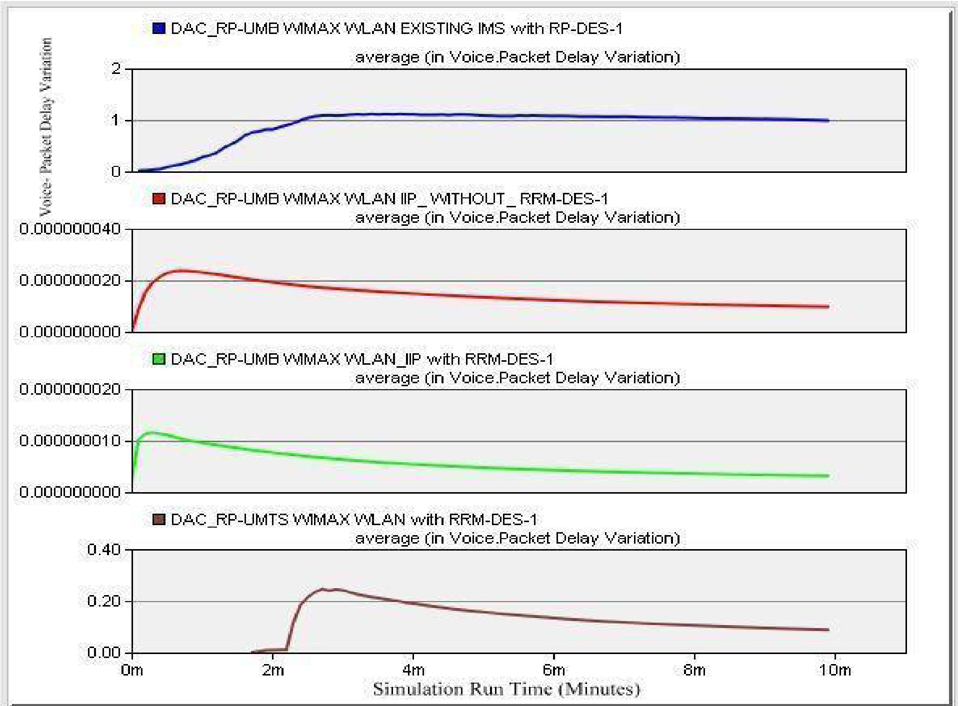


Figure 12 Voice application – packet delay variation (see online version for colours)



Figures 11 and 12 represent the packet end-to-end delay (sec), and packet delay variation for voice application respectively. For a voice call to take place without dropping, the packet end to end delay should be below 100 mSec. Packet end to end delay of voice signal takes into account the total delay involved in the transmission of the voice packet from the server to the client and is expressed as in equation (7).

$$T_{EED[s]} = T_{n[s]} + T_{e[s]} + T_{d[s]} + T_{c[s]} + T_{dc[s]} + T_{dj[s]} \tag{7}$$

where

T_n is the network delay

T_e is the encoder delay

T_d is the decoding delay

T_c is the compression delay

T_{dc} is the decompression delay

T_{dj} is the Dejitter buffer delay.

From the simulation, it could be observed that the delays for all scenarios are less than the prescribed value theoretically. But in view of the IIP architecture, the end to end packet delay is 10% lesser than the IMS network. It can be observed that voice packet delay variation is beside less for the IIP implemented network compared to that with IMS network. As a complete pool of available bandwidth is allocated to the RT services, there is no delay in the transmission and reception of packet using DAC-RP scheme and hence it is minimum compared to the network without Resource provisioning. The fusion of the control and ASs as a single layer in the core of IIP interacts efficiently with the multimedia servers to reduce the time of transition, and thus reduces the end to end delay and packet delay variation of the voice application.

Comparing the performance metrics of the NRT and RT applications obtained for the interworking scenarios mentioned above, the IIP architecture with DAC-RP provides evidence to show its compatibility to support all the multimedia and interactive application, than the IMS. Further it reduces the delay metrics thus reducing the call blocking probability of the network, and increases the capacity of the network. As the channel partition is made dynamic, the dedicated channels could accommodate the changes in the traffic conditions. It was also observed that the response time for NRT and delay of RT services of the IIP is lesser even for increased number of the user than IMS, due to its unified flat architecture. The DAC-RP scheme, to allocate the desired channel for efficient utilisation of radio resource is observed from the result obtained, and it is evident that the response time taken for EMAIL, FTP and HTTP application and the Jitter, end to end delay and packet delay variation for voice application are reduced compared to the scenario without RRM. The comparison of the interworking network with UMTS, substantiate the features of UMB, to provide interoperability of networks and flexibility to support higher data rate. Yet the complexity of the proposed network lies in the performance analysis done for 400 user nodes, and with increased number of servers, resulting in increased time for simulation.

5.3 Mathematical validation

The mathematical analysis of the performance metrics considered in this paper, such as jitter, packet delay variation, packet end to end delay, MOS and response times of the EMAIL, FTP and HTTP does not follow a fixed analysis. The arrival time and execution process are does not follow a patterned distribution mechanism for all users and differs for each and every user. And for an interworking network with 400 users, and with each user assigned with random vector trajectory, obtaining the time factors stating the start and end of execution of application for each user is not fixed; hence the mathematical model forms an out off scope of this work.

Table 3 R factor, MOS and user satisfaction level measure

R factor	*MOS*	*User satisfaction level*
90 < R < 100	4.3 < MOS < 5	Very satisfied
80 < R< 90	4.0 < MOS < 4.2	Satisfied
70 < R < 80	3.6 < MOS < 4.0	Some user not satisfied
60 < R < 70	3.0 < MOS < 3.5	Many users not satisfied
50 < R < 60	2.5 < MOS < 3	No users are satisfied

However to validate the obtained simulation results, the mathematical analysis for MOS is defined in this section. It is the range that rates perceptual voice quality, based on ITU-T recommendations. MOS value typically ranges from 5–1, with five the best and one being the worst. MOS value is obtained by mapping it to the R factor value that ranges from 0 to 100. The relation between R factor and its equivalent MOS is stated in

Table 3. Mathematically R factor and MOS are an equivalent factor which depends on the basic SNR rate, the impairments caused due to voices signal, delay, the network equipments and an advantage factor which defines the tolerance value of the communication system [equation (8)] (Ribadeneira, 2007).

$$R = R_0 - I_s - I_d - I_{e,eff} + A \tag{8}$$

R_0 represents the basic SNR level, which includes the noise due to circuit, environment and subscriber line noise. I_s, is the impairment caused due to the voice signal, contributed due to the loudness rating of the telephone set, number of quantisation distortion units, and side tone loudness unit. I_d is the delay impairment factor, which is a number based on the measured one-way delay. It is contributed by the total delay present in the network and the values of talker, and the listener echo loudness rating. It could be typically considered based on the delay value, such as for a delay of $\geq$ 50 ms and $\leq$100 ms it is 1.5, for $\geq$200ms it is 7.4 and, 31 for delay more than 400 ms. $I_{e,eff}$ is the effective equipment impairment factor, added from the distortions caused due to codec and the end to end packet destructions. Depending on the codec type used, the value ranges from 0 to 70. The advantage factor, A, represents the user tolerance to the degradation of the voice quality. It is a factor having values depending on the type of the communication system such as, wire bound with value zero, cellular mobility in building with five, ten for mobility across geographical area, and the maximum of 20 is assigned to the location hard to reach. For a PCM G.711 Codec with 64 Kbps, the R factor can be taken as a function of the maximum network delay as given in equation (9) (Perlicki, 2002).

$$R_{G.711} = 92.68 - 0.1T_{n,\max} - 15.90 * \delta T_{n,\max} - 164.75 - 22\ln + 0.2pl \tag{9}$$

where $T_{n,\max}$ is the maximum network delay obtained for the network on whole, and pl is the packet loss percentage. Based on these terms that define the R factor of VoIP signal, the mathematical value for the UMB-WiMAX-WLAN_IIP interworking network with RRM, can be obtained for validation of the simulation metrics. The total network end to end delay from Figure 11 states a value of 80 ms for the considered scenario of IIP interworking with proposed DAC-RP. Assuming the packet loss percentage to be negligible (equal to zero), applying the values to the equation (9) above, we can obtain the R factor as, considering the values in seconds, the theoretical R factor will be calculated as

$$\begin{aligned} R &= 92.68 - (-0.0079 * -0.084) - 22\ln(1 + 0.2 * 0) \\ &= 92.68 - 0.0006 - 0 \\ &= 92.679 \\ &\cong 93 \end{aligned}$$

By mapping the calculated R factor to Table 3, the MOS value can be analyses to lie in the range of 4.3 to 5, providing the uses with high satisfaction. But comparing the manipulated value with the simulation, the scenario with IIP and proposed DAC-RP, shows an MOS value of 4.10 (from Figure 10), which is due to the RT exponential simulation time assigned for the start and end of VOIP application.

6 Conclusions

An efficient RRM mechanism is the vital solution to proficiently utilise the available radio resource of a heterogeneous interworking network. In this paper, a DAC-RP scheme has been implemented in a novel UMB-WIMAX-WLAN hybrid interworking network over unified IIP architecture providing users an enhanced QoS and experience over varied applications. IIP designed to offer flawless service to users is validated through comparing the results of scenarios interconnected with IMS and IIP. The IIP with flat horizontal architecture enhanced the function of the existing IMS, through direct interaction between servers of different layer. Each added AS performs its own functions and communicates with the call control layer elements right away via direct SIP signalling, depending upon the service invoked by the end user, thus reducing the latency. The DAC-RP algorithm based on a complete partitioning of available resource to RT and NRT application, make users available with always best connected radio resource by a complete dedicated group of channels. Its performance is endorsed through obtaining the response time for e-mail, FTP and HTTP applications and jitter, MOS, Packet delay variation and end-to-end delay for the voice application. The promoted features of the IIP architecture to reduce the signalling message between servers minimise the response time and delay metrics of the involved applications. The enhanced ability of the UMB network to serve the increasing demands of the future broadband technologies is validated by comparing with scenario simulated for an UMTS-WiMAX-WLAN interworking network. The obtained results validate the effectiveness of the hybrid coupled UMB-WiMAX-WLAN interworking network with the IIP and DAC-RP scheme to outcome the scenarios with IMS; and UMTS network, by reducing the response times and delay metrics of the defined user applications. It provides the users with the access to common applications across the divergent interworking network and reduces the signalling overheads through a fused level of interaction. The vulnerability of the UMB-WiMAX-WLAN interworking network with IIP architecture and DAC-RP is well demonstrated by the support offered by it to serve a number of multimedia and interactive applications compared to the other architectures.

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