

Automated analysis of network QoS parameters for Voice over IP applications

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Abstract

This paper presents technology for monitoring and assessment of network QoS parameter for Voice over IP (VoIP) applications in inter-domain environment.

The technology is based on automated analysis and description of structures (patterns) of QoS parameter behaviour impacting the VoIP quality:

- “Quality impairment” patterns, i.e. “outliers” of network delay and packet loss, which predict degradation of VoIP quality, when the network QoS parameter is exceeding some “threshold” value, selected based on the rating factor R of the ITU-T E-model.

- Network delay structures, used for appropriate choice of “playback delay adjustment” algorithms.

These facilities are integrated in the spatio-temporal QoS pattern analyser developed in the framework of INTERMON toolkit architecture for inter-domain QoS analysis [1]. The QoS parameter patterns of Internet connections are detected based on active probing, emulating VoIP traffic for different codecs.

Based on this technology, the network administrator can automatically evaluate the network ability for VoIP communication, and select appropriate parameters for playback algorithms considering network delay patterns. The paper presents state-of-the art of the technology and related research. Using real measurement scenario in inter-domain environment, the discussed facilities for evaluation of network QoS parameters for VoIP application based on patterns are shown.

1. Introduction

The VoIP applications in Internet are important focus of industrial and academic research. Based on the growing importance of VoIP applications in wide area Internet, different measurement approaches and technologies are proposed in known researches to monitor and access network QoS parameters for VoIP applications (see, e.g. [3], [4], [7], [10] and [28]).

To assess the characteristics of inter-domain connections for VoIP traffic, this work propose the use of patterns, i.e. abstractions for structures of QoS parameter behaviour.

The work is part of the INTERMON project (see [1]) developing architecture for analysis of QoS in inter-domain environment integrating inter-domain routing (BGP-4 protocol), traffic and QoS parameter monitoring and modelling tools. The following aspects are considered in the INTERMON approach to monitor and access network QoS for VoIP in inter-domain environment:

- Monitoring of the connection characteristics for VoIP applications based on active QoS measurement of emulated VoIP traffic.
- Detection of “VoIP impairment” delay and packet loss patterns for network connections characterising the impact of the network delay and packet loss on the quality of VoIP (using E-Model, [16]).
- Detection of delay patterns of network connection, which can be used to support “heuristics” for playback delay adjustment parameters or algorithms at the receiver of the VoIP applications.
- Discovering the impact of inter-domain routing and topology on the network QoS impacting the VoIP quality.

INTERMON technology is designed to support actual challenges for automated QoS measurement and analysis for VoIP applications in wide area networking:

- Addressing the needs of ISP providers with new tools that support emerging standards like E-Model by analysing the network impairment patterns (see [41]).
- Mining and understanding of the network delay and packet loss characteristics of network connections, relevant for VoIP communication (quality, and playback delay adjustment) using patterns (see [2]).
- Considering the impact of inter-domain routing and link failures. It is done by possible interaction with other INTERMON tools like InterRoute ([40]).

This paper considers especially the pattern approaches (impairment patterns and playback delay algorithm impacting patterns), which are integrated in the INTERMON pattern analyser to access the characteristics of the inter-domain network connections for VoIP communication. Pattern analysis for VoIP is illustrated based on real world measurement scenarios.

2. Network QoS Parameter Pattern Analysis for VoIP

2.1. Background of Pattern analysis for VoIP

Patterns focus on “interesting” structures in data behaviour, which can be abstracted and used for efficient understanding, knowledge discovery, analysis and decisions. Patterns are used in different areas. Patterns like “Head and Shoulders” and “Symmetric Triangle” are useful technique for describing behaviour of stock prices. In ECG (electrocardiogram), the detection of “Heart Beat” pattern is a known procedure.

To describe time series data structures considering abstractions of structures, different methods can be used, as for instance linear approximation (see, e.g. [45] and [46]). The general facilities of spatio-temporal QoS pattern analysis in INTERMON as well as their interaction with the inter-domain topology and routing discovery is discussed in [2] and [31].

This paper is aimed at patterns that describe structures in QoS parameter time series data behaviour (end-to-end delay and packet loss) in order to access characteristics of network connections for VoIP communication.

We distinguish between basic and composite patterns describing significant QoS parameter structures of network connections for VoIP communication. The composite patterns consist of sequences of basic patterns (see [2]). Examples for basic QoS parameter structures, which can be detected and compared to study VoIP communication, are:

- “Increase/decrease” representing continuous increase/decrease in the parameter values.
- “Plain” describing QoS parameter behaviour (“oscillation”) in a given range.
- “Spike” depicting sudden increase in the QoS parameter value with a short duration.
- “Outlier” characterising abnormal QoS parameter behaviour (delay increase exceeding some threshold values compared with the R-factor, or empty pattern (no delay is measured) (see [47])).

An example of a composite pattern is a sequence of “plain” and “spike” patterns.

We differentiate patterns for frequently looking in a QoS monitoring database considering details only for the discussed structures that are “interesting” for:

- VoIP quality assessment (for instance, study of occurrence and frequency of “outliers”).
- Playback buffer adjustment heuristics (analysis of characteristics, occurrence and frequency of “spikes”, “plain”, “increase”, “decrease” structures).

We combine in this way heuristics (for playback delay adjustment, quality assessment) for VoIP quality with network QoS structures considering:

- Interval of times, called in data mining segments [47], where basic and composite patterns of network QoS patterns occur.
- Frequency of pattern found in a given time interval and per specific period.
- Dependencies of patterns, considering multivariate time series data patterns and their temporal and spatial relationships.

The study of connection delay and packet loss characteristics for VoIP communication using patterns is a more efficient technique, because:

- More abstract description by structures (e.g. “increase” / “decrease”, “plain”, “spike”, “outlier”), which is preferred by the network administrator to understand large volume of time series data, measured over long time period.
- Condensed, qualitative description of row QoS parameters to provide abstract presentation. Thanks to the background knowledge, the network administrator knows the kind of patterns that are important and worth looking for abstraction of events.
- Knowledge discovery using experiences. The administrator is able to identify frequently repeating patterns, corresponding to normal situations, and seldom patterns, indicating some abnormal state. The occurrence of such patterns is then the basis for taking of decision on the VoIP quality support.
- Learning dependencies of patterns in multivariate time series data, i.e. causal relationships. This includes dependencies of QoS patterns (delay, packet loss, delay jitter), BGP-4 protocol patterns, or patterns of network router failures.

2.2. QoS Parameter Pattern Analyser for VoIP Communication

The VoIP pattern analyser is part of the INTERMON architecture including facilities for automated detection and dependency analysis of patterns, useful for assessment of network QoS for VoIP communication.

The VoIP quality assessment technology includes:

- Active measurement of QoS parameters of network connections by emulation of VoIP traffic using CMToolset [21].
- Detection of VoIP quality “impairment” (delay and packet loss) patterns and delay patterns for tuning of playback adjustment algorithms.
- Management of patterns in a database for enhanced application tuning and support [2].

The following figure shows the integration of VoIP QoS pattern analyser in the INTERMON toolkit:

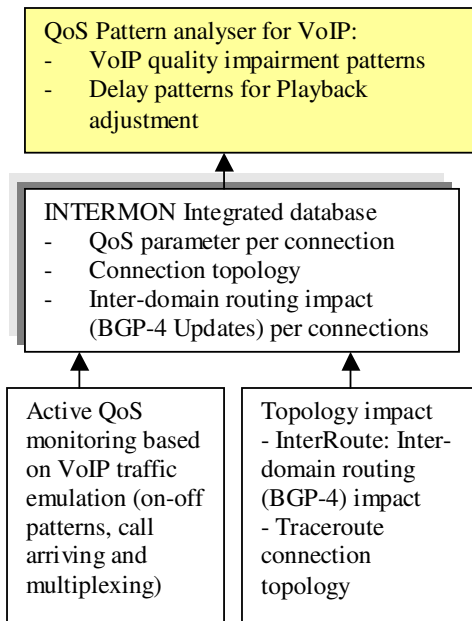


Figure 1: INTERMON approach to study QoS of inter-domain connections for VoIP applications

A particular goal of the pattern technology is to determine ahead of time using INTERMON tools and databases, whether the inter-domain network will provide good voice quality for specified end systems.

The VoIP pattern analysis uses the QoS measurement database of the active measurement toolkit CMToolset (see [21]). The QoS parameter measurement data of CMToolset is structured per generated traffic flow that is related to the end systems (hosts). The pattern analysis includes data mining functions for spatio-temporal comparison of data (per end-to-end connection and traffic) and filtering considering the incomplete measurements (due failures in the measurement system) before the pattern detection. Further integration of the VoIP technology for operational management should consider more efficient data base design and operation to reduce filtering time and consider lack of measurements due to failures.

Generally, the VoIP traffic emulation is based on:

- Call model: long distance calls, business calls.
- Modelling of Call arrival strategies.
- Processing according encoding techniques: G.711, G.723.1, G.723.2, G.728 and G.729.

The CMToolset is used for this purpose; it integrates traffic generation functions based on on-off patterns to model talk-spurts and silence gaps, as it is considered in

known researches (see, e.g. [9]). For instance, talk-spurt and silence periods can be modelled by exponential distribution with an average of 1.5s (see [8]).

The active QoS monitoring allows modelling of arriving strategies for long-distance calls, e.g. arrival times according to Poisson processes and call duration with exponential distributions, which are typical for description of VoIP active measurement scenarios. [8] discusses the emulation of the arrival of short business calls. Multiplexing strategies for long distance calls considering on-off patterns similar to [22] could be supported and their effect on voice traffic aggregation can be studied. Different encoding and conversation types for long distance encoding can be considered as they are discussed in [6].

2.3. User interface for VoIP pattern analysis and scenario

The user interface (GUI) includes functions to obtain patterns based on supporting different data mining tasks, which are useful for VoIP quality analysis. They are:

- Obtaining of QoS parameter time series data (end-to-end delay, packet loss) in spatio-temporal context.
- Mapping of network threshold parameters for delay and packet loss patterns using R-factor.
- Periods for which the patterns will be studied in the time interval, for instance frequency and occurrence per day, per week to consider the calendar effects and specific days.
- Selection of pattern type(s) to be studied: “outlier”, “increase”, “decrease”, “spike”, and “plain”.
- Description of parameters for the different types of basic patterns (such as gradients and threshold values).
- Pattern analysis options.

Spatio-temporal selection of QoS parameter data for the pattern analysis is based on providing of:

- Parameters for specified sender / receiver connection and time interval.
- Flow identification when more flows are emulated in the given time interval for the specified sender/receiver pair.

Similar to [43], we use the E-model and R-factor to select the threshold delay and packet loss / packet burst length, corresponding to some R-factor to study “outliers” of network delay and packet loss.

The patterns discussed in the following sections are obtained with the VoIP pattern analyser using QoS parameter values of an end-to-end connection between Madrid and Salzburg stored in CMToolset database. The QoS parameters are obtained for emulation of VoIP traffic with CMToolset – 20 ms sending interval and 160 Bytes packet length.

We give only examples on the usage of patterns for describing connection properties without to draw

conclusions on periodical pattern behaviour of the connection.

3. Detection of “Quality impairment” patterns considering E-model

3.1. E-model quality impairment

For the study and assessment of network QoS of today Internet for VoIP communication, the ITU-T E-model and R-factor for prediction of VoIP quality based on “impairment” parameters such as packet loss and delay ([12], [13], [14] and [16]) is used. E-model, recommended by the Telecommunication industry associations [5], is:

- A computational model standardised by ITU-T ([12], [13], [14], [15] and [16]) using transmission parameters to predict the subjective quality of voice.
- Computationally simple based on impairment factors and their statistical descriptions.
- A good way to predict the effect of delays and packet losses of the network on the VoIP quality.

[19] found that the E-model had the highest correlation with MOS. The validity of the E-model is checked for different encoding and conversation types in, e.g. [11] and [19]. The E-model combines different impairments into a single rating R-factor on a scale from 0 to 100, which is translated into MOS. R-factor with distortion due to codecs, delay and packet loss is defined as follows:

$$R = R_o - I_d - I_s,$$

where R_o is the highest possible ratio,
 I_d - effect of delay,
 I_s - loss and signal distortion.

Impairments, due to codecs, delay and percent packet loss, are defined in ITU-T Recommendation G.113, Transmission impairments, Appendix I ([13]). Also practical researches and experiences have discussed impairment of end-to-end delay and packet loss, on R-factor (e.g. [3] and [11]).

The E-model is used in this work to select the threshold values for automated detection and analysis of network delay and packet loss “outliers”, i.e. all these QoS parameter data which exceeding the threshold level corresponding to the given R-factor. The network “outlier” for VoIP study is a pattern that is related to the given level of R-factor and corresponding threshold. Based on proposed metrics, quantitative and qualitative analysis of outlier patterns on different network connections is possible in order to select the appropriate connection for VoIP communication.

3.2. Network delay impairment

The network delay impairment on the R-factor is based on the detection and analysis of network delay “outlier” data

related to some R-factor value, i.e. all the network delay data which exceeding a delay threshold d_{thr} corresponding to some R-factor.

Our goal is the analysis of “outliers” based on the network delay D_n .

Considering [44] and [39], the mouth-to-ear delay $D_{m,e}$ of VoIP, consists of fixed and variable parts, with D_n belonging to the variable part, as shown in figure 2.

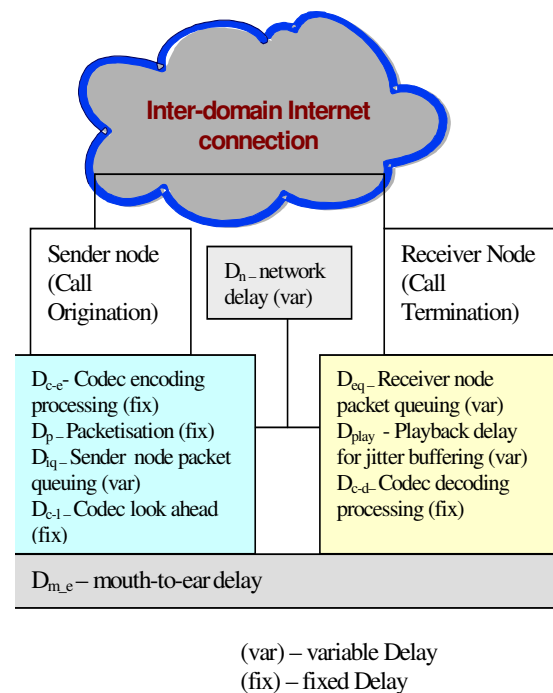


Figure 2: VoIP Mouth-to-ear Delay Analysis

For the goals of this work a simplified delay formula is used with focus on the network delay D_n and playback delay D_{play} :

$$D_{m,e} = D_c + D_p + D_n + D_{play}.$$

- where
- D_c is the processing delay due to the encoding scheme at sender and receiver,
 - D_p is the processing delay due packetisation,
 - D_n is the network end-to-end delay in inter-domain environment,
 - D_{play} is the playback delay due to jitter of the packets caused by network.

[36] includes study on these fixed parts of VoIP delay such as processing due to encoding and packetisation.

[39] discusses the composition of the D_n from the intermediate transmission, queuing and propagation delays caused by the routers of the end-to-end connection path, which will be not studied in this paper.

Threshold delay values d_{thr_m} , corresponding to the R-factor and different standards and codecs, are shown in the following table, derived from [3], [11] and [13]:

MOS	R-factor	G.711 d_{thr_m}	G.729 A d_{thr_m}	G.723.1 d_{thr_m}
4.34-4.5 best	90-100	160	-	-
4.03-4.34 high	80-90	250	150	-
3.60-4.03 medium	70-80	325	250	210
3.10-3.60 low	60-70	425	320	290
2.58-3.10 poor	50-60	-	420	370

Table 1: Threshold delay d_{thr_m} mapping into R-factor

It should be noted, that the threshold values in table 1 are the maximum values for the given ranges of the R-factor. These threshold values d_{thr_m} are related to the mouth-to-ear delay values, i.e. D_{m_e} . The network delay threshold d_{thr} is selected from the mouth-to-ear delay d_{thr_m} , mapping to the R-factor, subtracting the other fixed and variable parts of the delay, as shown in formula above, so that $d_{thr} \leq d_{thr_m}$.

Different types of conversation have specific switching speed and delay sensibilities, thus specific delay thresholds d_{thr_m} . The G.114 ITU standard defines, that D_{m_e} should not exceed 150 ms ([17]). [11] studied that the users perceive a linear degradation in the quality up to 177 ms. [18] concludes that the E-model simply some averaging of the different types of conversation for the mapping of delay values to the R-factor and MOS. [35] discusses different studies considering the impact of the type of conversation on D_{m_e} .

3.3. Analysis of network delay outliers

Let $\{X_t\}$ be the time series data consisting of the network delay parameter values evaluated for (emulated or real) VoIP packets on an end-to-end connection.

A sequence of $x_1, \dots, x_i, \dots, x_n$, belonging to $\{X_t\}$ build a delay “outlier” related to R-factor value with the corresponding threshold d_{thr_m} , when for all x_i , $i = 1..n$, hold that $x_i > d_{thr_m}$.

The analysis of outlier delay patterns for the purpose of the VoIP quality study includes the following metrics:

- Analysis of the occurrence and frequency of all delay “outliers” in a given time interval and for a given period (daily, monthly), i.e. the number of observations of “outlier” patterns $P_{di,k}$, $k = 1..m$, related to the time interval Δt :

$$F_{di} = \sum P_{di,k} / \Delta t.$$

- Relative length of all delay outliers segments (i.e. durations) in the observed interval Δt and per period Δt_p .

$$L_{di} = \sum L_{di,k} / \Delta t, \text{ where } L_{di,k} \text{ is the duration of the outlier pattern } P_{di,k}.$$

- Histogram for analysis of the delay values of the outliers, and their assignment to appropriate buckets (ranges) b_i , $i = 1..k$, of QoS threshold values.

A bucket b_i is defined by $[d_{thr_low,i}, d_{thr_high,i}]$ with $d_{thr_low,i}$ as lowest boundary and $d_{thr_high,i}$ as highest boundary. In the analysis, we consider as network delay threshold 160 ms corresponding to R-factor 90-100 in case of G.711 VoIP packet emulation. This builds the lower boundary of the first bucket. We select the following lower bucket boundaries to build the histogram for the analysis of the outlier values: 160 ms, 400ms, 1s, 1min. Values over 1min are often due to inter-domain routing problems.

The following figure shows an example for the analysis of delay outliers in a time window of 4-days in a real world inter-domain networking scenario between Madrid and Salzburg. The selected network delay outliers are related to the R-factor 90-100. Periods of 10 minutes are considered to aggregate the analysis on the outliers.

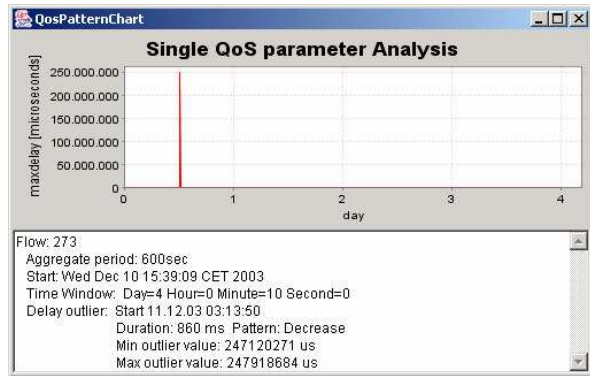


Figure 3: Delay outlier study in the given time interval

The “highest” outlier whom value exceeds the maximum threshold is given (outlier value about 4 minutes, it means that the packets are lost in this period). The relative length of the “highest” outliers (duration, x-axis) is negligible (860 ms); but this causes also the highest packet loss burst (see next section).

3.4 Network packet loss analysis

The network packet loss impairment on the R-factor is based on the detection of:

- Consecutive packet loss, i.e. packet loss bursts P_{li} ,
- Packet loss rate, i.e. all lost packets for a given time period.

The impact of the packet loss on the VoIP quality depends on the end-point's playback algorithm intelligence and encoding scheme.

We relate the detected packet loss rate and bursts to R-factor using tables giving us the packet loss thresholds corresponding to the R-factor. They are based on the:

- Packet loss rate mapping on VoIP quality for different codecs ([44]).
- Burst packet loss to the R-factor and MOS mapping for different encoding schemes as discussed in [3] and [19].

Impact of long packet loss bursts, called outages, on VoIP, which could be caused in inter-domain environment for instance by BGP-4 protocol failures or misconfigurations are studied in [9] observing:

- Their length, i.e. that the most outages are short, but a small portion of them are extremely long and make up the majority time of outages.
- Frequency and symmetry.

The focus of our analysis of burst packet loss is:

- Packet loss rate (considered mouth-to-ear) per time interval and period.
- Occurrence and frequency of packet loss bursts in a given time interval or period.
- Relative length of all packet loss burst segments (i.e. duration) in the observed interval Δt and per period Δt_p .
- Histogram for analysis of the duration of the detected packet loss bursts in the given period.

The following figure shows the statistics derived for packet loss rate considering periods of 10 minute in the time interval of 4-days:

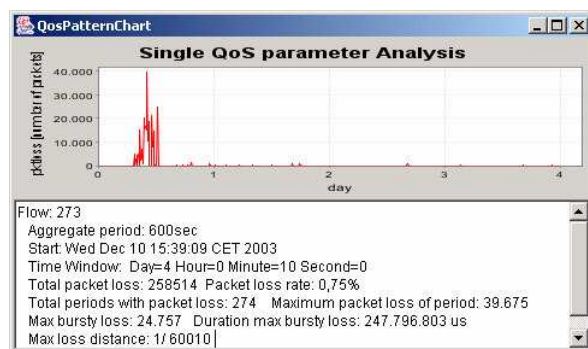


Figure 4: Burst packet loss analysis in the given time interval

The resulting packet loss rate over the 4 days is 0,75% which is mainly due to the greatest packet loss burst of 24.757 Pkts leading to packet loss rate of 41,28 % in the 10 min period beginning at 11.12.03 03:13:51. The duration of the maximum packet loss burst period is about 4 minutes.

The VoIP communication will be degraded in this time very strong (almost no voice is available, probably caused

by a major routing problem), while in the remaining period the packet loss will remain acceptable.

4. Network patterns impacting the playback delay adjustment algorithms

To deal with variations in network delay, the common practice is to use jitter buffer, which introduce additional end-to-end delay, e.g. playback delay. The jitter buffering holds packets that arrive before their scheduling playback time and discards those, which arrive later. In choice of playback delay, a trade-off is performed between increase in the mouth-to-ear delay to avoid loss of late packets and tolerable packet loss to keep the mouth-to-ear delay small. First we give an overview on some algorithms for playback (jitter) buffer adjustment, which are based on consideration of delay patterns.

Then we define delay patterns and appropriate statistics using gradient analysis approach to support the selection of appropriate delay adjustment algorithms.

4.1. Network delay behaviour considered in algorithms for playback delay adjustment

The role of network delay variation for adjustment of playback buffers is discussed in a number of researches as in [23], [24], [25], [26], [27], [29] and [33].

Adaptive playback algorithms designed to consider network QoS behaviour, are for instance:

- Exponential average (playback delay calculated from the approximated values for the mean delay and variance) ([23]).
- Fast exponential average using weighted mean delay ([23]).
- Spikes detection ([28], [24] and [30]).

Exponential average algorithms to adjust the playback delay are based on following formulas ([20], [23] and [25]):

$$\text{Delay change} \rightarrow d_{\Delta} = \alpha \cdot d_{\Delta(i-1)} + (1 - \alpha) \cdot n_i$$

$$\text{Variance} \rightarrow v_{\Delta} = \alpha \cdot v_{\Delta(i-1)} + (1 - \alpha) \cdot d_{\Delta i} \cdot n_i$$

Parameter α has critical impact on rate of the d_p , i.e. playback delay. [20] studied different values for α and observed that:

- Higher values of α are good only in the situations where network conditions are stable, e.g. slow delay and jitter increase.
- Smaller values of α are appropriate when network conditions are changing rapidly (sudden increase/decrease in delay).

There is no optimal fixed value of α for different network QoS behaviour; therefore the adaptation of α considering the network delay structure is required.

[30] uses a statistical approximation of the delay distribution with the aim not to react too quickly to short-lived network jitter. There are playback delay schemes that follow the delay profile ignoring the role of spikes

([23] and [27]). These schemes work well, if the variations of the delay take place at a much slower rate.

The impact of “spikes”, e.g. sudden and large increase in the delays of the time series data which are of small duration Δt , on the playback delay adjustment is considered in different works:

- [28] Discusses the learning of spikes when they occur, identifying their magnitude and frequency of occurrence above a certain threshold as important for selection of playback delay adjustment scheme.
- [24] aim is based on the distribution of traced delays, to determine the average network delay under normal condition using this information to react more effectively to delay spikes. During the spike, the first packet in the spike is used as the playback delay. After the spike, the playback delay is chosen by finding the delay corresponding to the q_{th} quintile of the distribution of the last received N packets.

Considering these approaches for playback delay adjustment using in advance for knowledge of network delay behaviour, we propose to use patterns to analyse the network delay behaviour for specific periods. This analysis should be the base for determination of appropriate playback strategy.

4.2 Gradient based Pattern analysis of network delay change rate

We study the delay patterns of network connections in inter-domain environment to detect their suitability for the different playback delay adjustment algorithms. As discussed in the previous section, there are some useful analysis of the network delay supporting the selection of specific algorithm / parameters for playback delay adjustment. These are:

- Rate of the increase and decrease of the network delay that is described by the gradients. It is used especially in the techniques discussed in [20], [25], [26]. They justified parameter selection of their algorithms based on the network delay dynamic.
- Plain delay structures within a given range values can also be used in addition to slow/increase decrease patterns for playback delay justification ([23], [27]).
- Spikes and their frequencies in a given time interval are important for playback delay algorithms considering them (see [24] and [28]).

From a mathematical point of view, the real measured process with time discrete continuous values is mapped by the plain and increase/decrease patterns into a simple approximated process with plain phases and increase/decrease phases described with gradients. Piecewise linear approximations methods to describe QoS structure behaviour (increase, decrease) in aggregated form ([45], [46]) can be used for abstraction of delay structures. Approximation considering maximum values is applied in the INTEMON concept and can be used to

select playback delay adjustment algorithm and what kind of parameters can be set.

The basic patterns used:

- Plain,
- Increase,
- Decrease,
- Spike.

Increase and decrease patterns are sequences of time series data which values are increased respectively decreased. The gradient indicates the rate with which the consecutive QoS-values rise/fall in the given time interval. Let the QoS values $x_1, \dots, x_i, \dots, x_n$ belong to $\{x_t\}$, which is an increase basic pattern with

$$0 = x_1 = x_{\min} < \dots < x_i < \dots < x_n = x_{\max}$$

and decrease basic pattern with

$$x_{\max} = x_1 > \dots > x_i > \dots > x_n = x_{\min} > 0.$$

The gradient m of an increase or decrease pattern is defined as:

$$m = \Delta x / n, \text{ where}$$

$\Delta x = x_n - x_1$ is the difference between maximum and minimum value,

n is the number of data in the time series sequence.

Spikes are sudden increases, which are of small duration. The difference of “spike” to “outlier” in the delay impairment pattern is that spike is defined for small duration Δt (because for this small duration there is no need for playback delay change) and the threshold could be set independent from the E-quality, i.e. the system could detect spikes based on evaluation of the standard deviation in comparison with sequenced values.

The following analysis shows the delay characteristics study, significant for the delay adjustment selection. We show as an example the encountered patterns in the studied 4-days time interval of inter-domain network QoS of for VoIP application. The following figure shows the maximum delay behaviour of the Madrid–Salzburg connection considering periods of 10 minutes.

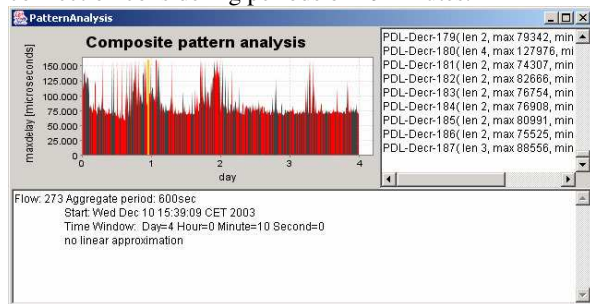


Figure 5: Maximum end-to-end delay of inter-domain connection for generated VoIP traffic

We consider the maximum delay behaviour per 10-minute period, because on this way we minimise the packet losses and approximate the delay buffers in a slower time intervals. The network outliers beginning of some lower

boundary level are discarded, because we do not want to impair the end-to-end delay with outlier delay. We take the R-factor reference of 160ms as the boundary for delay outlier discarding.

The following Figure 6 shows the 4-days network delay patterns with using of linear approximation algorithm with parameter of 20 ms which allows more efficiently to select the phases of the playback buffer adjustment:

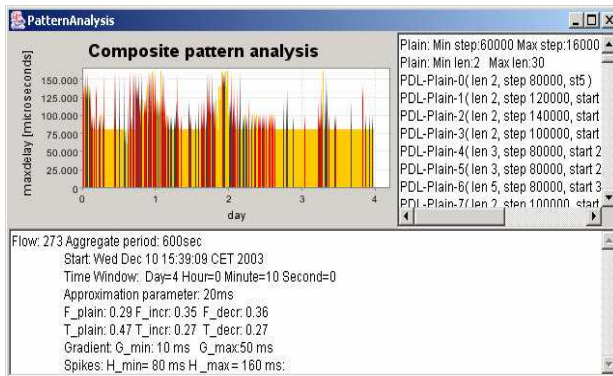


Figure 6: Using of linear approximation to describe network delay patterns for playback delay adjustment

The slow increase and decrease analysis is described by gradients which maximum values is 50 ms. Plain patterns as well as slow increase/decrease patterns interleaved with spikes are the prevailing patterns in the studied time interval and allow setting of algorithms which are described in [24] and [28].

“Turbulent” time intervals with rapid increase / decrease rate for this connection were of small duration.

5. Comparison of approaches to study Internet QoS for VoIP

There are a lot of researches, which are aimed to study VoIP quality in wide area Internet based on measurements (see, e.g. [4], [8], [7], [28], [32] and [37]).

One of the focuses is to analyse the impact of inter-domain routing (BGP-4 protocol) on network delay and packet loss of VoIP communication.

[8] discusses the link failures in a backbone network and the resulting BGP-4 behaviour on VoIP quality decrease concluding that link failures may be followed by long periods of inter-domain routing instabilities.

[32] is a similar work, focussed on the link failures and inter-domain routing impact on packet loss and delay characteristics. [10] analyses the measured single-hop delay from an operational backbone network and its impact on the VoIP quality.

[38] focuses on the impact of network QoS on different audio codecs as well as correlation of packet loss ratio and round trip time.

Further researches are aimed to study the network QoS parameter impact on standardised VoIP quality metrics. [3] considers the one-way transmission delay (propagation and serialisation delay), jitter, packet losses and codecs compression times to analyse and predict the quality of VoIP in IP networks, especially the impact of congestion on VoIP quality. The idea of the NetPredict ([3]) is to determine in ahead if the network will provide good voice quality for remote locations by taking into account impact of codecs (compression/decompression times, serialisation delay), network delay and loss and their impact on R-factor.

The measurement concepts are aimed to access availability of VoIP considering different factors and metrics. [11] investigates the use of the ITU-T's E-Model as a tool to relate several transport level metrics to an estimate of conversational voice quality; the reduction of the existing E-model in terms of transport-level metric for the purpose of monitoring of conversational voice quality was analysed.

[9] evaluates the VoIP service availability in the current Internet using several metrics including call success probability, overall packet loss probability, the proportion of time the network is suitable for VoIP service, and call abortion probability induced by network outages.

Scenarios and technologies for QoS evaluation of VoIP end-points in terms of mouth-to-ear delay are discussed in [6] and [42]. Mouth-to-ear delay analysis includes impacts such as clock skew, silence suppression behaviour and robustness to packet loss, playout delay adjustment and intelligence.

A lot of measurements consider the impact of underlying protocol and implementation issues of VoIP applications. [7] focuses to measure delay, interarrival times and path asymmetry impact for VoIP based on RTP/RTCP. [29] measures round trip delays of small UDP probe packets sent at regular time intervals to analyse the end-to-end packet delay and loss behaviour for VoIP.

An approach to derive an exact metric for numerical evaluation of the QoS of Internet connections is discussed in [34]. The quality of the connection in this work can be calculated as a vector of single weighted metrics, and the numerical representation of the overall connection quality is the product of the single values of the weighted metrics. These metrics, especially developed for verification of SLA of multimedia services, consider parameters like throughput, delay, jitter and packet loss ratio.

Considering these approaches, the benefits of our technology is the automation of the QoS monitoring and analysis of network QoS parameters (delay, packet loss) considering many approaches for automated analysis:

- Active monitoring based on different statistical approaches to generate traffic.
- Using the data mining approach of patterns to describe important network delay and packet loss impacts on VoIP Quality and application requirements (playback buffer adaptation).
- Mapping of monitored QoS metrics to VoIP quality (E-Quality Model) considering encoding schemes.
- Detection of Inter-domain routing impact using tool interaction in the INTERMON architecture.

Thus, we combine in one technology many automated approaches to study the network connection QoS behaviour for VoIP.

6. Conclusions

This work presents a new approach for evaluating the QoS parameter of Internet connections for VoIP applications using automated detection of patterns impacting quality and parameters of VoIP application. This approach, integrated in the INTERMON toolkit, allows considering:

- Mapping of network delay and packet loss "outliers" to the R-factor demands for VoIP quality.
- Network delay patterns for selection of playback delay adjustment.
- Impact of inter-domain routing by interaction in INTERMON integrated toolkit.

Further research is aimed on improvement of the technology to study and enhance VoIP quality based on patterns. Especially detection of QoS delay and packet loss "outliers" caused in the inter-domain environment, such as inter-domain route change, route flapping, traffic load, DoS,...) should be studied and considered in the assessment of network infrastructures for VoIP communication.

One of the future activities will be the characterisation of the outliers above using simulation tools in order to study the performance of several playout algorithms. The idea is for instance characterise by simulation the outlier due to a link failure between BGP routers and find the better playout algorithm in terms of the MOS parameter. First results about link failure outliers can be found in reference [47].

7. References

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