TELKOMNIKA Indonesian Journal of Electrical Engineering
Vol. 12, No. 11, November 2014, pp. 7832 ~ 7844
DOI: 10.11591/telkomnika.v12i11.6580

Handover Analysis of Data and VoIP Services in 802.11b/g/n Wireless LAN

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Abstract

IEEE 802.11b/g/n WLAN with conventional Extended Service Set (ESS) infrastructure doesn't support handover feature. Handover feature allows user to migrate services between APs without losing connection. Handoff latency is one problem in WLAN to perform real-time application such as VoIP. To bring seamless handover and QoS guarantee in ESS network, we propose five network optimization methods i.e.configuring overlapping area of APs, placing APs in one roaming domain, using the same SSID and security mode, choosing APs channel by margin of 5, and configuring APs as DHCP forwarder. Handover test is done by sniffing on the client that experienced handover. In the result, there are three steps in Layer 2 handoff, i.e. probing, authentication, and re-association with maximum handoff latency is 325.02 ms in data services and 67.412 ms in VoIP. Overall throughput is 1.955 – 3.268Mbps in data services and 200.704 – 230.4 Kbps in VoIP. In VoIP services, one way delay is 39.985 - 49.18 ms, one way jitter is 9.45 - 19.57 ms, and the packet loss is 0 - 0.548%. Overall, the built network system is able to guarantee QoS in handover case, both in data and VoIP services.

***Keywords**: WLAN, handover, QoS, Sniffing, VoIP*

1. Introduction

IEEE 802.11 Wireless LAN a.k.a. Wi-Fi have seen immense in last few years. Wireless LAN offer several advantages over fixed (or "wired") networks i.e. mobility, ease and speed of deployment, flexibility, and cost [1-2]. The popular and most used 802.11 standards for Wireless LAN or Wi-Fi now are 802.11b/g/n [3-4].

There are two operation modes in WLAN i.e.independent (ad-hoc) and infrastructure. Extended Service Set (ESS) Network is classified into infrastructure mode, built by multiple Basic Service Sets (BSS) that interconnected by Distribution System (DS). For the most part, DS are wired Ethernet [5].

The process of migrating connectivity from one Access Point (AP) to another is commonly referred to handoff, handover, or roaming [6]. Generally, ESS network conventional doesn't support handover feature. Handover feature increase the mobility factor in the Wireless LAN, but handoff latency is one of problem in Wireless LAN to perform real-time applications like Voice over Internet Protocol (VoIP) [7].

Seamless handover is needed to solve handoff latency case at real-time communication in Wireless LAN and guarantee QoS during handover [8]. To bring seamless handover in Wireless LAN, we propose five network optimization methods i.e.configuring overlapping area of APs, placing APs in one roaming domain, using the same SSID and security mode, choosing APs channel by margin of 5, and configuring APs as DHCP forwarder.

1.1. Supporting Handover to ESS Network

An Extended Services Set (ESS) is defined as two or more Basic Service Sets connected by a common Distribution System (DS) [9], as shown in Figure 1. IEEE 802.11 doesn't specify the DS technology.In nearly all commercially successfull products, Ethernet is

Received July 5, 2014; Revised August 19, 2014; Accepted September 5, 2014

used as the DS technology [1, 5]. The BSS cells coverage may overlap to provide roaming or handoff capabilities [10]. With overlapping area of BSS cells, the station (STA) is possible to migrate connectivity before lose the connection from old AP. Besides that, to speed up the handover process, all of the APs must configured in the same Service Set Identifier (SSID) and security mode [11]. The same SSID and security modeconfiguration can decrease the synchronization latency between AP and STA.

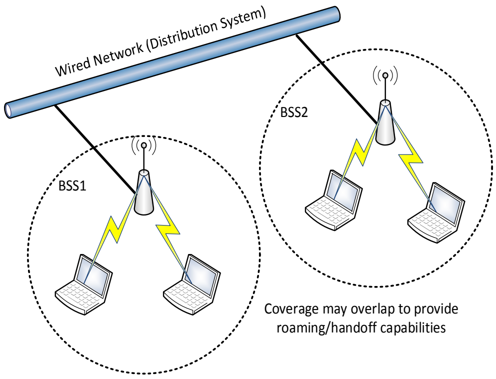

Figure 1. The Conventional Extended Services Set

There are two handover processes based on roaming domain, i.e. handover in one roaming domain and handover between roaming domains [5]. This domain is also referred as a Layer 2 network. APs that are in the same broadcast domain and configured with the same SSID are said to be in one roaming domain. Handover in one domain roaming shown in Figure 2. Handover between roaming domains mean handover between APs that located in different subnets or different networks. Handover between roaming domains can impact the application session at client and add the handoff duration.

To prevent IP segmentation in the STA when switching the APs services, the APs must be configured as Dynamic Host Configuration Protocol (DHCP) forwarder. By default, AP or Wireless Router configured as DHCP server [17]. DHCP forwarder means APs forward the DHCP protocol from one dedicated server. This technique can preventIP segmentation in the STA during handover or the IP address of the STA is the same, both before and after handover process.

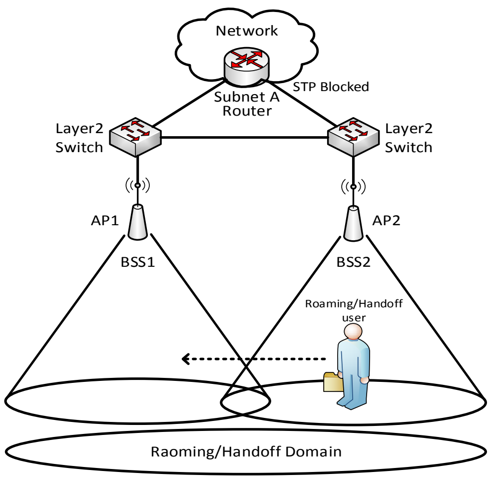

Figure 2. Handover in One Roaming Domain

1.2. DSSS non-Overlapping Channel

Direct Sequence Spread-spectrum (DSSS) is spread spectrum based on direct sequence that used in WLAN [10]. In DSSS channels, based on US and Europe standards, channel 1 for instance, operates from 2.401 GHz to 2.423 GHz (2.412 GHz ± 11 MHz), channel 2 operate from 2.406 to 2.428 GHz (2.417 GHz ± 11 MHz), and so forth. DSSS system with overlapping channels should not be co-located because there will almost always be a drastic or complete reduction in throughput [10]. Because the center frequency are 5 MHz apart and the channels are 22 MHz wide, channels should be co-located only if the channel number are at least five apart, such as 1 and 6, 2 and 7, etc. Therefore, there are three channels can be co-located, i.e. 1, 6 and 11, as shown in Figure 3.

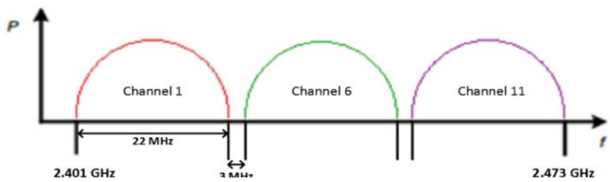

Figure 3. DSSS Non-overlapping Channel

1.3. Quality of Services Guarantee

The reliable network must provides Quality of Service (QoS) guarantee to clients. In data services, the system called very reliable if thethroughput is 99% from bandwidth. In VoIP services, a summary of the key QoS requirements and recommendations for VoIP are [12]:

a) One-way delay (mouth-to-ear) should be no more than 150 ms, it is the same as ITU-T G.114 recommendation [13];
b) One-way jitter should be targeted under 30 ms;
c) Packet loss should be no more than 1%;
d) Throughput should be guaranteed at 21–320 kbps per call.

1.4. Related Work

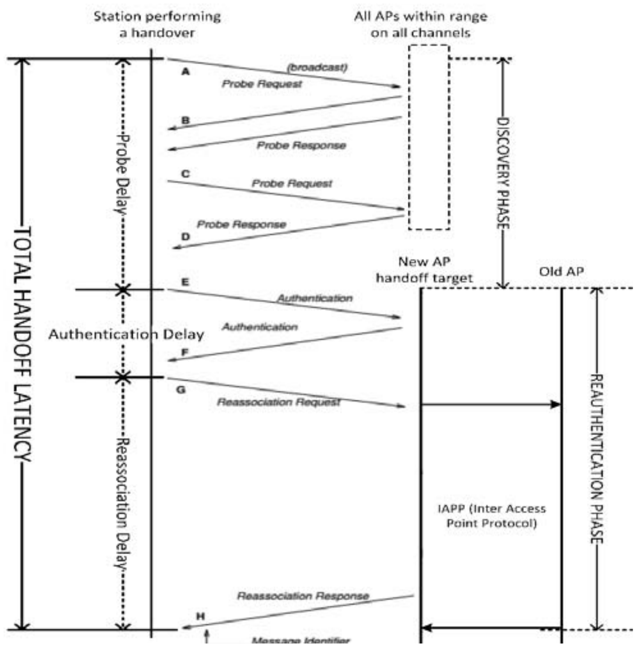

Figure 4. IEEE 802.11 Handoff Procedure and Handoff Latency

Many researches have been dedicated to analyze the handover procedure and improve the handoff performances in IEEE 802.11 WLAN [14-16]. In Mishra research [14] about an empirical analysis of handoffs using cards and AP from several vendors, there are two logical steps in handover process: discovery and re-authentication. Discovery is a scanprocess ortime to find the new AP to handover, a.k.a. probing process. Re-authentication process consist of authentication and re-association. The handover procedure and handoff latency illustrated in Figure 4.

In the experiment results [14], probe delay take 90% from total handoff latency. Total handoff latency best value is 53.3 ms and worst value is 420.8 ms. The best value occurs at cisco AP and Lucent STA and the worst value occurs at cisco AP and cisco STA.

H.S. Kim et al. [15] proposed selective channel scanning mechanism using neighbor graph. Neighbor graph is short scan algorithm which can give the nearest APs and their channels information. In the results, total handoff latency best value is 12 ms and worst value is 332 ms.The best value occurs at Selective Scanning with Unicast method in 1 neighbor and the worst value occursat Selective Scanning method in 2 neighbors.

Chung-Sheng Li et al. [16] proposed the neighbor graph cache (NGC) mechanism to reduce scanning latency while a mobile station tries to make a link-layer handover. The simulation results show that the handover delay by NGC is 2.614 to 50 ms and able to meet the criteria of VoIP application.

2. Research Method

Generally, the built network system for this experiment is 802.11 WLAN with Extended Service Set infrastructure that support seamless handover. As mentioned earlier, to bring seamless handover in this network system, we propose five network optimization methods i.e.configuring overlapping area of APs, placing APs in one roaming domain, using the same SSID and security mode, choosing APs channel by margin of 5, and configuring APs as DHCP forwarder. In this ESS network there is a local VoIP server that dedicated to VoIP testbed. The overall system design shown in Figure 5.

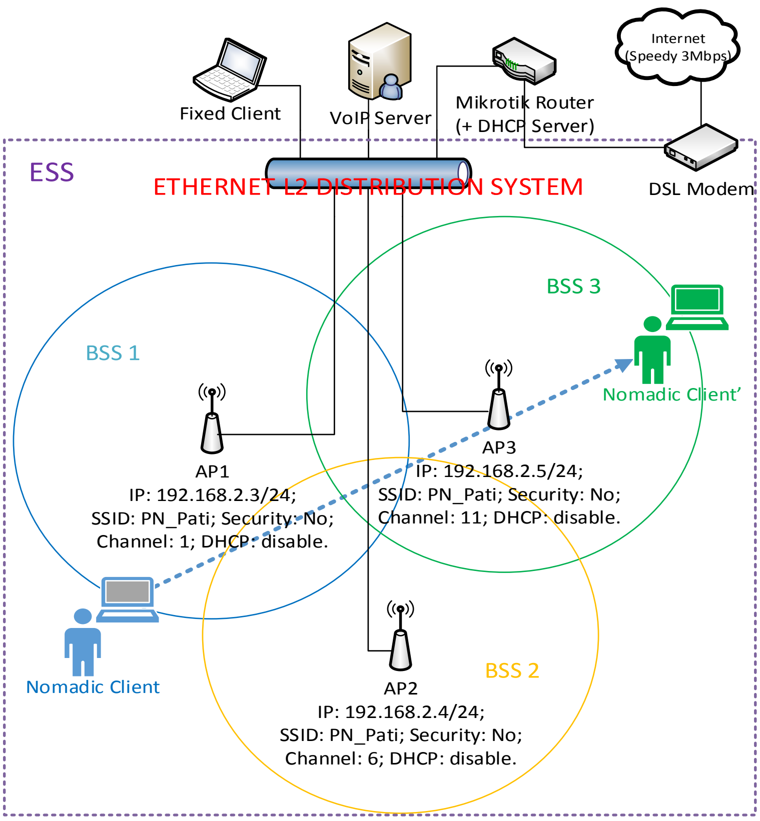

Figure 5 Overall Network System Design

2.1. The Network System Explanation

The experimentwere done in the “Pengadilan Negeri Pati” office at Pati, Central Java, Indonesia. The network system, as shown in Figure 5, built and tested in this place. The main part of this network system are Extended Service Set (ESS), VoIP server, Mikrotik router, and Fixed & Nomadic Client that be explained below.

2.1.1. Extended Service Set (ESS)

The ESS consist of three APs with overlap BSS cells. The APs interconnected by Layer 2 Distribution System (DS), it means the DS system built by Layer 2 device, such as switch. The APs place in one roaming domain and their IP address are classified in one subnet or ID Network (192.168.2.0/24). The SSID and security mode of all APs are in the same configuration, the SSID is PN_Pati and the security mode is open authentication or disable. The APs channel have margin of 5, i.e. AP1 in channel 1, AP2 in channel 6, and AP3 in channel 11. The DHCP feature in all APs are disable to perform DHCP forwarder.

The APs used in this experiment are TP-Link WR841ND v8 type. This product specifications are described bellow [17]:

a) Hardware: Atheros AR9341 Chipset, 535 Mhz CPU, 32 MiB RAM;
b) Firmware: TP-Link 3.13.33 Build 130506 Rel.48660n;
c) Standard: IEEE 802.11b/g/n, up to 300Mbps

(NB: In addition, to speed up the passive scanning process, the beacon interval of APs is set at minimum value= 40 ms. The default value in TP-Link is 100 ms)

2.1.2. VoIP Server

VoIP server is a personal computer desktop that installed Elastix v.2.4.0 64bit operating system. This personal computer (or this server) powered by AMD Phenom II X2 processor, 4GB RAM, and 500GB hardisk. This serverlocated in local network and serve the VoIP facility on the network system coverage, both wired and wireless.

2.1.3. Mikrotik Router

This device have three main functions, i.e. Routing, Network Address Translation (NAT), and DHCP server. This device powered by Intel Dual Core 2.7GHz processor, 1GB RAM, 250GB hardisk, Broadcomm BCM5704C Gigabit Ethernet, and v5.20 lv6 mikrotik routerOS.

2.1.4. Fixed and Nomadic Client

In this network system there are two clients with different characteristic, namely fixed and nomadic. The characteristicdifferences arethe mobility and the connection type. Thefixed client doesn’t move (connected to network using cable) and the nomadic client moves between APs coverage and experienced handover (connected to network using wireless). The specifications of clients are described below:

a) Fixed client: Asus A43SV powered by Windows 8 Pro 64bit and Realtek PCIe GBE Family, 10/100/100Mbps ethernet NIC;
b) Nomadic client: Sony Vaio VPCEB25FG powered by Windows 8 Pro 64bit and Atheros AR9285 Wireless Adapter.

2.2. Testbed Methodology

In this section, we described how to test the built network system in handover case, both in data and VoIP services. This testbed aims to test the ability of the built network system in delivering seamless handover feature and guarantee the QoS. To obtain valid data on the results of testbed, we performed network sniffing at the client. The sniffer software are Xirrus Wi-fi Inspector [18], Microsoft Network Monitor, Colasoft Capsa [19], and Wireshark [20-21]. The testbed parameters are shown in Table 1 below.

Table 1. Testbed Parameter

No.	Parameter	Value
1.	VoIP Codec	G.711 PCM µ-*law*
2.	Mobility speed	Normal walk ±1.2192m/s [22]
3.	Mobility path	Shown in Figure 6

2.2.1. Handover at Data Services

Handover at data services testbed is done by sniffing on the client that experienced handover (with parameter and path specified) within ±60 seconds sniffing time when the client was doing the download activity from googlevideo.com.

2.2.2. Handover at VoIP Services

Handover at VoIP services testbed is done by sniffing on the client that experienced handover (with parameter and path specified) within ±60 seconds sniffing time when the client was doing the VoIP telephone activity (between nomadic and fixed client).
(NB: sampling performed 3 times on each path, both for data and VoIP services)

2.2.3. VoIP QoS per Receive Signal Strength Indicatior (RSSI)

This testbed means Quality of Service (QoS) testing in VoIP based on RSSI decrement. We determinedthe RSSI decrement value is 10 dBm with the lower limit is range (-30) to (-40) dBm and upper limit is range (-70) to handover RSSI threshold. As an analogy, firstly we measured VoIP QoS at RSSI range (-30) to (-40) dBm, secondly we measured again at range (-40) to (-50) dBm and so on. VoIP call duration is ±60 per RSSI decrement. This testbed usedthe same parameter and path as the previous testbed.

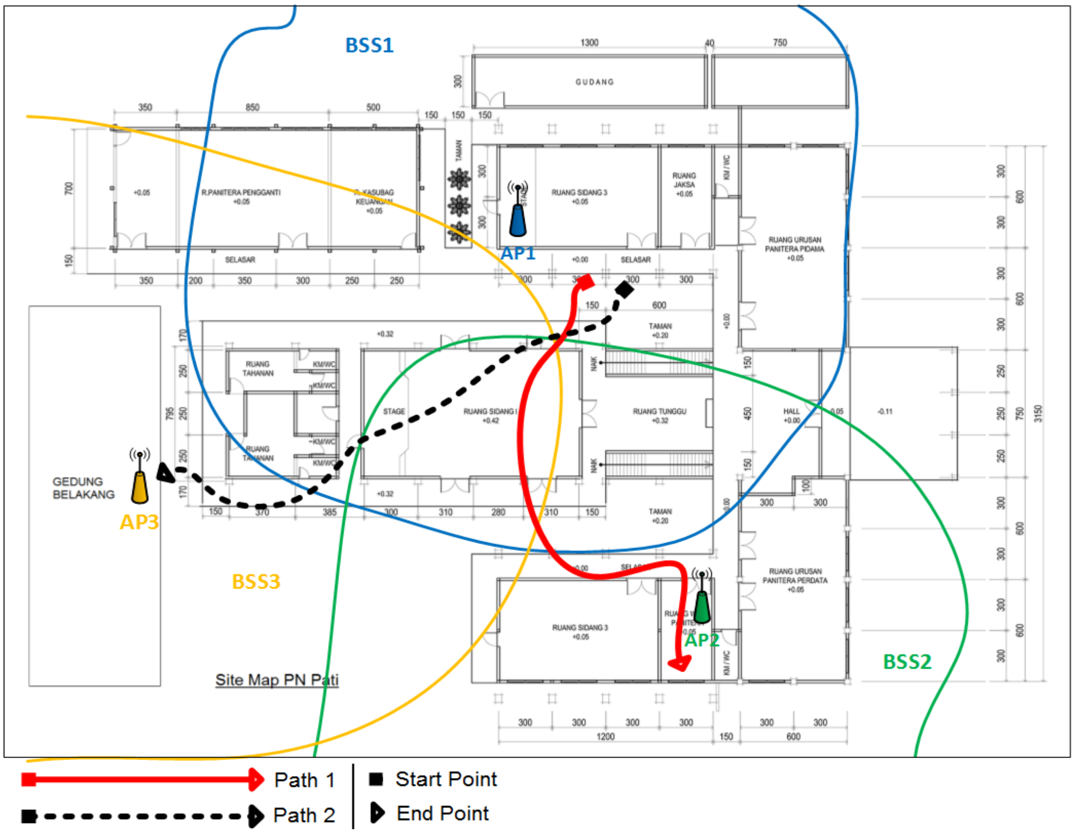

Figure 6. Path of Mobility

3. Results and Analysis

In this section, we described and analyzed the results of the testbed. The testbedresults and analysis are about handover process (in Layer 1, Layer 2, and upper Layer OSI perspective), the Quality of Service (QoS) during handover, and QoS per RSSI decrement.

3.1. Handover at Data Services in Layer 1 OSI Perspective

As shown in Figure 7 below, the x-axis shows time in seconds andthe y-axis shows RSSI in dBm. The first graph shows RSSI when client is served by AP1 (SSID: PN_Pati & MAC: 10-FE-ED-9C-04-DA) and the second graph shows RSSI when client is served by AP2 (SSID: PN_Pati & MAC: C0-4A-00-EC-93-10). Can be seen that nomadic client never loses the connection during handover. Once the nomadic client moves away from AP1 coverage and RSSI getting down until RSSI handover threshold (-72 dBm at this sample), in the next second, the nomadic client directly served by AP2 at -52 dBm RSSI without losing the connection. It means the handover has been successfully performed in Layer 1 OSI perspective.

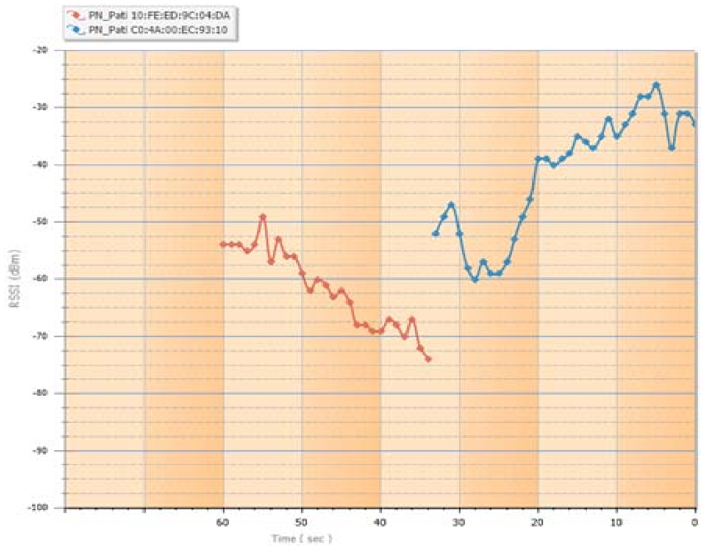

Figure 7. RSSI during Handover at Data Services (path 1, sample 1)

All of other sample results aren't much different with the first sample that explained in the previous, they showed the same procedure. The handover procedure in Layer 1 OSI perspective is once the client moves away from old AP and the RSSI getting down until RSSI handover threshold, in overlapping APs area, the client will switch directly to new AP services that have higher RSSI. Overall in the results, the RSSI threshold value is more than -70 dBm.

3.2. Handover at Data Services in Layer 2 OSI Perspective

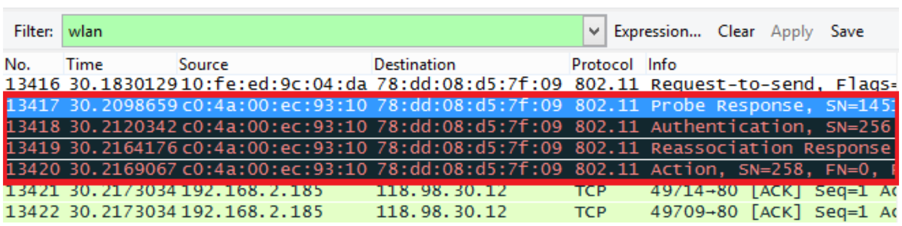

Figure 8. Layer 2 Handover Procedure (data services, path 1, sample 1)

Figure 8 shown the sniffing result using Microsoft Network Monitor and analyzed with Wireshark [20-21] at sample 1 path 1 testbed. Can be seen in the Figure 8 above, there are three logical steps in a handover: Probing, Authentication, and Re-Association. At packet number 13416, AP1 (MAC: Tp-LinkT_9c:04:da) still sent RTS frame to client (MAC: HonHaiPr_d5:7f:09). At the next packet, AP2 (MAC: Tp-LinkT_ec:93:10) sent probe response as response from client's probe request in active scanning. Then at packet number 13418, AP2 sent authentication message to validate the client before join the network. After the authentication successfully, at packet number 13419, AP2 sent re-association message to client that means allows client to re-join to the network. After that, the next 802.11 packet is contain action information and the data traffic (TCP) working again. It means the handover has been successfully performed in Layer 2 OSI perspective.

All of other sample results aren'tmuch different from the first sample that explained in the previous, they have the same procedure. The handover procedure in Layer 2 OSI perspective consist of three logical steps: Probing, Authentication, and Re-Association. By analyzing the timestamp of handover step at the sniffing results, we can get the handoff latency, include Probe Delay, Authentication Delay, and Re-association Delay [14]. Table 2 shown total handoff latency that generated in the testbed.

Table 2. The Handoff Latency at Handover of Data Service

Testbed		**Probe Delay (ms)**	**Auth. Delay (ms)**	**Re-ass. Delay (ms)**	**Total (ms)**
path	sample				
	1	29.022	4.383	0.489	33.894
1	2	282.93	4.282	6.167	293.379
	3	314.27	4.26	6.49	325.02
	1	41.896	4.519	20.983	67.398
2	2	80.986	4.282	6.712	91.98
	3	94.709	4.298	10.3	109.307

From Table 2 above, known that most latency is caused by probing phase. This result is similar toMishra et al. [14] research, that probing delay up to 90% from total handoff latency. The “big” latency in probing phase is caused by scanning channel time (frequency changes) before client switch the AP services.

Based on Table 2, total handoff latency at data service handover best value is 67.398 ms and worst value is 325.02 ms. By comparing this total handoff latency with the previous researches [14-15], the total handoff latency worst value in this testbed is better, however the best value is lower.

3.3. Handover at Data Services in Upper Layer OSI Perspective

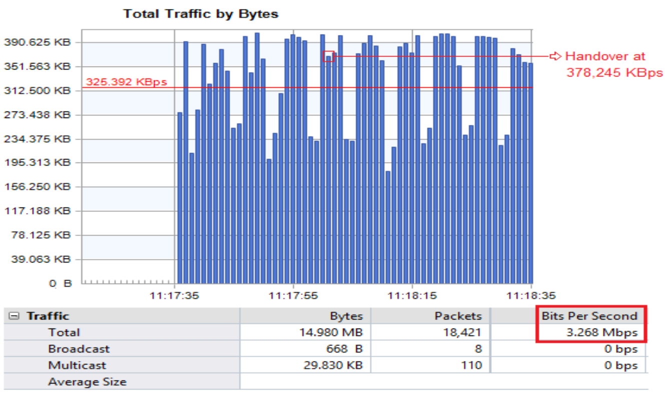

Figure 9. Throughput (data services, path 1, sample 1)

From Figure 9 above can be seen that the handover process in built network system doesn‘t have significant effect on throughput. The throughput at handover time stillat 378.245 KBps. However, when viewed from the fluctuation pattern of throughput, the throughput at handover time is lower than after and before. The overall throughput value in this testbed is 3.268 Mbps. This value is good because the ISP Speedy (internet connection source in this system)have bandwidthup to 3 Mbps. The throughput in all of samples at handover of data services shown in Table 3.

Table 3. The Throughput at Handover of Data Services

Testbed path	Testbed sample	Throughput at handover (KBps)	Overall throughput (Mbps)
1	1	378.245	3.268
	2	234	3.245
	3	216.24	1.955
2	1	12.5	1.866
	2	43.75	3.238
	3	32.083	3.2043

From Table 3 above, the worst throughput at handover occurs in sample 1 path 1 testbed, but its value still at 12.5 KBps or 100 Kbps. It is not abig problem because overall throughput still reach 1.866 Mbps. Overall, the built network system give QoS guarantee in data services during handover in upper Layer OSI perspective.

3.4. Handover at VoIP Services in Layer 1 OSI Perspective

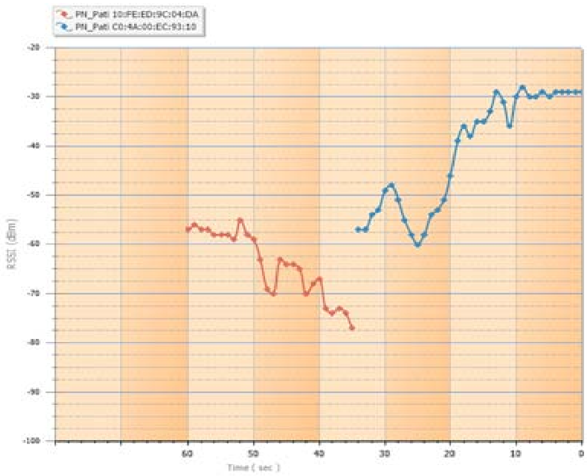

Figure 10. RSSI during Handover at VoIP Services (path 1, sample 1)

Generally, the handover procedure in Layer 1 OSI perspective at VoIP and data service have same pattern, as shown in Figure 10. Once the client moves away from old AP and RSSI getting down until RSSI handover threshold, in overlapping APs area, the client will switch directly to new AP services that have higher RSSI without losing the connection. Overall, the RSSI threshold value in VoIP services handover is more than -70 dBm.

3.5. Handover at VoIP Services in Layer 2 OSI Perspective

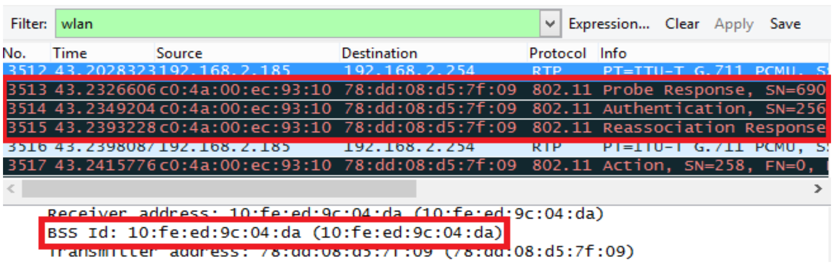

Figure 11. Layer 2 Handover Procedure (VoIP services, path 1, sample 1)

As shown in Figure 11 above, the handover procedure in Layer 2 OSI perspective between data and VoIP services are identical. This result is shown the same patternin all of sample results. There are three logical steps in handover of VoIP servicesprocess: Probing, Authentication, and Re-Association. Table 4 shown the handoff latency value at handover of VoIP service.

Table 4. The Handoff Latency at Handover of VoIP Service

Testbed		Probe Delay (ms)	Auth. Delay (ms)	Re-ass. Delay (ms)	Total (ms)
path	sample				
1	1	32.088	4.402	2.255	38.745
	2	31.68	4.45	3.753	39.883
	3	19.269	4.26	2.763	26.292
2	1	40.443	4.273	0.6	45.316
	2	46.755	15.98	4.677	67.412
	3	26.158	4.362	2.573	33.093

Based on data above, it can be said that this results are identic with the data services testbed results, thatmost handoff delay is caused by probe delay. The probe delay is up to 90% from total handoff latency. The handoff latency in this tesbed isrange of 26.292 to 67.412 ms and the average is 41.79 ms.By comparing this handoff latency value withthe handoff latency in previous researches [14-16], this handoff latency value is the best. This result indicates thatour system sucessfully deliver seamless handover for VoIP application better than previous researches [14-16].

3.6. Handover at VoIP Services in Upper Layer OSI Perspective

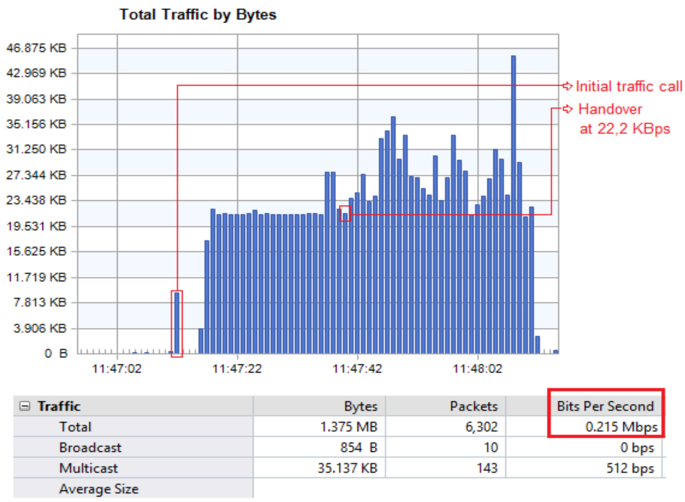

Figure 12. Throughput (VoIP services, path 1, sample 1)

As shown in Figure 12 above, the handover at VoIP services in built network system doesn't have significant effect on throughput. At handover time, the throughput still reach 22.2 KBps. However, when viewed from the fluctuation pattern of throughput, the throughput at handover time is lower than after and before. Overall throughput in this testbed is 0.215 Mbps or equivalent with 220.16Kbps. The throughput in all of samples at handover of VoIP services shown in Table 5 below.

Table 5. The Throughput at Handover of VoIP Services

Testbed path	Testbed sample	Throughput at handover (KBps)	Overall throughput (Kbps)
	1	22.2	220.16
1	2	22.2	206.848
	3	22.1	214.016
	1	25.83	215.04
2	2	22.43	200.704
	3	22	230.4

From Table 5 above, known thatoverall VoIP throughput value is 200.704 - 230.4 Kbps. The worst throughput value at handover occurs at path 1 sample 1 testbed, it is 22 KBps or 176 Kbps.Generally, all of VoIP throughput values is good because it is larger than the codecs bit rate that used in the testbed. The codec that used in the testbed is G.711 PCMU with 64 Kbps bit rate [23].

3.7. End-to-end VoIP Quality of Service (QoS) During Handover

End-to-end VoIP QoS in the built network system means VoIP quality of service from fixed client to nomadic client, the analogy shown in Figure 13. To obtain end-to-end VoIP QoS values is done by analyzing two segments: nomadic client <-> server and server <-> fixed client. Table 6 below shown the values of end-to-end QoS in handover at VoIP services testbed.

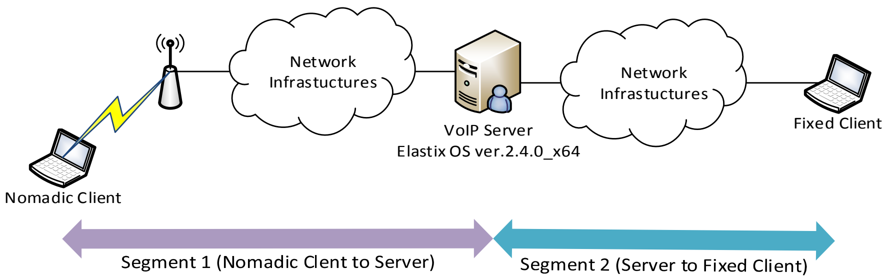

Figure 13. End-to-end VoIP QoS Analogy

Table 6. End-to-end VoIP QoS in Handover at VoIP Services

Testbed		End-to-end$_1$ QoS			End-to-end$_2$ QoS		
Path	Sample	Delay (ms)	Jitter (ms)	Loss (%)	Delay (ms)	Jitter (ms)	Loss (%)
	1	40.12	9.45	0	40.28	19.57	0.55
1	2	40.19	18.08	0.02	40.19	18.17	0.51
	3	49.18	17.98	0	40.37	18.32	0.03
	1	40.13	18.56	0	39.98	18.04	0.03
2	2	39.99	17.88	0	40.02	19.23	0.07
	3	40.28	18.13	0.16	40.27	18.08	0.54

Notes:
End-to-end$_1$ QoS= nomadic client→ fixed client
End-to-end$_2$ QoS= fixed client→nomadic client

By comparing the QoS values in table above (Table 5 & 6) with QoS recommendations for deliver a good VoIP services [12-13], the QoS values are still in the recommendations. The one way delay is 39.985 - 49.18 ms, one way jitter is 9.45 - 19.57 ms, packet loss is 0 - 0.548%, and overall throughput is 200.704 – 230.4 Kbps. It means that the overall built network system is able to bring seamless handover with QoS guarantee.

3.8. End-to-end VoIP Quality of Service (QoS) per RSSI

In this section, we described the result of QoS per RSSI testbed.The result shown in Table 7 for path 1 testbed and Table 8 for path 2 testbed.

Table 7. End-to-end VoIP QoS per RSSI Decrement at Path 1 Testbed

RSSI	QoS end-to-end$_1$			QoS end-to-end$_2$			Overall Throug. (kbps)
	Delay (ms)	Jitter (ms)	Loss (%)	Delay (ms)	Jitter (ms)	Loss (%)	
R1	39.98	17.93	0	39.97	18.41	0	161.79
R2	39.99	18.42	0	39.86	18.59	0	227.33
R3	40.08	18.51	0	39.97	17.83	0	254.98
R4	39.98	18.09	0	39.98	17.99	0	187.39
R5	40.14	18.47	0.02	40.19	18.16	0.48	152.06

Table 8. End-to-end VoIP QoS per RSSI Decrement at Path 2 Testbed

RSSI	QoS end-to-end$_1$			QoS end-to-end$_2$			Overall Throug. (kbps)
	Delay (ms)	Jitter (ms)	Loss (%)	Delay (ms)	Jitter (ms)	Loss (%)	
R1	39.99	18.17	0	39.99	21.66	0	178.688
R2	39.99	17.92	0	39.97	21.3	0	176.128
R3	39.98	17.92	0	39.93	18.4	0	173.056
R4	39.98	17.89	0	39.87	17.83	0	171.008
R5	40	17.64	0	40.01	9.75	0.03	200.192

Notes:
R1= (-30)-(-40) dBm; R2= (-40)-(-50) dBm; R3= (-50)-(-60) dBm; R4= (-60)-(-70) dBm; and R5= (-70) dBm - handover.
End-to-end$_1$ QoS= nomadic client→ fixed client.
End-to-end$_2$ QoS= fixed client→nomadic client.

From the table above, can be seen that the VoIP QoS values include delay, jitter, packet loss, and throughput doesn’t have a straight correlation with the RSSI decrement. It happens because the nomadic client, at this testbed, doesn’t move and have stable RSSI despite the RSSI decrement.

The worst delay occurs at (-70) dBm – handover RSSI and the packet loss only occurs at (-70) dBm – handover RSSI. It means that handover process add the VoIP delay value and make a packet loss. However, overall QoS value still recommended to deliver a good VoIP services [12-13].

4. Conclusion

To bring seamless handover feature in ESS network, we have proposedfive network optimization methods i.e.configuring overlapping area of APs, placing APs in one roaming domain, using the same SSID and security mode, choosing APs channel by margin of 5, and configuring APs as DHCP forwarder. As described in the result and analysis, this method successfully brought handover feature for data and VoIP services. Overall, the built network system successfully gave seamless handover feature with QoS guarantee.

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