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PERFORMANCE OF VOICE OVER IP (VOIP) OVER A WIRELESS LAN (WLAN) FOR DIFFERENT AUDIO/VOICE CODECS

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Abstract. Capacity and Quality of Service (QoS) are two of the most important issues that need to be resolved before the commercial deployment of VoIP over wireless LAN (WLAN). The capacity is highly dependent on the chosen speech codec. Thus, several codecs are studied (namely, G.711 and G.723.1 and G.729) to determine their effects on the available capacity supported. On top of that, factors affecting QoS such as packet loss, jitter, throughput, and delay for various capacity networks are studied in this paper. This was done by simulating VoIP traffics over the WLAN using Network Simulator 2 (*ns2*) and predicting voice quality based on E-model. The simulation measurements were verified by the theoretical analysis. In conclusion, G.711 codec allows up to 5 simultaneous VoIP nodes, G.723.1 codec allows up to 15 nodes and G.729 codec allows up to 5 nodes for a voice quality greater than R=70 and distance 10 meter from VoIP nodes to access point (AP).

Keywords: Quality of service (QoS), wireless LAN (WLAN), voice over IP (VoIP), voice codec

Abstrak. Kapasiti dan Kualiti Perkhidmatan (QoS) adalah antara dua isu penting yang perlu diselesaikan sebelum penggunaan VoIP melalui WLAN dapat dikomersialkan. Kapasiti banyak bergantung kepada jenis *speech codec* yang digunakan. Oleh sebab itu, beberapa jenis *codec* dinilai (G.711, G.723.1 dan G.729 'codec') untuk menentukan kesannya terhadap kapasiti yang mampu ditanggung. Selain daripada itu, faktor-faktor lain yang memberi kesan kepada QoS seperti kehilangan paket, *jitter*, *throughput*, dan lengah masa dalam beberapa kapasiti rangkaian yang berbeza dikenal pasti. Ini dilakukan dengan menjalankan penyelakuan trafik VoIP ke atas WLAN menggunakan 'Network Simulator 2 (*ns2)*' dan menganggarkan kualiti suara berdasarkan E-model. Keputusan daripada penyelakuan disahkan menggunakan analisis secara teori. Kesimpulannya, *codec* G.711 membenarkan akses 5 nod VoIP secara serentak, G.723.1 membenarkan sehingga 15 nod dan G.729 membenarkan sehingga 5 node bagi kualiti suara melebihi R=70 dalam jarak 10 meter daripada *access point* (AP).

Kata kunci: Kualiti perkhidmatan (QoS), LAN tanpa wayar (WLAN), suara melalui IP (VoIP), *codec* suara

1.0 INTRODUCTION

Voice over IP (VoIP) involves digitization of voice streams and transmitting the digital voice as packets over conventional IP-based packet networks like the Internet, Local Area Network (LAN) or wireless LAN (WLAN) [8]. The goal of VoIP is to provide voice transmission over those networks. Although the quality of VoIP does not yet

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match the quality of a circuit-switched telephone network, there is an abundance of activity in developing protocols and speech encoders for the implementation of the high quality voice service [10],[11],[12],[14]. In WLAN, as VoIP technology is still in the early stages of commercial deployment, it is necessary to examine if VoIP over WLAN can provide a Quality of Service (QoS) comparable to that of the existing PSTN and cellular networks. So, it is essential to determine the number of simultaneous users a WLAN can support simultaneously without significantly degrading the QoS and also analyze the delay, jitter and packet loss of VoIP over WLAN.

The QoS on VoIP network partly depends on the types of voice codec used [2]. The primary functions of a voice codec are to perform analog/digital voice signal conversion and digital compression. H.323 specifies a series of audio codec ranging in bit rates from 5.3-64 kbps [9]. Among three commonly used codec in Internet telephony are G.711, G.723.1, and G.729. These codecs differ in their coding rate (bps), frame rate (frames/s), algorithmic latency that will influence the speech quality or Mean Opinion Source (MOS) in a VoIP network.

In this paper, we simulate a VoIP network in a 802.11b WLAN by using *ns2* [1],[3],[7] to make a measurement on VoIP channel characteristic such as delay, jitter, throughput, packet loss contributing to QoS for varying number of nodes with three different codecs which are G.711, G.723.1 and G.729. The results obtained from simulation were analyzed to obtain the performance of VoIP over WLAN network. Finally, estimation on channel capacity of VoIP over WLAN were done by using theoretical analysis, throughput measurement analysis and ITU-T G.107 E-model analysis for voice quality to determine how many simultaneous VoIP channels can the current capacity of a WLAN supports.

This paper is organized as follows: Section 2 is the introduction to the VoIP which provides information of the benefits, applications, technical aspect of VoIP such as protocol stack, coding and traffic. Section 3 shows the methodology for the simulation which uses the *ns2* as simulation tools. System for experimental such as network topology, voice codec parameters, traffic, WLAN parameters, operating range and voice quality prediction tool used in this simulation (E-Model) are explained in details. Section 4 gives the result and analysis from the simulation. It also includes the theoretical analysis which verified the simulation result. Section 5 concludes this paper.

2.0 BACKGROUND

2.1 VoIP over WLAN System

Figure 1 describes a VoIP system implemented in the wireless LAN (IEEE 802.11b). As depicted in the figure, the speech source alternates between talking and silence period, which is typically considered to be exponentially distributed. Before transmitted over packet switched networks, the speech signal has to be digitised at the sender; the

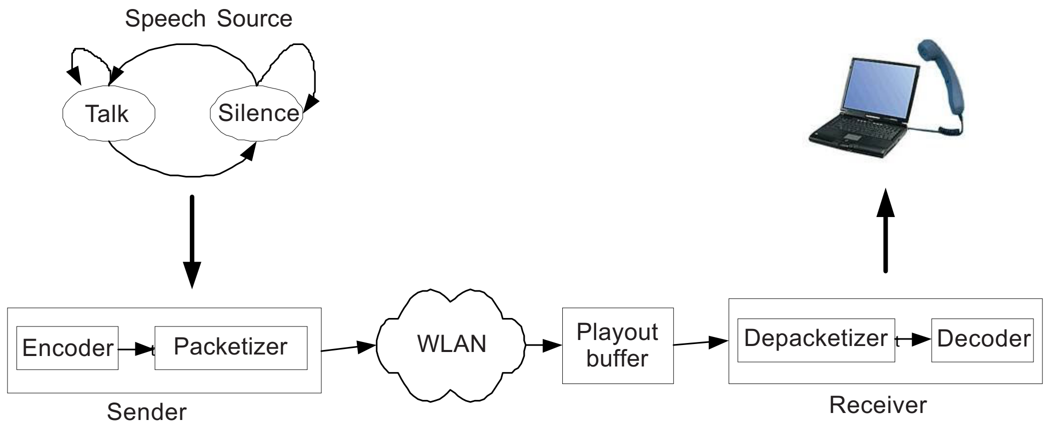

Figure 1 VoIP over WLAN system

reverse process is performed at the receiver. The digitalization process is composed of sampling, quantization and encoding. There are many encoding techniques that have been developed and standardized by the ITU such as G.711, G.729 and G.723.1. The encoded speech is then packetized into packets of equal size. Each such packet includes the headers at the various protocol layers such RTP 12 bytes, UDP 8 bytes, IP 20 bytes, 802.11 34 bytes and the payload comprising the encoded speech for a certain duration depends on the codec deployed.

As the voice packets are sent over IP networks and wireless channel, they incur variable delay and possibly loss. In order to provide a smooth playout delay, at the receiver, a playout buffer is used to compensate the delay variations. Packets are held for a later playout time in order to ensure that there are enough packets buffered to be played out continuously.

2.2 VoIP Protocol Stack

Figure 2 shows the basic IP network protocol stack used to implement VoIP. In order for the Internet to provide useful services, Internet telephony required a set of control protocols (H.323) for connection establishment, capabilities exchange as well as conference control.

H.323
RTP, RTCP, RSVP
UDP, TCP
Network Layer (IPv4, IPv6)
Data Link Layer
Physical Layer

Figure 2 VoIP protocol stack

H.323 is a standard that specifies the components, protocols and procedures that provide multimedia communication services such as real-time audio, video, and data communications over packet networks, including Internet Protocol (IP) based networks. Real-Time Transport (RTP) protocol provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast networks.

The RTP control protocol (RTCP) is used to monitor the quality of real-time services and to convey information about participants in an on-going session. There are components called monitors, which receive RTCP packets sent by participants in a session. These packets contain reception reports, and estimate the current quality of service for distribution monitoring, fault diagnosis and long-term statistics. Both TCP (Transmission Control Protocol) and UDP (User Datagram Protocol) enable the transmission of information between the correct processes (or applications) on host computers.

IP is responsible for the delivery of packets (or datagram) between host computers. IP is a connectionless protocol and it does not establish a virtual connection through a network prior to commencing transmission because this is the task of higher level protocols. IP makes no guarantees concerning reliability, flow control, error detection or error correction. The result is that datagram could arrive at the destination computer out of sequence, with errors or not even arrive at all.

2.3 Audio Codec

2.3.1 G.711 codec

In wireless networks, G.711 is applied for encoding telephone audio signal at a rate of 64 kbps with a sample rate of 8 kHz and 8 bits per sample. In an IP network, voice is converted into packets with durations of 5, 10 or 20 ms of sampled voice, and these samples are encapsulated in a VoIP packet.

2.3.2 G.723.1 codec

G.723.1 codec belongs to the Algebraic Code Excited Linear Prediction (ACELP) family of codec and has two bit rates associated with it: 5.3 kbps and 6.3 kbps. The encoder functionality includes Voice Activity Detection and Comfort Noise Generation (VAD/CNG) and decoder is capable of accepting silence frames. The coder operates on speech frames of 30 ms corresponding to 240 samples at a sampling rate of 8000 samples/s and the total algorithmic delay is 37.5 ms. The codec offers good speech quality in network impairments such as frame loss and bit errors and is suitable for applications such as VoIP.

2.3.3 G.729 codec

G.729 codec belongs to the Code Excited Linear Prediction coding (CELP) model speech coders and uses Conjugate Structure - Algebraic Code Excited Linear Prediction (CS-ACELP). This coder was originally designed for wireless applications at fixed 8 kbit/s output rate, not including the channel coding. The coder works on a frame of 80 speech samples (10 ms) and the required look ahead delay of 5 ms. So the total algorithmic delay for the coder is 15 ms.

2.4 VoIP Traffic

Voice traffic has a very stringent delay constraint. It has active talking periods where the source is sending out periodic voice packets or the talker is speaking and silence periods where no voice packets are generated or the speaker is silent. Most standard voice encoding has a fixed bit rate and a fixed packetization delay [2], [12]. There are thus producing a stream of fixed size packets. This packet stream is however only produced during talk-spurts and the voice coder sends no packets during silence periods. The behavior of a single source is easily modeled by a simple ON-OFF model shown in Figure 3. During talk-spurts (ON periods), the model produces a stream of fixed size packets with fixed inter-arrival times (T).

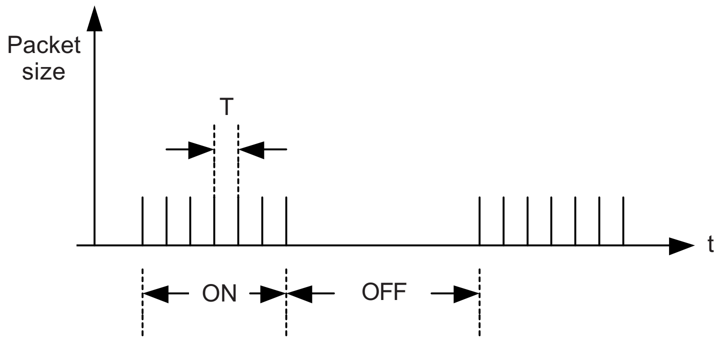

Figure 3 Characteristics of a single source

3.0 SIMULATION WORKS

3.1 Approach

Based on the flow chart in Figure 4, the main source file is simulated with voice traffic file and certain position of mobile node file. Three widely used codecs for VoIP application are simulated, which are G.711, G.723.1 and G.729. Then, the measurements of delay, jitter, throughput, and packet loss are sieved out from trace output file by using AWK file. There are four AWK files that had been created: *measure-*

delay.awk, *measure-thruput.awk*, *measure-packetloss.awk* and *measure-jitter.awk*. Tracegraph [13] is used to draw the graphs based on the data from the simulation output files. Finally, E-model will be used to calculate the Transmission Rating Factor, R and Mean Opinion Score (MOS) value to obtain the maximum number of VoIP users in a single cell WLAN with acceptable R value. Futhermore, the measurement of throughput obtained from simulation is analyzed as a method to determine minimum of nodes can be support in single cell WLAN by connecting to the same access point (AP).

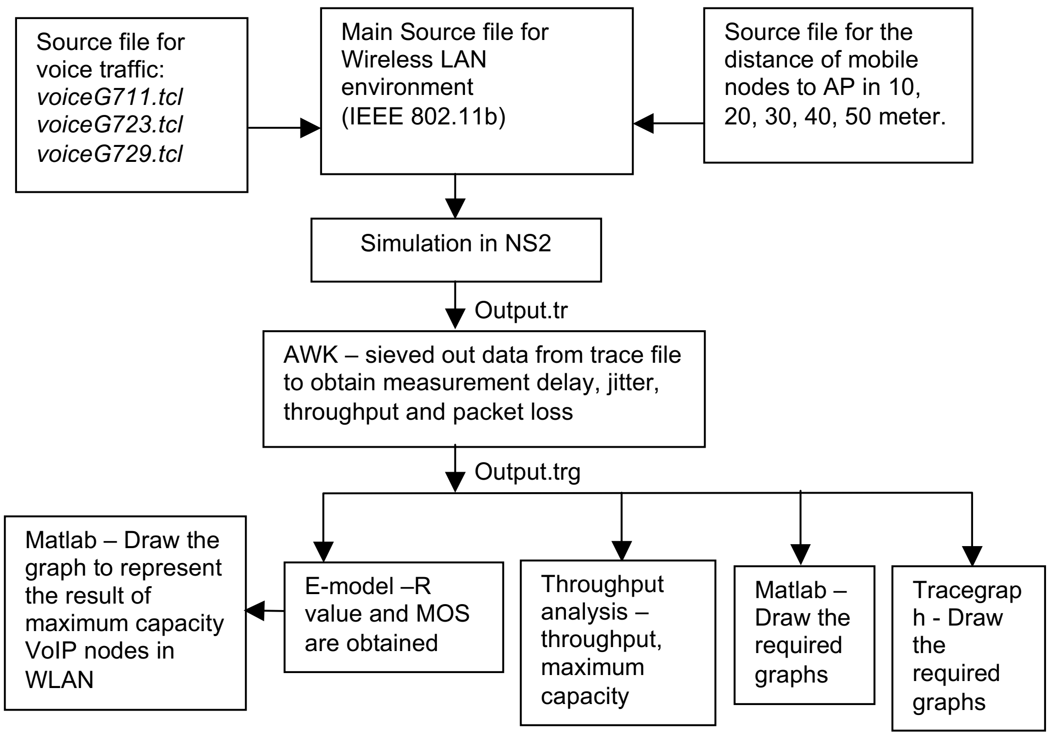

Figure 4 Simulation overview

3.2 Simulation Framework

3.2.1 Network Topology

In the simulation, all VoIP nodes are assumed to be at an equal distance to the AP as shown in Figure 5 for a distance *d* to the AP with 6 VoIP nodes. There are four data rates defined for 802.11b transmission at 2.4 GHz: 1, 2, 5.5, and 11 Mbps. In these cases, data rate depends on how much distortion presents in the environment as a function of distance to AP. Simulations are done with the distance of VoIP nodes from AP fixed at 10, 20, 30, 40 and 50 meter.

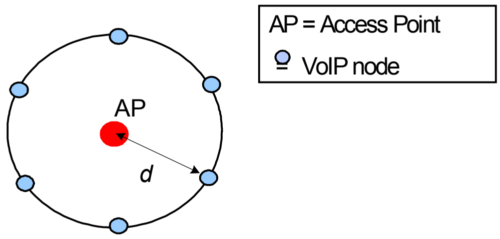

Figure 5 Network topology with 6 VoIP nodes

3.2.2 Audio/Voice Codec

The main characteristics of the codec used in the simulation are summarized in Table 1.

Table 1 Audio/Voice codec parameters

Parameters	G.711	G.723.1	G.729
Bit rate (Kbps)	64	6.3	8
Framing interval (ms)	10	30	10
Payload (Bytes)	80	24	10
Packets/s, N_p	100	33	100

The standard method of transporting voice packets through WLAN network requires the addition of three headers which are IP, UDP and RTP. An IPv4 header is 20 octets, a UDP header is 8 octets and RTP header is 12 octets. A total of 40 octets are therefore sent each time a packet containing voice payload is transmitted.

3.2.3 VoIP Traffic Model

With silence suppression, VoIP traffic is modeled as an ON-OFF Markov process. The alternative periods of activity and silence are exponentially distributed with average durations of $\frac{1}{\mu}$ and $\frac{1}{\lambda}$, respectively. Typically, the average activity cycle is 42.6 %, as recommended by the ITU-T P.59 specification for conversational speech [6]. When the source is in the "ON" state, constant rate source for each codec, denoted by "CBR/UDP" is generated at a constant interval. No packets are transmitted when the source is "OFF".

Voice traffic source file that contains ON-OFF Markov model of voice sources, random variables (uniforms & exponentials) and CBR source which was created by C.N. Chuah on 10/21/1998 is used in this simulation. These files were obtained from following website http://www.ece.ucdavis.edu/~chuah/research/voip/nscode/voice.tcl

3.2.4 WLAN Parameters

The network simulator will be used to form an appropriate network topology under the MAC (Media Access Control) layer of the IEEE 802.11b. According to the IEEE 802.11b protocol specifications, the parameters for the WLAN are shown in Table 2.

Table 2 Parameter values of 802.11b DCF

Parameter	**Value**
DIFS	50 μsec
SIFS	10 μsec
Slot Time	20 μsec
CW_{min}	32
CW_{max}	1023
Data Rate	1,2,5.5,11 Mbps
Basic Rate	1 M bps
PHY header	192 μsec
MAC header	34 bytes
ACK	248 μsec

3.3 E-Model

E-Model [6],[7] provides a powerful method of assessing whether a WLAN data network is capable and ready to carry VoIP calls as well as performing voice-readiness testing.

An E-model calculation considers all of the following factors: delay, percentage of packets lost, delay introduced by the jitter buffer, and the behavior of the codec. Once the R value is calculated from these factors, an estimate of the MOS can be directly calculated from it. Furthermore, the maximum number of simultaneous of VoIP calls that can be handled by the WLAN will be determined.

3.3.1 Mean Opinion Score

The leading subjective measurement of voice quality is the MOS, as described in the ITU Recommendation P.800. The mapping between audio performance characteristics and a quality score makes the MOS standard valuable for network assessments, benchmarking, tuning, and monitoring. From Table 3, MOS can range from 5 (Excellent) down to 1 (Bad).

A MOS of 4 or higher is generally considered toll quality (as per voice call in PSTN). A MOS below 3.6 results in many users who are not satisfied with the call quality.

Table 3 The mean opinion score scale

MOS	Quality Rating
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

3.3.2 Mapping between MOS and E-Model

R factor values range from 100 (desirable) down to 0 (unacceptable) [4], [5], [6]. Once the value of R is calculated from these factors, an estimate of the MOS can be directly calculated using the following formula:

$$MOS = 1 < 1 + (0.035*R) + (R(R - 60)*(100 - R)*7.0e\text{-}^{06}) < 4.5 \tag{1}$$

From Table 4, R factor values from the E-model are shown on the left, with their corresponding MOS values on the right. The likely satisfaction level of human listeners is shown in the middle.

Table 4 Mapping between R values and estimated MOS

R	User Satisfaction	MOS
90-100	Very Satisfied	4.3-4.5 (Desirable)
80-90	Satisfied	4.0-4.3 (Desirable)
70-80	Some users dissatisfied	3.6-4.0 (Acceptable)
60-70	Many users dissatisfied	3.1-3.6 (Acceptable)
50-60	Nearly all users dissatisfied	2.6-3.1 (Not recommended)
0-50	Not recommended	1-2.6 (Not recommended)

3.3.3 E-model Parameters for Simulations

In the simulations, the values for each E-model parameters are as in Table 5 and Table 6. Table 5 lists the default values given by the standard. Meanwhile, Table 6 gives the value for the parameters that are used in the simulation.

Table 5 Default E-Model parameter values for the simulation

Parameter	Abbr.	Unit	Default value
Sending Loudness Rating	SLR	dB	+8
Receiving Loudness Rating	RLR	dB	+2
Sidetone Masking Rating	STMR	dB	15
Listener Sidetone Rating	LSTR	dB	18
D-Value of Telephone, Send Side	Ds	-	3
D-Value of Telephone, Receiver Side	Dr	-	3
Talker Echo Loudness Rating	TELR	dB	65
Weighted Echo Path Loss	WEPL	dB	110
Number of Quantization Distortion Units	qdu	-	1
Circuit Noise referred to 0 dBr-point	Nc	dBmp	–70
Noise Floor at the Receiver Side	Nfor	dBmp	–64
Room Noise at the Send Side	Ps	dB(A)	35
Room Noise at the Receiver Side	Pr	dB(A)	35
Expectation Factor	A	-	5

Table 5 Default E-Model parameter values for the simulation

Parameter	Abbr.	Unit	Default value
Sendi			
Parameter	Abbr.	Unit	Value
Absolute Delay in echo-free Connections	Ta	ms	T = Ta
Round Trip Delay in a 4-wire Loop	Tr	ms	Tr = 2T
Equi pment Impairment Factor	Ie	-	G711 : 0 G723.1m : 15 G729 : 12
Packetization Delay (Voice Frame Duration)	T_{PACK}	ms	G711 : 10 G723.1m : 30 G729 : 10
Look Ahead Delay	T_{LA}	ms	G711 : 0 G723.1m : 7.5 G729 : 5
Network Delay	T_{NW}	ms	Simulated
Access Delay	T_{WLAN}	ms	Simulated
Jittering Delay	T_{JITT}	ms	Simulated

4.0 RESULTS

4.1 QoS Measurement Analysis

The QoS measurements resulted from simulation are analyzed in this section.

4.1.1 Packet Loss

Packet loss is expressed as a ratio of the number of packets lost to the total number of packets transmitted. Packet losses results when packets sent are not received at the final destination. The percentage of packet loss for different coding technique at certain operating range is depicted in Figure 6 to 8.

Packet loss is an important parameter affecting the performance of the network. For the simulation analysis, G.711 suffers dramatically from the packet loss compare with G.723.1 and G.729. Generally, packet loss is related with the packet length, which is proportional to transmission time associated with each packet. Furthermore, the time

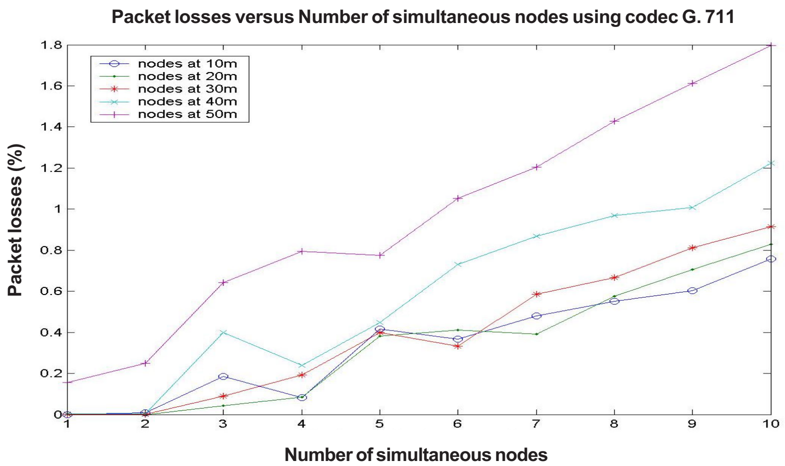

Figure 6 Percentage packet losses vs. number of simultaneous nodes using G.711 codec as a function of distance to AP

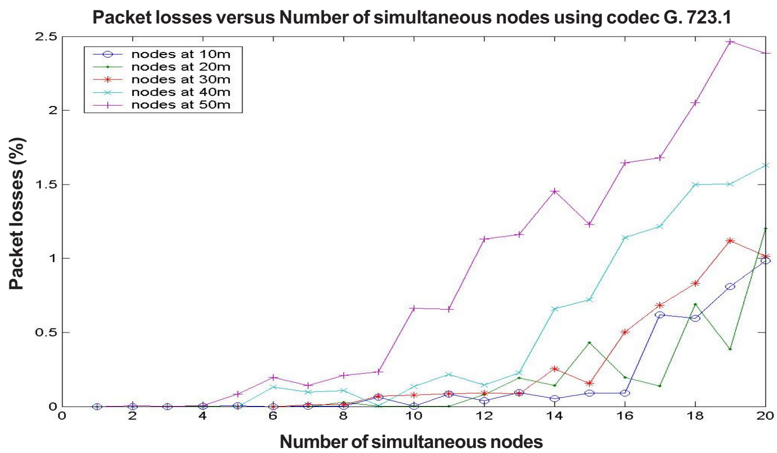

Figure 7 Percentage packet losses vs. number of simultaneous nodes using G.723.1 codec as a function of distance to AP

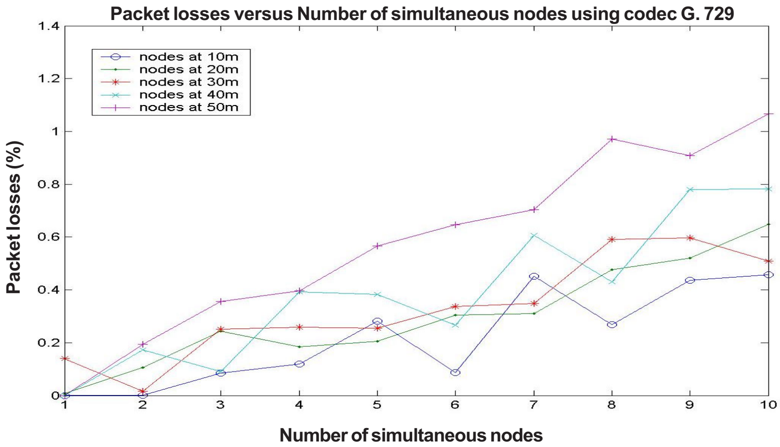

Figure 8 Percentage packet losses vs. number of simultaneous nodes using G.729 codec as a function of distance to AP

intervals between packets are shorter in G.711, which worsens the performance in terms of dropped packets.

Most of the packet losses come from the transmission failure of the AP. The reason is that 802.11b is designed in a way that every node has to wait for a random amount of time before it tries to send a packet. When the data is concentrated into the AP in the center, at some instant the AP will hold many packets that need to be injected in the network. However, there still have other nodes that are trying to flood packets, so in a fair manner, these packets will be accumulated in queue. If the state of the queue is defined as the number of waiting packets in the queue, this queuing system is unstable. It will eventually overflow and start to drop packets by a Drop Tail manner, which is the default setting in the simulation.

4.1.2 Throughput

The throughput (measured in bps) corresponds to the amount of data in bits that is transmitted over the channel per unit time. The throughput for different codec systems at certain operating range is shown in Figure 9 to 11. From these figures, the number of nodes and their distance from the access point will affect the effective throughput. G.711 gives the highest throughput for the same number of simultaneous nodes and distances.

4.1.3 Jitter

In the simulation, jitter is measured by using the formula below:

$$\text{Jitter} = \frac{\text{current packet received time} - \text{last packet received time}}{\text{differential of sequence number between two packet}} \tag{6}$$

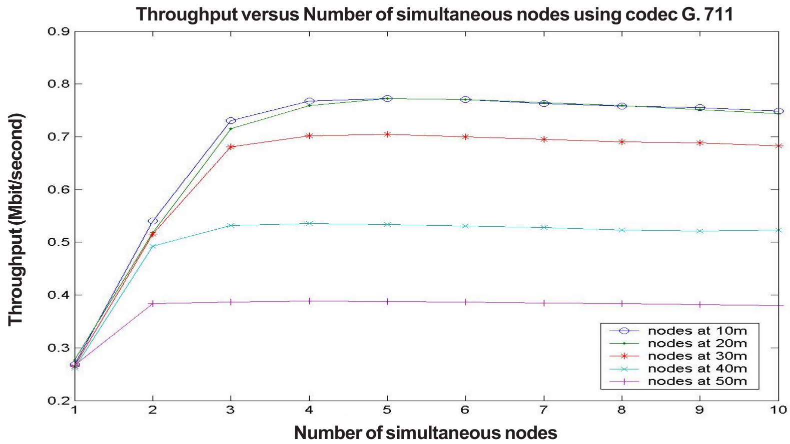

Figure 9 Throughput (Mbps) vs. number of simultaneous nodes using G.711 codec as a function of distance to AP

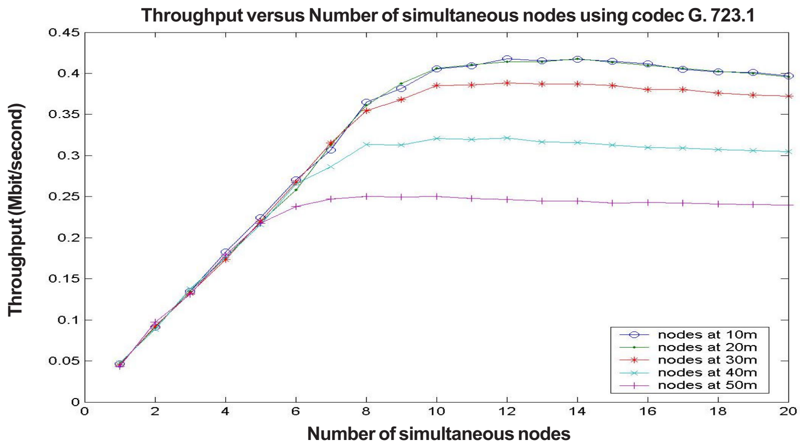

Figure 10 Throughput (Mbps) vs. number of simultaneous nodes using G.723.1 codec as a function of distance to AP

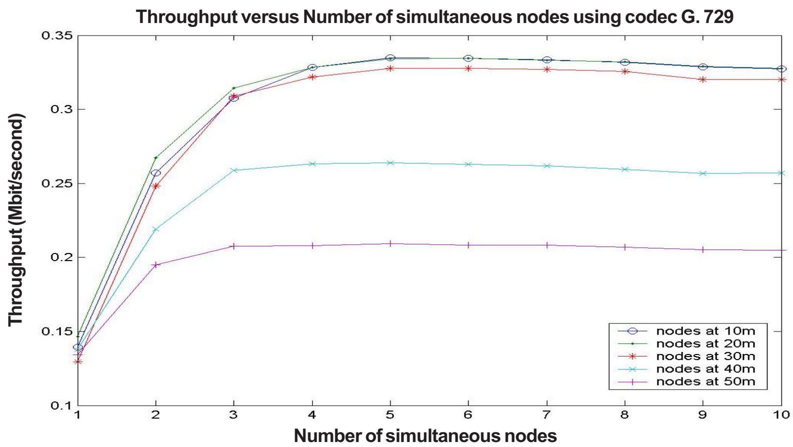

Figure 11 Throughput (Mbps) vs. number of simultaneous nodes using G.729 codec as a function of distance to AP

Jitter is defined as a variation rate in the delay of received packets. From Figure 12 to 14, we can notice that irrespective of the packet size and amount of data sent, the jitter values does not vary much. However, jitter delay for G.723.1 is bigger than G.729 and G.711 as many small size packets are generated with variant inter-arrival time and hence the jitter between packets is significant. As the maximum packet size is increased to 120 bytes for G.711, the jitter is less significant as a smaller number of packets with less delay variations are generated.

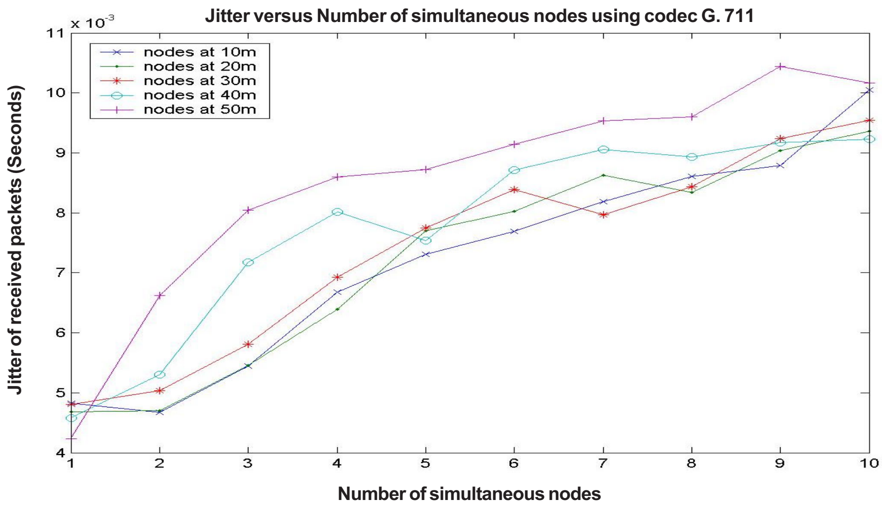

Figure 12 Jitter (second) vs. number of simultaneous nodes using G.711 codec as a function of distance to AP

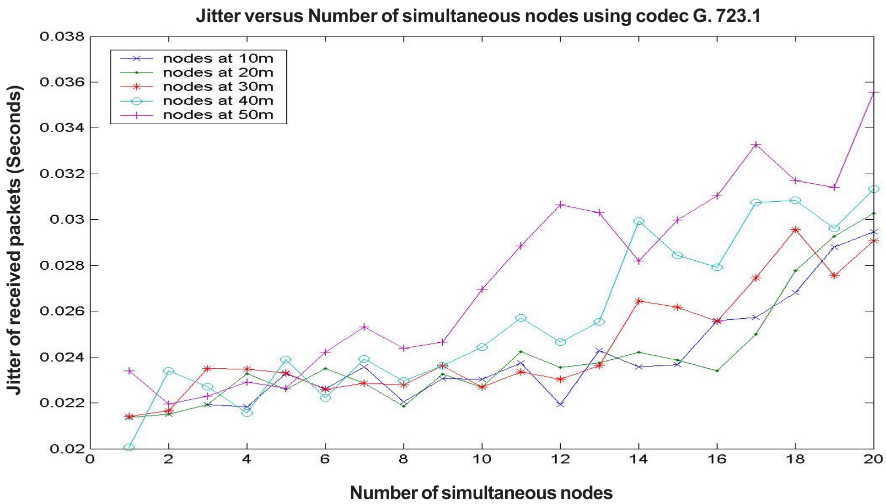

Figure 13 Jitter (second) vs. number of simultaneous nodes using G.723.1 codec as a function of distance to AP

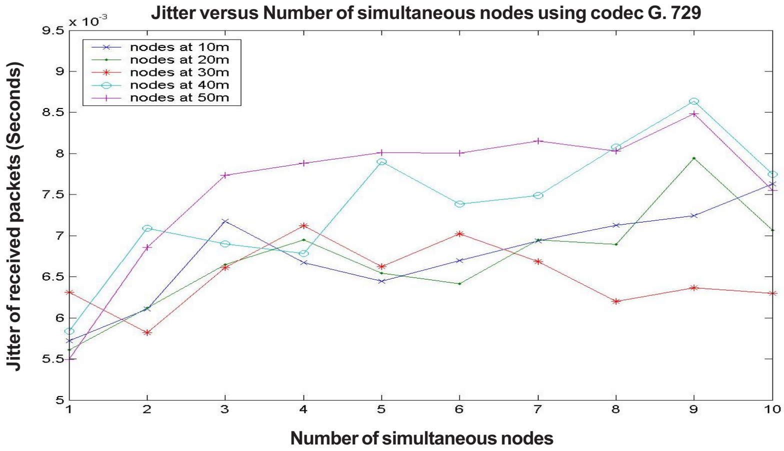

Figure 14 Jitter (second) vs. number of simultaneous nodes using G.729 codec as a function of distance to AP

4.2 Capacity Analysis

In the simulation, there are three analytical methods used to obtain maximum number of VoIP calls that can simultaneously take place in a WLAN cell. The channel capacity as a function of the chosen codec and of the distance of the VoIP nodes to the AP which is evaluated from theoretical analysis are used to compare with the simulation result of throughput analysis and E-model analysis. Conclusions are made based on the E-model that is an efficient tool to predict the voice quality.

4.2.1 Theoretical Analysis of VoIP Capacity

Let n be the maximum number of sessions that can be supported. The transmission times for downlink and uplink packets are T_{down} and T_{up}, respectively. Let T_{avg} be the average time between the transmissions of two consecutive packets in a WLAN. That is, in one second, there are totally $\frac{1}{T_{avg}}$ packets transmitted by the AP and all the stations. So,

$$\frac{1}{T_{avg}} = \text{number of streams} * \text{number of packets sent by one stream in one second} \quad (7)$$

For a VoIP packet, the header overhead, OH_{hdr} consists of the headers of RTP, UDP, IP and 802.11 MAC layer:

$$OH_{hdr} = H_{RTP} + H_{UDP} + H_{IP} + H_{MAC} \quad (8)$$

At the MAC layer, the overhead incurred at the sender is

$$OH_{sender} = DIFS + averageCW + PHY \quad (9)$$

If it is the unicast packet, the overhead incurred at the receiver is

$$OH_{receiver} = SIFS + ACK \quad (10)$$

where average CW = slotTime*(CW_{min}–1)/2 is the average backoff time when there are no other contending stations. We ignore the possibility of collisions and the increase of backoff time in subsequent retransmissions after a collision in the analysis here. This means that the VoIP capacity to be derived is an upper bound on the actual capacity. However, contention overhead is negligible compared with other overheads, and the analytical upper bound is actually a good approximation of the actual capacity, as will be verified by the simulation results later. Thus,

$$T_{down} = T_{up} = \frac{(Payload + OH_{hdr})*8}{dataRate} + OH_{sender} + OH_{receiver} \quad (11)$$

In the ordinary VoIP case, n downlink and n uplink unicast streams is considered. On average, for every downlink packet, there is a corresponding uplink packet. So,

$$T_{avg} = \frac{T_{down} + T_{up}}{2} \quad (12)$$

From (7), we have

$$\frac{1}{T_{avg}} = 2n * N_p \quad (13)$$

where N_p is the number of packets sent by one stream per second. Within a Basic Service Set (BSS), there are two streams for each VoIP session. The values of DIFS, PHY, SIFS, ACK for 802.11b are listed in Table 2. Meanwhile, N_p and Payload values for different codec are listed in Table 1. Table 7 shows the results of deriving capacities VoIP on WLAN when G.711, G.723.1 and G.729 codecs are used.

Table 7 Maximum VoIP nodes supported for different codec and channel capacity (Theory)

	Maximum simultaneous VoIP nodes		
Bit rate	**G.711**	**G.723.1**	**G.729**
11 Mbps	5.4	17	5.7
5.5 Mbps	4.8	15.7	5.4
2 Mbps	3.5	12.4	4.4
1 Mbps	2.5	9.4	3.3

4.2.2 Simulation Analysis of VoIP Capacity

By using values of maximum achievable throughput from simulation, VoIP capacity in WLAN can also be evaluated. The following formula is used for getting the average packets sent from AP and all VoIP nodes in one second.

$$\frac{1}{T_{avg}} = \frac{\text{Maximum Throughput}}{\text{dataRate}} \tag{14}$$

Then, formula 14 is applied to get the maximum supportable VoIP nodes in the 802.11b. Table 8 indicate the result of capacity of VoIP nodes over WLAN for different codec.

Table 8 Maximum VoIP nodes supported for different codec and channel capacity (Simulation)

	Maximum simultaneous VoIP nodes		
Bit rate	**G.711**	**G.723.1**	**G.729**
11 Mbps	4.0	12.13	4.13
5.5 Mbps	3.65	11.24	4.00
2 Mbps	2.76	9.47	3.25
1 Mbps	1.98	7.40	2.50

4.2.3 Simulation Analysis (E-model) of VoIP Capacity

The E-Model calculates the R, using the network impairment factors, which were obtained from simulation such as delay, jitter, and packet loss (Figure 15). Table 9 provides the results of capacities VoIP over WLAN from the E-Model calculation when codec of G.711, G.723.1 and G.729 are used.

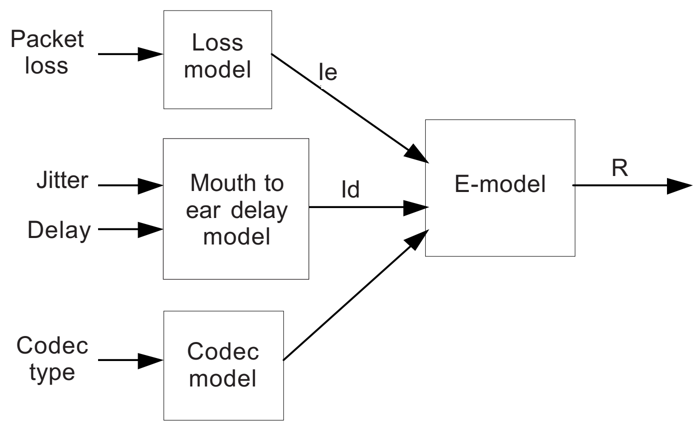

Figure 15 Calculating R factor from simulation result using E-Model

Table 9 Maximum VoIP nodes supported for different codec and channel capacity (using E-Model)

	Maximum simultaneous VoIP nodes		
Bit rate	**G.711**	**G.723.1**	**G.729**
11 Mbps	5.0	14.8	4
5.5 Mbps	4.5	13.2	4
2 Mbps	3.0	10.5	2.8
1 Mbps	2.5	10.0	2.5

The performance of the G.711, G.723.1 and G.729 codecs from the E-model analysis are shown respectively in Figure 16 to 18 as a function of the distance to AP. With all codecs, there is a high degradation of the capacity with distance. With G.711, going from 10 m to 50 m reduces the capacity from 4 VoIP calls to 2 calls. With G.723.1, while 14 simultaneous calls are possible at 10 meters, 10 calls can be made at 50 meters. On the other hand, performance of G.729 is quite similar with G.711 where from 4 calls supported at 10 meters down to 2 calls at 50 meters.

These values are quite low when related with the available physical data rate in the cell, especially at 10 meters, where physical rate is 11 Mbps. In fact, IEEE 802.11b suffer from a huge overhead, due to the RTS/CTS handshake, the acknowledgement, and the MAC header with 1 Mbps used to transmit the control packets and the physical header. Moreover, for each voice frame, a RTP/UDP/IP header has to be added. The proportion of this overhead is particularly high for small data packets.

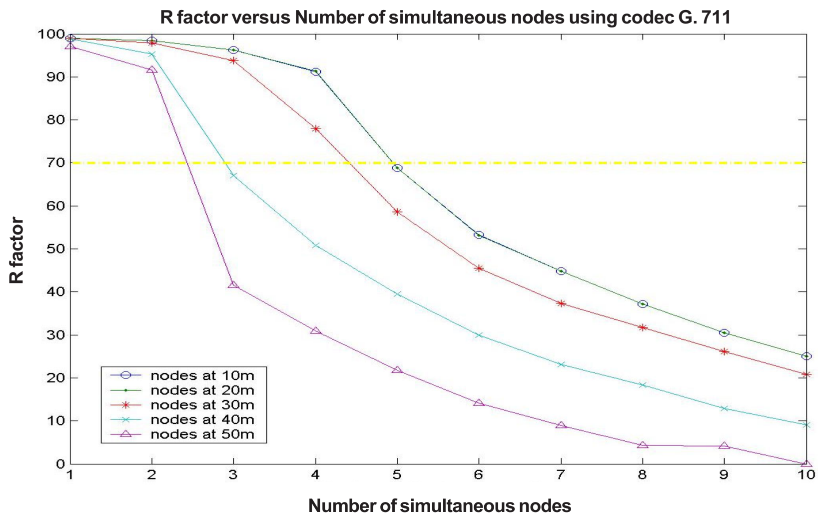

Figure 16 R factor vs. number of simultaneous nodes using G.711 codec as a function of distance to AP

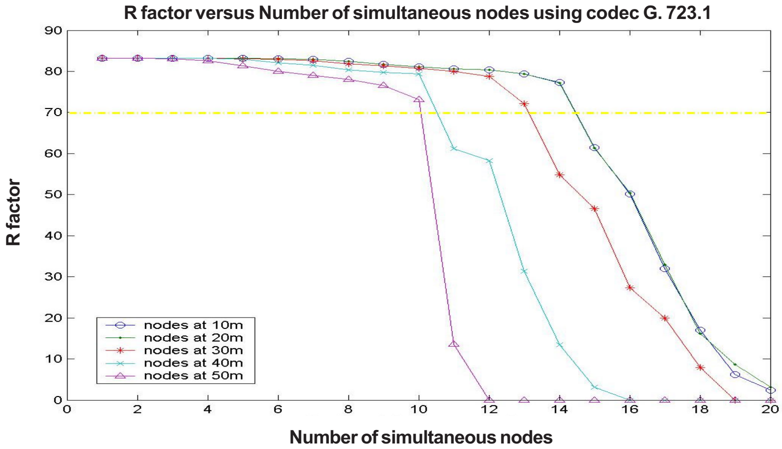

Figure 17 R factor vs. number of simultaneous nodes using G.723.1 codec as a function of distance to AP

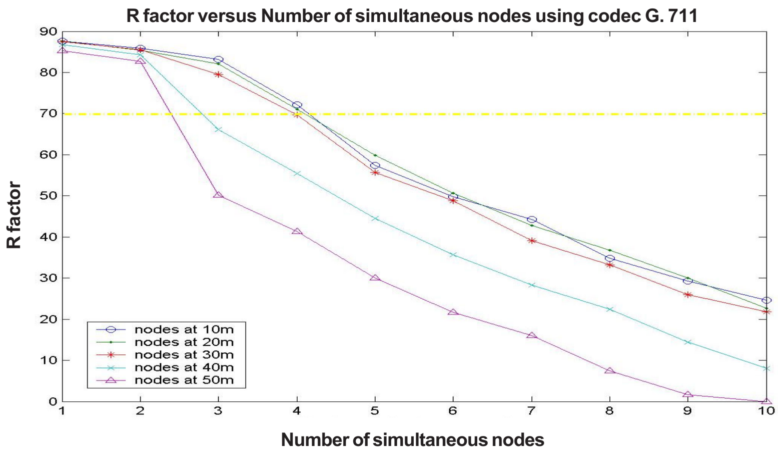

Figure 18 R factor vs. number of simultaneous nodes using G.729 codec as a function of distance to AP

This phenomenon had been considered in theoretical analysis VoIP capacities. However, maximum supportable VoIP calls for various bit rates which obtain from theory analysis are higher than E-model analysis. This is due to the fact that the possibility of collisions, retransmission, and packet loss are ignored in the theoretical analysis. When there is no packet loss in the link, capacity increase to around 6 calls at 10 meters while around 3 calls at 50 meters with G.711 or G.729. While, the biggest degradation can be seen with G.723.1, from 17 calls at 10 meters down to 9 calls at 50 meters.

The maximum achievable R-Factor for each codec in a function of data rate mode is shown in Table 10.

Table 10 Maximum achievable R factor for different codec as a function of bit rate

	Maximum simultaneous VoIP nodes		
Bit rate	**G.711**	**G.723.1**	**G.729**
11 Mbps	98.9407146	83.2004431	87.6440089
5.5 Mbps	98.9258353	83.1976402	87.4776081
2 Mbps	98.7898322	83.2251705	86.7205701
1 Mbps	97.0992373	83.1247210	85.3282811

5.0 CONCLUSION

From the capacity and QoS analysis, the limits of each VoIP codec's used in the WLAN can be determined. It can be concluded that:

(1) ITU-T Recommendation G.711 codec is the preferred choice for encoder, as this avoids both delay and additional impairments, hence have toll-quality voice.
(2) Using G.729 as higher compression speech codec did not increase the number of channels that could be handled compared to G.711. The reason is that AP congestion depends much more on the number of packets the AP has to process than on the actual bandwidth.
(3) Unless a very high voice quality requirement precludes its use, G.729 as low bit rate codec is shown to allow a capacity greater than or equal to that when G.711 is used, for a given quality requirement.
(4) G.723.1 has the ability to provide the highest capacity for VoIP calls. Voice packets are small and sent very frequently which explains the low throughput for voice packets. Besides that, the G.723.1 has some features to deal with packet-loss. So, for a very busy network, it is better to choose G.723.1 as it also gives a lower bit rate.

E-Model as voice quality with maximum capacity prediction tool provided the capacity performance which matches theoretical and simulation (throughput) analysis quite well. Overall, with G.711, a maximum of 5 VoIP nodes with R factor of 98 can be supported in a single WLAN cell. Meanwhile, 15 nodes with R factor of 83 using G.723.1 and 5 nodes with R factor of 87 using G.729.

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