

### 3.1 Introduction

In this chapter OPNET 17.5 is used to simulate different scenarios namely IEEE 802.11a and n for three parameters in IPv4 and IPv6 environment (Throughput , delay and jitter) has considered to evaluate the network performance for IPv4 and IPv6 which are all Quality of services Measures .

### 3.2 Simulation Scenarios

To test our previous QoS strategy as a comparative method, in this study:

- Using OPNET 17.5 to implement our VoIP network.
- Setting up several nodes.
- Use an exponential traffic source to re-create a typical voice conversation over VoIP.
- Different protocols will be use.
- Measuring throughput, end-to-end delay, packet loss and jitter.
- Plotting our results and comparing them to our theory based predictions.

In our deployment all nodes configured to use the G.711 codic, Its formal name is Pulse code modulation (PCM) of voice frequencies, it is commonly used in VoIP application., this codic transmits information at a rate of 64kbps [18] .The network infrastructure is WLAN .OPNET 17.5 was used to simulate four different scenarios namely IEEE 802.11a, and n According to traffic analysis, three parameters: Throughput, delay and jitter has considered to evaluate the network performance IPv4 and IPv6. In each scenario there a small network represents light traffic and a large network with background traffic to generate the VoIP.

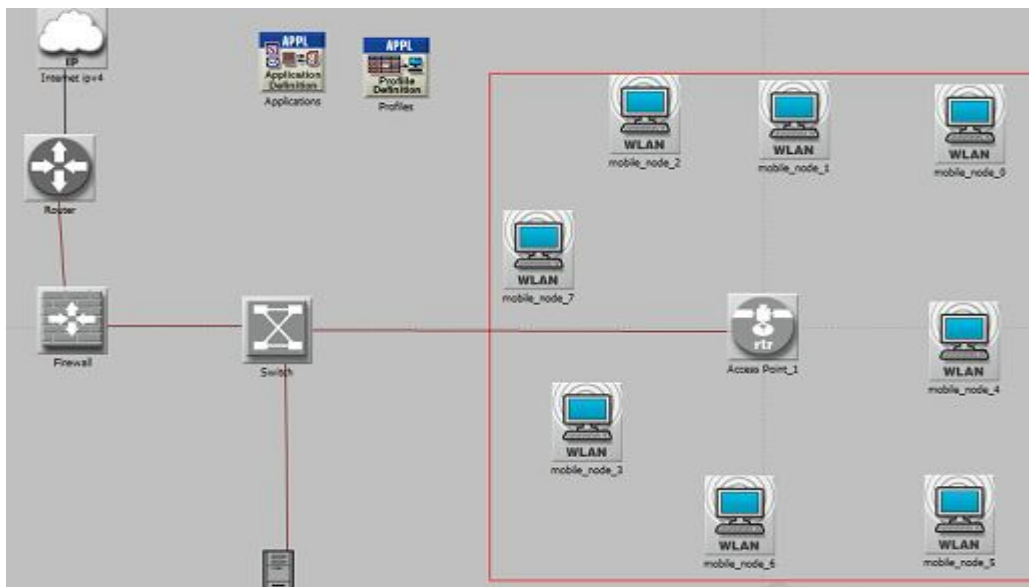
### 3.2.1 Network Designed Components:

This section discusses the main network components used in the suggested network models running on OPNET 17.5 and the devices used in. the **small network**: 8 WLAN work station, SIP server, 16 port switch Ethernet , Firewall, router, and 100 Base T full duplex for wired connection where used to build IP backbone for ipv4 and ipv6.

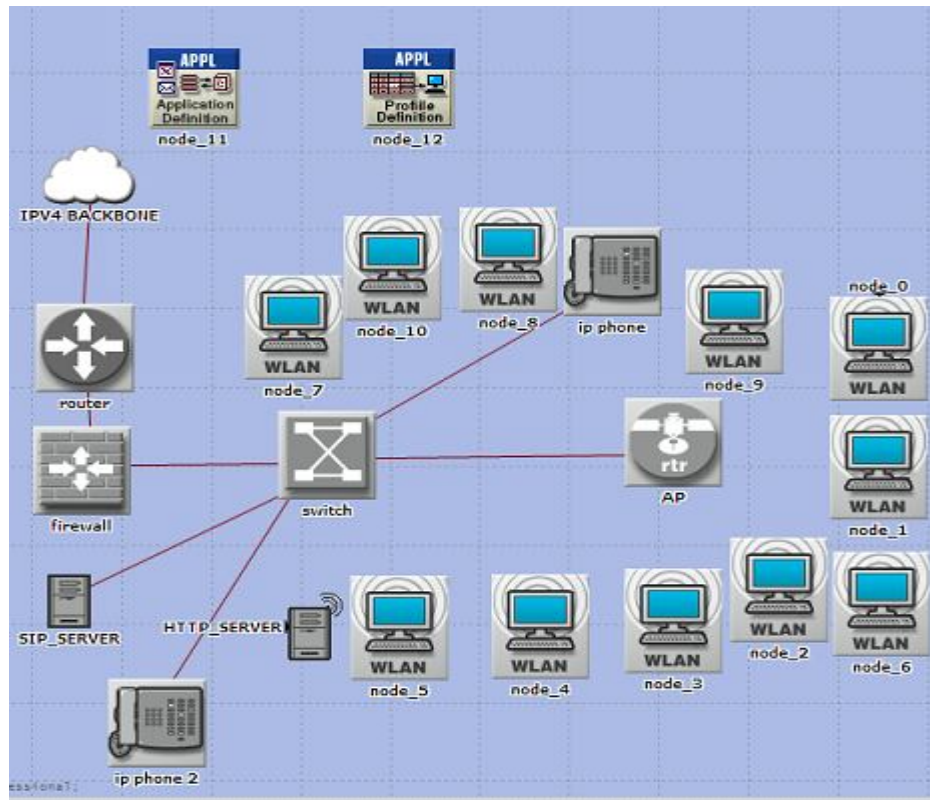
For **Large network**: 11 WLAN work station ,two wired IP phone , two servers (HTTP server ,video conference server for background traffic) ,SIP server, 16 port switch Ethernet, Firewall, router, and 100 Base T full duplex for wired connection where used to build IP backbone for ipv4 and ipv6.

#### Scenario (1)

Data rate 54 Mbps, Data rate 600 Mbps and numbers of nodes that use technology 802.11a, n using IPV4 small and large network in 240sec. see **Figure (3.1)** and **Figure (3.2)**



**Figure (3.1) The configuration of IPV4 Network showing VoWiFi small network**



**Figure (3.2) The configuration of IPV4 Network showing VoWiFi large network**

**Scenario (2):**

Data rate 54 Mbps, Data rate 600 Mbps and number of nodes that use technology 802.11a ,n using IPV6 small and large network in 240sec.see **Figure (3.3)** and **Figure (3.4)**

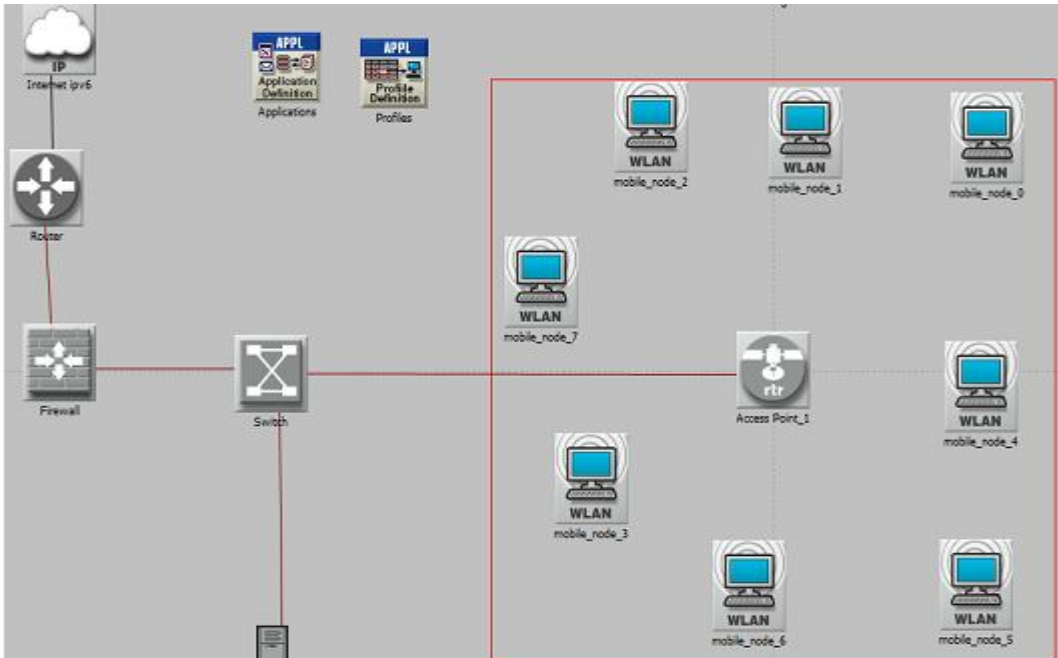


Figure (3.3) The configuration of IPV6 Network showing VoWiFi small network

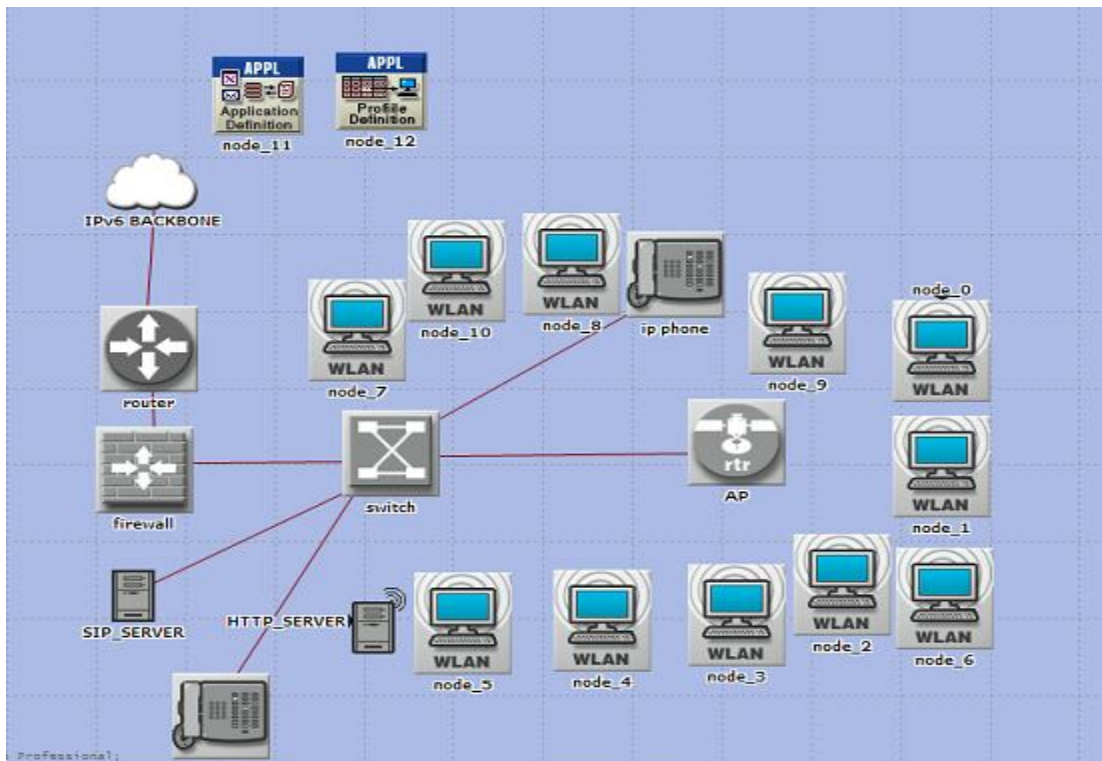
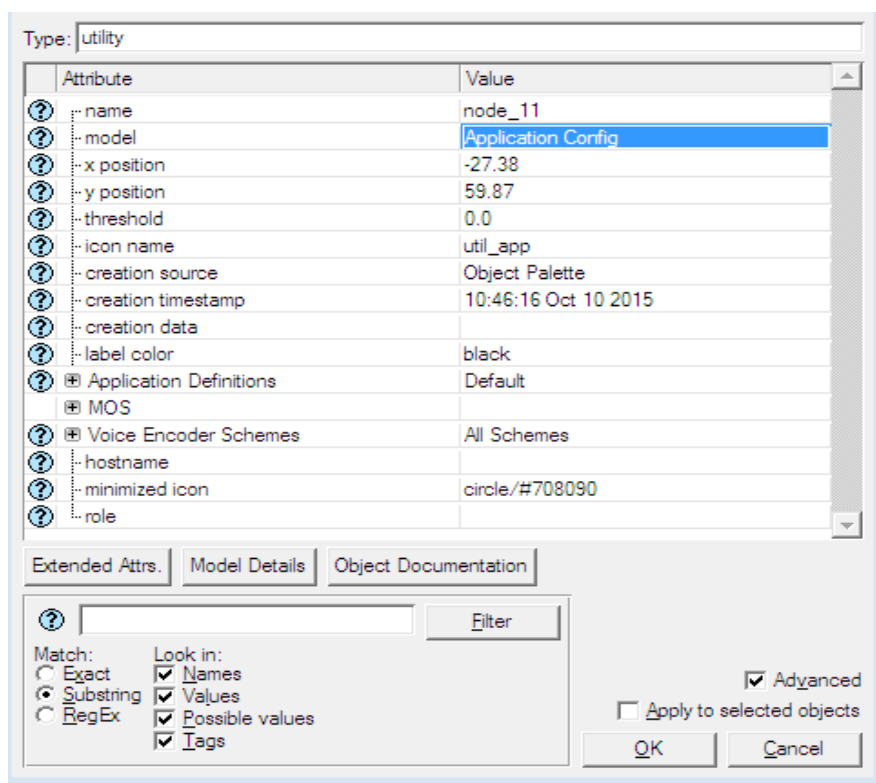


Figure (3.4) The configuration of IPV6 Network showing VoWiFi large

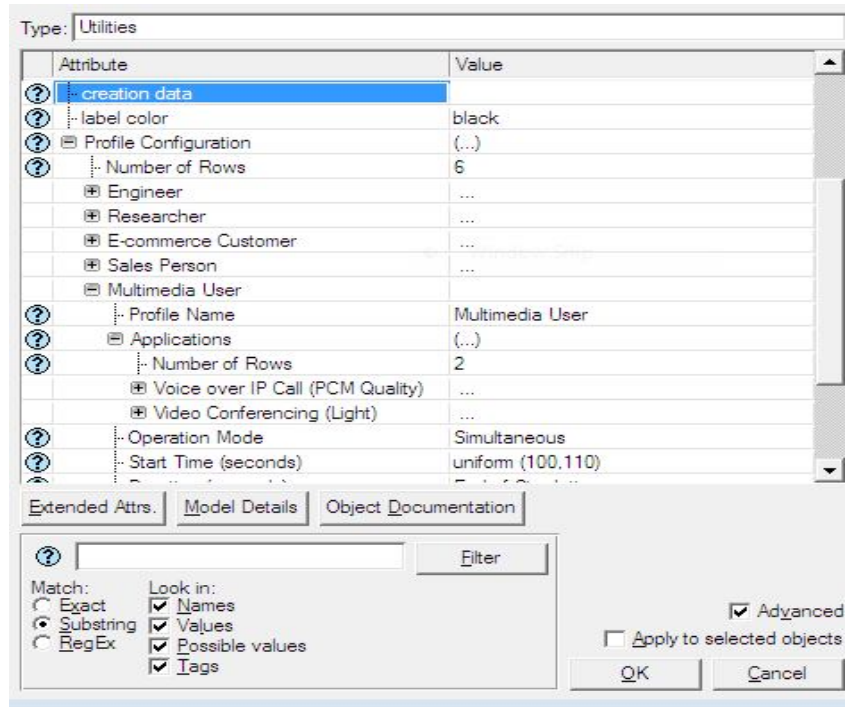
## network

## 3.2.2 Application parameter configuration

The **Application \_ Configuration** include a name and a description table that specifies various parameters for the VOIP application see **Figure (3.5)**. The specified application name is used while creating user profiles on "**Profile\_ Configuration**" object. The **Profile\_ Configuration** is used to create user profiles. These user profiles can be specified on different nodes in see **Figure (3.6)**.



**Figure (3.5) Profile\_ Config attribute dialog box**



**Figure (3.6) VoIP application configuration**

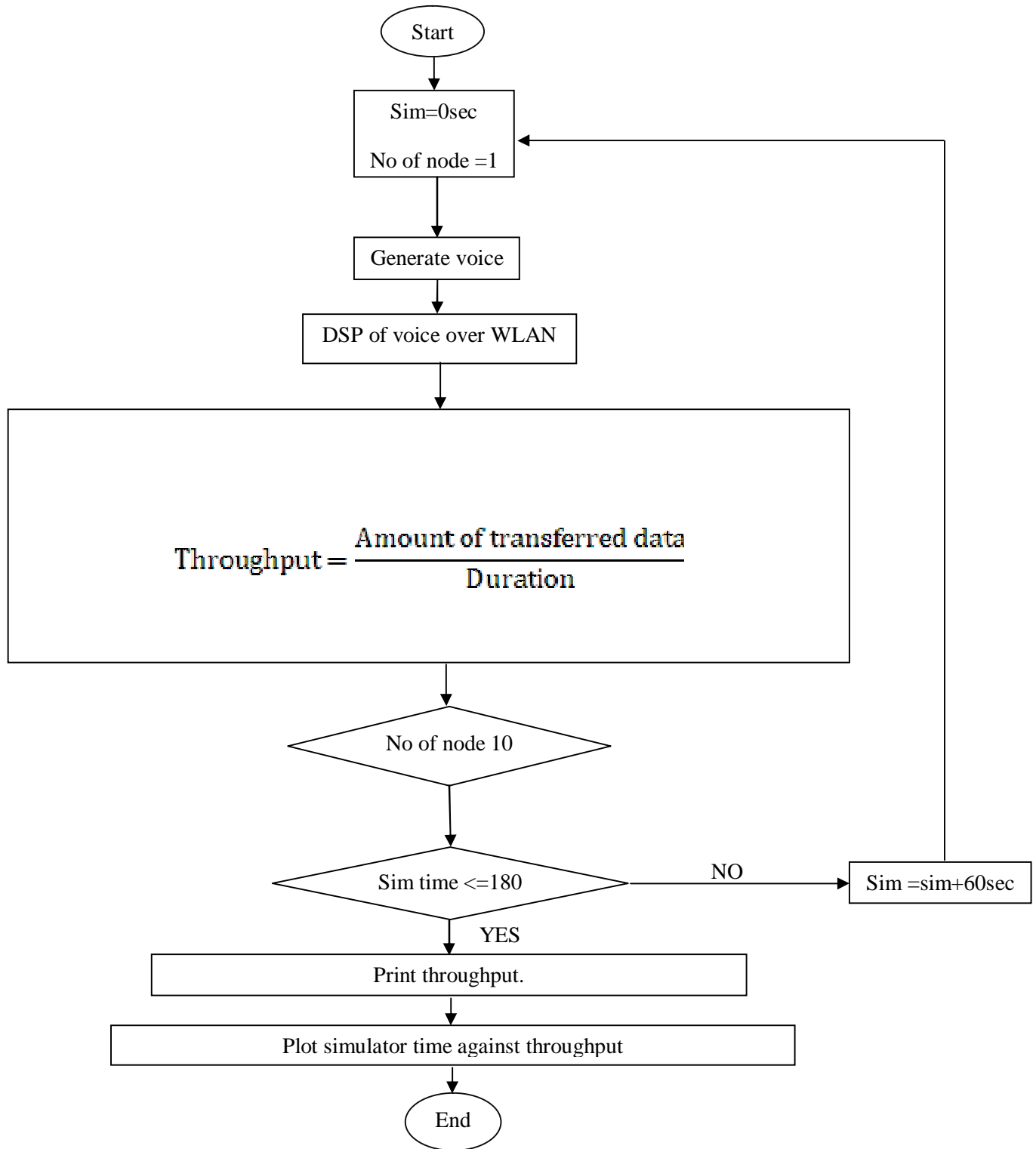
### 3.2.3 Simulation To Measure QoS

OPNET is the simulator tool used for designing the network and deploying VOIP technology. used to simulate two different scenarios namely IEEE 802.11 a, & n according to traffic analysis, for three parameters: Throughput, End To End Delay and jitter has considered to evaluate the network performance for IPv4 and IPv6 using UDP as transmission protocol.

In **figure (3.7)** from the start the first event the program read the parameter from the initial value and create an array parameter: amount of data transferred from the first time to the last time of the simulation and send it with no of node (flow ID) this data can used to calculate and plot, to get the throughput for VoIP node filtered the received packet and it occur time using the mathematical equation to get throughput **E (2.1)** that mentioned in chapter 2.

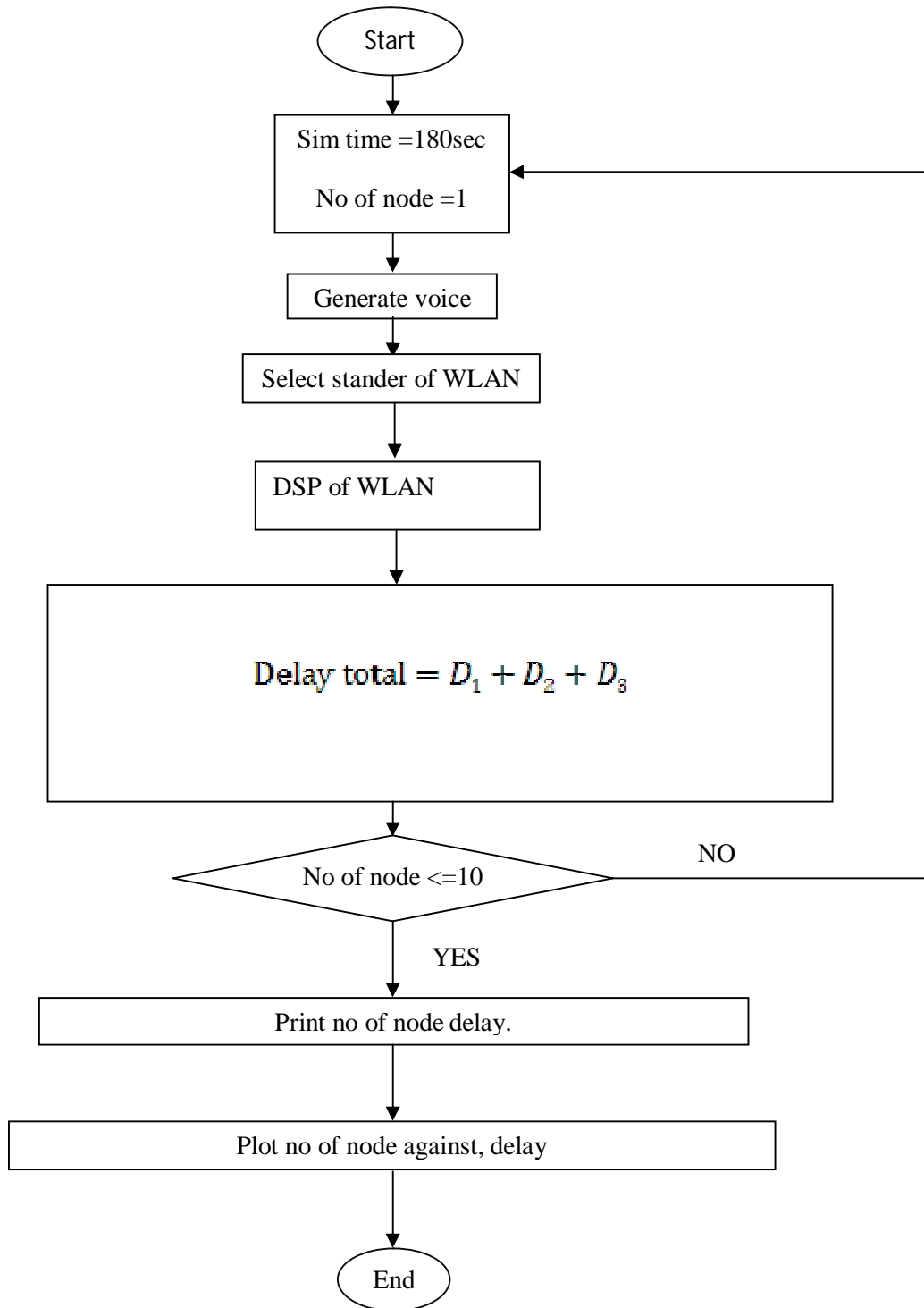
In **figure (3.8)** start the first event the program read the parameter from the initial value and creates an array parameter: Delay at the source, Delay at the receiver, Network delay from the first time to the last time of the simulation based on the time interval and the event of nodes queuing packet and nodes receiving packet End To End Delay calculate using mathematical equation **E (2.2)** that mentioned in chapter 2.

In **figure (3.9)** start the first event the program read the parameter from the initial value and creates an array parameter: random jitter, deterministic jitter, bit error rate from the first time to the last time of the simulation depend on deferent between delay packet and time. Jitter can calculate using mathematical equation **E (2.3)** that mentioned in chapter 2.

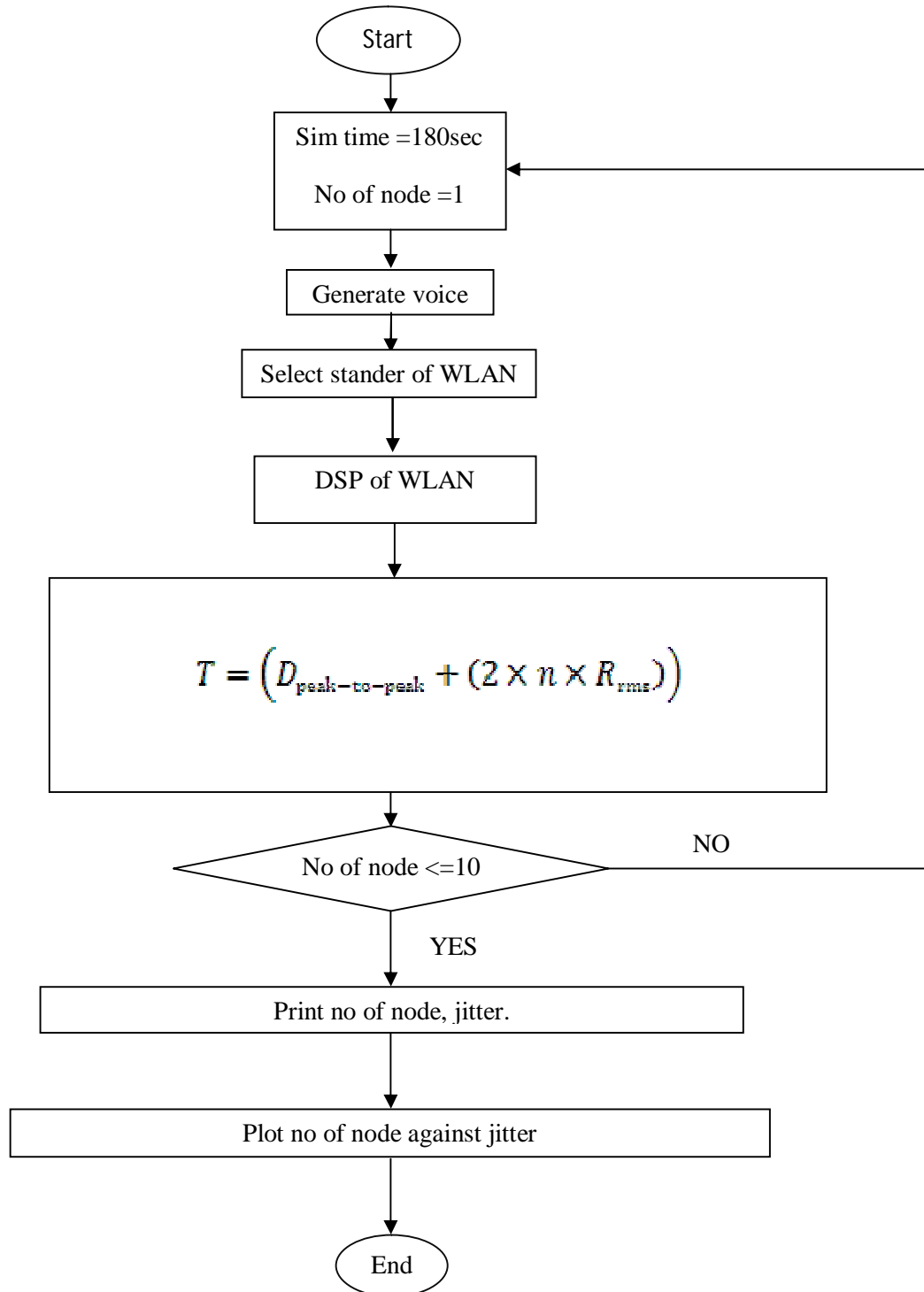


**Figure (3.7): Simulation flowchart Throughput vs. simulation time**





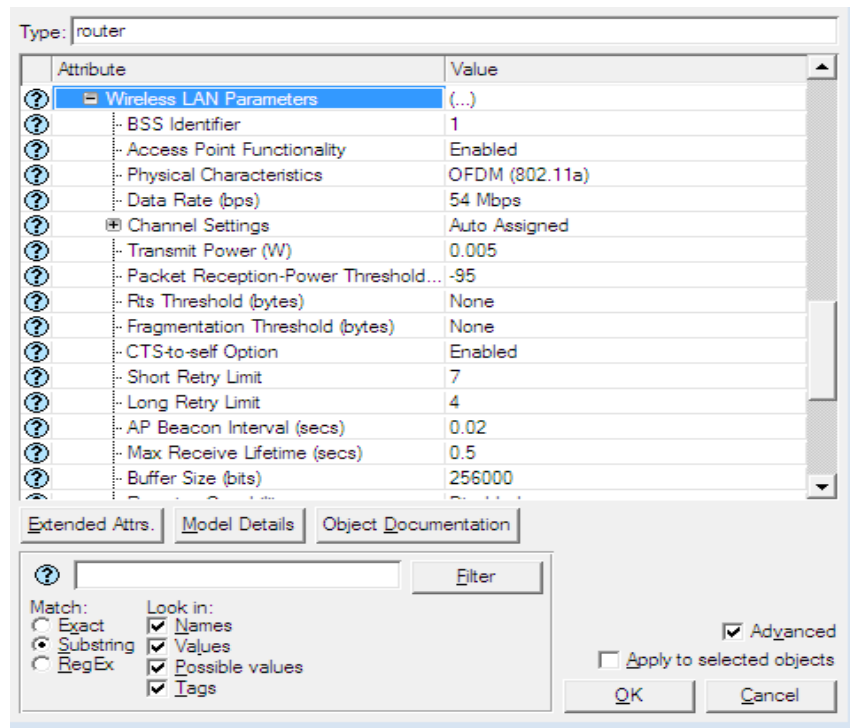
**Figure (3.8): Simulation flowchart End To End Delay**



**Figure (3.9): Simulation flowchart jitter**

### 3.2.4 Simulation Parameters:

The Simulation environment setup parameters which affect the performance of our system are depicted in **Figure (3.10)**, **Figure (3.11)** and **Table (3.1)**



**Figure (3.10):** Example for Wireless network parameters

The image shows a configuration window with two main sections: 'Node selection' and 'Traffic Details'.  
 In the 'Node selection' section:  
 - 'Full mesh between all nodes' is selected with a radio button.  
 - 'To all other nodes from:' is set to 'router' in a dropdown menu.  
 - 'From all other nodes to:' is set to 'router' in a dropdown menu.  
 - 'Include only VoIP gateways as source and destination nodes' is unchecked.  
 In the 'Traffic Details' section:  
 - 'Call volume:' is 1000, 'Units:' is Erlangs.  
 - 'Average call duration:' is 300 seconds/call.  
 - 'Voice flow duration:' is 3600 seconds.  
 - 'Encoder scheme:' is G.711.  
 - 'Type of service:' is Best Effort (0).  
 - 'Set start time' is unchecked, with a date/time field showing 13:13:17.000 Jan 14 2016.  
 - 'Include overhead (bytes)' is checked, with a dropdown set to UDP/IP.  
 At the bottom, there are buttons for 'Advanced Options...', 'Create', and 'Cancel'.

**Figure (3.11): Codec and transmission protocol configuration**

**Table ( 3.1) Simulation Environment parameter**

<b>Numbers of nodes</b>	10 nodes
<b>Network scale</b>	Office
<b>Specify size</b>	100*100 m2
<b>Technology</b>	IPv4 and IPv6 Wi-Fi (IEEE802.11 a, n)
<b>Data rate</b>	54 , 248 Mbps
<b>Link model</b>	100 Base T full duplex
<b>Application</b>	Voice over IP call (PCM Quality)
<b>Voice encoding</b>	G.711
<b>Duration of simulation</b>	180 second