5G MULTI-LAYER ROUTING STRATEGIES FOR TV WHITE SPACE SECONDARY USER ACCESS

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Abstract: As mobile applications and services have developed, the dramatic growth in user data traffic has led to the legacy channels becoming ever more congested with the commensurate requirement for more spectrum. This has motivated both regulatory bodies and industry to investigate innovative strategies to increase the existing spectral efficiency. Prominent examples include both *Long Term Evolution* (LTE) which employs *orthogonal frequency-division modulation* technology to improve bandwidth efficiency, and heterogeneous networks, which facilitate the offloading of data traffic between technologies such as from LTE to Wi-Fi and vice versa. Furthermore, as 5G mobile technology and related standards mature, there is an impetus to address the issue of *secondary user* (SU) spectrum access in which *TV White Space* (TVWS) is the prime contender. Two nascent viewpoints have emerged as to how this will evolve: i) greater coverage, ii) increased throughput allied with lower latency. This paper presents a novel TVWS framework that successfully fulfils both criteria to ensure 5G services can both exploit TVWS spectrum and protect the benefits of SU access and quality-of-service provision by using a routing strategy on the *Access Network Discovery and Selection Function* server to dynamically determine the most suitable heterogeneous technology for the new framework.

1 Introduction

The unused television (TV) bands which have arisen from the transfer from analogue to *Digital Terrestrial* TV (DTT) are commonly referred to TV White Space (TVWS) [1] [2]. These have been created by the localised allocation of DTT frequencies, so frequencies not allocated in a particular geographic area are available for usage by, for example, 5G *cognitive radio networks* (CRN), services and applications. Regulators like the *Office of Communications* (OFCOM) in the UK and the US *Federal Communications Commission* (FCC) have recently adopted proposals to allow new broadband devices to operate within TVWS provided the *primary user* (PU) is not impacted. To guarantee this, appropriate PU detection mechanisms need to be deployed, such as the *generic enhanced detection algorithm* (GEDA) [1], [2] so no active PU channels are used for TVWS access.

This paper addresses the unequal *radio frequency* (RF) coverage problem [1], where the RF transmit power of both fixed and mobile *secondary users* (SU) nodes can vary up to some prescribed maximum value. Fixed SU (forward link) nodes, however, can have a higher RF power allocation than their mobile (reverse link) counterparts because of their larger antenna to avoid hidden node problems [1]. The corollary of this coverage imbalance caused by the combination of regulatory RF power allocation and antenna height, is that ultimately the mobile SU governs the overall coverage. To compensate for this asymmetric coverage in the forward and reverse links and maximise coverage in both directions, an ad hoc routing strategy must be creatively employed in the latter i.e., from the mobile to the fixed node.

In most cases, regulatory *base station* (BS) transmitter power specifications [2] are higher than the mobile powers, which when coupled with the mobile antenna heights being lower than the BS, means the BS service area is always significantly greater. This mandates some form of routing to enable the TVWS SU mobile to occupy the same service area as the BS. Consequently, the proposed network structure has a forward link directly connected to the SU mobile nodes while the reverse link comprises multiple routes to the BS. Using a routing network from the mobile to the BS, means maximising the probability of a packet reaching its destination so not to waste bandwidth circulating packets which will be lost. The proposed strategy maximises coverage and SU *quality-of-service* (QoS), by using the following cross-layer parameters: link distance (layer 1), *time-to-live* (TTL) in layer 3 and the *QoS class identifier* (QCI) in layer 4. The heart of the new TVWS access topology is an IEEE802.11af *Wireless Local Area Network* (WLAN) [2], using *orthogonal frequency-division modulation*, with up to four channels bonded in either one or two contiguous blocks. To facilitate TVWS framework access, an *Access Network Discovery and Selection Function* (ANDSF) [3],[4] is implemented which is a 3GPP network element which uses the LTE infrastructure to establish a session by evaluating key parameters like the maximum coverage per QCI. The ANDSF server then determines the best available heterogeneous technology for the session to be anchored to, namely either LTE or TVWS SU (IEEE802.11af).

As a network continually changes due to node mobility and RF propagation conditions, the packet route will similarly change. Routing information has therefore to be regularly updated to avoid packet loss, with various *mobile ad hoc network* (MANET) routing protocols [5], [6] being available, notably *Dynamic Source Routing* (DSR), *Ad hoc On-demand Distance Vector* (AODV) and MPLS (Multi-Protocol Label Switching). DSR, AODV and MPLS-AODV, which is AODV used over MPLS, can all be applied to an IEEE802.11af model to achieve a symmetrical service area in the forward and reverse links. However, due to the diverse properties of these protocols, different QoS provision are afforded for different types of data traffic. The new access strategy evaluates the wireless properties along with the chosen QCI and allocates the most appropriate technology to maximise the user QoS. If TVWS IEEE802.11af is assigned due to being within the capture distance for the specific QCI, then the *maximum transmission unit* (MTU) size and TTL are selected to minimise the *packet error rate* (PER) and packet delay and ensure the QCI is always upheld in the forward link.

The remainder of this paper is organised as follows: Section 2 reviews the relevant mobile routing literature, while Section 3 details the simulation test models adopted including their respective noise strategies. Sections 4 and 5 respectively investigate BS coverage and mobile service area, while Section 6 evaluates the differences between the various MANET routing protocols. Section 7 introduces two case studies, also providing their respective results. Section 8 details the new algorithm to improve QoS within ANDSF with some concluding comments given in Section 9.

2 Mobile Routing Literature Review

This section provides a short review of the three main MANET routing protocols [5], [6].

DSR [5] is a simple protocol where all routing information is maintained by each individual node. It is specifically designed for multi-hop link use for mobile nodes and allows the network to be entirely self-organizing, without the need for network administration. The protocol has route discovery and route maintenance phases, which work collaboratively to enable nodes to discover and maintain routes to destination nodes. The protocol is demand-driven, so routing overheads are scalable to only what is required, but a key shortcoming is that packet transmission can only occur once a route to a destination node has been found.

In contrast, the AODV [5] routing protocol is solely designed for MANETs. It offers efficient adaptation to dynamic link conditions, low resource overheads, low network utilization, and determines unicast routes to destination nodes within the MANET. Route table entries are dynamically setup at each intermediate node as the packet is transmitted towards the destination so reducing traffic overheads.

MPLS [9] provides a connection-oriented QoS by utilising a condensed label structure at layer 2. In comparison to layer 3 *internet protocol* (IP) packet switching, which does not support connection-oriented QoS, this has the advantage of reducing the end-to-end delay due to faster label processing. Also, both DSR and AODV can be used within the MPLS framework to form MANET sub-protocols.

The ensuing sections will specifically consider the DSR, AODV and MPLS with AODV (MPLS-AODV) protocols embedded into an IEEE802.11af model to achieve symmetrical service areas in the forward and reverse links. The modelling strategy adopted will now be outlined.

3 Test Models

To reflect real-world scenarios, the routing model must assume a dynamic multi-nodal architecture and be able to determine IEEE802.11af SU QoS using PER and delay for DSR, AODV and MPLS-AODV with a *User Datagram Protocol* (UDP) transport layer. The model must also allow different data traffic parameters like MTU size and packet rate, so the INET frame model [7] [8] was selected to fulfil these requirements.

INET is based on the OMNet++ [7] [8] routing platform and simulates IEEE802af with Manetrouting. It models a fixed node (BS) along with several mobile nodes in a predefined area termed the *playground* [7]. By considering various scenarios, parameters including TTL, RF power, routing protocol and number of mobile hosts can be adjusted so changing network behaviour, with these changes then being measured using the PER and packet delay metrics.

The effect of noise on PU performance has been analysed in [2], so the focus in this paper is on the critical impact of noise on the SU performance. The noise regime of the test model has two components: (i) adjacent channel interference (Ch_{N+1}) and (ii) adjacent DTV area co-channel interference. Since the GEDA PU detection system [2] is used, no PU channel is allocated for SU access within a specific area, so co-channel noise is not a factor in the same DTV area.

As for adjacent channel interference, a radius is defined around a DTV PU transmitter so that Ch_{N+1} can be allocated to a TVWS SU without causing interference to the SU. To illustrate this, consider the Mendip DTV transmitter case study in [2], where a 3Km radius is used to determine the signal strength (-17dBm) from the model. The transmission mask for the DTV standard [10] then gives adjacent channel suppression of -83dB, and a SU interference signal of -100dBm at 3Km from the PU transmitter.

For the adjacent DTV area co-channel interference in the same Mendip DTV case study [2], an interference signal of -116dBm exists at the edge of the DTV area for possible impact on SU in an adjacent area. The corollary from this analysis is that the background noise value of -100dBm is used in all the routing models because it reflects the worst-case scenario.

4 BS Service Area Analysis

To appreciate MANET routing protocol behaviour when embedded into an IEEE802.11af model, the BS service area which forms the routing boundary is determined by three parameters:

1. Maximum *Effective Isotropic Radiated Power* (EIRP) used for a BS SU as specified by the relevant regulator [2] i.e., 17dBm and 30dBm for the UK and US respectively.
2. PER P_p.
3. The modulation scheme adopted to provide the requisite throughput and corresponding *Signal-to-Noise Ratio* (SNR) to attain the prescribed P_p.

To define the BS service boundary, the worst-case PER is used and to determine this value, the 3GPP [9] QCI is applied. The rationale for this is that QCI reflects the packet forwarding behaviour in LTE networks, and so represents a pragmatic solution for defining TVWS SU QoS classification. It also means it can be easily integrated into the LTE core network. The various QCI categories and related parameter settings are shown in Table 1 and are extracted from the 3GPP standards [9], for various data services using both *guaranteed bit rate* (GBR) and non-GBR data resource types.

The lowest BS PER defines the worst-case routing area for a mobile SU, which occurs when $P_p=10^{-6}$. This determines the service boundary by converting it into a matching *bit error rate* (BER) P_e [11] as follows:

$$P_e = 1 - \left(1 - P_p\right)^{\frac{1}{N}} \tag{1}$$

where N is the packet length, which for IP packets is normally 128, 256, 512, 1024 or 1500 bytes. The respective BER for a range of SNR values using 4, 16, 64 and 256 QAM modulation techniques is taken from [11]. For a 1500-byte packet, (1) gives $P_e = 8.33\times10^{-11}$ for 256-QAM which is used in IEEE802.11af, giving a SNR threshold of 35dB, from which the maximum distance D between a SU transmitter and receiver can be determined.

Since the SU network uses much less power than the PU, the predominant propagation component will be the

Table 1 3GPP QCI Category Specifications and related parameter values [9]

QCI	Resource Type	Priority	Packet Delay Target (ms)	Packet Error Rate (PER) Target	Example Services
1	GBR	2	100	10^{-2}	Conversational Voice
2		4	150	10^{-3}	Conversational Video (Live Streaming)
3		3	50	10^{-3}	Real Time Gaming
4		5	300	10^{-6}	Non-Conversational Video (Buffered Streaming)
5	Non-GBR	1	100	10^{-6}	*IP Multimedia Subsystem* (*IMS*) Signalling
6		6	300	10^{-6}	Video (Buffered streaming) TCP-based applications (www, e-mail, chat, ftp, p2p file sharing, progressive video)
7		7	100	10^{-3}	Voice, Video (Live Streaming) Interactive Gaming
8		8	300	10^{-6}	Video (Buffered streaming) TCP-based applications (www, e-mail, chat, ftp, p2p file sharing, progressive video)
9		9			

line of sight (LOS) with reflection. This contrasts with the PU, where it is a combination of LOS, reflection and diffraction and so for this reason Rician fading [12] is chosen for the SU propagation channel because it emulates a predominant LOS with reflection.

To baseline the *peak coverage distance* (*D*) in the forward link so the new SU mobile coverage model (reverse link) has a maximum coverage target, the *free space loss* (FSL) is used [2].

$$FSL\ (dB) = 20Log(D) + 20Log(f) + 20Log\left(\frac{4\pi}{c}\right) \qquad (2)$$

where D is the distance between the SU transmitter and receiver (m), f the frequency (Hz) and c the speed of light ($3\text{x}10^8$ m/s). The receiver signal at the demodulator is now calculated using the *receiver actual noise* (RAN):

$$RAN = 10Log(k \cdot T_o \cdot B) - NF \qquad (3)$$

where B is the relevant DTT bandwidth (8MHz and 6MHz respectively for the UK and US), k is Boltzmann's constant ($1.38 \text{ x } 10^{-23}$), T_0= 290 ^{o}K (ambient temperature of 17^{o}C) and NF is the receiver noise figure (7.5dB). Thus, with EIRP=17dBm and SNR=35dB [11], D can be derived from:

$$SNR + RAN = EIRP - FSL + G_T + G_R \qquad (4)$$

where G_T and G_R are respectively the transmitter (0dB) and receiver antenna (2dB) gains.

Hence, for the UK scenario and using a TVWS frequency of 706MHz which is unused by the PU, and EIRP=17dBm, this translates to a SU coverage radius of 400m. For the corresponding US scenario, with a TVWS frequency of 629MHz which again is not used by the PU and the same EIRP, the coverage area radius is 517m.

The next section explains how the new QCI service structure is implemented using physical, transport and IP layer measurements to provide the appropriate QoS provision for SU mobile nodes.

5 Mobile Node Service area

The key motivation for this work is that the BS forward link uses a single-hop with no routing protocol due to the EIRP value disparity between the BS and mobile node [1]. By employing multi-hop routing in the reverse link, the BS service area becomes the target coverage for the SU mobile node service area, though in practice, by using QCI PER and packet delay metrics, this may not be achievable. This is because in the reverse link, the SU mobile uses a lower EIRP and so relies on routing which in turn depends on the population density to achieve the desired PER and packet delay. In the next section the reverse link behaviour from multiple SU nodes to the BS is analysed for the DSR, AODV and MPLS-AODV protocols, to facilitate coverage equalisation in both directions and in so doing, deliver a consistent SU QoS.

The routing simulator OMNeT++ applies the SNR to the PER data to mimic the behaviour of an IEEE802.11af mobile SU and to calculate the coverage per QCI category (see Table 1). This information is then embedded within the ANDSF policy server to monitor network performance.

6 AODV v DSR v MPLS-AODV Routing

To critical analyse the comparative differences between the AODV, DSR and MPLS-AODV protocols, the BS coverage radius for the UK scenario is used, which from Section 4 is up to 400m. This is not only used to determine the best routing protocol, but to examine the maximum service envelopes for differing QCI categories and the requisite ANDSF algorithm parameters.

The routing simulator applies a square routing boundary which is an equivalent routing area for the BS whose coverage radius is half the side of the square boundary. A variable packet rate between 0.25s and 0.5s is randomly chosen for each of the four simultaneous data sessions using 128bytes per UDP packet. This equates to a packet rate of 2 to 4 packets/s which will supply a UDP transport layer data speed in the range 2048bps to 4096bps per a mobile user session. To ensure the maximum hop count is achieved for accurate results, the TTL in the IP/MPLS header is set to 40 which is much greater than necessary. The various wireless parameters used in both the UK and US are given in Table 2.

Table 2 UK and US wireless parameters settings

Parameter	Value
Frequency (MHz)	UK = 706, US = 629
EIRP (dBm)	UK = 4, US = 16
Modulation Scheme	IEEE802.11af 256QAM
WLAN Data Rate	36Mbit/s
Mobile Node Mobility	Random (1 to 20 m/s)
DTT Bandwidth (MHz)	UK = 8, US = 6

A decisive factor affecting the performance of a routing protocol is the number of intermediate routing nodes in the routing area. The assumption is to use accepted community metrics (The World Bank, 2016) relating to the number of mobile routing devices in an area using a country's population per Km^2 (M_{Km}) and the number of mobile subscriptions per 100 people (S_{100}). $A_{network\ area}$ is the active network area under investigation. If it is assumed there are 4 major operators managing TVWS devices and that the mobile subscriber population P_{Pop} within a coverage area is uniformly distributed, then:

$$P_{Pop} = A_{network\ area} \cdot \left(\frac{M_{Km}}{4}\right) \cdot \left(\frac{S_{100}}{100}\right) \qquad (5)$$

P_{Pop} is the total number of mobile nodes and is calculated at each coverage radius, with Table 3 showing P_{Pop} and corresponding radius results for the UK scenario. This information is used as simulation parameters for the number of mobiles in a specific coverage radius for the AODV, DSR and MPLS-AODV protocols with the comparative PER and packet delay results respectively plotted in Figs. 1 and 2.

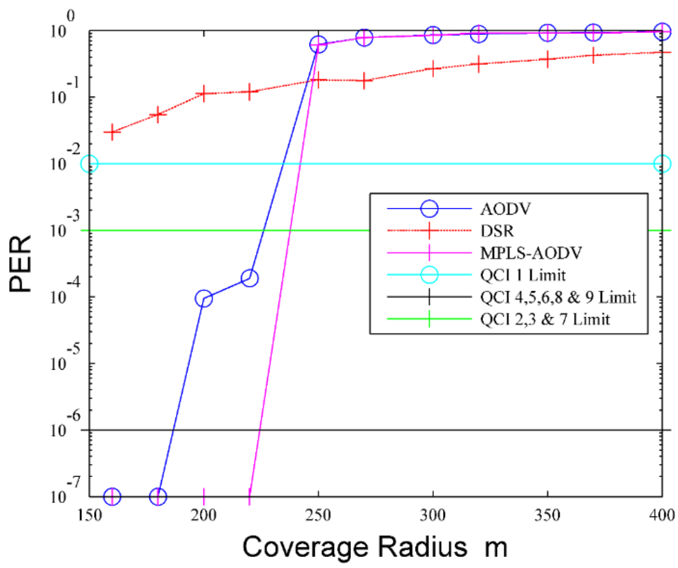

***Fig. 1** PER against Coverage Radius*

Fig. 1 reveals notable PER improvements for AODV compared to DSR, which fails to uphold any of the PER requirements in the QCI standards defined in [9]. MPLS-AODV has superior PER over AODV up to the QCI 1 limits because of lower packet latency as routing decisions are made on the MPLS label at Layer 2 rather than the IP Layer 3. This has the effect of reducing packet errors as routes change when nodes move.

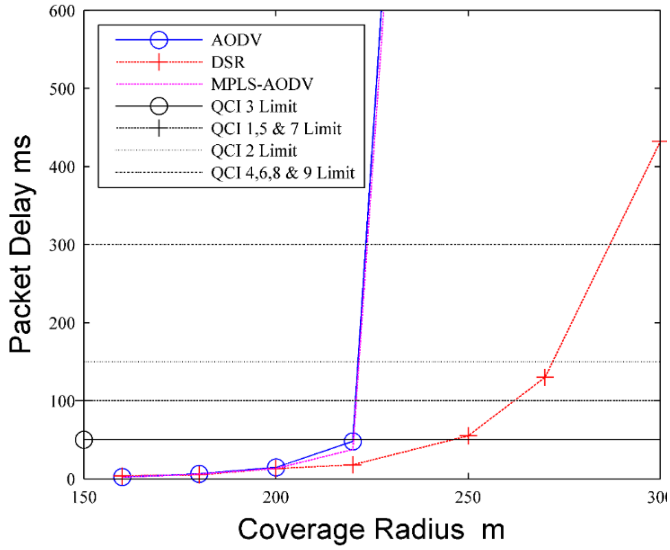

Fig. 2 *Packet delay against Coverage Radius*

Table 3 Mobile population per coverage radius per UK operator

Mobile subscriber population (P_{Pop})	Coverage Radius (m)
42	400
36	370
32	350
27	320
24	300
19	270
16	250
13	220
11	200
9	180
7	160

The relatively poor DSR performance stems from the mobile nodes moving in an irregular manner so when a route is established, the end-to-end route can change which may reduce the SNR on certain links to the point that a particular route is no longer viable and PER becomes unacceptably high. For both AODV and MPLS-AODV, a packet is sent to the nearest routable node, which in turn forwards the packet onto other nodes until it reaches the BS, so they are more resilient to route changes. The PER for both AODV and MPLS-AODV increases with coverage radii due to the longer hop distance which results in decreased SNR, even when the mobile population also increases.

The corresponding packet delay results for AODV and MPLS-AODV are displayed in Fig. 2 and show there is no significant difference between the protocols up to the QCI 4, 6 8 and 9 limits. Interestingly, DSR also provides good delay results, however the reason for this is the small number of packets delivered, as evidenced in Fig. 1, so those packets that are delivered will have low latency.

In critically evaluating the respective PER and delay results, a pragmatic conclusion is that MPLS-AODV delivers consistently lower PER for an analogous packet latency compared to either AODV or DSR so justifying its choice as the preferred protocol to uphold the QCI QoS requirements in [11]. The next section investigates the criteria to maximise the coverage radius for MPLS-AODV at various QCI settings to guarantee a prescribed QoS provision for SU.

7 SU Coverage Performance using MPLS-AODV

Two case studies, one for the UK and the other for the US, are presented to demonstrate the coverage radius for a TVWS SU using the MPLS-AODV protocol between active, intermediate and BS nodes. These case studies encompass the majority of available DTT standards which make the results interchangeable for most countries. The IEEE802.11af standard, [2] is used with the PER and packet delay measurements giving the coverage radius for differing QCI levels in Table 1. Once the coverage radii results are collected for each QCI category, they are used in an access algorithm in ANDSF to either allow transmission or redirect to an alternative technology such as LTE.

7.1 UK Case Study

The aim is to maximise the coverage radius for the various QCI levels in [9], using the relevant UK parameter values in Table 2, while the assorted UDP and IP parameter settings being given in Table 4.

Before explaining how UDP parameters are employed in the coverage radii simulation model, the mobile population must be determined using (5), with Table 5 displaying the corresponding mobile populations for different coverage radii from the BS.

To critically evaluate QoS provision, 4 concurrent data sessions are established, 1 per mobile node using the

Table 4 UDP and IP parameters

Number of concurrent data sessions	UDP Maximum Transmission Unit (MTU) (bytes)	Application Data Rate per Data Session (kbit/s)	UDP Packet Rate (packets/s)	TTL (s)
4	128	32	31.25	40
4	256	32	15.625	40
4	512	32	7.8125	40
4	1024	32	3.90625	40
4	1500	32	2.6667	40

population values in Table 5. Each session involves a 32kbit/s application [11] that supports the session initiation protocol, voice over LTE and over-the-top voice-over IP client applications, together with either an internet browsing or email application running in parallel. These 4 concurrent IP sessions have been specifically designed to rigorously demonstrate the network's QoS performance across the gamut of QCI categories [9].

To maximise performance, various UDP MTU packet lengths have been employed to reflect differing network effects including packet loss and delay, whilst avoiding packet fragmentation. The normal Ethernet MTU packet length is 1500bytes, but if the network endpoints use different MTU sizes, there is a point where MTU size can be optimised for wireless performance. There is thus a nexus between using small packets for low PER and larger packets which avoid fragmentation in packet delay. Figs. 3 and 4 show the PER and packet delay parameters respectively and provide insight into how the network can maximise the BS coverage radius with reference to the QCI categories in Table 1.

Fig. 3 shows the PER for various QCI categories with different MTU sizes, with the three horizontal lines being the 10^{-2}, 10^{-3} and 10^{-6} PER thresholds. The best performing MTU packet size is 512bytes and 1500bytes at the three respective thresholds, which are QCI 2, 3 and 7 up to 210m away from the BS and QCI 4, 5, 6, 8 and 9 up to 205m away from the BS and for QCI 1 215m. For the 128byte MTU size, more packets need to be transmitted to achieve the overall bit-rate leading to an increased probability of a packet being transmitted at a low SNR so increasing the PER as evidenced in Fig. 3. The MTU size of 512bytes and 1500bytes represents a pragmatic solution in terms of packet size, so lowering the error probability by minimising the number of packets sent, while the packet duration is of necessity small compared to node mobility to ensure a minimal PER whenever a route changes mid-packet. PER alone is deficient however, in assessing routing quality since packet delay is also considered in the QCI standards. For a MTU packet size of 1024bytes, from Fig. 3 the PER at 120m increases dramatically, though it only reflects the loss of a single packet due to collisions, since IEEE802.11af does not detect contention in the air interface, and UDP lacks a retransmission capability.

Table 5 UK mobile subscriber population per operator

Coverage Radius (m)	Mobile Subscriber population
240	15
220	13
200	11
180	9
160	7
140	5
120	4
100	3

The corresponding set of packet delay versus coverage responses are displayed in Fig. 4 for the same set of MTU sizes and QCI categories. Again, the horizontal plots are the various delay thresholds for specific QCI categories. The results again confirm an MTU size of 512 bytes outperforms all other MTU sizes, so this is evidently the best choice for any IEEE 802.11af based wireless network.

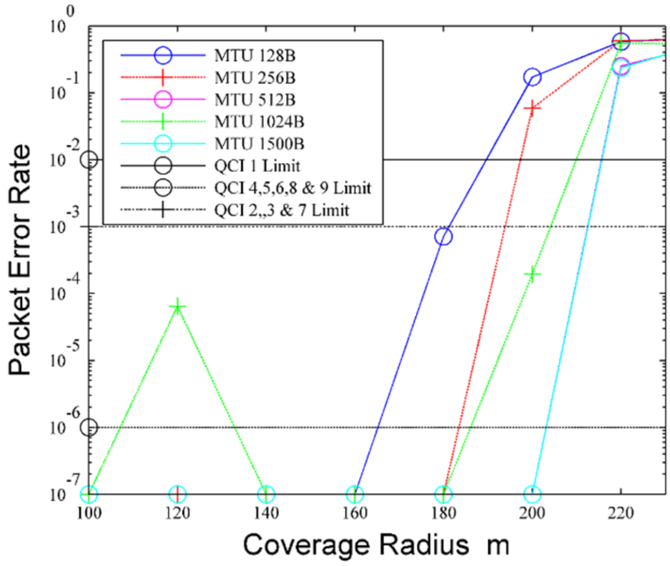

Fig. 3 *UK Packet Error Rate Results*

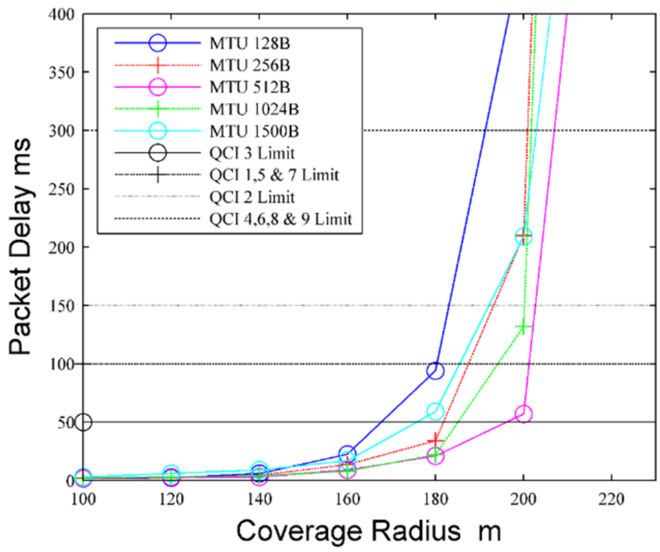

Fig.4 *UK Packet Delay Results*

For the smaller 128byte MTU, the processing time increases as does the packet delay. Conversely, in the 1500byte case, because the packet rate is lower than the node movement then route integrity is impacted leading to a higher packet delay, so a 512byte MTU size achieves the maximum coverage radius. Table 6 correlates the QCI categories with the maximum distances from the BS for a 512byte MTU, considering both PER and packet delay, where the latter is the key parameter because it consistently gives lower radii values than PER.

Table 6 UK QCI against supported distance from BS for an MTU of 512bytes

QCI Category	Distance from BS (m)
1	205
2	202
3	197
4	201
5	205
6	201
7	205
8	201
9	201

To help interpret these results, a further experiment was undertaken using the same experimental set-up, to determine the maximum hop count for all QCI categories that can support services at the maximum distance of 200m from the BS, for an MTU size of 512bytes and coverage radius of 200m. The simulation is repeated with TTL decremented by 1 for each subsequent execution run until the PER reaches the values defined in Fig. 3. When this occurs the minimum hop-count is TTL+1 which for the UK case is found to be 13. In other words, this is the number of hops beyond which further increases will not reduce the PER. The next section will examine the corresponding analysis for the US scenario.

7.2 US Case Study

The major difference between the UK and US case studies is the wireless parameter values (Table 2) [2], notably the mobile transmit power (EIRP) and DTT bandwidth.

The same UDP setup is used as the UK case study (Table 3) with the US mobile subscriber population per coverage radius per operator using (3), being displayed in Table 7. The corresponding PER and packet delay curves shown in Figs. 5 and 6 respectively.

Fig. 5 reveals that MTU sizes of 512 and 256 bytes both achieve a PER = 1×10^{-6} up to a coverage distance of 750m, while 256 and 128 bytes only achieve this PER value at 700m. Again, as in the UK scenario, MTU sizes of 128, 1024 and 1500 bytes increase the probability of errors as more packets are transmitted.

The corresponding packet delay results in Fig. 6 all follow the same trend as the UK scenario except the MTU 256 and 512 bytes sizes, which achieve the same results and are within the QCI bounds for all categories. This is because of the increased SU RF power, so giving the same result as the PER in Fig. 5. Since from a PER perspective, both MTU lengths of 512 and 256 bytes are able to support all QCI categories up to 750m and since 512 bytes consistently achieves both the lowest PER and packet delay, this value determines the maximum hop count, which for the US scenario is 10 in comparison to 13 for the UK.

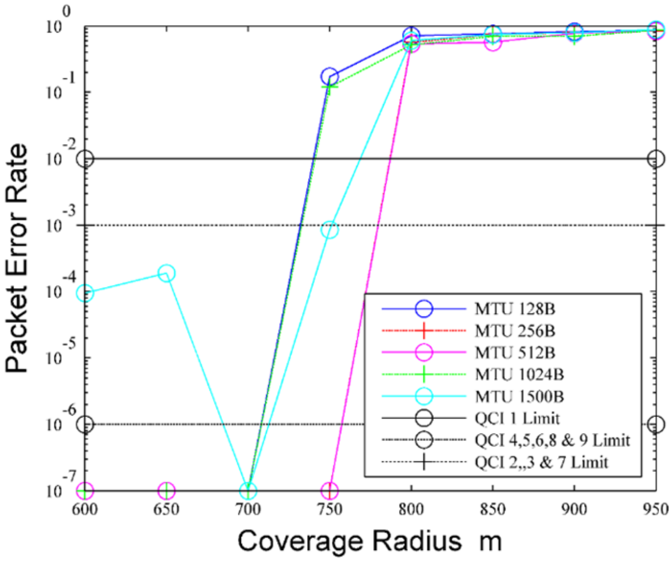

***Fig. 5** US PER versus coverage radius results*

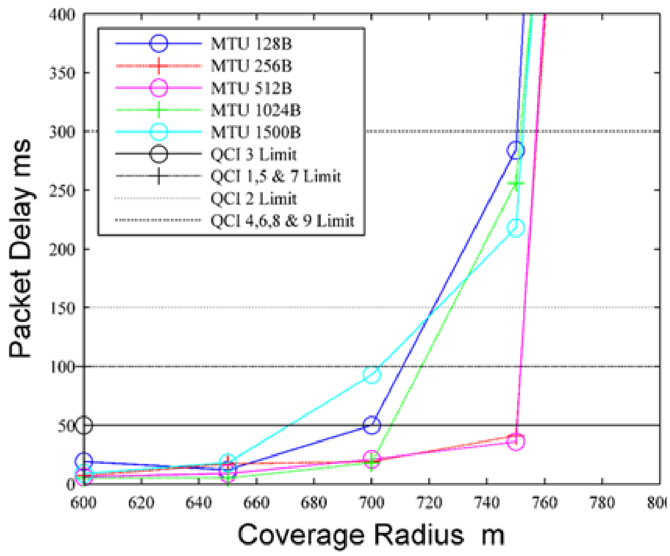

***Fig. 6** US packet delay versus coverage radius results*

Table 7 US Mobile Subscriber Population per Mobile Operator

Coverage Radius (m)	Mobile Subscriber population
950	27
900	24
850	22
800	19
750	17
700	15
650	13
600	11

7.3 Results Discussion

The key observation distilled from these results is the US coverage area able to be supported by this routing strategy (Table 8) is considerably larger than the UK. This is directly attributable to the FCC setting a mobile EIRP value ≈16 times greater, with the corollary being a mobile node can reach a BS in fewer hops, 10 instead of 13 hops, so representing a processing saving of more than 30%.

Table 8 US QCI against supported distance from BS for an MTU of 512bytes

QCI Category	Distance from BS (m)
1	750
2	755
3	749
4	755
5	750
6	755
7	750
8	755
9	755

8 Maximising QCI QoS using MPLS-AODV

Heterogeneous networks allow a call request from one technology with ANDSF deciding which technology the call is established on, with LTE being the default technology as it has a greater range than WLAN. The request from the *user equipment* (UE) should detail the QCI category required for the UE application along with *global positioning service* (GPS) location data, which is sent to the ANDSF where the access rules are executed. These rules determine which access technology to use and allocate the nearest resource ID for the UE to access. In a WLAN example, this will be the *service set identifier* (SSID) with which the UE sets up a traffic connection using IEEE802.11af parameters for the *evolved packet core* (ePC) [3], [4], which backhauls the traffic via the ePC.

The ANDSF policy algorithm to support IEEE802.11af and the assorted QCI categories (Table 1) is now discussed, where it is assumed the ANDSF standard in [3],[4] is the heterogeneous mechanism for technology selection.

The new ANDSF policy algorithm has been validated for both the UK and US case studies (Section 7), to implement an IEEE802.11af network with MPLS-AODV as the routing protocol. The various control parameters are defined in Table 9, while the pseudo-code representation of the ANDSF access algorithm is given in Algorithm 1.

The initialisation information for ANDSF includes the distance of the UE from a specific WLAN BS and is acquired by GPS alongside the Haversine distance [11]. This distance is compared with the maximum QCI service distance for the PER and packet delay results, and the lowest value used to decide if IEEE802.11af technology will service the UE at the specific QCI. Steps 1-6 in Algorithm 1 implement the Haversine distance [11] between two GPS coordinates, while Steps 7-14 compare this value with the maximum coverage distance for the specified QCI category using Tables 6 and 8 If it is greater than the maximum QCI service distance from a PER and packet delay perspective, then access is denied over an IEEE802.11af network, otherwise access is permitted and the SSID along with the transport and MPLS layer parameters, TTL and MTU size are sent to the mobile UE.

9. *IEEE802.11af ANDSF policy algorithm*

Table 9 ANDSF access control parameters

λ_1	Longitude of IEEE802.11af BS (radians)
Ψ_1	Latitude of IEEE802.11af BS (radians)
λ_2	Longitude of mobile UE in connection request from mobile UE (radians)
Ψ_2	Latitude of mobile UE in connection request from mobile UE (radians)
QCI	QCI category from mobile UE
D_{QCI}	Maximum distance from BS at which QCI category can be supported
N_{SSID}	SSID of BS identified by ANDSF (Algorithm 1)
N_{TTL}	Time-to-Live (TTL)
N_{MTU}	MTU Size (bytes)
R	Earths Radius in km (6371)

Algorithm 1 Pseudo-code representation for the ANDSF IEEE802.11af access algorithm

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Inputs: $\lambda_1, \Psi_1, \lambda_2, \Psi_2, QCI, D_{QCI}, R$
Outputs: $N_{SSID}, N_{TTL}, N_{MTU}$
$diff\lambda = \lambda_2 - \lambda_1$
$diff\varphi = \varphi_2 - \varphi_1$
$$a = \left(\sin\left(\frac{diff\varphi}{2}\right)\right)^2 + \cos\varphi_1 \cdot \cos\varphi_2 \cdot \left(\sin\left(\frac{diff\lambda}{2}\right)\right)^2$$
$$c = 2 \cdot \tan^{-1}(2\,(\sqrt{a}\,, \sqrt{(1-a)}))$$
$d = R \cdot c$
IF $d > D_{QCI}$ **THEN**
    No IEEE802.11af Access
ELSE
    IEEE802.11af Access Allowed
    N_{SSID} **= SSID of BS Identified**
    N_{TTL} **= TTL for country**
    N_{MTU} **= MTU (512bytes)**
END IF

By implementing these parameters, the BS distances in Tables 6 and 8 are upheld so maximising the probability of a packet being received. These distance values are the same as those obtained by using the GPS coordinates of the BS and mobile. It above all means a TVWS SU will not attempt to transmit a packet which will fail, so consuming valuable resources by needlessly circulating packets around the network until the TTL expires. The ANDSF algorithm then dynamically selects the most appropriate technology for the prevailing propagation conditions and the related QCI level required by the UE, so enhancing the overall QoS provision for the SU.

9 Conclusion

With bandwidth scarcity still a major bottleneck for 5G technologies, this paper has presented a novel TVWS IEEE 802.11af compliant access framework that enables a 5G network to fulfil its bandwidth and latency requirements by using a heterogeneous network arrangement that offloads data traffic according to *QoS class identifier* criteria. This not only enables 5G services to exploit TVWS spectrum, but crucially protects both SU access benefits and QoS provision by means of a routing strategy realised on the *Access Network Discovery and Selection Function* (ANDSF) server, which determines the most suitable heterogeneous technology to use. Since regulators allocate lower SU mobile powers, to achieve equi-distant coverage in both the forward and reverse links, an innovative routing approach is mandated. The new TVWS access framework accommodates this using a cross-layer routing algorithm to make access decisions based on both user QoS requirements and the distance of a SU from the BS. It critically addresses the inherent imbalance of SU transmit powers in the IEEE802.22, OFCOM and FCC standards, by allowing lower SU mobile powers, while concomitantly maintaining the coverage radius via a multi-hop MANET routing strategy in the reverse link.

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