Implementasi dan analisa perbandingan qos pada unified communications jaringan voip dengan menggunakan secure sip metode SRTP dan SSL = Implementation and comparision analysis of qos in unified communication for voip network using secure sip with SRTP and SSL method

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Abstrak

Kebutuhan informasi dan semakin majunya perkembangan teknologi mendorong lahirnya Teknologi Informasi dan Komunikasi yang dapat memberi efektifitas dan efisiensi kerja manusia. Akses internet yang juga mudah dijangkau, kapan saja, dan dimana saja memicu lahirnya integrasi semua teknologi dan aplikasinya. Telepon, e-mail, instant messaging, bahkan video conference, mulai didorong agar dapat saling terintegrasi dan sinkron sehingga berbagai jenis aplikasi dan coba.

Didapatkan bahwa QoS berupa delay, jitter, packet loss, dan throughput tidak mengalami perubahan yang signifikan dan masih memenuhi standar ITU-T. Nilai delay yang didapatkan sebelum pengamanan SIP sebesar 33,974 ms, sedangkan setelah implementasi secure-SIP naik menjadi 39,964 ms. Untuk nilai jitter dengan dan tanpa menggunakan secure-SIP sekitar 0,6 ms. Tidak ada paket yang hilang, dalam hal ini nilai packet loss sebesar 0%. Sedangkan nilai throughput sekitar 50 paket/detik.

......Information needs and more advanced technological developments led to the establishment of information and communication technology which allowed human to work more effectively and efficiently. Internet access which available in anytime and anywhere, also triggered the integration of all means of technology and its applications. Telephone, e-mail, instant messaging, even video conference, are all begin to be integrated and synchronized so that they all could be accessed all at once, in a single device or application. As a new technological advancement, Unified Communications is expected to not only serve the business world, but

also education, to increase the effectiveness and efficiency in conducting their daily activities. But this integration and ease comes with some unfavorable aspect. Security aspect becomes a very important part to support quality of services. This final paper was implemented the IP telephony for VoIP application with security scenario on different SIP protocol: RTP (non-secure), SRTP, and SSL. Then, the the performance and security on IP Video Telephony after and prior to the implementation of secure-SIP method was analyzed.

The results show that QoS in forms of delay, jitter, packet loss and throughput, did not reveal significant changes and is still within the standard of ITU-T. Delay measurement prior to the SIP securing is 33,974 ms, whereas after the implementation of secure SIP, it increase to 39,964 ms. As for the jitter measurement, with or without secure-SIP, is approximately 0,6 ms. No packets are lost, so the value of packet loss is 0%. Throughput is about 50 packet/second