Analysis and Measurement of Session Setup Delay and Jitter in VoWLAN Using Composite Metrics

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ABSTRACT

In this paper we look at the signaling for session mobility using the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP). Bearing in mind that perceived voice quality is sensitive mostly to delay variation (also called jitter), we work on bounding that value for providing a better voice Quality of Service over Wireless LAN (WLAN). For mobility we use a hybrid SIP-MIP stack. The basic idea is to combine the strengths of MIP for doing fast handovers and the strengths of SIP for powerful session adaptation capabilities. Numerical calculation figures as well as real implementation results are provided. It can be observed that the session update cycle during mobility can be modified in such a way that session delay variation bounds achieved are quite low and there are almost no spikes. We use composite performance metrics that we defined in previous work to analyze the correlation between the data patterns for session signaling delay in the classical case and our proposed case to better highlight the achieved performance delta.

Categories and Subject Descriptors

C.2.6 [Computer-Communication Networks]: Internetworking.

General Terms

Algorithms, Performance, Design, Experimentation.

Keywords

Session initiation protocol, session description protocol, quality of service, wireless LAN, IP mobility.

1. INTRODUCTION AND MOTIVATION

Voice is sensitive to jitter more than to any other network Quality of Service (QoS) parameter. Especially in wireless networks that are multi-hop or that involve small or medium-sized cells (e.g. IEEE 802.11 networks), where the link is more error prone and

transmission is more difficult to manage and coordinate, voice traffic faces a major challenge. Voice over Wireless LAN (VoWLAN) and how to improve or provision its quality has recently become an important research issue.

For simply nomadic users, the quality of VoWLAN depends on the service level delivered by the visited access point. The performance of that particular access point then depends on the operator or service provider to whom it belongs and also on the load it experiences. For mobility scenarios however, where the user roams in 802.11 open access networks [25], there is a frequent handover taking place using Mobile IP (MIP) and session adaptation at OSI/ISO Layer 5 using the SIP protocol. We provide some insights on using hybrid SIP-MIP mobility in Section 3 of this paper.

Now for a mobile pedestrian user, having an average speed of 4-5 km/hr or 1.3 m/sec, he would traverse a distance of 50 meters in 38 seconds and a distance of 100 m in 76 seconds.

There is plenty of time for a mobile terminal to perform a handoff from one access point to another. However, for a vehicular speed of 50 km/h or 13.9 m/s, an access point coverage range of 50 meters would be traversed in about 3.6 seconds and one of about 100 m in approximately 7.2 seconds. So the point is that the higher the mobility speed, the harder it is to perform a handoff and the less time it is desirable that a session setup or update consumes on the radio link. Real-time applications in particular have strict requirements on network QoS parameters.

To solve the challenges posed by the problem in question, it is necessary to design an architecture that enables different signaling paths for improved session management. Furthermore, it has to be ensured that all parameters needed to build or update a multimedia session (in this case voice) are available in real-time. This is shown in Section IV of the paper.

Session setup time in multimedia using e.g. SIP is affected by the quality of the used link, and SIP uses a retransmission timer that doubles in size upon the occurrence of errors [6]. Therefore, it makes sense to model session signaling in such a way that the overall setup or update time is smaller, reducing the probability of an error occurring.

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This paper is organized as follows. After this introductory section, Section 2 analyzes related work in the area. Then Section 3 discusses multimedia session setup identifying where there is room for improving performance. Section 4 shows a high-level architecture of Open Access Networks (OAN) based on WLAN, and then explains the improved signaling scheme that is used to bound jitter for voice traffic. Section 5 then analyzes the results of the newly proposed scheme quantitatively and finally Section 6 concludes the paper.

2. RELATED WORK

A wide range of papers has been analyzed in parallel to conducting our work. In fact, we observed that as SIP is adopted in various access technologies, it faces a challenge in each access technology it is deployed in. VoIP over 3G is thoroughly discussed by Prasad et al in [6]. The authors analyze the different phases in setting up a SIP-based voice session and explore the different cases via numerical analysis in order to obtain an estimate on what it would cost in terms of time and resources to establish an appropriate VoWLAN or video-over-WLAN session in 3G networks.

The authors in [13] explore statistical multiplexing and try to suppress the large overhead in VoWLAN. They perform voice multiplexing using a polling mechanism in the contention-free period and a deterministic priority access for voice traffic in the contention period. They also work on reducing the overhead for voice traffic.

In [14] McGovern et al. address the problem of link adaptation using media codecs in WLAN. Different codecs are believed to result in different levels of congestion, and thus the system the authors propose switches back and forth between codecs to reduce the measured or perceived congestion level.

Bacciu et al. [12] presents a fuzzy approach to determine a soft admission control mechanism for Voice-over-IP services over Wireless LANs. A framework is defined where the provider may express the network status and the client their preferences by the means of an approach based on Fuzzy Set Theory.

Brännström et al. [4] proposes a mobility support system integrating the benefits of application-layer SIP mobility with network-layer MIP mobility where a cross-layer information system exchanges context for mobility adaptation.

Åhlund et al. [31] proposes a Running Variance Metric performance metric of wireless local area network used to predict relative traffic loads of available access points at the network layer.

The different approaches in this section either take a step in introducing combined metrics in mobility scenarios or try to solve the congestion and bad coupling of voice traffic and IEEE 802.11. We move on in the next sections to show how hybrid SIP-MIP mobility is used on a single stack with access point controllers brokering for airtime resources after every handover, and then measure the system performance. Assessing system performance in the case of SIP classical session negotiation over MIP, and then with parameter injection [3] using specialized limited signaling and more intensive computations, allows us to show a delta in performance. The key that enabled our quantitatively improved result is the architecture with proper placement of components, and specialized signaling that lead to lower session update delays

and thus lower errors and shorter occupation of the airtime needed for VoWLAN session initiation or resumption.

3. HYBRID MIP-SIP MOBILITY MECHANISMS

The concept of hybrid Session Initiation Protocol and Mobile IP joint mobility scheme has been addressed in different contexts. A pioneering work in this area is by Tafazolli et al [33]. A MIP stack on top of which SIP resides can exploit the architecture as if using two superimposed worlds: the MIP world and the SIP world in a synchronized and aligned fashion.

We used a hybrid Mobile-IP and Session Initiation Protocol stack as a mobility solution for multimedia. MIP is used to perform fast handover and association to a new access point and to acquire a new IP address. Then we used with the Birdstep MIP client [29] a specially modified API that sends updates and event notifications upwards. In this way, the SIP mobility module which operates on logical layer 5 in the ISO/OSI model can then adapt a multimedia session (either voice or video) to the new network conditions and after MIP has handled the basic mobility part.

This hybrid stack has been used in different contexts [33] but not in the way we use it in this work. The basic idea is to combine the strengths of MIP for doing fast handovers and the strengths of SIP for powerful session adaptation capabilities. The reason for this is the fact that multimedia is often handled with SIP or peer protocols and WLAN environments have relatively small cells, so when traversing them and holding a VoIP call, the session update delay upon incurring handovers is a significant parameter. We analyze this value and show how real measurements of an existing prototype depict the improvement in performance.

4. MULTIMEDIA SESSION SETUP ANALYSIS

Nomadic mobility where a user moves from a location to another and then uses the Internet in a stationary manner is a common scenario. This is handled nowadays with the 3G or CDMA modems available from operators and service providers. On the other hand, WiFi grew beyond the hotspot scenario to become the main technology for open access networks (OAN) [5, 22, 25, 26]. OANs enable mobility within a so called distribution system where access points are connected e.g. via a virtual operator to provide continuous network coverage that enables the user to seamlessly roam from one AP to another.

Now bearing in mind that 802.11b and g WiFi cells are larger than those of 802.11a, but still relatively small, high speed mobility is hard to support. For this reason, we analyzed the cases with pedestrian speed mobility and low to moderate speed vehicular mobility, up to 50 km/h.

Pedestrian speed is for a user roaming with his terminal and traversing many WiFi access points in the process. Low and moderate speed vehicular motion is for some types of transportation which have speed limitations due to technical or regulatory reasons.

- the first column presents mobility speed in km/hr
- the second column presents mobility speed in m/s
- the third column shows t_1 , the time in seconds it takes a terminal to traverse an access point with coverage diameter 100 meters

- the fourth column shows t_2 , the time in seconds it takes a terminal to traverse an access point with coverage diameter 50 meters

Table 1. Motion speeds and average durations of stay within a WiFi access point.

Case	km/h	m/s	t_1 (sec)	t_2 (sec)
Walking	5	1.38	76.92	38.46
Riding	15	4.17	23.98	11.99
Vehicular	20	5.56	17.99	8.99
	25	6.94	14.41	7.20
	30	8.33	12.00	6.00
	35	10.29	10.29	5.14
Moderate Vehicular	40	11.11	9.00	4.50
	45	12.50	8.00	4.00
	50	13.89	7.20	3.60

As Table 1 shows, there is enough time-room for a station to perform a handover, but the session update or setup delay can sometimes become comparable with the station's duration of stay within an access point if the error rate on the wireless part is too high [6].

Theoretical calculations in [6] show that the session setup delay for SIP in 3G networks used with UDP is about 4.61 seconds when using the Radio Link Protocol (RLP) and a transmission rate of 9.6 Kbps, and 2.9 seconds when using a transmission rate of 19.2 Kbps. The setup delay time for a SIP-based session is a few seconds over a narrow-band rate. For WLAN (broadband), this setup time can vary but would still be in the range of a 100-200 milliseconds as measurements show in [3].

To establish a session, the sequence: INVITE, 183 SDP, PRACK (final SDP), 200 OK, 180 RINGING, 200 OK, and ACK is used. SDP stands for session description protocol [27].

The same parameter sequence is necessary when updating a session due to mobility or upon the need to change session parameters (e.g. CODEC, resources). The only difference between a session update and a session initiation is that the former uses a SIP RE-INVITE whereas the latter uses SIP INVITE. Therefore, we make the observation that even a session update in WLAN which involves pure signaling can substantially disrupt or degrade a voice call when traversing many cells. VoWLAN (Voice over WLAN) this has to consider different signaling alternatives that reduce signaling delay and provide more bounded jitter with fewer spikes and a lower jitter on the average, yielding a much more acceptable perceived audio quality in IEEE 802.11.

Table 2. Message sizes and frame count per message.

Messages	Payload (bytes)	Message (bytes)	# frames, 9.6 kbps	# frames, 19.2 kbps
SIP INVITE	700	728	37	19
SIP 183	835	863	44	23
SIP PRACK	538	586	30	16
SIP 200 OK	545	573	29	15
SIP 180	349	377	19	10
SIP ACK	300	328	17	9

The total payload size in bytes for a SIP session setup or update is: $700+835+538+545+349+300 = 3267$ bytes. This is a lot for a "lightweight" signaling protocol. Calculating the size of the total data required, which includes payload and overhead yields: $728+863+586+573+377+328 = 3455$ bytes.

Then it is obvious that a modified signaling scheme that spares some of the messages listed in Table 2 would significantly reduce session setup time and also the probability of errors occurring during updating the session when roaming between APs.

The higher the used rate, the fewer frames are needed to transmit the required information. As the rate in Table 2 is slightly more than doubled from case1 to case 2 being first 9.6 and then 19.2 Kbps respectively, the number of frames needed shrinks to about a half. Therefore as the rate is increased, the number of frames decreases proportionally. WLAN links are error prone. Thus for a particular Frame Error Rate e.g. 0.01%, the probability of getting an error gets smaller as the number of transmitted frames decreases. This makes session signaling delay cycles for sessions with higher rates lower.

5. SUGGESTED ARCHITECTURE AND IMPROVEMENTS

5.1 QoS in Open Access Networks

[17, 22] describe a self-contained resource allocation mechanism that was deployed on AP controllers in the architecture to perform rate control. Regulating the QoS budget of a station is done at an IP level. A tuple of parameters indicating the average packet size and the average packet rate is assigned to a station and updated dynamically by the QoS broker residing on each AP. So as the load on an access point increases, the broker redistributes capacity between stations.

Figure 1. Basic architecture.

Now using Table1 and Figure1 we obtain Equation 1.

Equation 1. Total signaling delay.

$$delay_{total} = 2(proc_{dc} + prop_{dw} + proc_{dw}) + prop_{db}$$

Figure 1 labels the various delays with symbols, so we get a one cycle delay of:

With $proc_{dc}$ being the processing delay on the client, $prop_{dw}$ being the propagation delay towards the AP, $proc_{dw}$ is the

processing delay on the WLAN AC controller, and $prop_{ab}$ being the propagation delay on the backend, respectively. The value $delay_{total}$ stands for one cycle of E2E delay between two clients using SIP for a VoWLAN session. The processing delay for various messages depends on the type of the message received or sent but take pretty close values.

For the classical case of negotiating session parameters either when setting up an initial session or upon updating the session settings due to mobility or resource reallocation, 6 cycles are needed, one for each of the rows in Table 2. Hence, the entire signaling cost is approximately 6 times the $delay_{total}$ variable.

In order to measure this value for a session setup, we used Java timers within our SIP-stack based on the JAIN-NIST [28] base version. The timers, which are based on threads, are started when a frame processing starts and stopped when the other party (correspondent client) finishes its processing. This way, the E2E delay in a real-implementation of the system depicted in Figure 1 is measured. Section 5 of this paper outlines some figures from the measurements made on the system.

5.2 Experimental Setup and Signaling Scheme

Our MIP used for the experiments used Eager Cell Switching (ECS) as an Agent Discovery and Motion Detection (ADMD) method. ECS is designed to trigger immediate handover to the new foreign agent (FA) as soon as the mobile node (MN) receives an Agent Advertisement (ADV) message from that FA. Mobile IP is used on both cases of the experimental MIP-SIP stack, so the delta achieved in delay reduction and jitter bounding is due to improved signalling in the SIP part using the parameter injection algorithm presented in [3].

As seen in the previous subsection, there is a relatively high number of messages going back and forth until a session is established in SIP. This propagation is prone to errors in WLAN causing substantial delays as calculated in [6].

Looking more closely at the RFCs in this area, such as [7, 18, 19, 21, 27], the following branch-case can be observed as mentioned in RFC 3264 [27]: “An Offer/Answer Model Session Description Protocol”: “If multiple formats are listed, it means that the offerer is capable of making use of any of those formats during the session. In other words, the answerer MAY change formats in the middle of the session, making use of any of the formats listed, without sending a new offer”.

In other words, after establishing an initial session between two peers A and B engaged in a multimedia (video or voice or both) session, then when one node moves, it is allowed by RFC 3264 to modify the data stream and piggyback the new session settings with the new stream. This signaling procedure is depicted in figure 3, the “session update” part.

Figure 2. Experimental setup.

We used four access points placed 50 meters apart as depicted in Figure 2. Human testers were used for walking and traversing the different 802.11b APs. Windows XP equipped laptops were used as test clients with a hybrid MIP-SIP mobility client developed jointly with Birsdstep [29]. When the signal strength of the next AP was stronger than that of the one the client was associated with, a handover was triggered (pure MIP handover). After performing the hard handover in Mobile IP and the handoff is complete, the Mobile IP API sends a trigger to activate the “REINVITE” message on the SIP part of the stack. The QoS controller on the access point assigns a new resource pair to the client upon joining that AP. Average Packet Size (APS) and Average Packet Rate (APR). Details concerning architecture, mathematical computations and implementation for realtime resources as well as new session parameters are in [3, 16].

After a client switches APs, MIP performs an update and then the new network parameters are fed into the SIP module via the AP QoS controller as well as upwards by the MIP API. “The RE-INVITE” is launched with the newly computed parameters. For a multimedia session SIP proxy on each controller is updated with the new location (IP address) of the callee. As the user continues to move, at every handover, there is a MIP-handover followed by a SIP-SDP re-invite. The MIP handover delay is represented logically by the following time span: let the time-stamp of the last received packet (can be traced via ICMP) on the first interface be t_1 , and the time-stamp of the first received packet on the new interface be t_2 , then the handover cycle delay is $t_2 - t_1$. SIP session signaling delay is measured via thread based timers. After the trigger from the MIP API reaches the SIP-part, a SIP-SDP session update cycle is launched. In this cycle, there is a computation part and session re-launch part with new parameters. Once the session is re-launched with a data stream obtained on the stack with updated session parameters, the timer is stopped.

Figure 3. Modified signaling procedure.

The AP controller module that acts also as a QoS broker for roaming clients re-assigns a new rate for the client as it joins this access point [3].

Figure 4. Delay components in the architecture used.

As Figure 4 shows, there is a delay component due to Mobile IP handovers and then on the upper part of the hybrid stack, there is the SIP delay. On each client, MIP and SIP interact with each other via an API. MIP implementation and API has been provided by Birdstep [23] and we developed the SIP stack on top for session management. After the session settings are re-computed and a new Real-time Protocol (RTP) stream is re-launched on the stack, the cycle is closed and this is the session signaling delay in SIP.

The next section will numerically compare the results for the jitter and delay variables.

6. QUANTITATIVE ANALYSIS

In this section, we test the two signaling cases for a SIP-session update for a voice call between two parties A and B. A is mobile, roaming from one access point to another whereas B is stationary.

In both cases the mobile nodes A and B both use a hybrid MIP-SIP stack as depicted in Figure 4. The core difference between case1 and case2 is that in the former, the classical SIP negotiation path is used to update the session parameters of the VoWLAN

call. In the latter case, the signaling path shown in Figure 3 is followed. Basically, what is done in case2 is that: after joining a new AP during mobility, the QoS controller module attached to the AP allocates a new rate to the client. The client then re-computes the new session parameters for all multimedia sessions including the VoWLAN session [3, 5]. After this computation is done, the data stream based on RTP (real-time protocol) is launched from A to B and the updated session parameters are piggybacked. Avoiding unnecessary negotiation after every time a new AP is joined reduces delay as shown in Figure 5.

Figure 5. Session update delay times comparison for two different signaling methods.

The fact that a lower session update time is obtained in the second case that modifies the data stream avoiding negotiation automatically means that the user will experience a milder perturbation in the VoWLAN session. Perceived audio quality is furthermore more sensitive to session jitter, or delay variation. Using the delay values in Figure 5, we compute the jitter values and plot them in Figure 6.

Figure 6. Jitter values for the classical and the modified signaling cases for VoWLAN.

Network QoS parameter values have been grouped into intervals based on the perceived quality by end-users. According to [15], jitter values between 0-20 milliseconds show a good performance and yield a “good” perceived audio quality. Values between 20-50 milliseconds yield an “acceptable” audio quality and values above 50 milliseconds indicate a poor performance and poor perceived voice quality.

Figure 6 depicts the jitter mean value for the different session update mechanisms. The mean value is around 23 milliseconds in the classical case and 11.5 in the modified case. As a result, using

the classification of jitter into intervals based on the perceived audio quality, the classical signaling case falls into the “acceptable” range whereas the modified scheme which bypasses negotiation and updates the data stream for VoWLAN on each handover directly falls within the “good” range.

Using the metric: $m_1 = \text{delay} + k$; with $k = 5$, we obtain Figure 7 shown below. The two signaling schemes (negotiation and injection as in Figure 3) used with the hybrid SIP-MIP stack show a lot of similarity as Figure 5 and Figure 6 show. This similarity is then well reflected by the parallel curves using Metric 1 in Figure 7.

Figure 7. Metric 1 is linear in terms of delay and jitter.

Analyzing session setup and update delay cycle times in the two different cases of negotiation and injection (Figure 5), we observe from the available measured data set that:

Equation 2. Negotiation-injection linear metric relationship.

$$\text{delay}_{\text{negotiation}} = k_1 \cdot \text{delay}_{\text{injection}} + \epsilon_1$$

$$\text{jitter}_{\text{negotiation}} = k_2 \cdot \text{jitter}_{\text{injection}} + \epsilon_2$$

Where k_i is a linear constant and where all k values are close to each other. ϵ is a small correction variable that indicates the delta spread or variance of the data between the first and second schemes as shown in Figure 7 and Figure 8. The two above equations are also valid for the mean values of the session delays and jitters.

Equation 3: Coefficient calculation for used metrics from measured data.

$$k_1 = 350.93 / 300.90 = 1.166$$

$$k_2 = 22.63 / 11.49 = 1.97$$

Using the metric $m_2 = \text{delay} + \text{jitter}^2$ (Figure 8) shows a diversion between the curves, and does not say much about two schemes except that the injection mechanism outperforms the negotiation mechanism with hybrid mobility. The behavioral relationship is nevertheless better shown (to be a behavioral pattern and distribution similarity) is better seen using metric m_1 .

Figure 8. Metric 2 adds linear delay and quadratic jitter.

We hence see that for each particular pattern or scenario use-case, a specific composite metric is suitable. In our case m_1 turns out to be more suitable than m_2 due to the resemblance and correlation between the two data patterns for session setup delay.

7. CONCLUSION AND FUTURE WORK

We have shown that it is possible to use a different signaling mechanism in open access WLAN-based networks for a better performance. When replacing classical P2P negotiation via SIP messages with resource control at the access point, all the mobile stations needs to do is to re-compute their session resources after a handover and then launch an updated stream. This technique yields lower delay values and more strongly bounded jitter. The key issue here is fewer bytes and messages for session setup or update in the parameter injection scheme compared to negotiation. Jitter reduction from around 23 milliseconds on the average down to around 11.5 milliseconds yields a much better perceived audio quality for VoWLAN. Optimizing session setup delay cycle times and the variation of those cycle times (what we call session jitter) contributes to stabilizing voice-over-WLAN and makes it an attractive application and service for virtual operators and service providers. Furthermore, the metrics used in our previous publications allow us to graphically convey the system behavior and pattern in session delay distribution and bounds. The demonstration was performed for a pedestrian speed experiment, but the work can be extended to low and intermediate speed vehicular scenarios.

Future work includes activities such as using more metrics e.g. by Andersson et al, Brannstrom et al, and Ahlund et al [4] that variously combine jitter and delay in order to formally represent each signaling scheme and evaluate it. In addition, scalability tests for VoWLAN can be done to see how different signaling schemes perform with increasing load.

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