AI-BASED QOS PROFILING FOR NGN USER TERMINALS

Frank Weber, Woldemar Fuhrmann, Ulrich Trick1, Udo Bleimann, and Bogdan Ghita

Research Group for Telecommunication Networks, University of Applied Sciences Frankfurt/M., Frankfurt/M., Germany
{weber,trick}@e-technik.org
University of Applied Sciences Darmstadt, Darmstadt, Germany
{w.fuhrmann,u.bleimann}@fbi.h-da.de
Centre for Information Security & Network Research, University of Plymouth, Plymouth, UK
bghita@jack.see.plymouth.ac.uk

ABSTRACT

This paper addresses the identification of NGN (Next Generation Networks) user terminals experiencing similar QoS (Quality of Service) conditions. This information is useful regarding the identification of adequate QoS monitoring points in order to provide a resource-saving QoS monitoring approach. An ART 2 neural network has been evaluated for the comparison and classification of sequences of consecutive jitter (delay variation) values experienced by packets of simultaneous multimedia over IP data streams. A test bed has been set up, and test results are presented within this paper.

KEYWORDS

NGN, QoS, Jitter, Artificial Neural Network, ART 2

1. INTRODUCTION

Quality of Service (QoS) is currently considered to be one of the key features of the NGN concept [1] [2]. Unfortunately, as stated in [3] and [4], the active control of network resources within the IP transport network, as performed according to the NGN QoS architecture [5], results in a considerable amount of resource management traffic which is not scalable with the number of NGN subscribers and their individual communication behaviour. In order to address this issue, an integrated framework for comprehensive QoS control in SIP-based NGN has been introduced in [6]. A characteristic of this framework is the continuous collection of information on the IP transport network performance as experienced by any NGN subscriber terminal.

As a possible solution to the above mentioned scalability problem, previous research [7] proposed the dynamic selection of certain user terminals which represent reference points for QoS conditions similarly experienced by a number of user terminals. Hence, because only these selected user terminals have to be queried for the transmission of QoS-relevant information, this approach results in a comprehensive but resource-saving QoS monitoring concept.

This paper proposes the application of an ANN of the type ART 2 (Adaptive Resonance Theory) according to [8] for the assignment of NGN user terminals to virtual groups by the similiarity of effective QoS conditions. Initial tests have been accomplished and are introduced within this paper.

2. NEXT GENERATION NETWORKS (NGN) AND QUALITY OF SERVICE (QOS)

In 2004, the ITU-T (International Telecommunication Union – Telecommunication Standardization Sector) released its definition of NGN in [1]. According to [1], [2], and [9] the term NGN stands for a telecommunication network concept that can be characterised by a number of key features including, amongst others, “Packet-based data transport” and “Quality of Service support”. Although the term “Packet-based data transport” does not refer to any particular technology or protocol, IP (Internet Protocol) is the most likely network protocol choice for an NGN environment according to [9], [10]. The use of SIP (Session Initiation Protocol) for NGN service provisioning and signalling is widely accepted, and also suggested in [11].

2.1. The NGN architecture

Figure 1 shows the principle structure of an NGN.

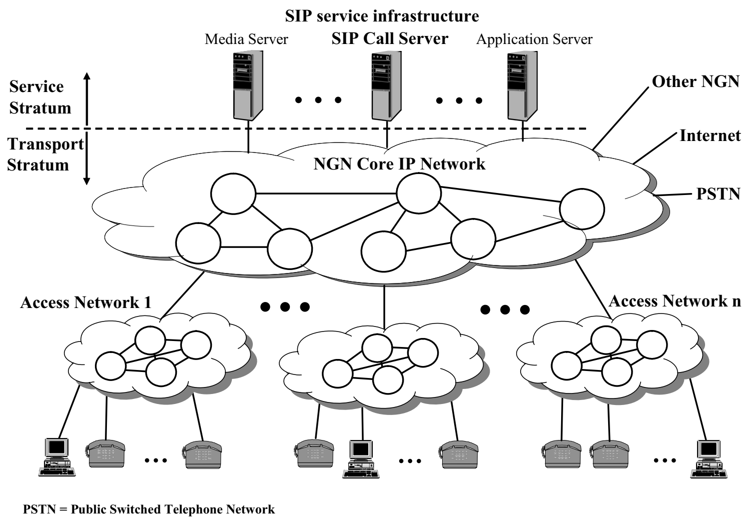

Figure 1. Principle structure of a SIP-based NGN

In the NGN architecture considered within this paper, SIP was chosen as the service signalling protocol. An NGN can be logically divided into a service stratum and a transport stratum. Utilising SIP for service control and signalling, an NGN’s service stratum provides services and applications to the NGN subscribers, such as (in the simplest case) the initiation of a VoIP (Voice over IP) session. In terms of SIP, the service stratum comprises a SIP server infrastructure. The NGN’s transport stratum provides IP connectivity and IP transport to the user terminals (such as VoIP phones). It consists of any arbitrary IP transport infrastructure, including both access and core networks. The user terminals are connected to interfaces (such as a DSL interface (Digital Subscriber Line)) provided by several access networks. By transmitting IP packets over this interface, the user equipment uses SIP to communicate with the NGN service stratum (e.g., to setup media sessions to other users’ end systems). Once a media session is established, media data are exchanged peer-to-peer between the involved NGN user terminals. Hence, after session initiation, the service stratum is not involved in the media data exchange.

2.2. QoS for real-time telecommunication services

For services provided within telecommunication networks, the term QoS has been defined as the "collective effect of service performance which determine the degree of satisfaction of a user of the service" [12]. According to [13], for packet-based media data transport, the quality of a real-time based telecommunication service as experienced by a service user directly depends on the network performance of the respective transport network. In [14] the network performance of an IP transport network is characterised by the packet loss ratio, the transfer delay, and the transfer delay variation (jitter). According to [15]-[18], these network performance parameters substantially influence the QoS of a real-time communication service as experienced by its users.

2.3. Integrated framework for comprehensive QoS control in SIP-based NGN

The authors proposed in a previous study [6] a framework for QoS control, aiming to address the scalability issues related to QoS provision in SIP-based NGN, as described in [3] and [4]. Within this framework approach [6], all action required for the control of the QoS affecting media sessions is performed within the NGN service stratum (i.e., cross-strata communication is avoided). Therefore, the framework has to be provided with an integrated mechanism for the collection of information on the QoS affecting any ongoing and future media session.

This information consists of delay, jitter, and packet loss values affecting the packets of a respective media data stream. The information is best collected close to the receiving user terminal of the respective data stream to consider the sum of effects appearing on the entire network path between sender and receiver. In order to minimise the resulting QoS monitoring traffic, only selected user terminals (representing a QoS reference point) are queried for information on the QoS conditions experienced by ongoing media sessions. This requires the identification of virtual groups of user terminals whose members experience similar QoS conditions, and hence, can be represented by a specific reference point. The assignment of user terminals to their respective virtual group, resulting from the comparison of QoS conditions experienced, is proposed to be performed by the use of an Artificial Neural Network (ANN) to allow for an improved real-time processing behaviour. The principle of assigning media streams to virtual groups (or classes, respectively) by the use of an ANN is introduced in section 4 of this paper. A detailed description of the overall framework functionality of QoS information collection is provided in [7].

Furthermore, the integrated framework introduced in [6] provides mechanisms for the maintenance and recovery of media sessions' QoS. These mechanisms are mainly based on the passive control of transport network utilisation by (amongst others) advanced Call Admission Control and codec downgrades in respect of the objective session significance. The developing of the effective QoS conditions is observed continuously, and the reaction on changing QoS conditions is performed based on a control loop. Further details on the overall framework functionality are described in [6].

3. ART 2 ARTIFICIAL NEURAL NETWORKS

Artificial Neural Networks (ANNs) are used in numerous technical applications in order to perform complex tasks such as pattern recognition (or pattern classification), function approximation, prediction/forecasting, optimisation, content-addressable memorising, cybernetics, as well as clustering/categorisation. The latter application is denoted as unsupervised pattern classification in [19].

ART 2 (Adaptive Resonance Theory) neural networks can be described as unsupervised-learning neural networks with the ability to compare analogue continuous value sequences with the objective to classify the sequences by their similarities [8]. This is performed by self-organisation of stable recognition codes generated from the input value sequences. An input sequence, also referred to as a pattern, is interpreted as an n-dimensional vector by an ART network, where n is the number of values comprised by the respective input pattern.

An ART 2 ANN provides n input units and m output units, the latter of which represent m individual output classes. If an arbitrary number of n-dimensional patterns is presented to an ART 2 network, after a predefined number of learning cycles, the network tries to map each pattern to one of m output classes by accomplishing a multi-step comparison process for each pattern. Patterns showing typical similarities are to be assigned to the same output class.

A number of setup parameters are provided by ART 2 ANN, of which the vigilance parameter ρ is the most effective. Depending on the degree of deviation in value patterns providing questionable similarities, the exactness of the assignment process can be influenced by this parameter. Further details on the theory of ART 2 neural networks can be found in [8].

4. TEST SCENARIO

The following subsections provide the descriptions of the test bed used for the evaluation of an ART 2 neural network for the assignment of VoIP streams to classes by the similarity of their experienced QoS characteristics. From the classification of data streams, the classification of user terminals can be derived within an NGN service stratum.

4.1. Test bed setup and extraction of QoS characteristics

The purpose of the tests performed was to evaluate whether concurrent multimedia over IP packet streams exchanged within an NGN can be assigned to different classes, each representing certain QoS characteristics. The QoS characteristics are assumed to be mainly influenced by the QoS conditions effective within the access networks of both, the sending and the receiving host of a respective multimedia stream. In our experiments, the QoS characteristics of packet streams were found to be best represented by jitter values (see section 2.2) because of their susceptibility to changes within the IP network load utilisation.

For the accomplishment of the tests, an NGN test bed has been set up, in which concurrent VoIP (Voice over IP) data streams (each providing constant packet sending intervals) were established. Figure 2 shows the test network layout, emulating a simplified NGN scenario.

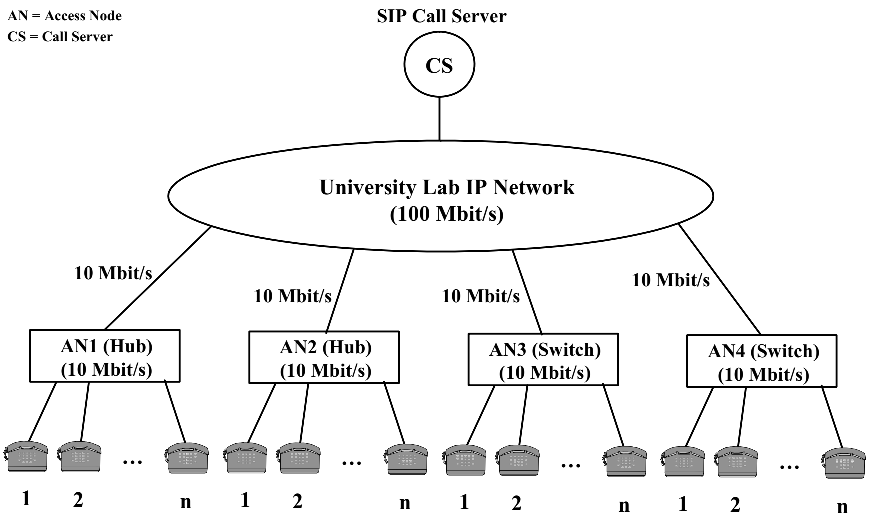

Figure 2. Test network layout

The NGN service stratum is represented by a SIP Proxy/Registrar Server. The transport stratum consists of an IP core network (University Lab IP Network, 100 Mbit/s) and four relocated access nodes (AN) (2 hubs (AN1 and AN2) and 2 switches (AN3 and AN4)), each representing an independent NGN access network (see section 2.1). Each access node was connected to the core network via a symmetric 10 Mbit/s Ethernet link. Different numbers of VoIP (Voice over IP) terminals (SIP hardware IP phones) were connected to the access nodes via symmetric 10 Mbit/s Ethernet links. For the media transmission, for all VoIP streams, RTP (Real-time Transport Protocol) was used. The voice data were encoded with the G.711 μ-law voice codec. The media sequence interval was set to a sequence length of 20 ms per IP packet. In addition to the VoIP streams, each access node was stressed with additional random TCP traffic exchanged between the respective access node and the core network.

Within this test bed, different VoIP communication scenarios (VoIP streams sent between defined terminals connected to defined access nodes) were arranged. Different simultaneously existing communication situations were considered in all communication scenarios. Each communication situation comprised different numbers of concurrent VoIP streams sent among the considered ANs. Table 1 shows the assignment of simultaneously existing communication situations arranged per communication scenario, and, within the correlation fields, the numbers of VoIP streams considered per communication situation. Within Table 1, for communication situations comprising streams exchanged between two different access nodes, the numbers of streams for each communication direction is provided, separated by a slash.

Table 1. Communication situations concurrently arranged per communication scenario

	Communication situations considered per communication scenario						
Communi-cation scenario	Internal streams AN1	Streams AN1↔AN2	Streams AN1↔AN3	Streams AN2↔AN3	Streams AN2↔AN4	Internal streams AN3	Internal streams AN4
I			5/5		7/7		
II	6			5/5			8
III	3	4/4		3/3		3	8

To obtain the QoS characteristics of the VoIP data streams, during the tests, all streams were simultaneously captured IP packet-wise at their respective receiving user terminals by the use of pcap (packet capture) trace functions integrated in the respective VoIP phones. The traces were performed to allow for later analysis. Note that, by the pcap trace function, each recorded packet is provided with a timestamp, indicating the time of arrival of the respective packet at the receiving host. See [20] for further details on the pcap trace function.

In a further step, the capture files recorded by the VoIP phones were analysed subsequently for each communication scenario. Wireshark software [21] was used for this purpose. From all captures, the packet-by-packet jitter (delay variance) characteristics of the respective VoIP stream were extracted by calculating the variation of the inter-arrival time for each pair of consecutive IP packets. Hence, for each IP packet of all concurrent VoIP streams, an individual jitter value was achieved, so that jitter value sequences could be composed. Subsequently, to achieve comparability, all jitter sequences obtained from VoIP streams associated with a respective communication scenario were synchronised in time.

In order to generate value sequences that could be processed by an Artificial Neural Network in real-time, the sequences were cut to a length representing one second of the associated VoIP stream (jitter values obtained from 50 subsequent IP packets with a payload sequence of 20 ms each). In a further step, all jitter sequences were smoothed by a running mean algorithm, taking into account five consecutive jitter values.

From the running mean algorithm, for each VoIP stream, one smoothed jitter sequence was obtained, each consisting of 46 discrete analog values. Within the following, a sequence of consecutive jitter values (representing the smoothed jitter sequence of one VoIP stream) is called a pattern.

4.2. ART 2 neural network implementation and application

For the comparison and classification of the jitter sequence patterns (see section 4.1) associated with a respective communication scenario, an ART 2 neural network (see section 3) was implemented by the use of JavaNNS (Java Neural Network Simulator; [22]). The neural network was provided with 46 input units, so that every jitter pattern could be presented to and processed by the neural network at once in full length.

To compensate potential inaccuracies within the classification process, the neural network was provided with ten output units, each of which could represent a specific jitter characteristic. The expected numbers of output classes to be assigned per communication scenario can be read from Table 1 (see section 4.1). Within the tests performed, it was assumed that, provided 100 percent classification accuracy, jitter characteristics resulting from the same communication situation could be mapped to one class (for VoIP streams sent among user terminals connected to the same AN) or two classes (one for each communication direction for communication involving two different ANs), respectively.

For each communication scenario, a multitude of classification runs were performed. Therefore, the jitter patterns obtained from the analysis of the VoIP streams associated with the respective communication scenario were presented to the ART 2 neural network as a set of pattern sequences. The ART 2 vigilance parameter ρ was varied among the runs in order to evaluate the most exact classification result per pattern set. Furthermore, different sequential orders of the jitter patterns within a respective pattern set were tested. The number of learning cycles to be performed by the ART 2 ANN before the actual classification was set to 100 for all runs.

5. RESULTS

The following subsections discuss the results obtained from the jitter pattern classification performed by the ART 2 neural network. As shown within the result tables, per communication scenario, each VoIP stream was mapped to exactly one class by the ANN. For each stream, the tables provide a separate column ("Stream No."), numbered in the order in which the patterns were presented to the ART 2 ANN within the respective pattern set. The classes, each representing a group of VoIP streams whose jitter characteristics show similarities, are presented as lines within the result tables. Hence, in the tables, the "x" symbol shown for each stream indicates to which class a respective stream has been mapped by the ANN.

Note that the mapping between a class number and its referred jitter characteristic can not be defined a priori, but comes as a result of the unsupervised ART 2 ANN learning process, which is followed by the final jitter pattern classification. I.e., if a pattern presented to the ANN within a pattern set does not show sufficient similarities with other patterns from the same pattern set, a new class is instantiated. Also note that the numbering of the classes does not represent a specific jitter mean value or QoS level, respectively.

5.1. Results from communication scenario I

Tables 2.a) and 2.b) show the assignment of VoIP streams to classes as classified by the ART 2 neural network. The vigilance parameter ρ was set to 0.99.

The two communication situations enclosed by scenario I were quite clearly distinguished by the ANN. All VoIP streams between user terminals connected to AN1 and AN3 were assigned to class 1, while only three of 14 VoIP streams between user terminals connected to AN2 and AN4 were not associated with class 3. Hence, regarding the assignment of VoIP streams to communication situations in scenario b, an accuracy of 87.5 percent was achieved. However, within the different communication situations, no distinction was made between the different stream directions.

Increasing ρ to 0.995, streams running from AN2 to AN4 could be clearly distinguished from streams running in opposite direction. However, no interrelation among the streams from AN2 to AN4 could be found.

Table 2.a). Communication scenario I: Assignment of VoIP streams by ART 2 ANN (ρ = 0.99).

Communication situation		Streams between AN1 and AN3									
Stream direction		AN1→ AN3					AN3→ AN1				
Assigned Class	**Stream No.**	1	2	3	4	5	6	7	8	9	10
1		x	x	x	x	x	x	x	x	x	x

Table 2.b). Communication scenario I: Assignment of VoIP streams by ART 2 ANN (ρ = 0.99).

Communication situation		Streams between AN2 and AN4													
Stream direction		AN2→ AN4							AN4→ AN2						
Assigned Class	**Stream No.**	11	12	13	14	15	16	17	18	19	20	21	22	23	24
1		x			x										
2			x												
3				x		x	x	x	x	x	x	x	x	x	x

5.3. Results from communication scenario II

Tables 3.a) and 3.b) show the assignment of VoIP streams to classes as classified by the ART 2 neural network. The vigilance parameter ρ was set to 0.9.

In principle, the communication situations "Internal streams AN1" / "Streams between AN2 and AN3" could be distinguished by the class assignment as performed by the ANN. Most VoIP streams among user terminals connected to AN1 were assigned to class 1, while most streams between user terminals connected to AN2 and AN3 were assigned to class 4. Within the latter communication situation, no definite differentiation was achieved between streams running from AN2 to AN3 and vice versa.

VoIP Streams of the third existing communication situation ("Internal streams AN4") were not clearly distinguished.

Table 3.a). Communication scenario II: Assignment of VoIP streams by ART 2 ANN (ρ = 0.9).

Communication situation		Internal Streams AN1						Streams between AN2 and AN3									
Stream direction								AN2→ AN3					AN3→ AN2				
Assigned Class	**Stream No.**	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1		x	x		x	x							x				
2				x											x		
3							x										
4								x	x	x	x	x		x		x	x

Table 3.b). Communication scenario II: Assignment of VoIP streams by ART 2 ANN (ρ = 0.9).

Communication situation		Internal Streams AN4							
Stream direction									
Assigned Class	**Stream No.**	17	18	19	20	21	22	23	24
1			x			x			
2				x					
3		x					x	x	
4					x				x

5.4. Results from communication scenario III

Tables 4.a) and 4.b) show the assignment of VoIP streams to classes as classified by the ART 2 neural network. The vigilance parameter ρ was set to 0.9.

Table 4.a). Communication scenario III: Assignment of VoIP streams by ART 2 ANN (ρ = 0.9).

Communication situation		Int.Streams AN1			Streams between AN1 and AN2								Streams between AN2 and AN3					
Stream direction					AN1 → AN2				AN2 → AN1				AN2→ AN3			AN3→ AN2		
Assigned Class	**Stream No.**	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17
1		x		x													x	x
2			x											x				
3					x	x	x		x	x	x	x	x		x	x		
4								x										

Table 4.b). Communication scenario III: Assignment of VoIP streams by ART 2 ANN (ρ = 0.9).

Communication situation		Int.Streams AN3			Internal Streams AN4							
Stream direction												
Assigned Class	**Stream No.**	18	19	20	21	22	23	24	25	26	27	28
1						x		x	x	x		
2												
3				x								
4												
5		x	x									
6					x							x
7							x				x	

In principle, the communication situations “Internal streams AN1” and “Streams between AN1 und AN2” could be distinguished by the class assignment as performed by the ANN. Two out of three VoIP streams (streams 1 and 3) among user terminals connected to AN1 were assigned to class 1, while most streams between user terminals connected to AN1 and AN2 were assigned to class 3. Within the latter communication situation, no definite differentiation was achieved between streams running from AN1 to AN2 and vice versa.

From the third communication situation, “Streams between AN2 and AN3”, three out of six streams (streams 12, 14, and 15) were also assigned to class 3, which can be explained to be due to the influence of the QoS characteristics affective in AN2.

From the fourth communication situation, “Internal Streams AN3”, two out of three streams (streams 18 and 19) were assigned to class 5 as its only representatives within this communication scenario.

The fifth communication situation, “Internal Streams AN4”, could not be clearly distinguished from the first communication situation, as streams from both situations were assigned to class 1.

Changing the order of jitter sequence patterns within the pattern set presented to the ART 2 ANN, with ρ set to 0.875, a clear differentiation of “Internal Streams AN4” from the other communication situations was achieved. However, the streams of the other communication situations could not be distinguished from each other.

6. CONCLUSIONS

Within this paper, the application of an ART 2 neural network for the identification of NGN user terminals experiencing similar QoS conditions was discussed. A test setup NGN implementation was created and used for verification, and different communication scenarios, each including various communication situations with a various number of VoIP streams, were arranged. The resulting jitter data were preprocessed and presented to the ART 2 neural network.

The test results show that, in principle, an ART 2 neural network can successfully be applied to assign VoIP streams (and hence, user terminals) to classes by the respective jitter characteristics experienced. It was found that the accuracy of the classification depends on various factors, including individual characteristic features of the respective jitter sequences. Hence, in order to normalise the assignment conditions for all value sequences to be classified, we suggest to use dedicated functions (such as Fourier Transformation or Fast Wavelet Transformation (FWT)) to extract comparable characteristics from the jitter sequences.

Furthermore it was also found that, within the series of tests performed, the accuracy of the classification of sequences by the ART 2 neural network was strongly influenced by the order in which the sequences were presented to the neural network within the unsupervised learning process.

In a further step we plan to adopt the herewith introduced QoS recognition mechanism into an NGN simulation environment, based on network simulation software such as ns2. Therefore, the extraction and preprocessing of the jitter data will be automated, as well as the presentation of the data sets to the ANN. Based on this setup, the framework for comprehensive QoS control, as introduced in [6] (see section 2.3) will be implemented.

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