AN INTEGRATED QOS METRIC FOR HANDOFF DECISIONS IN
5G AND LTE NETWORKS

by

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An Integrated QoS Metric for Handoff Decisions in 5G and LTE Networks

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Dedication

I dedicate this thesis to my parents and my sister.
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Abstract

AN INTEGRATED QOS METRIC FOR HANDOFF DECISIONS IN 5G AND LTE NETWORKS
Srinivas Sandhya Rani Siva Raju, MS
George Mason University, 2017
Thesis Director: Dr. Bijan Jabbari

Growing demand for wireless data and the spectrum congestion in the existing 4G LTE networks have led the researchers to focus on ways to utilize the large bandwidths available in the above 6 GHz spectrum. The technology that would be implemented in this spectrum has been termed as 5G is expected to roll out by the year 2020. While many models have been proposed and researched on the implementation of 5G, relatively less work has been done on one of the crucial aspects of cellular communications: handoffs.

Handoff is the process of transferring connection of a mobile user from one node to another, when the previous node cannot provide proper service. Current handoffs are performed, based on the signal strength. In this process, as the signal from the current node drops, the user then has to connect the node to a higher signal strength. While this method could work in micrometer wave spectrum, the propagation characteristics of a signal are different in the millimeter wave spectrum. Frequent drops in the signal level, even due to the user movement would mean that number of handoffs performed would be very high and would also lead to the ping-pong effect.

To avoid this, the use of an integrated QoS metric for the handoff decision has been proposed. In this method the handoff is performed not based on the signal strength level but
by considering past values of the each base station has been made. For the weighting criteria the normalized variance of SINR values of each base station, and the probability of missed detection has been considered. This ensures that the mobile user would connect to a more stable base station thus avoiding frequent unnecessary handoffs. A simulated approach has been taken to compare the performance of the weighted handoff with the traditional signal strength approach based handoff. The simulations were done in a 5G radio propagation framework using Matlab and in the LTE model of the Network Simulator 3 (NS3) software. The results show that the number of handoffs performed were lesser in the weighted handoff as compared to the traditional handoff, which was observed in both the 5G propagation simulation in Matlab and in the LTE model of NS3. In addition to this, analysis on the effects the weighted handoff would have on different network QoS parameters-the mean delay, the packet loss ratio and sum of jitter, has been done on NS3 LTE. Results show that the new algorithm doesn’t just abide by the LTE QoS, but also shows an improvement.
Chapter 1: Introduction

The increase in number of mobile users and the mobile IP data traffic has put a strain on the existing spectrum. Efforts to handle this increased usage and spectrum scarcity has led to focus on the next evolution in cellular communications called 5G which will be implemented in the high frequency, mmWave spectrum. In this chapter, an introduction to the issues present in the propagation of waves in the mmWave spectrum has been given. A overview of the handoff procedure and its evolution in cellular communications has been explained. Based on these two topics, the problem addressed by this thesis has been stated, followed by the organization of the thesis.

1.1 5G Radio Propagation

It has been predicted that by 2021 mobile data traffic will increase seven fold, accounting from just 7% of the total IP traffic in 2016 to 17% of the total traffic in 2021 [5]. The number of mobile users is also predicted to increase by a CAGR of 47% by 2021 [5]. This has led to a strain on the current available spectrum. Increased mobile IP traffic and strain on available spectrum has prompted research on vastly unused frequencies above 6 GHz also called the mm Wave spectrum. The technology that would be implemented in this spectrum has been termed as 5G technology and is expected to roll out by the year 2020. The fundamental issue with implementing 5G, is the properties of mmWaves that make is difficult to propagate. Studies have shown that mmWave at 60 GHz suffer 28 dB more losses than 2.4 GHz [6]. mmWaves undergo high attenuation in their path, which limits their

\[ \text{Licensed bands by FCC as of 2017: 27.5–28.35 GHz, 37.6–38.6 GHz and 38.6–40 GHz} \]
range of use mostly upto 200 m. A few studies have been performed by Korea [7], Nokia Siemens Networks [8], University of Surrey, England [9], and the NYU-Wireless Institute [4] to analyze the propagation of mmWaves in indoor and outdoor environments and to come up with a proper architecture to implement 5G.

**Propagation in mmWaves:** While there are different pathloss formulae used to calculate the pathloss, the general Friis formula for pathloss can be given as:

\[
\text{Free space path loss } L = \left( \frac{4\pi df}{c} \right)^2
\]

\[
L_{db} = 20 \log_{10}(d) + 20 \log_{10}(f) + 92.45
\]

where \( f \) is the frequency in GHz, \( d \) is the distance in km. This formula shows that the path loss is directly proportional to the frequency. As we move from the traditional 2.4 GHz bands to above 10 GHz bands, the path loss increases and the acceptable distance a wave can travel reduces vastly. Even a change from 2 to 4 km can increase pathloss by 6 dB. On top of that, above 10 GHz the power restrictions on devices are also different and for example in Europe above 10 GHz the transmission power cannot cross 30 dBm. These challenges make propagation in mmWaves a difficult issue. Hence up until now, mmWave bands have only been used indoors and mostly in line of sight communications (LOS). Propagation losses in the mmWave spectrum can be classified as:

### 1.1.1 Atmospheric Losses

One of the limiting factors inherent in the use of millimeter waves (30-300 GHz) is the considerable atmospheric attenuation caused by the absorption phenomena due to rain drops, water vapor and oxygen [6], [1]. With reference to an ideal free-space link, the effect of the atmospheric absorption is usually expressed in terms of specific attenuation, and is
customarily measured in dB/km. The main atmospheric phenomena causing signal loss is explained below.

**Gaseous Absorption:** The atmospheric gases that mostly affect mmWave propagation are water vapor and oxygen. Water molecules have an electric dipole moment whereas oxygen exhibits a magnetic dipole moment. When interacting with radio waves at a certain frequency, the above moments produce forced rotations of the molecules that cause dissipation of part of the energy carried by the propagating field. This absorption is maximum at certain frequencies which correspond with the resonance frequencies of the gas molecules. Hence at certain frequencies radio waves show higher attenuation in the environment. The effects of water vapor and oxygen on radio frequencies are shown in Fig 1.1.

![Figure 1.1: Specific water vapor and oxygen attenuation versus frequency, 10-400 GHz, T = 20o C, p_atm=1 atm, h=0 km, δ_w = 7.5 g/m3 [1]](image)

But in between such frequencies, there are windows of frequencies that can be used for
mmWave communication. Water vapor exhibits three absorption peaks due to molecular resonance phenomena at 22, 183 and 323 GHz. Unlike oxygen attenuation, propagation at these frequencies must be avoided at all costs. Molecular oxygen exhibits 45 distinct resonance peaks between 48 and 72 GHz, that all merge in a seamless attenuation curve at low altitudes (high atmospheric pressure) due to molecular collisions. However unlike water vapor, researchers are trying to exploit the absorption effects of oxygen, and active research has been going in 60 GHz band especially, as the co-channel interference obtained here greatly reduces the outage probability and the need for separation between signals [1].

60 GHz wireless systems operate at an oxygen absorption peak, reaching a maximum of 15 dB/km absorption at sea level. This high level of attenuation severely limits link distances, making 60 GHz useful for only short-distance transmission. Some equipment vendors offer outdoor communication links that deliver 1 Gb/s data reliably over distances of 400–800 m using 30–60 cm antennas. The high atmospheric attenuation adds advantages such as high-frequency reuse and secure communications because of the difficulty in eavesdropping. Also for out-of-atmosphere communications, for example, between satellites, 60 GHz excels as the oxygen absorption limitations disappear and essentially free space communication conditions exist.

Rain: Attenuation by rain depends on the size of rain drops, the amount of rain and the shape of the drops. While it is not possible to completely avoid this effect, statistical data on the history of rainfall can be used to predict the number of times rainfall will exceed ρ which is called the complementary distribution function of the rate of rain. If the specific attenuation $\alpha_o = a\rho^b$ at a rate of rainfall $p(\rho_o) = 0.01\%$ then the generic attenuation for a rate of $\rho$ can be given as [1]:

$$\alpha_{min}(\rho) = \alpha_o \left[ \frac{p(\rho)}{0.01} \right]^{-v}$$  \hspace{1cm} (1.3)
where

$$v = \begin{cases} 
0.33 & \text{if } 0.001\% < p(\rho) < 0.01\% \\
0.41 & \text{if } 0.01\% < p(\rho) < 0.1\% 
\end{cases}$$

The 70/80 GHz bands operate in an atmospheric window where clear air absorption is less than 0.5 dB/km, meaning that links can transmit across many miles (however actual practical limits are much shorter). The tradeoff of using this band is that through extensive data on rainfall patterns, one could predict when the attenuation could be highest. Since rain fall attenuation can be around 30 dB, this counts as a outage situation and hence communication is shut down for a short period as such outages would be for a shorter duration. While initial data by the ITU shows a 99.99% availability of a 1 Gb/s link, it still faces outages with a minimum duration of at least 5 mins [1]. The reason the 70/80 GHz, is being considered is that, apart from effects of rain, its small 1 mm wavelength band is practically unaffected by other factors like water vapor, sand, airborne particles.

1.1.2 Environment Obstacle Losses

The short wavelength of mmWaves makes it more susceptible to reflection than diffraction making it difficult to use over long distances especially if there are obstacles in the path. Due to this Non-Line of Sight (NLOS) communication is difficult to achieve at such small wavelengths.

Scintillation: It is the distortion represented by random fluctuations of the refraction index due to tropospheric turbulence that affect the electromagnetic waves during their propagation through low-atmosphere layers, and induce fluctuations of the received signal amplitude and phase. It is usually characterized by the variance and the amplitude spectrum of the signal fluctuations. Experimental measurements show that the scintillation effect, though enhanced in the absorption band, can be neglected in the vast majority of practical
applications.

**Foliage Loss:** Though not as dominant as the others, can be a significant factor in mmWave communications and it can be represented by the formula which is applicable for depth up to 400 m:

\[ \text{Loss } L = 0.2f^{0.3}R^{0.6}\text{dB} \quad (1.4) \]

where \( f \) is the frequency of the signal and \( R \) is the depth of the foliage. This relation is applicable from 0.2 GHz to 95 GHz. Also any animals or birds present could degrade line of sight for mmWaves causing outages.

**Scattering/Diffraction:** If there is no Line of Sight (LOS) path between the transmitter and the receiver, the signal may still reach the receiver via reflections from objects in proximity to the receiver or via diffraction or bending. The short wavelengths of millimeter-wave signals result in low diffraction. Like light waves, these signals are subjected more to shadowing and reflection. NLOS communication is possible only if the reflected multi-path wave has considerable energy. However for mmWaves, the waves diffuse than reflect resulting in lower power received at the receiver. The Table 1.1 shows the average attenuation mmWaves face by different types of materials and objects

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<th>Attenuation dB/cm</th>
<th>Obstacle</th>
<th>Attenuation dB</th>
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<tr>
<td>Concrete</td>
<td>6.67</td>
<td>Pedestrian</td>
<td>7-8</td>
</tr>
<tr>
<td>Glass</td>
<td>6.05</td>
<td>Tree branches</td>
<td>10</td>
</tr>
<tr>
<td>Stone</td>
<td>5.73</td>
<td>Car</td>
<td>10-14</td>
</tr>
<tr>
<td>Marble</td>
<td>1.25</td>
<td>Bicycle</td>
<td>4</td>
</tr>
<tr>
<td>Wood</td>
<td>4.22</td>
<td>Motorcycle</td>
<td>4</td>
</tr>
<tr>
<td>Plasterboard</td>
<td>1.51</td>
<td>Bus</td>
<td>16</td>
</tr>
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</table>


1.2 Handoffs in cellular communications

The procedure of handoff and a short survey on how handoff is performed in different technologies is provided here [2]. While the core idea behind the handoff might seem similar in a few of the below listed technologies, certain parameters like, the architecture, range of communications technology, and QoS requirement changes.

Handoff Procedure  Handoff is the procedure where there is transfer of the mobile connection from one source (ex: base station) to another. The reason a handoff is implemented varies on different factors but is mostly based on the drop in signal strength, increase in latency, network provider configurations, user speed. Handoff plays a crucial part in wireless networks and affect a lot of parameters like the Quality of service (Qos), capacity, data rate.

Hence generally handoffs are required to follow a set guidelines to avoid poor performance:

- The handoff latency should be as low as possible. Longer handoff latency could lead to packets being dropped and degraded cell service.
- The average number of handoffs should be limited. This is often represented by the ratio of the total handoff attempts to the number of successful handoffs, which should be maximized. Repeated handoffs are also called ping-pong handoff as the mobile user is connected and reconnected frequently sometimes to the same cell.
- A handoff should have as little effect on the quality of service as possible. The probability of dropping a call and blocking should be minimized while network services is balanced between handoffs.

Handoff in GSM  Handoffs in GSM systems can be divided into four types based on the movement between Base Station Controller (BSC) and Mobile Switching Center (MSC).
1. **Intra-Cell handoff**: The MS changes the channels (time slots) in the same cell or the same base transceiver station (BTS).

2. **Inter-Cell handoff within the same BSC**: The handoff when a user travels from the current cell to another cell controlled by the same BSC.

3. **Intra-MSC handoff**: The handoff when a user travels between two cells belonging to two different BSCs controlled by the same MSC.

4. **Inter-MSC handoff**: The handoff when the MS travels between two cells belonging to two different BSCs controlled by two different MSCs. Handoffs in GSM, could be initiated by the mobile user or by the networks provider, which many network providers would be able to define based on multiple parameters like delay, strength.

**Handoff in WLAN** WLAN techniques were made for the 2.4 GHz and 5 GHz band and their processes is described by IEEE 802.11. Handoffs in WLAN technology takes place in 3 steps.

1. **Discovery**: The stage when the MS scans to the discovery signals sent by the Access Points (AP). The MS can listen in passive mode (scanning all broadcasts sent by AP) or active mode (probing a particular APs which will then respond).

2. **Re-authentication**: After the discovery phase, once the most suitable AP is found, the MS starts the authentication procedure with that AP.

3. **Association**: Once the authentication has been completed, the MS begins it association with the new AP by sending its information about bit rate, ID etc.

The major drawback of handoffs in WLAN is the latency induced by the handoff which accounts for about an extra 50 ms, which doesn’t make it a seamless handoff especially if the MS is running any delay sensitive applications.
Handoff in UMTS  As UMTS is the 3rd generation of mobile networks (also called 3G), and an upgrade over GSM, its handoff process is in many ways similar to GSM. However UMTS has the added functionality of WCDMA in its physical layer which allows it to further classify its handoff into 2 types as shown in Fig 1.2:

1. **Hard Handoff:** This is also called as break-before-make handoff. Whenever a handoff takes place, the MS first loses its connection to the older Base station and then is transferred to a newer one. Hence this reduces handoff latency greatly and is also used in LTE networks.

2. **Soft Handoff:** This is also called make-before-break handoff. Whenever a handoff takes place, the MS is first connected to the new Base station before it loss connection with the older one. Hence here the older base station needs to transfer the connection and information of the MS to the new base station before it can cut it off. This type of a handoff takes longer than the hard handoff and consumes more network resources.

Figure 1.2: Hard vs Soft handoff
Because of the importance of the 3G architecture and its interoperability with other networks, its handoff procedure has been widely modified and used in different networks. The basic steps to perform this handoff consists of 6 steps, which have been modified in their own ways for different results, but the core procedure is somewhat similar.

(a) Step 1: MS sends measurement report to the node in response to measurement request

(b) Step 2: Based on the reports sent by the MS, the Radio Network Controller (RNC) decides to whether perform a handoff or not.

(c) Step 3: RNC asks MSC to reserve resources if the decision to make a handoff is selected.

(d) Step 4: The MSC respond with a acknowledgement (ACK) signal.

(e) Step 5: On receiving the ACK signal, the node then passes a handoff command to the MS

(f) Step 6: Handoff is performed by sending a handoff access message.

**Handoff in WiMax** Wireless Interoperability for Microwave Access (WiMAX) as the broadband wireless access technology is built on OFDM technology with good performance even in NLOS environment. WiMax works in two ways: Fixed and Mobile. Fixed WiMax as the name suggests has been optimal in a movement restricted LOS and NLOS environment with no handoff support. Mobile WiMax or Nomadic WiMax is based on 802.16e standard to provide support for mobile users with support for handoffs. It supports 3 types of handoffs namely-Hard handoff (default), Macro Diversity handoff (MDHO) (optional soft handoff), and Fast Base Station Switching (FBSS) handoff (optional soft handoff). While the hard handoff has the advantage of low complexity, it suffers from a high handoff latency between
60–300 msecs. The other soft handoffs have been shown to give much better performance as mentioned in [2] showing a latency at most upto 50 msecs.

1.3 Problem Statement and Proposed Solution

While the current handoff algorithm does provide a good quality of service, it cannot be used the same way in mmWaves. The propagation of signals in the mmWave spectrum is much different, facing high attenuation and atmospheric absorption. In this thesis an alternative, joint metric, handoff decision approach has been proposed. Instead of taking just the signal strength values as the criteria for a handoff, the modification decides to store continuous previous values of each base station. Then the calculation of the variance of the signal to noise and interference ratio of each base station, along with the probability of missed detection, is stored as a weighted parameter. This weighted value of each base station is monitored by a central controller, which then decides which base station the handoff would be done to.

The simulation of the modified algorithm has been done in a mmWave environment and a LTE setup. For the mmWave model, stochastic geometry has been used in Matlab. The performance of the new algorithm in comparison to the traditional handoff algorithm has been done using the number of handoffs performed as the criteria. For the LTE setup the NS3 LTE Lena project was used. Comparison of the new algorithm has been done with the native NS3 algorithm using the number of handoffs along with 3 QoS parameters viz the mean delay, the packet loss ratio and the sum of jitter. The results show that the new handoff method reduces the number of handoffs executed while also improving the QoS of the network and could prove to be a solution to the handoff issue in the mmWave 5G technology.
1.4 Organization of thesis

The rest of the report is organized in the following way. In Chapter I the characteristics of mmWave spectrum was explained followed by a brief survey on past handoff algorithms. In Chapter II emphasis is put on the current handoff procedure used by LTE and the issues with using them in mmWaves has been explained, the Weighted handoff model has also been explained. In Chapter III the methodology used to simulate a handoff scenario in a 5G radio propagation environment, in Matlab has been explained and the results obtained in Matlab have been explained. In Chapter IV, the Network Simulator 3 architecture is introduced, and simulation in NS3 has been explained and the results obtained is shown. In Chapter V, a conclusion by addressing the entire work of this thesis and certain proposed future steps that can be taken, are presented.
Chapter 2: Handoffs in current system

2.1 Introduction

In this chapter the handoff procedure in current LTE standard will be briefly explained and the analysis of the problem with using the current handoff in mmWave is explained. Following on work done previously in [10], the handoff rates in a mmWave scenario for different obstacle scenario is derived and plotted. The proposed joint decision weighted handoff model is introduced and the system model used for the simulation is explained.

2.2 Handoff in LTE

4G LTE is the current standard of cellular communications in most parts of the world. Able to achieve high data rates, it can handle applications like online gaming, video calls with ease. The LTE architecture consists of eNodeB(eNBs), Mobility Management Entity (MME) and System Architecture Evolution (SAE) which includes the Software Gateway (SGW). Unlike 3G, LTE doesn’t support soft handoffs and only works with hard handoffs. In LTE, network controlled, UE assisted handoff is performed, wherein the eNBs makes the handoff decisions without the involvedment of the MME/S-GW. All necessary handoff information is exchanged between the eNBs via X2 interface. A handoff complete message is sent to the MME/S-GW once a new connection is established between UE and the target eNB, which helps to reduce the signaling load on the S1 interface. The entire handoff process in LTE has been shown in Fig 2.1. Similar to 3G, handoffs in LTE may be classified by the target system, frequency or by the method they are performed.
Intra eNodeB handoff refers to a case where the source and target cell reside in the same eNodeB.

Inter eNodeB handoff depicts a situation where the two target cells are located in two different eNodeBs. X2 or S1 handoff process needs to be initiated.

Inter eNodeB handoff with MME change. X2 handoff process cant handle an MME relocation, so S1 procedure must be used instead.

2.3 Handoff in mmWave

Because the research in mmWave technologies is still going on, relatively less work has been in the field of handoffs in mmWaves. In [11] The authors have proposed a handoff method in Virtual Cellular networks (VCN), where instead of having dedicated cells for users, different nodes form a virtual network around each user and utilize the entire bandwidth for each user,
which potentially increases capacity per user. However this method also depends on the total signal strength and threshold over which the VCN is evaluated. A handoff procedure for railway passengers in mmWave spectrum has been proposed in [12], wherein several nodes are distributed in the train which connect to a central controller which provides services to each node. In it, the handoff proposed, is basically done by the base station detecting a fall in signal strength and the central controller which knows the path and direction of the train, decides the next controller the nodes must connect to. While this method does improve capacity in trains, it depends on the central controller to keep track of the movement and location of each and every node and to know which controller would fall in its path next. In [13] an indoor cellular architecture has been proposed, in which each user is connected to a base station which is connected to a Supervisory Host (SH). The SH maintains a record for each user inside, and allocated spectrum to each base station using Wave Division multiplexing (WDM) while continuously monitoring its signal strength. When it detects a fall in the signal level it initiates a connection with a new base station. This again works on the principle of a central computer storing information on the signal strength of the user at each point and then deciding the handoff.

2.3.1 Issues using current handoff in mmWave

Most of the handoff algorithms explained above, rely on the assumption that a mobile once in the network is served by the nearest BS or strongest BS. This is due to only considering the path loss exponent model of radio propagation and to remove the effect of fading. While this method might apply for below 6 Ghz waves, above 6 Ghz the effect of fading needs to be taken into account. As explained in Chapter 1, propagation in mmWave spectrum varies due to many factors like blockages, movement of users, environment changes. In fact as shown in Fig 2.2, even a movement of few meters on the users part could potentially drop the SINR value by more than 20 db. Based on the simulations done in [14], a single access
point would be able to cover a distance of only a single room, hence even a complete indoor coverage would require many access points. To consider a handoff algorithm solely on the maximum SINR value of each BS would lead to a higher number of handoffs which degrades network performance.

Figure 2.2: SINR at 60 GHz [2]

A typical handoff is performed when the signal strength level of one station drops below a certain threshold level and there is a neighboring station which has a higher level, as shown in the Fig 2.3. Although hysteresis values could be used for better efficiency by reducing the number of handoffs, the higher the hysteresis value the greater would be the handoff delay. And for the mmWave scenario where the SINR could drop from 20 dB to 0 and rise again, a rather high hysteresis value would be required and since delay is an crucial aspect of 5G, increasing delays has to be reduced.
2.3.2 Handoff Rates in mmWave

From [10] an explanation of the problem of increasing handoffs in mmWave systems for different scenarios and an general estimation of the number of handoffs (based on only signal strength) a user would have to perform in a typical mmWave outdoor environment has been explained. Though the scenarios are not practical enough, they do give an idea on the rate of handoffs, for connections which depend highly on a LOS connection for each user.

**Fixed obstacle in the path of the user** In this scenario the user is walking in a street and is connected to base station 1. As an obstacle is encountered the signal strength drops and the user makes a handoff to base station 2 (assuming there are no other base stations on the other side of the street). Once the user is clear of the obstacle he once again makes a handoff to the original base station. In this way if there are many such obstacles, the user would keep changing base stations depending on the position of the obstacles on the street \((h)\) and the obstacle spacing \((s)\). Based on the assumption that each obstacle forces atleast 2 handoffs from the user, the interval between handoffs can be given as:

\[
I_{\text{handoffs}} = \frac{1}{2v \left( \frac{h}{s} \right) \left( 1 - \frac{h}{w} \right)}
\]
where \( v \) = speed of the user,
\( s \) = center to center spacing between obstacle
\( h \) = position of the obstacle
\( w \) = width of the street

From this equation the rate of handoffs per second can be calculated and has been plotted in Fig 2.7 for varying obstacle spacing and user speeds. It was found that for pedestrian speeds there is a handoff every second whereas for vehicular speeds this further reduces.

**Moving vehicles in the path of the user** In this scenario the assumption is that the obstacles are not fixed but are moving like buses or trucks. If a user is connected to base station 1, then when a truck passes the LOS connection between the user and BS is blocked, a handoff would occur. However once a truck leaves, the user would once again make a handoff back to original BS. In this way for every moving obstacle the user would make a handoff twice the rate of which would depend on the interarrival time of the vehicle. Hence the handoff rate in this case can be given as:

\[
\text{Handoff rate} = \frac{2}{T}
\]
where $T$ is the truck inter-arrival time in seconds. The rate of handoffs in Fig 2.5 has been plotted, for different inter-arrival times, assuming the speed of the truck as 30 km/hr. It was found that in this situation a handoff would occur every 0.17 seconds.

**Pedestrians in the path** Since the nodes would be mounted at a height more than a pedestrian’s, it can be safely assumed that the obstruction due to a pedestrian would be when the pedestrian is quite close to the user i.e towering over the user. Now modeling pedestrian movements and their effects on signal level itself is altogether different study, here an assumption is made that the pedestrian obstacles are in the following ways:

- A single pedestrian i.e a single person walks between the user and the node either while going in the opposite direction or walking faster ahead of user. The handoff rate in this scenario would depend on the pedestrian density in the sidewalk and can be estimated in similar to the vehicles scenario. It was observed here that with a pedestrian density of 3 in a 5 meter distance, there was 1 handoff per sec.

- Crowd of pedestrians or if there are sudden bursts of people weaken the signal strength between user and the node. Here it was assumed that all pedestrians including the
Orientation of user changes  While handoffs due to user orientation might not be a major cause, it is quite difficult to model such a scenario. Hence different scenarios were assumed and handoff rates were calculated. If the orientation of the user changes due to sudden movements, then the rate of handoff might occur in bursts typically in the range of ‘0.24–0.32 handoffs/sec’. If the change is due to the user’s normal activity without an impetus, i.e. without an burst of movement, then for an average orientation turn of 90 degrees, the handoff rate would be ‘1.33 handoffs/sec’. While these scenarios might be a bit general in nature they do give a look into the rate of handoffs in such scenarios. The plot for the rate of handoffs for the above explained situations has been plotted in Fig 2.7.
2.4 Joint decision Handoffs

Handoffs in mmWave systems has not been researched quite well as of now and using the existing signal strength based handoff system is clearly not a viable choice especially in mmWave systems. Hence the use of a handoff decision based on parameters which have a direct/indirect relation with the QoS of the network has been suggested. Such handoffs can be called a joint decision/weighted parameters handoff. Weighted handoffs or multi-attribute handoffs, have been researched quite alot in Vertical handoffs involving Wi-Max/WLANs’ [15]. The main idea behind using the weighted handoff is to reduce handoff effects on the network according to certain QoS parameters chosen either by the service provider or the user. In the weighted approach, weighted preference is assigned to each input parameter (depending on the preference) and a cost value is calculated for each base station. To ensure the service continuity and to maintain the promising QoS, the decision on which base station to make a handoff has to be taken properly. Typically attributes included in such algorithms are the cost, delay, available bandwidth, user policies, applications etc. While considering these attributes to make a handoff decision has been shown to work in vertical networks, in horizontal networks there has been less study. Since the received
signal strength handoff is not an appropriate choice in mmWaves, a joint decision algorithm is proposed. In order to reduce the frequency of handoffs, rather than to allow the UE to connect to a base station on its higher signal strength, it is important to decide the base station based on channel stability, quality and occupancy.

2.4.1 Simple Additive Weighting

The technique used to calculate the weight is the Simple Additive Weighting (SAW) or Weighted Sum Method. It is a simplified and most prominently adapted multi-attribute decision mechanism. In SAW a score of evaluation is calibrated for each alternative (base station) by using the product of the scaled value assigned to the alternative of that particular attribute and the weights of relative importance directly assigned by decision maker followed by addition of each criteria. For this method, a direct rating on the standardized scales has been done exclusively in pure qualitative attributes. For each attribute $X_{ij}$ the score of base station $i$ is obtained by multiplying a weight $w_j$ to each attribute

$$V_{ij} = \sum_j w_j X_{ij} \text{ for } i = 1, 2, \ldots n \quad (2.3)$$

The decision matrix for 3 base stations $A_1, A_2, A_3$ for attributes $X_1, X_2, X_3$ can be given as:

$$V = \begin{pmatrix} X_{1j} & X_{2j} & X_{3j} \\ w_1 & w_1 & w_1 \\ w_2 & w_2 & w_2 \\ w_3 & w_3 & w_3 \end{pmatrix}$$

Based on the weights assigned to each attribute, the choice of base station selection differs. For some applications like FTP speed and quality is more important than stability hence the more importance can be given to the channel quality, whereas for applications like VoIP
or Video calling, connectivity is more important than a little loss in information, hence the
stability has to be given more importance.

2.5 System Model

In the proposed handoff algorithm, the attribute we choose are the variance of the SINR
of each base station and the probability of missed detection of each base station. The
variance has been used because as the user moves, the signal level would keep varying
either due to the movement of the user, environment changes. As was explained in [14],
even slight movements on the users part might cause a big drop in the signal level. These
drops trigger handoffs frequently, and even lead to the ping-pong handoff effect. To reduce
such handoffs, it is better to use the behavior of the base station as a parameter instead
of just the signal strength. As the user keeps moving, the values of the SINR of each
base station are monitored. The normalized variance of the SINR of each base station is
a direct indication of the performance of the base station compared to other base stations,
for the user, not just at that instance but along the entire path. Base stations with a higher
variance in SINR, would indicate the path along it to be prone to more signal strength
level drops or variations, whereas base stations with a lower variance in SINR, would be
considered as the more stable option. Further as more users values for each base station is
stored, the weight for each base station would improve wrt to others. But instead of using
just the variance of SINR, calculation of the probability of missed detection has also been
added. This is because, along with a base station’s stability, the quality of the channel at
a particular instance has to be considered. With the increase in number of mobile users,
interference among users is an important parameter to be considered. As mobile usage
fluctuates throughout the day, the number of users occupying the same base station at a
particular time varies. Even if a base station shows stable performance, interference among
users degrades service which might lead to some users not getting optimal service or being dropped, forcing further handoff. Hence along with the variance, the detection of multiple shared users on the base station is also taken into consideration. The missed detection probability is an indication if there are other users present on the same channel. The lower the probability of missed detection the better is the existing user detected. Hence the new user can make a handoff to a channel with less interference to existing ones.

System Description: In the proposed system, the signal received at the base station is constantly monitored and stored. Since the base stations work in cooperation, the signal strength at each base station is periodically sent to a ‘Central Controller’. The central controller keeps track of the system performance. In this case the performance of all the base stations that come under it, and makes the decision for the UE on which base station is the best one for it to connect to, for the initial access and subsequent handoffs. It first calculates the direction in which the UE should make its initial access connection by comparing the SINR:

\[
SINR_{\text{best}} = \max(SINR_1, SINR_2, SINR_3, \ldots)
\] (2.5)

Based on the base station with the largest SINR value, it instructs the UE to change its antenna direction in that direction, to connect to the best base station as shown in Fig 2.8. Now that the best ‘initial’ base station has been decided, the central controller has to monitor channel behavior to decide the base stations the moving UE should connect to. Here it is assumed that the UE is mobile and hence would at some point disconnect from its current base station and connect to a new one. It decides the handoff by maintaining the parameters such as the SINR values, variance of the SINR each channel and the probability of missed detection of each channel. The central controller maintains a routing table of values for each base station, sent each time. Since the SINR of each base stations is received

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constantly, the central controller can decide based on the movement of the UE, as to which base station has a better network quality for it to connect to when the time comes. Moreover since the variance of the UE is maintain and updated periodically, it allows to better capture the dynamics of the channel and bias the cell selection strategy of delay-sensitive applications towards more robust cells. The two parameters used are:

- Variance of the SINR values of all base stations
- Better probability of detection at each base station

The variance of the SINR values of each base station is calculated over a certain fixed period of time using the standard variance formula:

\[
var_1 = variance(SINR_1^1, SINR_1^2, ..., SINR_1^N) \tag{2.6}
\]

\[
var_1 = \frac{\sum (SINR_1)^2}{t} - \left( \frac{\sum SINR_1}{t} \right)^2 \tag{2.7}
\]
where the subscript ‘1’ denotes the base station 1 for ‘N’ SINR values considered here. The base station which has a lesser normalized variance value is considered as a better base station as its SINR values doesn’t fluctuate as much as the one with more variance. The central controller maintains a table of the weight values of each base station and decides based on the weight, of the neighboring base stations, the best one the UE should connect to. Also to consider the quality of the signal from the base station at that point, the probability of missed detection is taken into consideration. If there are more users connected to a base station, the arrival of a new user could force some of the users to vacate the spectrum leading to frequent spectrum handoffs, which at times can be avoided. Hence the probability of missed detection is considered here, for if the interference by the new users does not reduce the minimum transmission power required for the existing user, a handoff can be avoided. The probability of missed detection has been calculated using the energy detection method (explained in the next chapter) and can be given as:

\[ P_{md} = 1 - Q\left(\frac{\gamma}{\sqrt{\sigma^2/N}} - \frac{A}{\sqrt{\sigma^2/N}}\right) \]  

(2.8)

The weights for the algorithm is decided by the normalized, variance of each SINR and the probability of missed detection of each base station, which are calculated periodically by each Sounding reference signal (SRS) received. At each point when the UE has to perform a handover, it compares the weights of all the base stations. The base station with the minimum weight is then selected for the handover. Weight parameter for base station ‘n’ can be calculated as:

\[ \text{Weight } W_n = \min[\alpha \text{Variance}(\text{SINR})_n + \beta P_{md}] \]  

(2.9)

While there are many weighted averaging methods like TOPSIS, GRA, MEW, WPM available, for this simulation, the Simple Average Weighting (SAW) method is used, wherein an
'importance' is assigned to each input parameter i.e \( \alpha \) and \( \beta \). To measure the performance of the new model, simulations have been done in Matlab and in NS3 and has been explained.

\section*{2.6 Conclusion}

In this chapter, issues with handoff in mmWave spectrum has been analyzed. Considering different obstacle scenarios, the handoff rates in a general mmWave radio propagation scenario was calculated. It was shown that the current handoff scheme with mmWave base stations, would induce a high handoff rate with unnecessary handoffs. A modified handoff decision scheme based on the normalized value of the variance of the SINR and the probability of missed detection was proposed. The joint decision metric was calculated using a Simple Additive Weighting method, and was explained.
Chapter 3: 5G Radio Propagation Framework

3.1 Introduction

Since there are no mmWave simulators as of now which could be used to compare handoffs, a general framework was used to simulate a mmWave system in Matlab using the model given in [16]. The simulated model provides a good representation, of the signal characteristics as would be seen in a 5G mmWave radio propagation system. In this chapter, the simulated framework, with a Random Walk Mobility Model, has been explained. Further, derivations for the stationary distributions of the nodes, for the mobility model, used in the simulations, has been given. In the simulation both the strongest cell handoff algorithm as well as the joint decision handoff algorithm was used. For comparison between the modified and native handoff algorithm, the number of handoffs performed over increasing UE duration has been done.

3.2 Coverage Analysis in Matlab: SINR and Rate

Since, mmWaves are very sensitive to obstacles in the path of propagation, modelling a coverage area for such frequencies cannot be taken from previous channel models. For this system, certain concepts of stochastic geometry is taken and modelled for the 5G network parameters. Using the method proposed in [16] received SINR has been modelled with base stations placed according to the Spatial Poisson point process distribution. Stochastic geometry has rarely been used to design network models, but in an mmWave system, since a dense network of antennas for directional beamforming is considered, using stochastic
geometry and the modelling of blockages was done according to a random distribution. The line of sight (LOS) probability for each base station is done based on its distance from the user i.e users further away from a base station have a higher chance of being NLOS than LOS. A Bernoulli distribution model is used to decide the receiver path loss characteristics based on the LOS and NLOS probability distribution. The system includes a LOS distribution network, wherein the 5G propagation parameter values given in [4] for the city of New York has been used.

**Poisson Point Process (PPP):** In general a Poisson Point process is defined as a random object containing points randomly located in it. Spatial point processes, are used to represent the locations of random points or in this cases base stations, inside a plane. The Poisson distribution plays an important role in spatial point processes, because it gives the points the characterization of total independence and randomness. Unlike the Poisson distribution in the 1 dimension, in spatial point processes, Poisson distribution in 2-dimensions (the 2 co-ordinates of the locations) has to be used. The general Poisson point process can be given by the equation given below:

\[
P\{N(a,b) = n\} = \frac{[\lambda(x,y)]^n}{n!} e^{-\lambda(x,y)}
\]

(3.1)

where \(|\lambda(x,y)|\) denotes the mean or intensity. A non-homogeneous Poisson process defined in the plane \(R^2\) the intensity function is given by:

\[
\lambda(x, y) = \int_B e^{-(x^2+y^2)} dx \ dy
\]

(3.2)

where \(B\) is the bounded area of \(R^2\) of which \(x\) and \(y\) are the 2 Cartesian co-ordinates.

**Coverage Area Characteristics:** The advantage of using the PPP is the general randomness and independence it provides, wherein usually deciding the coverage area requires
knowledge of size, location, dimensions. Using the PPP, a general coverage area whose base stations density can be varied in the values of $|\lambda|$ of the PPP, and SINR can be calculated as a function of its LOS probability has been explained. The general system model for the stochastic framework has been shown in Fig 3.1.

![Figure 3.1: 5G Radio Propagation Framework in Matlab](image)

The random distribution of obstacles can be used to design both outdoor and indoor propagation, but here only the outdoor case is used for performing the simulations and analysis. Further it is assumed that interference between indoor and outdoor base stations is not present, hence an outdoor user can connect only to a outdoor base station. Similarly, the LOS and NLOS probability distributions can be used as given in [16], but here only the design for the LOS function was considered, where-in the distribution of all the blockages, are stationary and considered impenetrable. The area is designed for an outdoor scenario wherein the LOS probability of a base station is modeled as a distance dependent probability function where the locations of the base stations are assumed to be a non-homogeneous PPP.
Based on this system, simulations were carried out to calculate the SINR, rate, variance of SINR and the probability of missed detection, for each base station. To simplify the calculations the following assumptions were made:

- The LOS function is defined as its step function
- The obstacles are assumed to be buildings i.e stationary objects
- The base stations form a PPP with density ‘$\lambda$’ on the plane
- The mobile equipment user is outdoor and not surrounded by a concrete or glass wall
- Both the receiving user and the base stations make use of directional antennas

The LOS probability function to calculate the SINR is given as-

$$SINR = \frac{|h_o|^2 M_r L(R_o)}{(\sigma^2/M_t) + \sum_{l>0} D_l(\epsilon_l)|h_l|^2 L(R_l)}$$  \hspace{1cm} (3.3)

The variables used in the above equation is explained below:

1. $L(x)$: It is the pathloss gain which varies according to the value of ‘$x$’ which in this case is the distance denoted by $R_o$ or $R_l$. Different pathloss laws can be given for different links, where the general formula is given as:

$$L(x) = B(p(x))C_l R^{-\alpha_l} + (1 - B(p(x)))C_n R^{-\alpha_n}$$

where $B(p(x))$ is a Bernoulli random variable with parameter $p(x)$, which here is the LOS probability of independent non-homogeneous PPP base stations. $C_l$ and $C_n$ are the intercepts of the LOS and NLOS path loss. $\alpha_l$ and $\alpha_n$ are the pathloss exponents

2. $M_t, M_r$: They are the main lobe gain at the transmitter and the receiver respectively.

From the data given in [4] the values of the transmitter and receiver main lobe gain
has been considered.

3. $h_o$ and $h_l$: They are the independent small scale fading effects caused on the signal due to the various scatters and reflected waves arriving. For mmWave network it is more efficient to use Nakagami fading instead of the traditional Rayleigh or Riccian.

4. $\sigma$: It denotes the thermal noise generated. It is assumed it to be 0.6 for the simulation, which is a general value of thermal noise.

5. $D_l$: It is the total directivity gain from the base station to the receiver. Since a directional antenna array system was used, the directivity has to be taken into account for calculating the SINR values. The directive gain varies as the distance ‘l’ which is the distance from the base station to the user varies and FBR $\epsilon_t$.

6. $\epsilon_t$: It is the front to back ratio (FBR) of the antenna. For the directional sector antennas the FBR is defined as the ratio of the power transmitted in the forwards direction to the power transmitted in the backward direction.

The data rate of the channel for each base station has also been calculated using Shannon-Hartley theorem, which states that the channel capacity in bit/sec can be calculated using the formula:

$$R = B \log_2 \left( 1 + \frac{S}{N + I} \right)$$

where $R$ is the channel capacity calculated in bits/sec, $B$ is the bandwidth of the channel in Hz, $S$ is the signal strength in watts and $N$ is the average noise power in watts.

### 3.3 Probability of missed detection

Cognitive radios enable usage of the network based on how many existing users are connected, reducing interference between secondary and primary users. The main goal of
cognitive radios (CR) is to establish a communications link and maintain its robustness under conditions where the spectrum availability is constantly changing. It is important to get information on not just the channel quality but also its occupancy as interference between users could degrade cell connection leading to drops. Two parameters that play a crucial part in deciding how traffic will be bestowed is the probability of missed detection viz is the probability of a new user not detecting the presence of existing users in the channel when a user is present and probability of false alarm viz is the detection of a user when not present. The probability of missed detection depends on the SINR value of the signal, the channel noise characteristics and the detection threshold. This enables better UE connection to a base station at a particular point. A lower probability of missed detection would mean better occupancy of the channel and lesser interference introduced to the primary user present. Since the probability of false alarm is higher for a lower threshold, and the probability of missed detection is lower for a lower threshold, a trade-off is needed to be made between the two as shown in Fig 3.2.

![Figure 3.2: Probability of missed detection for increasing SINR](image)

Figure 3.2: Probability of missed detection for increasing SINR
A simple way to understand detection in cognitive radio is by working with these two equations:

\[
y[n] = \begin{cases} 
  w[n] & \text{if } H_0 \\
  w[n] + x[n] & \text{if } H_1
\end{cases}
\]  

(3.5)

where \(w[n]\) is the Additive White Gaussian Noise (AWGN), \(y[n]\) is the detected signal and \(x[n]\) is the transmitted signal. \(H_0\) is the absence of a primary signal whereas \(H_1\) is the presence of the primary signal in the channel. Using this, the probability of false alarm \(P_{fa}\) and the probability of detection can be inferred \(P_d\). From the above two equations:

\[
P_{fa} = P(\text{Signal Detected } | H_0)
\]

\[
P_d = P(\text{Signal Detected } | H_1)
\]  

(3.6)

Consequently the probability of missed detection is the probability of not detecting a signal when it is present.

\[
P_{md} = 1 - P_d
\]  

(3.7)

The various signal detection systems used include the matched filter detection, energy detection, cyclostationary detection and eigen value based detection which provide better performance but with increasing complexity.

### 3.3.1 Derivation of the Probability of Missed Detection

As stated above, in solving any binary hypothesis problem two conditions are considered:

\[
P_0 = P(H = H_0)
\]

\[
P_1 = P(H = H_1)
\]  

(3.8)
where the probability densities for a signal ‘y’ called the likelihood functions and can be conditioned as:

\[ H_0 = p_{y|H_0}(y|H_0) \]
\[ H_1 = p_{y|H_1}(y|H_1) \]  

(3.9)

Now for a noisy channel where a single information signal ‘s_m’ is sent the received signal y is:

\[ y = s_m + w \]  

(3.10)

where ‘m’ corresponds to \( H_0 \) or \( H_1 \) and ‘w’ is an independent Gaussian random variable with variance \( \sigma^2 \). From this the probability density function can be calculated as:

\[ p_{y|H_0}(y|H_0) = N(y; s_0, \sigma^2) = \frac{e^{-(y-s_0)^2/(2\sigma^2)}}{\sqrt{2\pi\sigma^2}} \]
\[ p_{y|H_1}(y|H_1) = N(y; s_1, \sigma^2) = \frac{e^{-(y-s_1)^2/(2\sigma^2)}}{\sqrt{2\pi\sigma^2}} \]  

(3.11)

Now based on that we can calculate the probability of missed detection and probability of false alarm. The basic detection formula as shown above is given by:

\[ y[n] = \begin{cases} w[n] & \text{if } H_0 \\ w[n] + x[n] & \text{if } H_1 \end{cases} \]  

(3.12)

Compared to the basic pdf of signal, in this case \( s_0 \) corresponds to a signal being absent or ‘0’ and \( s_1 \) corresponds to a signal being present or ‘A’. Hence

\[ H_0 : y \sim N(0, \sigma^2) \]
\[ H_1 : y \sim N(A, \sigma^2) \]  

(3.13)
Based on this the individual pdfs’ of ‘y’ correspond to tail probabilities in a Gaussian distribution and the functions can be defined using the ‘Q-Function’. Hence the probability of detection can be given as:

\[ P_d = P\{y > \gamma | H_1 = A \} \]  \hspace{1cm} (3.14)

where \( \gamma \) is the detection threshold.

\[ P_d = Q\left( \frac{\gamma - A}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.15)

And the probability of false detection is given by the detection of a signal when no signal is present i.e

\[ P_{fa} = P\{y > \gamma | H_0 = 0 \} \]  \hspace{1cm} (3.16)

\[ P_{fa} = Q\left( \frac{\gamma}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.17)

Therefore the probability of detection can be expanded as-

\[ P_d = Q\left( \frac{\gamma}{\sqrt{\sigma^2/N}} - \frac{A}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.18)

\[ P_d = Q\left( Q^{-1}(P_{fa}) - \frac{A}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.19)

Hence the probability of missed detection can be shown as:

\[ P_{md} = 1 - P_d \]  \hspace{1cm} (3.20)

\[ P_{md} = 1 - Q\left( \frac{\gamma}{\sqrt{\sigma^2/N}} - \frac{A}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.21)

\[ P_{md} = 1 - Q\left( Q^{-1}(P_{fa}) - \frac{A}{\sqrt{\sigma^2/N}} \right) \]  \hspace{1cm} (3.22)
where the term \( \frac{A}{\sqrt{\sigma^2/N}} = d^2 \) is called the deflection coefficient, which is a reasonable definition for the SINR. The above equation shows that the probability of missed detection depends on the probability of false alarm, wherein for a lower probability of missed detection a higher probability of false alarm would have to be accepted. Hence a trade-off needs to be made when deciding a proper probability of missed detection. Based on the above formula the probability of missed detection has been calculated by assuming a \( P_{fa} \) of 0.001 in the simulations.

### 3.4 Simulations and results in Matlab

To check the performance of the modified algorithm in an mmWave environment, a stochastic mmWave model was simulated in Matlab. The simulations done in Matlab have taken values from the experiments conducted by the NYU Wireless team in the city of New York [4], and is given in Table 3.1.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Matlab</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of UE</td>
<td>1</td>
</tr>
<tr>
<td>Number of eNB</td>
<td>3</td>
</tr>
<tr>
<td>Transmit Power</td>
<td>27 dBm</td>
</tr>
<tr>
<td>Mobility model</td>
<td>Random Walk Mobility</td>
</tr>
<tr>
<td>Fading model</td>
<td>Large Scale Nakagami</td>
</tr>
<tr>
<td>Antenna Gain (Receiver and Transmitter)</td>
<td>15 dBi</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>400 Mhz</td>
</tr>
</tbody>
</table>

### 3.4.1 Random Walk Mobility Model

The prediction of user movements is an important aspect of any MANET simulation, to get more practical and real life readings. There have been many models proposed which attempt
to mimic user movement from the real world, and choosing the right mobility model can have
significant effects on the simulation results [18]. A random walk mobility model was used
as shown in the Fig 3.3, where each node is assigned an initial location \((x_0, y_0)\), destination
\((x_1, y_1)\) and speed \(S\). The speed of the user is chosen independently and randomly from
the interval \((v_0, v_1)\) irrespective of the location and previous speeds. Additionally a ‘Pause
Time’ can also be added in which case it resembles a ‘Random Waypoint Mobility Model’.
The algorithm used for the simulations is given below and the stationary distributions of
the speed and location of the user has been explained [19].

![Figure 3.3: Random Walk Mobility Model](image)

**Stationary Distribution of Speed:** Consider a node traveling at a speed ‘s’ on a
path of length ‘l’, then the time spent on the path can be given as \((l/s)\). Therefore the
conditional density of \(f_s(s/l)\) of the speed for given path length, is proportional to \(f(s/l)\)
over path length ‘l’. Since the unconditional density of speed is proportional to the mean
path length $E[l]$, $f(s)$ can be given as:

$$f(s) = \begin{cases} 
\frac{1}{s \log(v_1/v_0)} & \text{if } v_0 \leq s \leq v_1 \\
0 & \text{otherwise}
\end{cases}$$

(3.23)

To sample the initial speed $s$ of the node we compute the cumulative distribution from the equation of $f(s)$ given above.

$$F(s) = \int_{v_0}^{s} f(t) dt$$  \hspace{1cm} (3.24)

$$F(s) = \frac{\log(s) - \log(v_0)}{\log(v_0) - (v_1)}$$  \hspace{1cm} (3.25)

The inverse of $F(s)$ can be given as:

$$F^{-1}(u) = \frac{v_1^u}{v_0^{u-1}}$$  \hspace{1cm} (3.26)

Hence by taking a uniform random variable $U$ between $(0,1)$ and using $S = F^{-1}(u)$, we can compute the initial speed of the node.

**Stationary Distribution of Locations:** The unconditional probability density function of the x-coordinate is an integral involving the joint density of $x_1$, $x_2$, $y_1$, and $y_2$ while the same can be applied to the y-coordinate as well. Since the speed is constant along a path, the location of the node is uniformly chosen between the start $(x_1, y_1)$ and end $(x_2, y_2)$ points. Hence the probability density of the x-coordinate of the node’s location is:

$$g(x|x_1, x_2) = \begin{cases} 
\frac{1}{|x_2 - x_1|} & \text{if } x \text{ between } x_1 \text{ and } x_2 \\
0 & \text{otherwise}
\end{cases}$$  \hspace{1cm} (3.27)
The proportion of time spent by a node on a path of a given length is proportional to the
length of the path. To see this, consider two paths of lengths $l_1$ and $l_2$. If nodes are traveling
on each of these paths, the expected times spent on the paths are $E(l_1/S) = l_1E(1/S|l_1)$
and $E(l_2/S) = l_2E(1/S|l_2)$. Since the speed $S$ is chosen independently of path length,
$E(1/S|l_1) = E(1/S|l_2)$, so the expected times on the paths are proportional to the lengths
$l_1$ and $l_2$. It follows that the joint density of the path endpoints $x_1$, $x_2$, $y_1$, and $y_2$ is given
by:

$$h(x_1, x_2, y_1, x_2) = k\sqrt{[(x_2 - x_1)^2 + (y_2 - y_1)^2]}$$

(3.28)

where $k$ is chosen to get the density integral to $l$ (i.e. the length of the side of the square
grid) and can be shown as:

$$1/k = \int_0^l \int_0^l \int_0^l \int_0^l \sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2}dx_1dx_2dy_1dy_2$$

Since the $x,y$ coordinates are identically distributed, the general formula of the unconditional
density of the coordinates can be given:

$$g(x) = 2\int_0^x \int_x^l \int_0^l \frac{k\sqrt{(x_2 - x_1)^2 + (y_2 - y_1)^2}}{x_2 - x_1}dx_1dx_2dy_1dy_2$$

(3.29)

While the above equation could be used to compute the endpoints, it is not feasible and
a simpler method was used by making a small assumption that the distance between the
2 endpoints are proportional to the joint probability density. The speed of a path $S$ is
independent of the path and hence the path length is dependent on the endpoints $(x,y)$.
The maximum path for a square of side length $l$, is $\sqrt{2}l$, and the path length between any
2 points inside the square has to be less than this length else, the endpoint coordinates are
recalculated. Once the condition is satisfied, the endpoints are selected and the initial point
is proportionally selected between them. The algorithm explaining the entire random walk
process has been given in Table 3.2.

Table 3.2: Random Walk Mobility Model Algorithm

<table>
<thead>
<tr>
<th>Steps for Random Walk Mobility Model</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Generate endpoints ((x_1, x_2, y_1, y_2)) uniformly</td>
</tr>
<tr>
<td>2. Calculate (r = \sqrt{[(x_2 - x_1)^2 + (y_2 - y_1)^2]/2l} )</td>
</tr>
<tr>
<td>3. Generate uniform random variable (U_1) between (0,1)</td>
</tr>
<tr>
<td>4. If (U_1 &lt; r) accept ((x_1, x_2, y_1, y_2)) else repeat Step 1</td>
</tr>
<tr>
<td>5. Generate uniform random variable (U_2) between (0,1)</td>
</tr>
<tr>
<td>6. Initial point of node is ((U_2x_1, (1-U_2)x_2)) and ((U_2y_1, (1-U_2)y_2)) whereas the endpoints are ((x_2, y_2))</td>
</tr>
<tr>
<td>7. Calculate speed of node based on its distribution given by Equation(3.26)</td>
</tr>
</tbody>
</table>

3.4.2 Results

In Matlab the number of handoffs performed by both algorithms was compared. A plot of the number of handoffs performed for increasing user movement duration has been plotted. The overall number of handoffs performed in the modified algorithm is significantly less as shown in Fig 3.4. As the total distance and time the user moves increases, the gap in the performance of both algorithms widen. This is because as more values are stored
the stability of base stations are better understood and the modified algorithm performs
lesser handoffs whereas the traditional handoff performs handoffs irrespective of the
past values and only based on signal strength.

![Figure 3.4: Average number of handoffs in Matlab](image)

**3.5 Conclusion**

The 5G radio propagation framework, simulated using Matlab was explained in this chapter.
The calculation of the SINR values of the base stations using this framework and the
probability of missed detection based on the SINR and signal threshold was derived. The
handoff algorithm, both native and weighted was used in the mmWave simulation. To
keep the results general, a random walk mobility model has been used. The stationary
distributions of the location and speed of the nodes have been derived and the simulated
path has been plotted. The average number of handoffs performed by both algorithms was
compared over increasing user movement and it was shown that the weighted algorithm
performs lesser handoffs as compared to the signal strength algorithm.
Chapter 4: LTE Simulation in NS3

4.1 Introduction

Network Simulator 3(NS3) is a discrete event based network simulator that is used for research purposes by simulating networks such as WiMax, Wi-Fi [3]. Codes in NS3 are compiled using the Python based WAF compiler. The LTE architecture was added to the NS3 software as part of the collaboration on the LENA LTE project. The software in place comes pretty close to emulating an actual LTE service though development is still taking place to further improve it. NS3 follows a software structure quite similar to the actual LTE architecture. However there are certain key differences (like periodic measurements) that are not supported as of yet in NS3. Measurements in NS3 are only event triggered, which means that an UE RSRP or RSRQ report can be used only if one of the 5 available events are triggered as shown in Table 4.1.

Table 4.1: List of event based triggering in NS3

<table>
<thead>
<tr>
<th>Event</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Event A1</td>
<td>Serving cell becomes better than threshold</td>
</tr>
<tr>
<td>Event A2</td>
<td>Serving cell becomes worse than threshold</td>
</tr>
<tr>
<td>Event A3</td>
<td>Neighbour becomes offset dB better than serving cell</td>
</tr>
<tr>
<td>Event A4</td>
<td>Neighbour becomes better than threshold</td>
</tr>
<tr>
<td>Event A5</td>
<td>Serving becomes worse than threshold1 AND neighbour becomes better than threshold2</td>
</tr>
</tbody>
</table>
4.1.1 Modelling in NS3

This section discusses about the different classes and models that were used to stimulate the system in NS3. Base stations and UEs’ in NS3 are denoted as nodes with topology helpers and models provided to configure the network.

- **Node Container**: Keeps track of the pointers to all the nodes or elements of the model ex: base station, UE. All the applications, IP protocol stack, mobility can be added to a node after it has been created using the Node Container.

- **Point to Point Helper**: Help install device and channel attributes. Data rate, delay, MTU are all installed using this class.

- **Internet Stack helper**: Like the Point to Point Helper, the Internet stack helper is needed to install the internet protocol on the nodes created by the Node container, the IP address, subnet mask for the nodes. Along with the Ipv4 InterfaceContainer, it helps create and maintain a list of the IP devices connected.

- **Position Allocator**: It is the class that decides the location of the nodes. Every eNB or UE, would need a position allocated to it. It accepts a vector of position co-ordinates (x,y,z) of each node. Any mobility that would be needed on the UE has to be added after it has a position allocated.

- **Random Walk Mobility model**: Its the mobility model class added to the UE. In each instance the UE moves with a speed and direction chosen at random with the user-provided random variables until either a fixed distance has been walked or until a fixed amount of time. The eNB on the other hand is designated a Constant Position model, wherein its location remains constant.

- **Application container**: Its the class which assigned the UDP socket to the client and
server applications. It starts at the start time of the simulator and end when the simulation ends.

- LTE Helper class: Helps create and configure all the LTE entities in our model. It handles the downlink/uplink channels, fading models, antenna model, the PHY, MAC and RRC on the nodes. More importantly it helps configure the MAC scheduler and the Handoff algorithm. Its sends the required attributes like hysteresis, TTT if mentioned to be used by the handoff algorithm.

### 4.1.2 Handoff in NS3

By default NS3 supports three handover algorithms:

1. **Manual Handover**: As the name suggests in this case the program performs a handover manually to another eNB irrespective of the UE measurements.

2. **A3 RSRP Handover**: Here A3 is taken from EVENT A3 which is trigger event for the eNB to evaluate the measurements from the UE. Event A3 is when the RSRP value of a neighbor eNB becomes higher than that of the serving eNB. The algorithm check if the neighbor RSRP is higher than the serving by a value called ‘hysteresis’ till a certain period of time called the ‘Time to trigger (TTT)’. If both conditions are met the handover is performed. This sort of a handover is also called the Strongest cell handover and is shown in Fig 4.1.

3. **A2-A4 RSRQ Handover**: In this method of handover, the first event is triggered when the RSRQ value of the serving cell drops below a certain threshold. The algorithm then checks the RSRQ values of the neighboring cells. If a neighboring cell has a value greater than the serving cell by a threshold, only then is a handover triggered and is shown in Fig 4.2.
Figure 4.1: A3 Handoff [3]

Figure 4.2: A2-A4 Handoff [3]
4.1.3 Handoff flow in NS3

The handover algorithm operates at the source eNodeB and is responsible in making handover decisions in an automatic manner. It interacts with an eNodeB RRC instance via the Handover Management SAP interface. The handover algorithm interface consists of the following methods:

- **AddUeMeasReportConfigForHandover**: (Handover Algorithm \(\rightarrow\) eNodeB RRC) Used by the handover algorithm to request measurement reports from the eNodeB RRC entity, by passing the desired reporting configuration. The configuration will be applied to all future attached UEs.

- **ReportUeMeas**: (eNodeB RRC \(\rightarrow\) Handover Algorithm) Based on the UE measurements configured earlier in AddUeMeasReportConfigForHandover, UE may submit measurement reports to the eNodeB. The eNodeB RRC entity uses the ReportUeMeas interface to forward these measurement reports to the handover algorithm.

- **TriggerHandover**: (Handover Algorithm \(\rightarrow\) eNodeB RRC) After examining the measurement reports (but not necessarily), the handover algorithm may declare a handover. This method is used to notify the eNodeB RRC entity about this decision, which will then proceed to commence the handover procedure.

**Note**: For the *AddUeMeasReportConfigForHandover*, the method will return the *measId* (measurement identity) of the newly created measurement configuration. Typically a handover algorithm would store this unique number. It may be useful in the *ReportUeMeas* method, for example when more than one configuration has been requested and the handover algorithm needs to differentiate incoming reports based on the configuration that triggered them. The entire structure is shown in Fig 4.3.
A handover algorithm is implemented by writing a subclass of the \textit{LteHandoverAlgorithm} abstract superclass and by implementing each of the above mentioned SAP interface methods. The modified handover algorithm was developed similarly except for instead of waiting for an event to be triggered in-order to get values, periodic measurements were taken. This was done by setting the Threshold for the EVENT A1 & A4 low so that the event is continuously triggered to get readings for all base stations.

4.2 Evaluation parameters

Evaluation of the mean delay, packet loss ratio and jitter has been done in NS3. Getting these stats was done using the Flow Monitor module designed for NS3. Evaluation was done in 2 cases, one when the load in the system is very low and when the load is very high (close to 1).
4.2.1 Mean end-to-end packet Delay

The End-to-end packet delay is the time taken for a packet to be transmitted across a network from source to destination i.e if the arrival instant of a packet at a network is \( t_1 \) and its departure from the network is \( t_2 \) then the end to end packet delay of the \( n^{th} \) packet can be given as:

\[
\text{End to End Delay } d_n = t_2 - t_1
\] (4.1)

The end to end delay is an ordered sequence of packet end-to-end delays from the 1st packet; and it is a stochastic process. Hence the mean end to end delay is expressed as the sum of the delays divided by the total number of packets.

\[
\text{Mean End to End Delay } \mu_D = \frac{\sum_{i=1}^{N} d_i}{N}
\] (4.2)

The end to end delay is divided into the following delays:

- **Transmission delay:** It is the amount of time required to push all the packet’s bits into the wire. In other words, this is the delay caused by the data-rate of the link. For packet length, \( L \), in bits and link bandwidth in bps. For a link of 10 Mbps and packet size of 1500 Bytes, the transmission delay is approx 1.2 msec.

\[
\text{Transmission time } = \frac{L}{R}
\] (4.3)

- **Processing Delay:** It is the time required by the devices to process the packet headers. This along with the queueing delay is a crucial part of most network delay based researches. According to TR 125 912, the processing delay in LTE is approximately 12.2 msec including the eNB and UE processing and S1-U transfer delays.

- **Propagation Delay:** As the name suggests, it is the time required by the packet to propagate from the source to the destination. For the length of physical link \( d \) and
propagation speed in the medium $\mu$ and can be generally assumed to be negligible.

\[ \text{Propagation time} = \frac{d}{\mu} \]  

(4.4)

- **Queueing delay:** The queueing delay is the time required for the packet to get serviced by the queue or it is the total time a packet spends waiting in the queue and the time taken to get serviced. As each device would have fixed service rate of packets, packets form a queue in the network. Before explaining the queueing delay, there is another important theorem called ‘Little’s Theorem’ that needs to be addressed. Little’s theorem gives a relation between the number of packets in a system and the delay as follows:

\[ \text{Number of customers } N = \lambda T \]  

(4.5)

where $\lambda$ = average arrival rate of packets and $T$ = time average of the customer delay. Little’s theorem basically says that for larger number of packets $N$, there will be larger delay. This principle can also be extended to the waiting time in the queue as:

\[ \text{Number of packets in queue } N_Q = \lambda W \]  

(4.6)

where $W$ is the average waiting time in the queue. Using Little’s theorem, the queueing delay in a standard M/M/1 system can be calculated.

**Delay in the M/M/1 queueing system:** The M/M/1 queueing system consists of a single queueing station with a single server. Packets arrive in Poisson process with a rate of $\lambda$ and are serviced with an exponential distribution of mean $1/\mu$. Analysis of most queueing systems are done using a Markov chain, which has the importance of a memoryless property i.e packets in the system arrive and get
serviced independently. The average packets in the queue is,

\[ N = \frac{\rho}{1 - \rho} = \frac{\lambda}{\mu - \lambda} \]  

On using Little’s theorem the average delay as:

\[ T = \frac{N}{\lambda} = \frac{\rho}{\lambda(1 - \rho)} \]

\[ T = \frac{1}{\mu - \lambda} \]

where \( \rho = \frac{\lambda}{\mu} \) and is called the utilization factor of the system. The queueing delay of a network approaches steady state time average in 200 sec. Increase in utilization factor \( \rho \) can increase the network throughput but also increases delay drastically. It is important to understand that, for a stable queueing system \( \rho \ll 1 \), for if \( \rho \to 1 \) the system would become unstable and would not be able to handle the packet arrival rate and the delay would rise sharply and the average number of packets in the system \( N \to \infty \). Hence for a stable system the value of \( \rho \) has to be less than 1. For a load of 0.1 and inter-arrival time of 10 msec, the queueing delay is 1.1 msec.

### 4.2.2 Delay due to handoffs

The delay due to handoffs is an issue of concern in the design of handover algorithms in cellular mobile communication systems. If handover does not occur quickly, the quality of service (QoS) may deteriorate below an acceptable level and if the handoff occurs quickly it would lead to instability. The time involved in the handover process is mainly due to the time taken for measurement and processing of the signal strength and the time taken for making a decision for handover initiation, if actual
switching time is ignored or in the case of LTE it is the time between the UE sending the measurement report indicating handover and the time the UE sends the RRC-ConnectionReconfigurationComplete to the target eNodeB. An important parameter affecting handoffs is the hysteresis value, a larger hysteresis reduces unnecessary handoffs but increases handoff delay. Following the analysis used by [20], the calculation of the handoff delay has been done, keeping in mind the usual definition of a handoff occurring due to signal strength. The mean of the signal levels received from the two base stations separated by $D$ km at a point $d$ from communicating base station can be formulated as:

$$
\mu_0 = K_1 - K_2 \log(d) \quad (4.10)
$$

$$
\mu_1 = K_1 - K_2 \log(D - d) \quad (4.11)
$$

where $K_1$ and $K_2$ are the path loss. The handoff occurs at a point where the difference between the two signals is $h - \sigma$,

$$(h - \sigma) = (\mu_1 - \mu_0) = K_2 \log\left(\frac{d}{D - d}\right) \quad (4.12)$$

where $h$ is the difference in pathloss and $\sigma$ is the standard deviation in the received signal from the base station. Therefore

$$
\frac{2d - D}{D} = \frac{10^{(h-\sigma)/K_2} - 1}{10^{(h-\sigma)/K_2} + 1} \quad (4.13)
$$

For a user moving at a speed $v$, the delay due to the handoff can be calculated as:

$$\delta_h = \frac{d - D/2}{v}$$
Therefore

\[ \delta_h = \frac{D}{2v} \left\{ \frac{10^{(h-\sigma)/K_2}}{10^{(h-\sigma)/K_2} + 1} - 1 \right\} \]  

(4.14)

Now based on the size of the cell the value of \( \delta_h \) can be changed:

- For a macrocell: \( \delta_{hM} = \frac{T}{2} + K_{rv} \left\{ \frac{10^{(h-\sigma)/K_2}}{10^{(h-\sigma)/K_2} + 1} - 1 \right\} \) where \( K_{rv} \) is the normalized cell size in km, which depends on the distance between base stations and speed of vehicle and \( T \) is the averaging period. Inference can be made that for a normalized cell size \( K_{rv} \), the hysteresis value affects the handoff delay. Hence for larger cell sizes it is preferable to use a smaller hysteresis value to reduce the delay.

- For a microcell: In a microcell, the size of the cell is quite small, however the drop in signal level happens quite quickly, typically even a turn could drop the signal level by more than 15 db. Hence the handoff delay in terms of the signal drop level \( L \).

\[ \delta_{h\mu} = \frac{T}{2} + \frac{T(h - \sigma)}{4L} \]

In both cases the term \( \frac{T}{2} \) is added due to the delay occurring by the time averaging period. Plotting a graph of the handoff delay wrt the hysteresis for a constant normalized distance is shown in Fig 4.4. From it the delay that would occur due to the hysteresis in a strongest cell handoff approach can be approximately calculated.
For a time average of 5 msec, and no hysteresis, the average handoff delay is 2.5 msec which brings the total mean end-to-end delay, in the modified algorithm to 17 msecs. Whereas the signal strength handoff with a hysteresis of 4 db would produce an additional handoff delay of 5.4 msec and the average end-to-end delay in the native handoff is 22.4 msecs.

4.2.3 Packet loss

Wireless channels are known to be generating burst packet losses/errors due to the nature of the connectivity. Packet losses that may occur during the transmission of data in a network could be due to several factors: channel errors, network congestion or delay deadline violation. Packet loss ratio is measured as a percentage of packets lost with respect to packets sent. Most basic networking equipment will use FIFO queuing for packets waiting to go through the bottleneck and they will drop the packet if the queue is full at the time the packet is received. Other full queue mechanisms include random eviction or weighted random eviction. The major reason for packet loss in IP networks is the buffer overflow at
the outgoing interface in network nodes. If the aggregate IP traffic from multiple incoming interfaces to the outgoing interface is too high, the buffer of the outgoing interface will be filled up and at some point in time the buffer is full and new packets will be dropped, causing packet loss inside the network. The packet loss ratio is calculated by the number of lost packets divided by the number of total packet. For number of packets lost \( P_{nl} \) of the total number of packets \( P_n \), the packet loss ratio percentage \( p_L \) can be given as:

\[
p_L = \frac{P_{nl}}{P_n} \times 100
\]

To characterize the performance of the packet loss in bursty wireless channels, the method proposed by [21] has been used. It models the packet loss probability based on the maximum allowable delay and error correlation in the system. For a low load system, the packet loss probability depends more on the error probability and error correlation of the channel than the max delay or packet arrivals. A two dimensional Markov model is used with state space \{\((i,j)| - A + 1 < i < D, 0 < j < C\}\} where A is the packet arrivals and the D is the maximum delay in the system, C which is simplified version of the Gilbert-Elliott model [22], is the highest error state number. The error state ‘C’ can be given as:

\[
C = \begin{bmatrix}
C_{0,0} & C_{0,1} \\
C_{1,0} & C_{1,1}
\end{bmatrix}
\]

In the above matrix, the state 0 can be denoted as a success state and the state 1 can be denoted as a failure state. The 2-D Markovian model includes states \{\((i,j)\} \text{ where } i \text{ denotes the probability of the state of the packet for delay and goes from the inter-arrival time to the maximum allowed delay, whereas } j \text{ is the probability of the packet in the error state and goes from 0 to 1. Considering the case of a system with high load and high correlated errors, which is typically a worst case scenario, a simple but intuitive formula for the packet loss is given. From the error state matrix, the transition from a success state to a failure}
state can be denoted as $r$, i.e. $r = C_{0,1}$, and the transition from a failure state due to high
delay to low delay as $s$, i.e. $s = C_{1,0}$ and the maximum delay allowed as $D$. Assuming a high
error correlation and heavy load the packet loss probability general formula can be given as:

$$p_l = \frac{r(\rho r - s + \rho s)}{x^D s^2 (\rho - 1) + r(\rho r - s + 2\rho s)}$$

(4.16)

where $\rho$ is the load of the system and $x = \frac{1 - \rho}{1 - \rho(1 - r) - (1 - \rho)s}$

From this equation subsequent equations for the packet loss ratio can be calculated. For
low load case, where the error probability is also low, the packet loss can be given as:

$$p_L \approx (1 - s)D r$$

(4.17)

where the term $r/s \approx \epsilon$ is called the error probability of the system or time fraction a packet
remains in the error state. The packet arrival rate is not taken into account since a low load
condition was assumed. This is because the queue length is usually zero at the beginning
of the error period if the load is not very high, the packet loss probability (i.e. fraction of
packet lost) is largely independent on the load. However for a very high load, the queue
length may not be able to reach zero (due to very few empty slots) before the next error
burst occurs. In this case, the packet loss probability can be substantially larger, as when
$\rho \to 1$ the packet loss probability $p_l \to 1$ i.e every packet goes from a state of failure back
again. When the maximum delay is very high the packet loss probability decreases as the
duration for the packets to stay in the queue increases. Based on the above formula, for
a maximum allowed delay of 10 secs, error probability $\epsilon : 0.01$ and $s : 0.1$, the packet loss
ratio comes around 0.35%.
4.2.4 Sum of jitter

Jitter is an undesirable effect caused by the inherent tendencies of IP networks and components. Jitter is defined as a variation in the delay of received packets. The sending side transmits packets in a continuous stream and spaces them evenly apart. Because of network congestion, improper queuing, or configuration errors, the delay between packets can vary instead of remaining constant. Most devices use a jitter buffer at the gateway of the receiver to compensate for this. These buffers based on the algorithm they are running on, can reorder the packets in the correct order to make the effect of jitter seem negligible for the user. The size of the buffer and the amount of delay accepted can be configured by the user, hence high delay systems are more susceptible to the effects of jitter. The sum jitter is the total jitter experienced. It is an important and crucial parameter to be considered for network performance as high jitter can cause lip-sync errors, packet loss due to buffer over/under flow and buffering issues for all downlink transmissions especially with video playbacks. The sum of jitter depends on the packet arrivals, delay, and service time in the system. While jitter is an important aspect of any network QoS, modeling jitter is quite difficult and not much research has been dedicated to its analysis, as compared to the packet delay and packet loss. Previous works include [23] where they have calculated the jitter faced in differentiated service like the expedited forwarding and priority queueingweighted fair queuing, in [24] the evaluation of jitter with varying packet arrival rates has been given by keeping the utilization factor the same. While these papers do give a good modelling of the jitter they calculate it for a tagged periodic cell stream going through nodes of an ATM network so that the service time is constant which is not necessarily the case with IP traffic. Here the analytical model given in [24] which explains the jitter behavior wrt the variation to the delay and utilization factor upto a limit, is presented. Since jitter is defined as the
variation in the packets transfer delay:

\[ J_i = T_{j+1} - T_j \]  

(4.18)

or by the average or expected value

\[ J^{(N)} = E \left[ \sum_{n=1}^{N} |T^{(N)}_{j+1} - T^{(N)}_j| \right] \]  

(4.19)

Considering the stream of packets to be a Poisson process with:

- \( t_j \) = arrival time of packet \( j \)
- \( I_j \) = inter-arrival time of packet \( j \)
- \( r_j \) = departure time of packet \( j \)
- \( W_j \) = waiting time of packet \( j \)
- \( S_j \) = service time of packet \( j \)

Hence the transit time of packet \( j+1 \) can be given as:

\[ T_{j+1} - T_j = \max(S_{j+1} - T_j, S_j - I_{j+1}) \]  

(4.20)

Now since random variables \( T_n, S_{n+1} \) and \( I_{n+1} \) are independent, the jitter can be calculated as:

\[ J^{(N)} = E \left[ \sum_{n=1}^{N} |T^{(N)}_{j+1} - T^{(N)}_j| \right] \]  

(4.21)

If all the distributions are taken as exponential, Service time \( S(s) = \mu e^{-\mu s} \)

Inter-arrival time \( I(i) = \lambda e^{-\lambda s} \)

Transit time \( T(x) = \eta e^{-\eta s} \) where \( \eta = \mu - \lambda \)

The end-to-end jitter for a single node can be found as:

\[ J = \frac{1}{\eta} \left( 1 - e^{-\eta t_k} (\eta t_k + e^{-\eta t_k}) \right) \]  

(4.22)
where $\eta \tau_k \approx (1 - \rho)/\rho$. The general formula for jitter can hence be given as:

$$J \approx \frac{1}{\eta} \left[ 1 - e^{(\rho-1)/\rho} \left( \frac{1-\rho}{\rho} + e^{(\rho-1)/\rho} \right) \right]$$  \hspace{1cm} (4.23)

Assuming the distributions are exponential, i.e $T(x) = \eta e^{-\eta x}$, $S(s) = \mu e^{-\mu s}$ and $I(i) = \lambda e^{-\lambda i}$ the jitter in terms of the load, whether is lightly loaded ($\rho << 1$) or heavily loaded $\rho \rightarrow 1$. When $\rho$ becomes large, the value of $\eta \tau_k \approx 0$, hence taking the first order expansion of the exponential the value of the jitter can be given as.

- For lightly loaded i.e $\rho << 1$ : $J \approx \frac{1}{\eta}$

- For heavily loaded i.e $\rho \approx 1$ : $J \approx \frac{1}{\mu}$

While these formulae do not necessarily give the exact value as the simulated results of the sum of jitter found, they do explain the behavior of jitter wrt the utilization factor $\rho$.

For the steady state low load condition the sum jitter based on the delay in the modified handoff jitter value comes to 25.32 sec and for the strongest cell handoff jitter value comes to 30.04 sec as compared to 22.5 and 31.4 secs found in the simulation.

### 4.3 Results in NS3

To verify the advantages of the new model, comparison of the new method with the strongest cell handoff of NS3 has been done. The hysteresis value for the native NS3 algorithm was kept between at 4 dB. The random walk model was used with increasing simulation time. The final values were taken as the average, over distinct seed values. Both the algorithms were initially run upto 1200 seconds and the number of average number of handoffs performed by the user was noted in Fig 4.5.
The modified algorithm, NS3 performs a lower number of handoffs compared to the native algorithm at steady state. The simulation approaches steady state at 200 secs. The average handoffs in the modified handoff was 52 as compared to the 65 performed by the native handoff. Comparison of the 3 QoS parameters which are the mean delay, packet loss ratio and sum of the jitter, with both algorithms, has also been plotted.

In this chapter, the mean handoff delay was explained, as a function of the averaging time duration and the hysteresis. From the Fig 4.4, for a hysteresis value of around 4-5 dB, the mean delay of the system increases by around 5 msecs. Since the modified algorithm doesn’t have a hysteresis value, the additional delay is just the time averaging delay. Hence the native algorithm shows a mean delay of around 21.8 msecs as compared to 17.7 msecs as shown in Fig 4.6, and a difference of ‘4.1 msecs’. The analytical values of the mean delay has also been plotted. The analytical difference between the delay was calculated to 5.4 msecs.

The packet loss ratio is a function of not just the maximum delay allowed in the system but also the error correlation. The packet loss probability is dependent more on the queue
length and max delay. Hence at low loads, the queue is usually empty, so all packet trans-
missions are successful and correlation between burst of packet loss is lower. The packet
loss ratio in the modified handoff is 0.55% and for the native algorithm it is 0.4% as shown
in Fig 4.7 while the analytical packet loss ratio for an ideal case was calculated to be 0.35%.

The sum of jitter depends on the variations in the transit time for consecutive packets
in the queue. For a low load (in this case 0.1) the transit time would include the delay,
inter-arrival time and service time of the system. Hence as the load increases, the delay
increases and the jitter increases. In Fig 4.8 the total sum of the jitter for the modified
algorithm was around 22.5 seconds as compared to 31.4 seconds for the native handoff.
Whereas the analytical jitter was 30.04 sec for native and 25.32 sec for modified handoffs.

Figure 4.6: Mean delay in NS3
Further the plot for the mean delay, packet loss ratio and the sum of jitter for varying values of the utilization factor is given. For an increasing load, the number of packets in queue increases and the delay rises as the system cannot keep up with rate of packet arrivals.
Hence the delay rises with increasing utilization factor as long as $\rho < 1$, however when $\rho = 1$ the number of packets in the queue get so high that delay value would tend to $\infty$ as shown in Fig 4.9.

The packet losses are largely independent of the load until it reaches close to 1. When the load reaches 1, the number of slots available for packets reduces, hence the correlation between packet losses increases until the load becomes so high that every packet goes into the error state. The packet loss for increasing load has been plotted in Fig 4.10 and it can be be seen that when the delay becomes so high, practically no packets leave and hence the packet loss ratio reaches around 100% i.e complete packet loss.

![Figure 4.9: Mean delay vs. Utilization factor in NS3](image)

Figure 4.9: Mean delay vs. Utilization factor in NS3
The jitter in Fig 4.11 however shows a different response for increasing load as compared to the delay or packet loss. As the utilization factor increases the jitter seems to be increasing due to the delay in the system however as the load gets too high, the delay becomes quite high with very little variation. And the only jitter in the system is due to the variation in the service time of the system. Hence when the load becomes too high the jitter drops to the service time.
Figure 4.11: Sum of jitter vs. Utilization factor in NS3

4.4 Conclusion

In this section the NS3 simulations have been discussed. First a explanation of the NS3 LTE Lena architecture has been explained, the various models used for this simulation has also been listed. Both, the native and weighted algorithm was performed in the NS3 environment in a Random walk mobility model. The average number of handoff in the modified value is 52 as compared to 65 by the native handoff algorithm. Results were shown for the steady state mean delay, packet loss ratio and the sum of jitter. The modified algorithm shows an improvement of 20.3% in the mean delay, 28.14% in the packet loss ratio and 27.28% in the sum of jitter. Further the simulations were done against a varying utilization factor, wherein also the modified algorithm performs better than the native algorithm.
Chapter 5: Conclusion and Future Work

5.1 Conclusion

In this thesis the topic of handoffs in cellular networks especially the now emerging mmWave 5G network was covered. First the introduction of some concepts of 5G mmWaves, the issues related to its propagation and the technological solutions being pursued to enable it, was explained. Next, a brief survey of the various handoff methods in different cellular technologies especially the 4G LTE network was covered. The issues with using the traditional signal strength handoff in 5G was studied and performance predictions were made. Next, a modified joint decision handoff algorithm was proposed based on the normalized, variance of the SINR and probability of missed detection, which could serve as an better alternative for handoffs in mmWave 5G communications.

The proposed handoff was simulated in a 5G radio propagation model and the LTE scenario. To simulate a mmWave network, stochastic geometric approach was used for the coverage and rate analysis. The signal strength handoff and the modified handoff, was implemented and the results were compared. The weighted handoff algorithm shows a significant reduction in the number of handoffs executed. To further study the effects of the modified handoff the LTE model available in the NS3 Lena project was used. Comparison between the modified handoff algorithm with that of the NS3’s A3 strongest cell algorithm was simulated. The weighted handoff performs better than the native handoff by reducing the average number of handoffs at steady state from 65 to 52. Further comparison was shown on the 3 QoS parameters of the LTE network which are the mean delay, the packet
loss ratio and the sum of jitter. In all cases the weighted handoff performs better than the native handoff showing a improvement of 20.3% in the mean packet delay, 28.14% in the packet loss ratio and 27.28% in the sum of jitter.

5.2 Future Work

In this thesis an evaluation was done on the number of handoffs in a mmWave environment. Simulating such a handoff in a more sophisticated 5G radio propagation software will provide more insight into the extent of how the handoff would affect different QoS factors as it was shown in the LTE simulations. The Simple Additive Weighting (SAW), method was used to calculate the weights, whereas better methods like TOPSIS, GRA could prove to be better alternative into actually modelling the weight of each parameter and improving the QoS. Also the advantages of using a MIMO antenna system was not realized here, but with MIMO poised to be a part of the 5G architecture it could be considered in future simulations.
Bibliography
Bibliography


Curriculum Vitae

Srinivas Sandhya Rani Siva Raju received the B.E. (Bachelor of Engineering) in Electronics and Telecommunications in 2014 from the University of Mumbai. He then pursued his M.S. (Master of Science) in Electrical Engineering from the Department of Electrical and Computer Engineering at George Mason University. He has worked as a Teaching Assistant in the Electrical and Computer Engineering Department and as a Graduate Teaching Assistant in the Information and Technology Department. In the summer of 2017 he also worked as the IoT Prototype Development Intern in Société Européenne des Satellites (SES) Networks.