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# Performance Evaluation of Video, Voice and Web Traffic Over Heterogeneous Access Networks

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**Abstract** By simulation using NS-3 we evaluated the performance of voice, video and web traffic sharing a wireless access network connected to a wired core. We compared the performance in terms of end-to-end delay, end-to-end delay variation, average throughput and loss percentage. For the wireless access network, we considered cases when it consisted of a single technology type, e.g., WiFi (IEEE 802.11), WiMAX (IEEE 802.16) and LTE, and when it was heterogeneous, i.e., when the three technologies coexisted and simultaneously shared the same IP core. We attempted to ascertain the impact of this type of heterogeneity on end-to-end performance. It was found that this heterogeneity in the wireless access portion of the network can improve, degrade or have no impact on application performance depending on the network conditions and the application itself. Some key research challenges in Fifth Generation wireless communications are heterogeneous Cloud Radio Access Networks (HC-RANS), backward compatiblity with 4G/3G networks and providing low-latency and QoE. To achieve end-to-end QoS guarantees in such settings the interface with the core must also be addressed, especially when backward compatibility is to be assured. This simulation study attempts to highlight the impact of this type of heterogeneity on network performance.

**Keywords** Differentiated services · Quality-of-service · Performance evaluation · WiMAX · LTE · WiFi

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## 1 Introduction

The provisioning of quality-of-service (QoS) guarantees in telecommunication networks is becoming more and more crucial as the traffic flowing through such networks comprise not only data (as in earlier times) but also delay-sensitive voice and video. In fact, end-users may generate all three traffic types at the same time while engaged with the network. Multimedia traffic has become mainstream with a growing level of intolerance, among end-users, for poor network performance. Additionally end-users can access the network through a variety of technologies—both wired and wireless.

This challenge is exacerbated by the fact that this QoS for multimedia applications may have to be guaranteed across heterogeneous networks. To this end, substantial research has been conducted on the provisioning of end-to-end QoS differentiated services across heterogeneous networks that consist of wireless technologies in the access portion (which may have a wired or wireless backhaul) and a wired core.

To achieve QoS within the wireless access portion itself is a more challenging task than in the wired core. This is due to the nature of the medium (limited bandwidth, random fading, interference) and, there is also the prospect of mobility. (See [1].) A number of QoS mechanisms for the wireless domain have been explored. These include packet scheduling techniques on the air interface in both uplink and downlink direction, connection admission control, transport layer performance improvement against wireless losses, dynamic frequency channel allocation for spectrum management, logical link layer (LLC) techniques, load-aware multipath routing, video source coding, complementary MAC protocols, policy-based management solutions.

Contemporary wireless technologies such as LTE, WiMAX (IEEE 802.16) and WiFi (IEEE 802.11) have employed a subset of these techniques plus other advanced ones to provide robust service differentiation.

In [2], the authors made the observation that although access networks support QoS by service differentiation by the MAC layer or by radio bearers, each system has a different approach to end-to-end QoS and that interworking between the various QoS systems is not standardized. This was also recognized by Malila and Ventura [3]. According to the authors, the main challenge stems from the fact that different technologies (both in the access and the core) employ different QoS strategies and that although there are QoS standards for a given technology, there is no standard solution for end-to-end QoS provisioning in networks which consist of multiple technologies. This has implications with respect to service-level agreements since it would be difficult to provide end-to-end QoS guarantees.

In this paper we seek to address this by analyzing how end-to-end performance is affected when different wireless technologies (more specifically WiFi, WiMAX and LTE) coexist or share the common core. Our focus in this paper is not on mobility management, but rather on the co-existence of multiple wireless technologies subtended by the same core. Using NS-3, we evaluate the performance of voice, video and web traffic in terms of end-to-end delay, delay variation, throughput and packet-loss across such a heterogeneous network.

We present in Sect. 2 a brief review of the literature and in Sect. 3 some background on how QoS is enabled in LTE, WiMAX and WiFi. In Sect. 4, we describe the simulation methodology. The results and discussion are presented in Sects. 5 and 6, respectively. We conclude in Sect. 6.

## 2 Review of the Literature

Research with regards to the performance evaluation of wired-cum-wireless heterogeneous networks has been pursued along a number of intersecting dimensions. These include:

(1) The type, number and combination of wireless access networks interworking with the wired core, e.g., IEEE 802.11 (WLAN), IEEE 802.16 (WiMAX), LTE, GPRS, UMTS, HSPA, CDMA, Satellite, WSN, MANET
(2) The type and combination of application traffic being analyzed, e.g., voice (VoIP), video, TCP flows such as web traffic.
(3) The layers at which service differentiation was implemented: MAC, LLC, IP, transport and/or application, and whether such differentiation follows a decoupled or coupled approach. Examples of such works include [4, 5]
(4) The performance measures being analyzed, e.g., network QoS metrics such as delay, jitter, packet loss or end-user quality measures, i.e., QoE, such as MOS. [4]
(5) The presence or absence of architectures or techniques that enabled differentiated services within the core
(6) The role of signaling [6]
(7) The context or application space for such networks, e.g., military/tactical communications, disaster management [7, 8]
(8) The impact of channel effects peculiar to wireless systems such as fast and slow fading, error control and mobile
(9) The implications for mobility management in terms of handoffs (vertical or otherwise), call-establishment [2, 9–12]

There were simulation studies, performance evaluations using test-beds and theoretical analysis. With regards to the first dimension, i.e., type and number of wireless access networks, WLANs (mainly IEEE802.11 networks) were investigated in [1, 2, 7, 8, 11–21]. In particular, WLAN technology was investigated in conjunction with GSM/GPRS/UMTS in [2, 9, 10, 21] , in conjunction with WiMAX in [16], together with LTE in [11, 12], in ad hoc mode [1, 7, 17, 18]. Heterogeneous networks consisting of UMTS access and a wired core were studied in [5, 22]. WiMAX heterogeneous networks were studied in [3, 4]. The combination of Satellite, 3G Cellular, WiMAX and WLAN interfacing with a wired core was investigated in [20], and the combination TETRA, GSM/GPRS/UMTS cellular, WiFi and WiMAX I in [8]. However, in these last two, no numerical network performance analysis was performed.

Concerning the second dimension, video traffic was treated with in [3, 9, 15, 22], voice traffic in [1, 2, 10, 13, 17, 18]. For the most part these traffic streams were analyzed in the presence of competing flows. The work of [14, 16, 19] looked at voice and video traffic being transported in the same network. The work of [16] examined the network performance with video conferencing, voice, FTP and web-browsing. Skype, in particular, was examined in [4].

In [20], the authors described approaches to interworking satellite, third generation (3G) cellular, WiMAX and WLAN technologies for the purpose of providing fuller coverage and mobility management. The main thrust of the paper was to derive analytical cost models for both signaling and data traffic across the access networks integrated with the IP Multimedia Subsystem (IMS) networks. The authors did not investigate actual network performance in terms of, for example, end-to-end delay, packet loss and jitter.

The authors of [10] conducted an evaluation of Radio Access Technologies (RAT) selection algorithms using an all-IP heterogeneous wireless real-time testbed which consisted of three different Radio Access Networks (RANs): UTRAN, GERAN and WLAN. The core network was based on DiffServ technology with MPLS. Even though addressed in this paper was the issue of multiple access technologies sharing a DiffServ core, the focus was the effectiveness of the RAT selection algorithms rather than on the impact of access-heterogeneity itself on end-to-end performance.

In [12], the authors outlined three architectural approaches to integrating WiFi and LTE: (1) application layer integration for client-server communication, (2) core-network-based integration which can take into account radio link conditions as perceived by the end-user but not as a whole since it would require a heavy signaling burden, and (3) RAN-based integration which can enable multi-RAT cooperation. However, their focus was heterogeneity in the radio access network. In other words there is interworking among the different radio access technologies (e.g., WiFi and LTE) within the radio access network itself rather than at the core. This integration begins at the level of the end-user device (UE) which is outfitted with multiple interfaces to the radio access technologies. So that not only will WIFI-only nodes be able to access cellular services seamlessly from the LTE network, but even allow end-user devices to transmit data on both the LTE and WiFi interfaces at the same time. (See also [11].)

The aforementioned paper dealt with radio-access integration and the issue of heterogeneity at that level. However, in this paper, we seek to investigate core-network integration and the impact this has on the end-to-end performance of multimedia traffic when multiple access technologies (such as IEEE 802.11, LTE and WiMAX) share a common edge node at the boundary of the core network.

## 3 Background

In this section, we provide a brief summary of how LTE, WiMAX and WiFi provide service differentiation to traffic flows.

### 3.1 QoS in LTE

The EPS bearer that is first established when the UE connects to the network is referred to as the “Default” bearer. EPS bearers established thereafter are called “Dedicated” bearers. The default bearer would be Non-GBR (Non Guaranteed Bit Rate) and would have an associated QoS Class Identifier (e.g., nine (9) for TCP-based services). An Uplink Traffic Flow Template (UL-TFT) in the UE would map a Service Data Flow (SDF) to the EPS bearer established in the uplink direction. A Downlink Traffic Flow Template (DL-TFT) in the PGW would map an SDF to an EPS bearer in the downlink direction. The SDF is the actual connection for data transfer. Each SDF can be uniquely identified by the EPS bearer. Multiple SDFs can be mapped to the same EPS bearer and as a result will receive the same packet-forwarding treatment (QoS), e.g., scheduling policy, queue management policy.

Each EPS bearer has an associated QoS profile which includes the following parameters: QoS Class Identifier (QCI), Allocation Retention Priority (ARP),Maximum Bit Rate (MBR),Guaranteed Bit Rate (GBR). The QCI serves as a label for each class of service or packet forwarding treatment offered by the LTE network. The ARP determines whether a request for a bearer to be established or modified can be accommodated or rejected given resource constraints. It also is used by the network to determine which bearers would actually be dropped if resources should become limited (e.g., in a handoff procedure).

An EPS bearer can be either Guaranteed Bit Rate (GBR) or Non-GBR. For the GBR bearer, the MBR and GBR parameters are specified. The MBR is the maximum bit rate at which the GBR bearer can transmit. If it goes beyond this value, the traffic would be discarded. The GBR is the expected bit-rate for data transmission on the GBR bearer. The MBR and GBR parameters are not specified for Non-GBR bearers.

### 3.2 QoS in WiMAX

The common part sublayer of the WiMAX Medium Access Control (MAC) layer provides a connection-oriented service to the layers above (rather than a contention-based MAC as in WiFi). It establishes and maintains connections and assigns the requisite bandwidth on the uplink and downlink. Therefore, before being transported across the air interface, all user traffic must first be associated with an established connection [identified by a connection identifier (CID)]. A single CID can be used to transport an individual application traffic or traffic from a group of applications. There is a one-to-one mapping between each individual connection and an active service flow [identified by a service flow identifier (SFID)] specification which delineates the QoS requirements for the traffic (e.g., traffic priority, minimum reserved rate, maximum sustained rate, maximum traffic burst, latency, tolerated jitter, packet loss ratio, scheduling service etc.) in a QoS parameter set.

A service flow defines the QoS requirements for a unidirectional flow of traffic. Therefore there is one for the uplink and another for the downlink. Each service flow (and hence each connection) is associated with a particular scheduling service (defined in the service flow specification). The five main options for scheduling service are: Unsolicited Grant Service (UGS), Real-Time Polling Service (rtPS), Enhanced Real-Time Polling Service (ertPS), Non-Real-Time Polling Service (nrtPS) and Best-Effort (BE). These are also referred to loosely as service classes.

UGS is designed primarily for real-time, constant-bit-rate sources (fixed-size packets generated on a periodic basis), e.g., VoIP, E1. Real-Time Polling Service is designed for applications such as streaming-audio/video where variable-sized packets are generated at periodic intervals. Enhanced rtPS (ertPS) is designed for applications such as VOIP with activity detection—variable rate but periodic; nrtPS for delay-tolerant, jitter-insensitive data traffic with variable-sized data packets such as FTP and HTTP. Best Effort (BE) facilitates traffic that requires no minimum QoS guarantees of delay and throughput, and which is satisfied with whatever resources are available. Therefore, depending on the specified scheduling service, packets belonging to a particular connection (CID) are forwarded to the appropriate queue. These queues reside both at the Base Station (BS) and the Subscriber Station (SS) to facilitate both downlink and uplink scheduling respectively. However, the BS has both the downlink scheduler and the uplink scheduler. The BS controls the allocation of resources both on the downlink and on the uplink. Upon receiving bandwidth requests from the various connections (of the SSs in the network) on the uplink, the BS uses the uplink scheduler and allocates uplink resources via UL grants to the various SSs, and notifies them of their allocation on the downlink in UL-MAP messages.

### 3.3 QoS in WiFi

The MAC sublayer of the IEEE 802.11 standard defines two medium access coordination functions: distributed coordination function (DCF) which is mandatory and the point coordination function (PCF) which is optional. The DCF is asynchronous and contention-based and is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), whereas the PCF is synchronous and contention-free and uses a centralized polling-based access approach. DCF, on its own, can provide no service differentiation and therefore supports only best-effort service. The PCF function itself, though intended to provide support to realtime data streams, was found to be insufficient for QoS differentiation. To improve QoS support in WiFi, the following features were introduced via the IEEE 802.11e specification: HCF (Hybrid Coordination Function), EDCA (Enhanced DCF Channel Access-prioritized QoS), HCCA (HCF Controlled Channel Access-prioritized QoS plus a contention free period).

The EDCA forms part of the Hybrid Coordination Function of IEEE 802.11e, and allows for up to eight different levels of priority. Defined in IEEE 802.11e are four (4) Access Categories (AC)—voice (VO) (priorities 6 and 7), video (VI) (priorities 4 and 5), best effort (BE) (priorities 1 and 2), and background (BK) (priorities 0 and 3). To each AC is an affiliated queue to which frames of the different traffic streams are forwarded/mapped. Each AC acts as a single enhanced DCF entity with its own EDCA parameters: Arbitration Interframe Space (AIFS(AC)),minimum Contention Window $CW_{\min}(AC)$, $CW_{\max}(AC)$ and backoff counter . The smaller AIFS(AC), the higher will be the medium access priority. For each AC there is also defined a transmission opportunity (TXOP(AC)) which is an interval of time the AC can transmit without having to recontend for access. Therefore, multiple MAC PDUs may be transmitted within this interval.

# 4 Simulation

## 4.1 Simulation Topology

The network simulator used for these simulations was NS-3 [23]. The network topology chosen to conduct the performance evaluation consisted of three access networks (each having one access node and a gateway node) sharing a common core that consisted of four routers (two edge nodes that flank two directly connected core nodes in a dumbbell configuration). This is shown in Fig. 1.

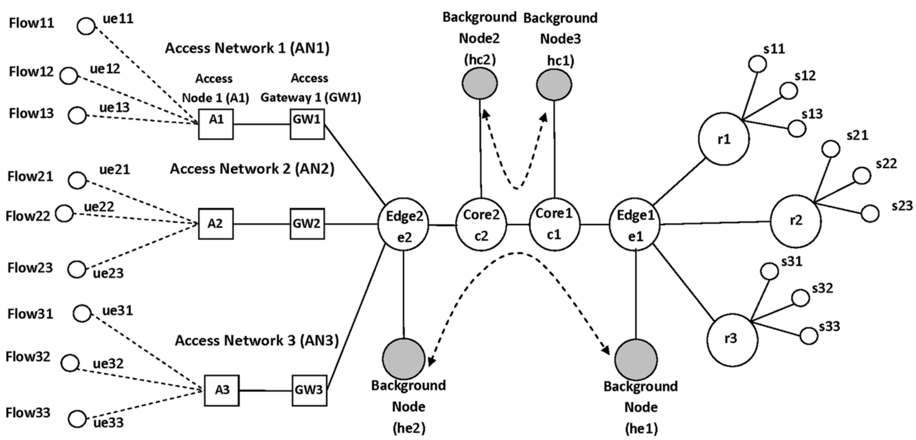



**Fig. 1** Network topology for simulations

Two sets of experiments were conducted. The first set examined the interactions of WiFi, WiMAX and LTE access networks. The second set explored the interactions among Mobile Adhoc WiFi (referred to as MANET), Infrastructure WiFi (referred to as "Infrastructure") and LTE.

Associated with the $i$th Access Network (ANi) was an access node (Ai) and a gateway node (GWi) where $i = 1, 2, 3$. Their names differed depending on the technology employed. For LTE, the access node would be the eNodeB (enB) and the gateway node would be the packet gateway (SGW/PGW). For WiMAX, the access node would be the base station (BS) and the gateway node, the ASN Gateway (ASN GW). For WiFi, the access node would be the access point (AP) and the gateway node an ordinary router.

Three flows originated from mobile stations (UEs) on each of these access networks and terminated on a remote host. So that in Fig. 1, Flow$_{ij}$ flowed from ue$_{ij}$ to s$_{ij}$ where $i = 1, 2, 3$ and $j = 1, 2, 3$.

For the first set of experiments there were 24 scenarios based on the access technologies (e.g., WiFi only, WiMAX only, LTE only, a mixture of WiFi, WiMAX and LTE, or a mixture of WiFi and LTE) and on the traffic mix (i.e., voice only, video only, web only, or all media). These scenarios are listed in Table 1. The simulation duration was 1800 s, with the first millisecond truncated to remove transients. All applications were started at 0 s.

**Table 1** List of scenarios for Experiment Set 1

| Scenario | Label | AN1 | AN2 | AN3 | Traffic mix |
|---|---|---|---|---|---|
| 1 | wifi | WiFi | WiFi | WiFi | Voice only |
| 2 | wifi | WiFi | WiFi | WiFi | Video only |
| 3 | wifi | WiFi | WiFi | WiFi | Web only |
| 4 | wifi | WiFi | WiFi | WiFi | All media |
| 5 | wimax | WiMAX | WiMAX | WiMAX | Voice only |
| 6 | wimax | WiMAX | WiMAX | WiMAX | Video only |
| 7 | wimax | WiMAX | WiMAX | WiMAX | Web only |
| 8 | wimax | WiMAX | WiMAX | WiMAX | All media |
| 9 | lte | LTE | LTE | LTE | Voice only |
| 10 | lte | LTE | LTE | LTE | Video only |
| 11 | lte | LTE | LTE | LTE | Web only |
| 12 | lte | LTE | LTE | LTE | All media |
| 13 | mixture | WiFi | WiMAX | LTE | Voice only |
| 14 | mixture | WiFi | WiMAX | LTE | Video only |
| 15 | mixture | WiFi | WiMAX | LTE | Web only |
| 16 | mixture | WiFi | WiMAX | LTE | All media |
| 17 | wifi2lte1 | WiFi | WiFi | LTE | Voice only |
| 18 | wifi2lte1 | WiFi | WiFi | LTE | Video only |
| 19 | wifi2lte1 | WiFi | WiFi | LTE | Web only |
| 20 | wifi2lte1 | WiFi | WiFi | LTE | All media |
| 21 | wifi1lte2 | WiFi | LTE | LTE | Voice only |
| 22 | wifi1lte2 | WiFi | LTE | LTE | Video only |
| 23 | wifi1lte2 | WiFi | LTE | LTE | Web only |
| 24 | wifi1lte2 | WiFi | LTE | LTE | All media |

For the second set of experiments there were 40 scenarios based on the combination of MANET, Infrastructure WiFi and LTE, and on the traffic mix (i.e., voice only, video only, web only, or all media). These scenarios are listed in Table 2. The simulation duration was also 1800 s. However, the applications that generated traffic were all started according to uniform random variable between 0 and 15 s. Measurements were started at 0 s.

For both sets of experiments, all link bandwidths were 100 Mbps and had a propagation delay of 2 ms except for the following cases:

- The bottleneck link between the core nodes, c1 and c2, was 1 Mbps
- The point-to-point link between the AP and the gateway node of the WiFi access network was 5 Mbps
- The bandwidth of the CSMA network connecting the base station with the ASN gateway in the WiMAX network was 10 Mbps
- The EPC data rate (i.e., between the eNodeB and the PGW node) in LTE was 10 Gbps with 0 ms of delay.

For the second set of experiments, the MANET parameters were as follows:

- There were ten nodes in the access network, although traffic originated from only three of them.
- Of the ten nodes, the AP was stationary, while the others were mobile with a node speed sampled from a Uniform random variable between 0 and 10m/s with zero pause time.
- The nodes were uniformly spread over a 20 m $\times$ 40 m area.

The default buffer-limit for all queues in the network was 100 packets.

### 4.2 Traffic Models

The voice traffic was modelled as an ON-OFF application with the ON-period being exponentially distributed with average duration of 1.5004 s and the OFF-period being exponentially distributed with average duration of 1.032 s. (See ITU-T P.59). During the on-period the data rate was 64 kbps, the packet size was 512 bytes, and UDP transport was employed.

The UdpTraceClient and corresponding UdpServer classes of NS-3 was used to realize the video traffic. The video trace file used was the Mr. Bean video (high quality MPEG 4) which can be retrieved from [24]. The maximum packet size was 1024 bytes.

The HTTP model [25] created by the Information and Telecommunication Technology Center (ITTC), the University of Kansas, was used to generate web traffic. The default parameter values were used. The connections were taken to be persistent.

The QoS class assignments as per the access technologies for each of these traffic types are shown in Table 3.

### 4.3 Parameters of the Access Technologies

For WiFi in the first set of experiments, the NS-3 default configurations were used, for example, Constant Speed propagation delay model, the Log-Distance propagation loss model with path-loss exponent equal to three, the NIST error-rate model, the AARF rate control algorithm. Active probing and QoS support were enabled. The Constant-Position mobility model was used.

**Table 2** List of scenarios for Experiment Set 2

| Scenario | Label | AN1 | AN2 | AN3 | Traffic mix |
|---|---|---|---|---|---|
| 1 | manet | MANET | MANET | MANET | Voice only |
| 2 | manet | MANET | MANET | MANET | Video only |
| 3 | manet | MANET | MANET | MANET | Web only |
| 4 | manet | MANET | MANET | MANET | All media |
| 5 | manet2infra | MANET | MANET | Infrastructure | Voice only |
| 6 | manet2infra | MANET | MANET | Infrastructure | Video only |
| 7 | manet2infra | MANET | MANET | Infrastructure | Web only |
| 8 | manet2infra | MANET | MANET | Infrastructure | All media |
| 9 | manet1infra2 | MANET | Infrastructure | Infrastructure | Voice only |
| 10 | manet1infra2 | MANET | Infrastructure | Infrastructure | Video only |
| 11 | manet1infra2 | MANET | Infrastructure | Infrastructure | Web only |
| 12 | manet1infra2 | MANET | Infrastructure | Infrastructure | All media |
| 13 | infra | Infrastructure | Infrastructure | Infrastructure | Voice only |
| 14 | infra | Infrastructure | Infrastructure | Infrastructure | Video only |
| 15 | infra | Infrastructure | Infrastructure | Infrastructure | Web only |
| 16 | infra | Infrastructure | Infrastructure | Infrastructure | All media |
| 17 | infra2lte1 | Infrastructure | Infrastructure | LTE | Voice only |
| 18 | infra2lte1 | Infrastructure | Infrastructure | LTE | Video only |
| 19 | infra2lte1 | Infrastructure | Infrastructure | LTE | Web only |
| 20 | infra2lte1 | Infrastructure | Infrastructure | LTE | All media |
| 21 | infra1lte2 | Infrastructure | LTE | LTE | Voice only |
| 22 | infra1lte2 | Infrastructure | LTE | LTE | Video only |
| 23 | infra1lte2 | Infrastructure | LTE | LTE | Web only |
| 24 | infra1lte2 | Infrastructure | LTE | LTE | All media |
| 25 | lteonly | LTE | LTE | LTE | Voice only |
| 26 | lteonly | LTE | LTE | LTE | Video only |
| 27 | lteonly | LTE | LTE | LTE | Web only |
| 28 | lteonly | LTE | LTE | LTE | All media |
| 29 | manet1lte2 | MANET | LTE | LTE | Voice only |
| 30 | manet1lte2 | MANET | LTE | LTE | Video only |
| 31 | manet1lte2 | MANET | LTE | LTE | Web only |
| 32 | manet1lte2 | MANET | LTE | LTE | All media |
| 33 | manet2lte1 | MANET | MANET | LTE | Voice only |
| 34 | manet2lte1 | MANET | MANET | LTE | Video only |
| 35 | manet2lte1 | MANET | MANET | LTE | Web only |
| 36 | manet2lte1 | MANET | MANET | LTE | All media |
| 37 | manetinfralte | MANET | Infrastructure | LTE | Voice only |
| 38 | manetinfralte | MANET | Infrastructure | LTE | Video only |
| 39 | manetinfralte | MANET | Infrastructure | LTE | Web only |
| 40 | manetinfralte | MANET | Infrastructure | LTE | All media |

**Table 3** QoS assignments of traffic across access technologies

| Traffic type | WiFi access class | WiMAX service flow type | LTE EPS bearer |
|---|---|---|---|
| Voice | 6 | UGS | GBR_CONV_VOICE |
| Video | 5 | RTPS | GBR_NON_CONV_VIDEO |
| Web | 1 | UGS | NGBR_VIDEO_TCP_DEFAULT |

The Log-Distance propagation loss model was used in the WiMAX access network. The physical layer model was the "SIMPLE_PHY_TYPE_OFDM" with modulation type being 16-QAM with 1/2 convolutional coding. The channel was SimpleOfdmWimaxChannel according to NS-3. The Constant-Position mobility model was used here as well. Of the three schedulers available in the WiMAX implementation in NS-3, i.e., SIMPLE, RTPS and MBQOS, the SIMPLE one was used.

In the LTE access network, the Log-Distance propagation loss model was used. The scheduler type was "PfPfMacScheduler".

In the second set of experiments, the Constant Speed propagation delay model and the Range propagation loss model with maximum range of 250 m was used for the MANET. QoS support was also enabled. The WiFi standard for the MANET was 802.11b with Direct-Sequence Spread Spectrum (DSSS) at 11 Mbps. The Constant-Rate rate control algorithm was also used. Optimized Link State Routing (OLSR) was used as the routing protocol in the MANET. The models and configurations for the Infrastructure WiFi and LTE networks were the same as that used in the first set of experiments.

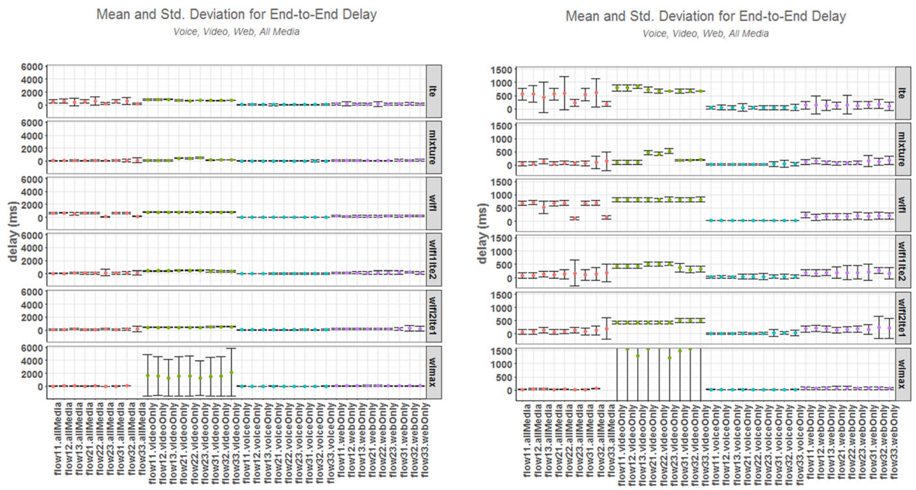
# 5 Results

## 5.1 End-to-End Delay

The mean end-to-end delay together with the standard deviation about the mean are plotted in error-plots for the different scenarios of Experiment Set 1. These are shown in Fig. 2b. The end-to-end delays for the WiMAX-only network with only video flows were found to be very large compared to those of the other scenarios. Therefore, in Fig. 2a, an enlarged version of the results is presented for greater detail of the other scenarios.

When there was only voice traffic in the network, the homogeneous WiFi-only case had the lowest average delay across the flows (i.e., 20.4 ms) as well as the lowest standard deviation (i.e., 1.75 ms). This was followed by the homogeneous WiMAX-only case with average delay of 25.5 ms and standard deviation 4.0 ms. The LTE-only case recorded the highest average end-to-end delay of 44.4 ms with a standard deviation of 86 ms. In the heterogeneous case, the average end-to-end delay for the WiFi flows increased to 24.4 ms and the standard deviation more significantly to 28.6 ms. Similarly, in the heterogeneous case, the average end-to-end delay for the WiMAX flows increased to 29.1 ms and its standard deviation more markedly to 28.8 ms. In the heterogeneous case, the average delay of the LTE flows dropped slightly (to 37.8 ms) and the standard deviation increased marginally to 93.55.

When there was only web traffic in the network, the homogeneous WiMAX-only case had the lowest average delay across the flows (69.7 ms) with an average standard deviation of 48.8 ms. However, in the heterogeneous case these web flows across the WiMAX access




**(a)** **(b)**

**Fig. 2** Plots of End-to-End Delay (Mean and Std. Deviation) of Voice, Video and Web Flows Using Different Access Network Types (i.e., LTE Only; WiFi Only; WiMAX Only; Mixture of One WiFi, One WiMAX and One LTE Access Networks; Mixture of One WiFi and Two LTE Access Networks; Mixture of Two WiFi and One LTE Access Networks)—Full View and Expanded View

network registered marginally higher delay (70.7 ms) but significantly higher standard deviation (60 ms). The web traffic in the homogeneous LTE-only case had an average delay across the flows of 140 ms with an average standard deviation of 193. However, the corresponding flows in the LTE-access portion of the heterogeneous network experienced an increase in the average delay (to 148 ms) but a decrease in the average standard deviation (to 160 ms). For WiFi, the end-to-end delay and standard deviation for the homogeneous case was 188 and 122 ms respectively, and in the heterogeneous case 118 and 87.8 ms respectively—which were both significant decreases.

For video-only traffic, the end-to-end delay averaged across flows was 813 ms in the WiFi-only (homogeneous case) and dropped to 103 ms in the heterogeneous case. The standard deviation remained basically the same (81 ms versus 80 ms). The video flows in the WiMAX-only (homogeneous case) had an average end-to-end delay of 1572 ms (with a 3058 ms standard deviation), but these values dropped for the corresponding flows in the heterogeneous case to 473 ms (with a 85 ms standard deviation). A similar situation occurred for the LTE-only (homogeneous) compared with the LTE in the heterogeneous network. In the homogeneous case, the average delay among the video flows was 723 ms (with a standard deviation of 70 ms), whereas in the heterogeneous case it was 189 ms (with a standard deviation of 22.7 ms)

When there is a mix of voice, video and web flows (i.e., the "all-media" traffic), the end-to-end delay (and standard deviation) averaged across flows dropped from 546 ms (and 97.6 ms) in the WiFi-only homogeneous case to 82.8 ms (and 79.0 ms) in the heterogeneous case. For WiMAX, the end-to-end delay (and standard deviation) increased from 37.4 ms (and 14.4 ms) in the homogeneous case to 72.0 ms (and 68.6 ms) in the heterogeneous case. For LTE, the end-to-end delay (and standard deviation) decreased from 474 ms (and 331 ms) to 104 ms (and 228 ms).

In summary, heterogeneity seemed to increase the end-to-end delay for voice flows (except for the LTE case) and standard deviation for voice flows for all cases. For web traffic, heterogeneity seemed to decrease end-to-end delay and its standard deviation for WiFi, increase both for WiMAX, and increase the delay but decrease the deviation for LTE. Video traffic benefited from heterogeneity markedly across all technologies in terms of decreased end-to-end delay and standard deviation.

For both WiMaX and LTE, it is the base station/eNodeB that schedules uplink access to the channel based on the temporal requests sent by the subscriber units. Therefore, before the subscriber units can send packets they must request then be notified of their allocations. This centralized, multi-step negotiation of resources does not occur for WiFi networks. Rather, the decentralized action of the CSMA/CA access mechanism seems to provide faster access to the core network. The impact of the centralized scheduling even in the homogeneous setting for LTE did not seem to provide as consistent a delay experience among its flows in the voice-only, video-only and web-only settings when compared to WiFi. The same could be said for the variability in delay with flows.

One would have expected that in the heterogeneous setting the aggressiveness of the WiFi would be to the detriment of the LTE and WiMAX networks. This was not found to be the case. Nevertheless the WiFi flows still enjoyed the lowest delay when compared to the LTE and WiMAX flows. Additionally, as one increased the proportion of LTE networks to WiFi networks in the wireless access from 0:3 to 3:0 (i.e., consider scenarios "wifi" → "wifi2lte1" → "wifi1lte2" → "lte"), it is the LTE flows that incurred greater variability, whereas the WiFi flows maintained lower variability in delay.

The voice and video traffic did not have built-in congestion control mechanisms. They both use UDP transport which itself provides no flow and congestion control. On the other hand, the web traffic uses TCP transport which responds to congestion. Additionally, video traffic is more bandwidth intensive than voice. Therefore, when the three types of traffic compete for bandwidth resources without service differentiation, the video would be the most aggressive absorbing the most resources. However, when the access technology provides adequate service differentiation, it would be expected that traffic flows perform just as well or even better in the multimedia environment (i.e., "allmedia") than in the non-multimedia setting (e.g., "voice-only", "video-only", "web-only"). In terms of end-to-end delay, this was found to be the case for the most part for the web-flows. The same could be said for the video flows. For the voice flows, however, the performance worsened in the multimedia setting in the WiFi-only homogeneous network and LTE-only homogeneous network.

The mean end-to-end delay with its standard deviation is plotted for the different scenarios of Experiment Set 2 in Fig. 3. It was expected that the additional overhead of routing in the MANET would have significantly impacted the end-to-end delay when compared to the Infrastructure WiFi and the LTE. However, there were no significant differences in end-to-end delay performance across the scenarios when the different combinations of MANET, LTE and Infrastructure WiFi access networks were considered. As in the first set of experiments, however, it was found that the LTE flows experienced greater variation in delay.

It should be noted that the end-to-end delay would have been determined for only those packets that reached their intended destination. For a more complete perspective the number of packets received as well as the packet-loss percentage could be investigated.

### 5.2 Packet-Loss Percentage

The packet loss percentage over the length of the simulation for the different scenarios are shown in bar graphs in Figs. 4 and 5. It was calculated as

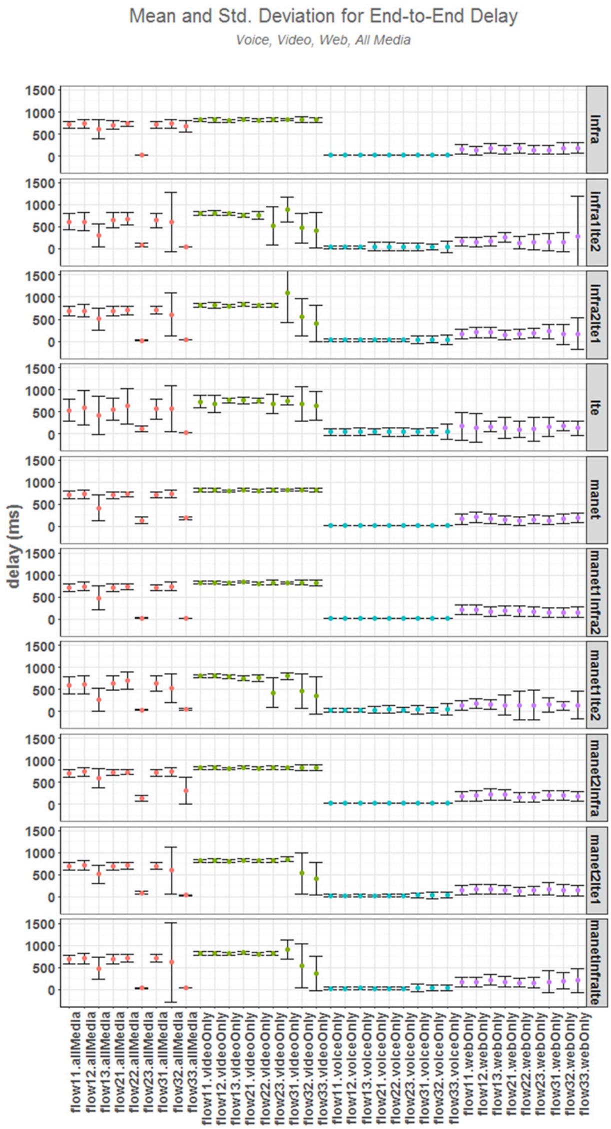



**Fig. 3** Plots of End-to-End Delay (Mean and Std. Deviation) of Voice, Video and Web Flows Using Different Combinations of Access Networks Consisting of LTE, MANET WiFi and Infrastructure WiFi

$$\text{Loss percent} = \frac{\text{Total loss}}{\text{Total loss} + \text{total received}} \times 100 \quad (1)$$

Web traffic losses were negligible across all flows and across all technologies in both the homogeneous setting and the heterogeneous setting. The maximum loss percentage was approximately 0.54%. The same could be said for voice only traffic for which the maximum loss percentage was 0.14%.

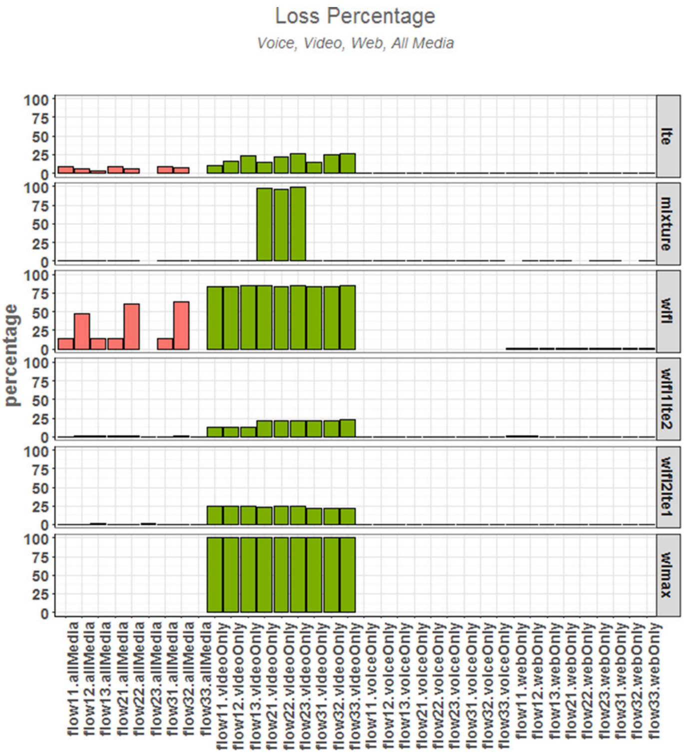



**Fig. 4** Plot of Loss Percentage of Voice, Video and Web Flows Using Different Access Network Types Consisting of LTE, WiFi and WiMAX Access Networks

When there were only video flows, however, the loss percentages were uniformly about 84–85% for the WiFi-only homogeneous case across all flows. The loss percentage was almost 100% for all the flows for the WiMAX-only homogeneous case. For the LTE-only homogeneous case, the loss percentage was less overall varying between 9.9 and 26.1%. However, in the heterogeneous case, the loss percentage for the WiFi-portion (maximum 0.44% and the LTE-portion (maximum 0.66%) was negligible when compared to their homogeneous counterparts. The WiMAX flows in the heterogeneous setting did incur much loss as in the homogeneous setting mentioned earlier.

With all media types flowing through the network, the loss percentages were comparatively negligible in the heterogeneous case (i.e., maximum 0.85%) and non-existent in the WiMAX-only homogeneous case. The voice flows and web flows in the multimedia traffic context incurred higher loss percentages than in their respective single-media context in the WiFi-only homogeneous case. On the other hand, the video flows incurred lower loss percentage. The same could be said in the case of LTE-only homogeneous network. However, the difference was significant but not as pronounced for the latter.

In the video-only scenarios, all flows seeking access at the access node (i.e., base station/AP/eNodeB) would be bandwidth intensive and because they are of one traffic class, they would all be competing against each other for that reserved for or assigned to video (e.g., the Access Category VI queue in WiFi and the rtPS queue in WiMAX) and not have access to the rest of the bandwidth resources. In the multimedia setting, there was

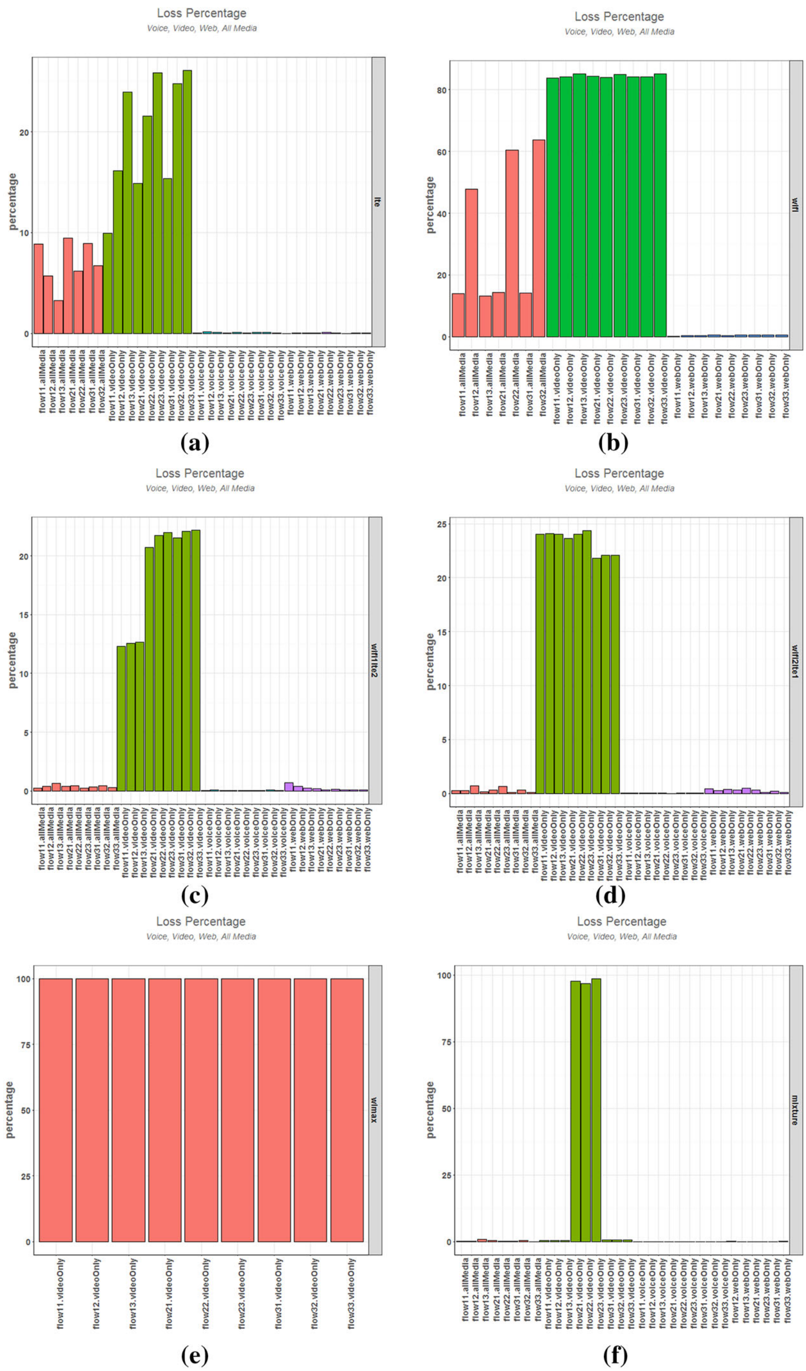



**Fig. 5** Loss Percentage of Voice, Video and Web Flows Using Different Access Network Types Consisting of LTE, WiFi and WiMAX Access Networks—Plots by Individual Access Network Types

only one video flow having to access its Access Category or rtPS queue, while the voice and web traffic flowed through their own assigned queues. Hence the lower loss percentages in that case. Additionally, the WiMAX video resource may have been underprovisioned in the first place.

Losses occurred at three main points in the network: at the output queue of subscriber station itself, at the access node and at the bottleneck router in the core (specifically at router “edge2”). For LTE and WiFi, most of the losses occurred at edge2. It was found that in the multimedia case, the voice and web traffic suffered higher loss percentages whereas the video traffic loss percentage decreased than in the non-multimedia case in the LTE and WiFi homogeneous networks. This may be due to the higher proportion of video packets being allowed through to the core that would compete with the less intensive voice and web flows.

Heterogeneity in the access network seemed to have lessened the percentage of losses across all flow types (i.e., voice, video, web) and across all access types (i.e., WiFi, WiMAX, LTE).

### 5.3 Total Number of Packets Received

The total number of packets received at the destination nodes over the length of the simulation for the different scenarios are shown in bar graphs in Fig. 6.

The total number of voice packets that flowed through the LTE-only, WiMAX-only, WiFi-only homogeneous networks and the heterogeneous network was roughly the same across the access-types, and uniform among the flows. The minimum was 15,562 packets and the maximum 17,177 packets.

For web traffic there was more variation among the flows for each access-type. Comparing the average number of packets received among the access-types, for LTE-only it was 22,937, for WiMAX-only it was lower 16,698 and for the heterogenous case it was even lower at 16,325. The average for WiFi-only was 28,714 but this was due to one flow having a very high thruput compared to the rest. Without this flow, the average would be 17,445 within the same range as the other cases.

For video-only traffic, the WiMAX-only network registered very low number of received packets (i.e., the minimum, 29, the maximum 118). For the WiFi-only homogeneous network, the average throughput was greater (i.e., the minimum, 23,992, mean, 25,217, maximum, 26,326) and roughly uniform across the flows. In the heterogeneous case, the WiFi-portion of the network had twice the throughput than that in the WiFi-only homogeneous setting (i.e., 54,765 packets versus 25,217). Similarly, the throughput across the LTE access in the heterogeneous case (54,432) was more than twice the average in the homogeneous (24,266). The average throughput across the flows also increased for the WiMAX portion in the heterogeneous case (1230 packets) over that of WiMAX-only homogeneous network (67 packets) (Fig. 6).

When there was multimedia traffic, the video traffic dominated, taking the highest proportion of the total traffic carried through the network. This was particularly so in the LTE-only and the WiFi-only case. But in the heterogeneous network, the proportion of video traffic while still the largest was reduced.

For the second set of experiments the pattern across the different access network configurations (i.e., Fig. 7) seemed the same for voice-only traffic. The same could be said for video-only traffic, web-only traffic and multimedia traffic. The same explanations offered for the first set of experiments may be plausible here. The MANET did not seem to

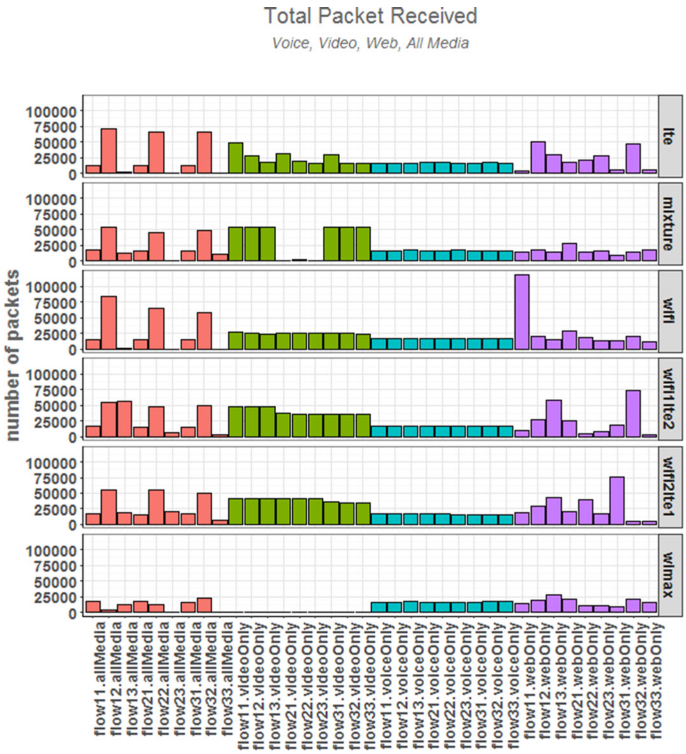



**Fig. 6** Total Packets of Voice, Video and Web Flows Received Using Different Access Network Types Consisting of LTE, WiFi and WiMAX Access Networks

lessen the throughput of the system and was not significantly impacted when it had to share core resources with Infrastructure WiFi and LTE.

In summary, it was found that heterogeneity in the access network did not significantly impact the voice traffic throughput in the voice-only setting and multimedia setting. It did however improve video throughput on average in the video-only setting and multimedia setting. But it reduced the number of web-traffic packets received than in the web-only case. In the multimedia setting, there was heterogeneity improved web traffic throughput for the most part compared to the LTE-only and WiFi-only homogeneous networks.

## 6 Discussion

Based on the aforementioned results, it can be seen that heterogeneity can improve, have no impact or impair the performance of voice, video, web and multimedia traffic in terms of end-to-end delay, end-to-end delay variation (or jitter), packet loss percentage and average throughput.

Interfacing with the droptail output queue of the ingress router would be the different medium access/scheduling schemes of the three access technologies (WiFi, WiMAX and LTE) comprising the heterogeneous network. The more aggressive a wireless medium access/scheduling scheme (provided that there are not bottlenecks on the wireless channel

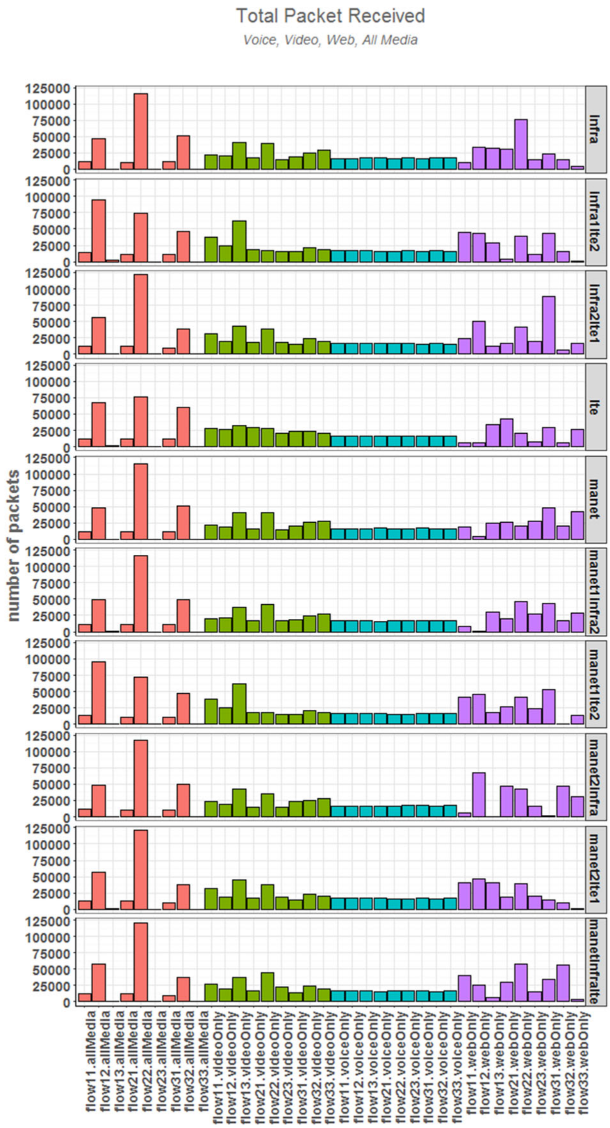



**Fig. 7** Total Packets of Voice, Video and Web Flows Received Using Different Combinations of Access Network Types Consisting of LTE, MANET WiFi and Infrastructure WiFi

itself or on the backhaul) the greater proportion of the output bandwidth of the core network it would receive at the risk of "starving" the others. The aggressiveness of such a scheme is dictated by its parameters, e.g., Contention Window size in WiFi.

For the case of single-media traffic (e.g., voice-only, video-only), the different schemes would not be competing based on priority queueing and inter-class scheduling but rather on their basic medium access protocols. Therefore if with heterogeneity the performance of these traffic flows is the same as or better than that in the homogeneous case, then comparatively the schemes would have been tuned equitably (in spite of their inherently different characteristics and parameters), ensuring uniform treatment of the same type of flows.

For the case of multimedia traffic, there would be the effect of the inter-class scheduling working in concert with the medium access protocol. So that in order for flows of a particular type/service class (e.g., voice) to perform similarly across the different access types, joint tuning of the scheduling protocols may be required in the heterogeneous network setting. This becomes even more challenging because it is the scheduling algorithm that is usually not standardized even within a given technology specification (since it serves as a differentiator among equipment manufacturers). However, without this joint tuning, there would remain a great level of uncertainty in end-to-end QoS "guarantees" when there is technology co-existence.

## 7 Conclusion

Using NS-3, we evaluated the performance of voice, video and web traffic in terms of end-to-end delay, delay variation, throughput and packet-loss across a heterogeneous network consisting of IEEE 802.11 (WiFi), IEEE 802.16 (WiMAX) and LTE access networks that share a common ingress router of a core IP network.

To determine the impact of heterogeneity, we compared the heterogeneous case to homogeneous cases consisting of WiFi-only access, LTE-only access, and WiMAX-only access. We also compared the impact on multimedia traffic with that of single-media traffic (e.g., voice-only, video-only and web-only).

It was found that this heterogeneity in the wireless access portion of the network can improve, degrade or have no impact on application performance. For example, it was found that heterogeneity seemed to increase the end-to-end delay for voice flows (except for the LTE case) and standard deviation for voice flows for all cases whereas video traffic benefited from heterogeneity across all technologies in terms of decreased end-to-end delay and standard deviation. Another result was that heterogeneity in the access network did not significantly impact the voice traffic throughput in the voice-only setting and multimedia setting. It did however improve video throughput on average in the video-only setting and multimedia setting. Additionally, heterogeneity in the access network seemed to have lessened the percentage of losses across all flow types (i.e., voice, video, web) and across all access types (i.e., WiFi, WiMAX, LTE).

Additional analyses that were performed, but due to space limitations were not presented in this paper, were to determine the impact of implementing Differentiated Services (DiffServ) in the core and the impact of competing flows in the core.

Some key research challenges in Fifth Generation (5G) wireless communications identified by Agiwal et al. [26] include network densification, Cloud Radio Access Networks (C-RANs), Heterogeneous C-RANs, backward compatibility with 4G/3G networks, providing low-latency and QoE. One of the main features of 5G is the coexistence of different technologies. Much emphasis right now is on heterogeneity at the radio level which would include the sharing of spectrum among technologies. However, to achieve end-to-end QoS guarantees, the interface with the core must also be addressed, especially when backward compatibility is to be assured. This simulation study attempts to highlight the impact of this type of heterogeneity on network performance.

## References


1. Domingo, M. (2006). Service differentiation via adaptive gateway discovery in ad hoc networks connected to wired networks. In F. Boavida, T. Plagemann, B. Stiller, C. Westphal, & E. Monteiro (Eds.), *NETWORKING 2006. Networking technologies, services, and protocols; performance of computer and communication networks; mobile and wireless communications systems, lecture notes in computer science* (Vol. 3976, pp. 13–24).
2. Koivula, M., Taramaa, M., & Ruuska, P. (2007). Differentiated services and vertical handovers supporting multimedia in heterogeneous networks. In *Proceedings of the IEEE 18th international symposium on personal, indoor and mobile radio communications (PIMRC 2007), 2007* (pp. 1–5).
3. Malila, B., & Ventura, N. (2013). Performance evaluation of a next generation network prototype using WiMax as an access network. In *Proceedings of 2013 IEEE EUROCON* (pp. 407–414).
4. Zhu, J., Vannithamby, R., Rodbro, C., Chen, M., & Vang Andersen, S. (2012) Improving QoE for Skype video call in mobile broadband network. In *Proceedings of the IEEE global communications conference (GLOBECOM), 2012* (pp. 1938–1943).
5. Friderikos, V., Wang, L., Iwamura, M., & Aghvami, H. (2004). A rate adaptation scheme for out of profile packets in a DiffServ enabled CDMA network. In Z. Mammeri & P. Lorenz (Eds.), *High speed networks and multimedia communications, lecture notes in computer science* (Vol. 3079, pp. 446–458).
6. Chassot, C., Lozes, A., Racaru, F., & Diaz, M. (2006). Signalling concepts in heterogeneous IP multidomains networks. In Y. Koucheryavy, J. Harju, & V. Iversen (Eds.), *Next generation teletraffic and wired/wireless advanced networking, lecture notes in computer science* (Vol. 4003, pp. 259–270).
7. Koo, H. L., Roh, B. H., Park, Y. K., Kim, S. E., & Kim, Y. S. (2009). Data delivery with end-to-end QoS guarantee in tactical network interconnected with wireless ad-hoc network and heterogeneous networks. In *Proceedings of the 11th international conference on advanced communication technology (ICACT 2009), 2009* (Vol. 01, pp. 804–809).
8. Orefice, P., Paura, L., & Scarpiello, A. (2010). Inter-vehicle communication QoS management for disaster recovery. In D. Giusto, A. Iera, G. Morabito, & L. Atzori (Eds.), *The internet of things*. New York, NY: Springer. https://link.springer.com/chapter/10.1007/978-1-4419-1674-7_20#citeas.
9. Vucevic, N., Bernardo, F., Umbert, A., & Budzisz, L. (2007). Evaluation of perceived QoS with multimedia applications in a heterogeneous wireless network. In *Proceedings of the 4th international symposium on wireless communication systems (ISWCS 2007), 2007* (pp. 102–106).
10. Umbert, A., Budzisz, L., Vucevic, N., & Bernardo, F. (2007). An all-IP heterogeneous wireless testbed for RAT selection and e2e QoS evaluation. In *Proceedings of the 2007 international conference on next generation mobile applications, services and technologies (NGMAST '07), 2007* (pp. 310–315).
11. Galinina, O., Pyattaev, A., Andreev, S., Dohler, M., & Koucheryavy, Y. (2015). 5G Multi-RAT LTE-wifi ultra-dense small cells: Performance dynamics, architecture, and trends. *IEEE Journal on Selected Areas in Communications*, *33*(6), 1224.
12. Andreev, S., Gerasimenko, M., Galinina, O., Koucheryavy, Y., Himayat, N., Yeh, S. P., et al. (2014). Intelligent access network selection in converged multi-radio heterogeneous networks. *IEEE Wireless Communications*, *21*(6), 86.
13. Li, J., & Cui, Q. F. (2006). The QoS research of VoIP over WLAN. In *Proceedings of the 2006 international conference on communications, circuits and systems, 2006* (Vol. 3, pp. 1782–1785).
14. Elmangosh, A., Ashibani, M., & Shatwan, F. (2007). The interworking between EDCA 802.11e and DiffServ. In *Proceedings of the IEEE international performance, computing, and communications conference (IPCCC 2007), 2007* (pp. 545–550).
15. Ke, C. H., Shieh, C. K., Chilamkurti, N., & Shirwadkar, U. (2006). A new framework for H.264 video delivery over heterogeneous networks. In *Proceedings of the international conference on networking, international conference on systems and international conference on mobile communications and learning technologies (ICN/ICONS/MCL 2006), 2006* (pp. 109–109).
16. Lee, M., Kwon, B., & Copeland, J. (2010). Cross-layer design of DiffServ architecture for end-to-end QoS in IEEE 802.16 multi-hop mesh/relay networks with IEEE 802.11e WLANs. In *Proceedings of the 2010 7th IEEE consumer communications and networking conference (CCNC), 2010* (pp. 1–5).
17. Domingo, M., & Remondo, D. (2004). Analysis of VBR VoIP traffic for ad hoc connectivity with a fixed IP network. In *Proceedings of the IEEE 60th vehicular technology conference (VTC2004-Fall), 2004* (pp. 2834–2837).
18. Domingo, M., & Remondo, D. (2004). A cooperation model between ad hoc networks and fixed networks for service differentiation. In *Proceedings of the 29th annual IEEE international conference on local computer networks, 2004* (pp. 692–693).

19. Villaln, J., Cuenca, P., & Orozco-Barbosa, L. (2007). On the capabilities of IEEE 802.11e for multimedia communications over heterogeneous 802.11/802.11e WLANs. *Telecommunication Systems*, *36*(1–3), 27.
20. Munir, A., & Wong, V. (2007). Interworking architectures for IP multimedia subsystems. *Mobile Networks and Applications*, *12*(5–6), 296.
21. Ryu, M., Kim, Y., & Park, H. (2008). Systematic QoS class mapping framework over multiple heterogeneous networks. In S. Balandin, D. Moltchanov, & Y. Koucheryavy (Eds.), *Next generation teletraffic and wired/wireless advanced networking, lecture notes in computer science* (Vol. 5174, pp. 212–221).
22. T. Pliakas, G. Kormentzas, & C. Skianis. (2006). End-to-end QoS issues of MPEG4-FGS video streaming traffic delivery in an IP/UMTS network. In A. Helmy, B. Jennings, L. Murphy, & T. Pfeifer (Eds.), textitAutonomic management of mobile multimedia services, lecture notes in computer science (Vol. 4267, pp. 247–255).
23. NS-3. NS-3 network simulator. https://www.nsnam.org/.
24. T.U.B. Telecommunication Networks Group. Trace files movies- mr. bean. http://www-tkn.ee.tu-berlin.de/research/trace/pics/FrameTrace/mp4/Verbose_bean.dat.
25. I. ResiliNets Research Group, U. Telecommunication Technology Center (ITTC), The University of Kansas Lawrence. HTTP model. https://codereview.appspot.com/4940041
26. Agiwal, M., Roy, A., & Saxena, N. (2016). Next generation 5G wireless networks: A comprehensive survey. *IEEE Communications Surveys Tutorials*, *18*(3), 1617.

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