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Dynamic Resource Allocation in 3GPP SIP Overlay Networks

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Abstract

The Session Initiation Protocol (SIP) is the IETF alternative for session related signalling. SIP will be used in the IP Multimedia Subsystem of 3rd Generation Partnership Project (3GPP) UMTS networks. To be able to guarantee Quality of Service (QoS) to customers in carrier grade networks it is required to ensure QoS for signalling. Resources provided by the multiservices transport network have to be allocated to the signalling traffic. This paper introduces a methodology that enables Dynamic Resource Allocation (DRA) on the basis of current message numbers. In particular, it can be used to dimension virtual SIP links (VSLs). The paper defines the DRA scheme, outlines its operation and gives simulation results that underline the advantages of dynamic resource allocation.

1. Introduction

The Session Initiation Protocol (SIP) [1] enables user location, session initiation and session management. The 3rd Generation Partnership Project (3GPP) [2] is a global initiative to develop standards and specifications for next generation UMTS networks. 3GPP has decided to use SIP as a signalling protocol for the IP Multimedia Subsystem (IMS) [3]. Guaranteed Quality of Service (QoS) on a large scale in carrier-grade networks requires QoS provisioning for signalling [4]. This is particularly important since the signalling traffic will share network resources with other network services in multiservice IP networks. Transport bearers will use QoS methodologies to protect traffic requirements of different services. A combination of Integrated Services (IntServ) [5] technologies at the network edges and Differentiated Services (DiffServ) [6] technologies in the core network appear to provide satisfactory resources [7]. For the remainder of this paper it is assumed that generic methodologies exist and that they provide QoS resources for signalling. It is also assumed that Service Level Agreements (SLAs) can be dynamically negotiated with the network.

In IP networks, SIP nodes are logically fully meshed. To be able to ensure the QoS on connections between SIP nodes the concept of Virtual SIP Links (VSLs) is used. VSLs are introduced in [8]. They are defined by their traffic specifications (TSpec) which have to be accepted by the transport QoS network. VSLs and SIP nodes form the virtual SIP overlay network (VSON). The aim of the methodology introduced in this paper is to define and negotiate VSL parameters dynamically. The discussions in the remainder of this paper assume that VSLs are used. VSLs are defined by mean rate, peak rate, burst size and minimum policed unit.

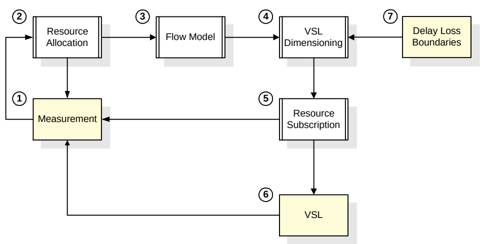

Figure 1: Dynamic Resource Allocation

Figure 1 depicts a block diagram of the DRA scheme. A function that implements this scheme has to be located in all SIP proxy servers. The plain grey boxes symbolise parameter inputs or outputs. The other boxes show the functions of this scheme. The first sub-function is the *measurement function* (Box 1). To be able to react to changes in the network, the load and other network conditions have to be known. VSON management parameters will be discussed in Section 2 in more detail. The *resource allocation function* (Box 2) judges the current measurement results and decides if further action is required. If no action is required, it returns to the measurement function. The resource allocation is introduced in Section 3. The *flow model* (Box 3) estimates the mean flow size. Section 4 discusses the flow model. Using these parameters and the *Delay and Loss Boundaries* (Box 7) the *VSL dimensioning function* (Box 4) calculates traffic specifications for the new VSL requirements. The *resource subscription function* (Box 5) negotiates the traffic specification with the transport QoS network. These two functions are discussed in Section 5. Once these parameters are accepted by the QoS enabled network the VSL (Box 6) is defined otherwise the function returns to the measurement. Remarks in Section 6 and simulation results in Section 7 conclude this paper.

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2. VSON Management Parameters

Various models and methodologies that are discussed in this paper require a number of local and global network parameters. Some parameters are node specific others describe the properties of whole network trees. Two different parameter types can be identified. The first type are fixed parameters that are defined by operator requirements, e.g. the number of messages that are allowed to be resent etc. These parameters are known domain wide. The second kind of parameters are local node and adjacent link parameters. For example, the probability that a message will be lost on its path through the network depends on adjacent links and on the network tree that is routed at this link.

Operator Requirements Network operators have defined sets of QoS requirements for their domain. These parameters have to be known by every local SIP proxy server. An example of such a parameter is the maximum percentage of sessions that are allowed to be rejected at the signalling network edge or the maximum delay that is allowed due to message queueing. These parameters are required to calculate the VSL specifications. They are depicted in Box (7) in Figure 1. Parameter-updates can be "piggybacked" on the update messages of the used SIP layer routing protocol or an arbitrary network management protocol is used to propagate the information.

Node Parameters Node parameters are specific for nodes and their adjacent links. Some parameters replicate the static node properties, others reflect dynamic changes. For example, parameters of these types are the amount a by which messages will increase/decrease when they pass these nodes (SIP VIA etc.). Other parameters include the loss on adjacent links and propagation delays. The major dynamic node parameter is the *message arrival rate* λ which is a measure of the network usage. Every node has to know the message arrival on all ingress and egress links. This can be measured by counting the number of messages per time. The flow model requires this information but it is also required by the message routing which is not discussed here.

3. Dynamic Resource Allocation

This section introduces the method how it is decided if resources have to be increased or reduced. SIP nodes monitor the traffic on all emanating VSLs. This is done in terms of current number of messages per time unit. It can be implemented with simple counters and timers. If this number crosses certain boundaries the resource subscription has to be changed. The system has to use at least two sets of thresholds which can be implemented as step-functions. One is used when the number of users decreases; the other function is used when the number of users increases. It is intended that the methodology be sensitive to the changes in the number of users using a connection but not to the normal statistical fluctuation. These fluctuations can either be estimated by observations of network traffic or it can be approximated by statistical analysis. The output of this sub-function is the number of messages for which the current link should be dimensioned.

Distance Estimation Between Two Thresholds If it is assumed that the measured arrivals follow a normal distribution for short intervals and long-term statistical effects are ignored, the required distance between two thresholds can be estimated using the standard deviation. The well known unbiased estimate for the variance of σ^2 is given by Equation (1).

$$S^2 = \frac{\sum_n (x - \overline{m})^2}{n-1} \tag{1}$$

where S^2 is the estimate of σ^2, n is the number of measurements, $\overline{m}$ is the average message count and x_i is the current measurement. The standard deviation can be used to estimate the likelihood that a measured value lay within the thresholds. For a thresholds of $\pm S$ the chance that a measured value is between the boundaries is 68 %, for $\pm 2S$ it is 95 % and for $\pm 3S$ it is 99.7 %. In the discussed case this means that if the thresholds are chosen to be 3 standard deviations of the mean value, there is a chance of 0.3 % that DRA detects a change in the mean value even the change is due to statistical fluctuations.

Threshold Functions The resource allocation can use hysteresis functions as they are known from other control technique problems. Figure 2 depicts a example of such a function pair. It depicts the *number of messages in percent of the maximum allowed number* versus the *percentage of resources that have to be requested from the network*. Function f_{up} (solid line) is used if the number of users increases and function f_{down} (dotted line) is used if the number of users decreases. The minimum number of users in this example is 10%. If the number of user rises above 10% additional resources are reserved. These are freed if the number of users drops below the applicable threshold. For practical applications f_{up} might be chosen to step earlier to the next level to allow for additional flexibility. The minimum distance between the functions has to be larger than the statistical variation as outlined above.

4. Calculating Flows

The mean flow size in VSONs is a major network usage parameter. The *flow model* depicted in box (3) of Figure

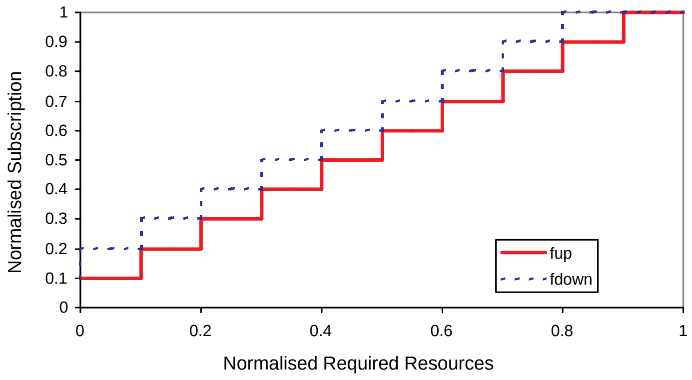

Figure 2: Dynamic Resource Allocation Functions

1 uses methodologies that are based on the model that was introduced in [9]. On the basis of local usage parameters which are available in these nodes, it is possible to estimate the mean flow size. The mean flow size is then used in the VSL dimensioning in Section 5.

Calculation of Message Numbers Figure 3 depicts an example connection between two nodes a and b. The mean size of messages leaving node a is denoted by $\overline{m}_a$, the number of original messages sent on the connection under the assumption of zero loss is denoted by $n_{a,b}$. The connection has a message loss probability of $P_E(a,b)$. The number of messages that were dropped and resent on connection a, b and beyond are denoted by $d_a^{a,b}$ and the number of messages that were dropped beyond b and traversed a, b is denoted by $d_b^{a,b}$. Depending on the local knowledge

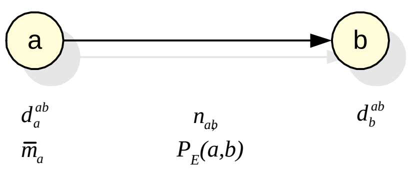

Figure 3: Example Connection

the flow on link a, b can either be calculated for node a by Equation (2) or for node b by Equation (3)

$$FL_{a,b} = \overline{m_a} \cdot \left(n_{a,b} + d_a^{a,b}\right) \tag{2}$$

In the case of node a, $d_a^{a,b}$ is known and in the case of node b, $d_b^{a,b}$ is known.

$$FL_{a,b} = \overline{m_a} \cdot n_{a,b} \cdot \left(1 + \frac{P_E(a,b) + d_b^{a,b}}{1 - P_E(a,b)}\right) \tag{3}$$

The next section shows how message size $\overline{m}_a$ can be calculated.

Average Message Size Estimation SIP messages vary considerably in size. They consist of a message header and a message body. The message body is usually used for the session description. The SDP protocol is used for this purpose. SIP uses messages with and without the message body. State full servers that are traversed by a request are recorded in the "Via" header field and in some cases in the "route-record" field. This causes the SIP messages to grow while they pass through the network. Since the observations are local the message size change is not important in this context. The current message size is the value of interest. A local node can calculate a recursive mean of all processed messages online. Equation (4) shows the calculation.

$$\overline{m}(t+1) = \overline{m}(t) + \frac{M(t+1) - \overline{m}(t)}{n} \tag{4}$$

The averaging count is denoted by n, i.e. the number of samples that are considered, the current message size is M and $\overline{m}$ is average. This value is simpler to calculate than the moving average. It does not require memory for the past n values. More recent measurements have a larger impact than older measurements.

5. VSL Dimensioning

The major parameters are the resources that are required by signalling. The resources are proportional to the number of messages on links. VSLs and their dimensioning were proposed in [8]. These VSLs are used to enable QoS provisioning. They are specified by mean flow size, delay and loss boundaries. The usage parameters are calculated by the *VSL dimensioning* sub-function and its specifications are formulated as TSpecs. The resource subscription function negotiates the TSpec on the basis of dynamic SLAs with the underlying transport network. Once these specifications are accepted, the new VSL is defined. If a request for additional resources is rejected, the current setup remains unchanged. Dimensioning requires two steps. First the required buffer size and normalised arrival rate are chosen; second the necessary peak rate is calculated. The message loss probability $P_E(\lambda_0, l)$ of a connection is determined by the arrival rate and the buffer size l [8]. The Message Loss Probability (MLP) of the leaky bucket that defines the VSL can be calculated by the combination of the MLP models for utilisations of one and less than one. The calculation is depicted in Equation (5).

$$P(\lambda_0, l) = min\left\{ l - (l + \lambda_0) \cdot e^{-\frac{\lambda_0}{l}}, \left(\frac{\lambda_0 \cdot e^{\lambda_0}}{e}\right)^{l+1} \right\} . \tag{5}$$

The buffer size in messages is denoted by l and the normalised arrival rate is denoted by λ_0. The calculation of the normalised arrival rate is shown in Equation (6).

$$\lambda_0 = \frac{p}{r} \tag{6}$$

where p denotes the peak rate and r denotes the mean rate. Since λ_0 is dimensionless, the rates can be either session per time unit or flows size per time unit. For example a peak rate of $p = 200$ messages and a mean rate of 100 messages yields $\lambda_0 = 0.5$. If a buffer size of $b = 5$ is chosen the message loss probability P_E equals 0.0778%. Once all parameters are calculated, the resources are requested from the underlying transport network.

6. Remarks

Routing Both the *dynamic resource allocation* as well as the *dynamic routing* are dynamic processes. To keep the network operation stable it is important that both schemes operate on different scales. The utilisation threshold of the routing has to be below the utilisation threshold for the DRA. This means that frequent fluctuations of the user numbers will be handled by the message routing rather than by the DRA. Long term traffic developments are handled by the resource allocation. The routing methodology will utilise a VSL close to the maximum virtual utilisation of 100%.

Note that a VSL is defined in a way that an utilisation of one is possible and the performance of the VSL is within the specification. The virtual utilisation is denoted in a fraction of the VSL size and not in relation to the resources. If the threshold of the DRA is set, e.g. to 60% virtual utilisation, the VSL size is incremented once the threshold is crossed. If this path is optimal the routing protocol will subsequently route more traffic on this VSL. The VSL will be further incremented if the threshold is crossed again.

Efficiency/Possible Savings To ensure QoS it is necessary to dimension the SIP resources in a way that the number of dropped messages is below a certain threshold. If the resource allocation is fixed, they have to be dimensioned for the maximum of admissible traffic f_{max}. If DRA is used only a fraction of this resources is required. The resource savings Δf can be calculated by Equation (7).

$$\Delta f = f_{max} - \int_t f(t)dt \tag{7}$$

where $f(t)$ are the resources that are required at time t. For discrete measurements the savings can be calculated by Equation (8).

$$\Delta f = f_{max} - \sum_t F(t) \tag{8}$$

where F describes the step function of the resource allocation.

7. Simulation Results

The DRA scheme was simulated with a discrete event simulator for the SIP protocol to show the dynamics of the operating scheme. The simulator uses the Mersenne Twister [10] with a period of $2^{19937-1}$ as a random number generator. For simplicity a session consisted of one request and one response. The initial session arrival rate was set to be 100 sessions per seconds and followed a Poison arrival process. The message size was evenly distributed between 300 bytes and 700 bytes. The arrival rate was changed every 10 seconds by 10 arrivals per second. The decision on increase or decrease of the arrival rate depended on a random number. For the first 250 seconds this number was set to be 0.5, for the next 300 seconds the ratio of increase to decrease was set to be 0.6, than it was kept at 0.5 for 150 seconds and for the remaining time the ratio was changed to be 0.4 until the original level was reached. The nodes were equipped with buffers that had a size of 5 messages. The worst case λ_0 was chosen to be 0.6. This yields for 1000 original messages 4.23 lost message in one measurement interval (Equation (5)).

Statistical analysis of the measurements for the first 250 measurements yields a mean of $\overline{m} = 525081$ bytes and a standard deviation of $\sigma = 25667$ (Equation (1)). Therefore 99.7% of all session arrivals should lie in the interval $[448080, 602082]$ ($\pm 4.9\%$). The simulation used intervals of $\pm 10\%$ for the initial resource size of 50000 bytes.

Figure 4 depicts the graphical output of this simulation run. The graph depicts the message count per second versus the time in seconds. The dotted line shows the resources that are reserved, the upper dashed line indicates the increase threshold and the lower line indicates the decrease threshold. It can be seen that as soon as the message count increases also the resource subscription increases. If the user numbers decrease also the subscription decreases. Figure 5 depicts message drops (left axes) and

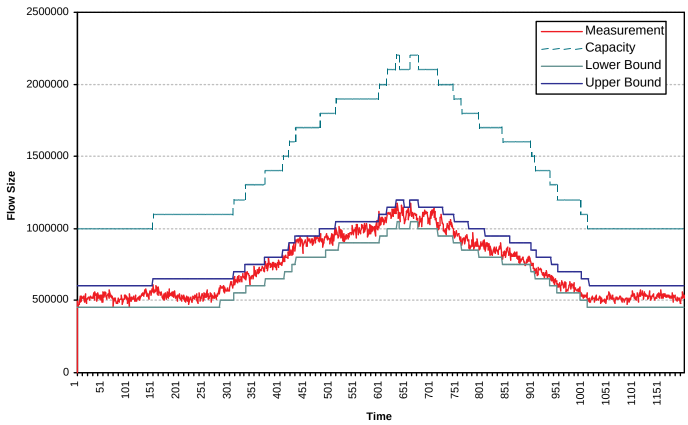

Figure 4: Example Simulation - Resources

Lambda (right axes) versus the simulation time for the same time interval than the figure above. The bold lines show the moving average over 20 measurements for both curves. Statistical analysis of the message drop count yields a 95% confidence interval of $[3.08, 3.41]$. Comparing the graphs for lambda and the message drop count indi-

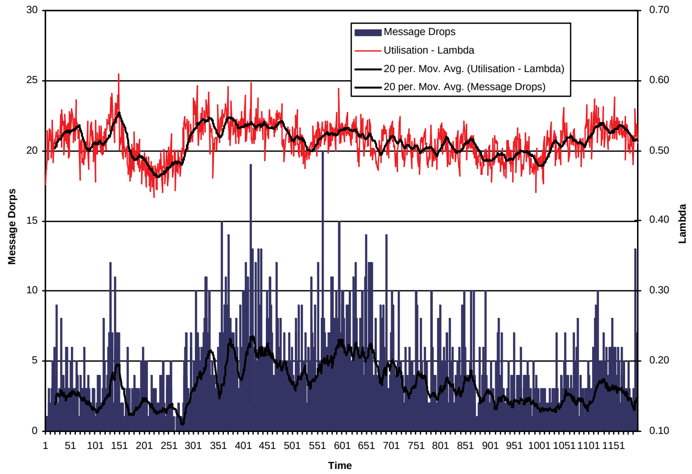

Figure 5: Example Simulation - Message Drop Count (left) and Lambda (right)

cates the expected correlation between both variables. If the traffic increases the utilisation and the drop count are slightly higher than when the traffic decreases. This is due to the fact that the traffic measurements are closer to the increase boundary during increase periods and therefore are located in the top part of the detection interval. The fluctuation of the utilisation depends on the size of this interval.

To fulfil the requirements in the static case, resources of 2200000 bytes had to be available for the duration of the simulation. Compared with the resources that were actually required by using the DRA scheme this yields savings of 35%. This is in particular of considerable order since signalling resources have to be of high quality and are therefore expensive.

8. Conclusions

The aim of dynamic resource allocation is twofold. Firstly it is a methodology to enable the QoS provisioning for the virtual SIP signalling network. Secondly it achieves the dimensioning automatically on the fly. It uses capabilities that mixed services IP transport networks provide. Dynamic resource allocation requires several supporting technologies. The underlying transport network has to support dynamic SLAs i.e. the ability to negotiate the resources with the network. It also requires that VSLs be used on the SIP layer.

In such a context, the dynamic resource allocation methodology allows the automated configuration of resources and ensures QoS for signalling. This enables the guarantee of QoS to customers in next generation carrier grade UMTS networks. DRA can yield considerable resource savings, which is in particular useful if the resources are billed on SLA bases. The DRA scheme can also be used in generic overlay networks.

9. Acknowledgements

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