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Comprehensive Survey on Speech Quality Adaptation Invoice Over Internet Protocol (VOIP)

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Abstract: ***Voice over Internet Protocol (VoIP) for telephony network is becoming an important tool for voice communication. It is one of the prominent and fastest growing telecommunication service based on an Internet protocol suite. VoIP enables the users to use the Internet as the transmission medium for voice communication Thissurvey aims to give a comprehensive review of the current state-of-the-art research on speech quality adaptation of VoIP systems at the application layer. It should be especially helpful for new researchers in the field to get a big picture of the diverse QoS control methods for VoIP and their effectiveness in improving speech quality. Pointers to relevant works in this field are also provided for further reference.***

Keywords: VOIP, SIP, QoS, RTP, RCTP

1. Introduction

Applications and services based on Voice over Internet Protocol (VoIP) experienced a rapid growth in the last two decades. At first, cost-efficiency and convergence benefits stimulated such a growth, but now users and operators are concerned more and more with the quality and dependability of VoIP services, since increases in scalability have brought together some hurdles, such as configuration complexity and management effort.

Because of uncertainty and non determinism inherent in IP networks, voice streams are impaired by packet loss, delay, and jitter, which directly affect user perception of speech quality. For years, this problem has been challenging researchers and practitioners, who have been designing and improving QoS control mechanisms for VoIP applications. Such mechanisms aim to make the best use of network and terminal resources to minimize the effects of network impairments on voice quality. Among the several proposed QoS control mechanisms for VoIP, some of them seek to adapt thevoice flow or other VoIP-related parameters in accordance with significant changes in the network, end users' preferences, or service providers' requirements.

Adaptive systems in general respond to changes in their internal state or external environment with guidance of an underlying control system. Particularly, VoIP systems are likely to require dynamic adaptation because of the decentralized control nature of IP networks and the stochastic nature of data packet delivery. Although the existing adaptive solutions for QoS control of VoIP exhibit some sort of feedback, they do not provide explicit focus on it. For instance, most of them do not consider transient and steady-state performance. Such negligence is a critical obstacle for validating and verifying these solutions [Muller¨ et al. 2008].

Besides adaptive control theory, QoS control mechanisms for VoIP share analoies to reliability theory and autonomic communications initiative (AutoComm). Therefore, exploiting and deepening the formalism of these areas could help in designing more robust, reliable, and scalable VoIP services.

The main objective of this article is to review the works in the literature that use some sort of adaptation to provide QoS control for VoIP applications. Adaptation may take various forms, so our most important guideline is that the system structure exhibits at least one closed feedback loop, which is composed of four key activities: monitoring, analysis, planning, and execution. We reinterpret these works from the perspective of adaptive systems' requirements and remark on the research challenges that need further investigation.

Thus, the scope of this survey is restricted to those adaptive solutions that handle the voice flow (data plane) at the application layer aiming to improve speech quality. We do not include those that exclusively manage call signaling or billing information, are oriented to security issues of the voice flow, or concentrate in networking issues of QoS control.

This survey aims to give a comprehensive review of the current state-of-the-art research on speech quality adaptation of VoIP systems at the application layer. It should be especially helpful for new researchers in the field to get a big picture of the diverse QoS control methods for VoIP and their effectiveness in improving speech quality. Pointers to relevant works in this field are also provided for further reference.

2. Background Information

In this section, we review some background information about VoIP fundamentals, speech quality evaluation, QoS management in IP networks, and principles of self-adaptive software. This will provide us with a common basis of understanding for discussing the VoIP adaptation mechanisms presented in later sections.

2.1. Data Flows in a VoIP Call

A typical VoIP call basically comprises three data flows.

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1) Signaling flow enables VoIP endpoints to communicate with each other, set up and tear down calls, and renegotiate session parameters during the call.
2) Media flow transports the sender's voice encoded by some codec algorithm. It is conveyed by the Real-time Transport Protocol (RTP)
3) Media control flow, carried by the RTP Control Protocol (RTCP), provides control and quality monitoring service for RTP transport. Three RTCP reports are of major importance for adaptive purposes: sender report (SR) and receiver report (RR), which carry the basic transmission and reception statistics from the active sender(s) and receiver(s) in a session; and extended report (XR) [1], which carries quality information, such as MOS and delay at the dejitter buffer.

2.2. Speech Quality Evaluation

As noted by Shannon, the fundamental problem of communication is reproducing, at the receiver, a message sent by the source, either exactly or approximately. In VoIP-based systems, because of the lossy compression performed by speech codecs, the receiver will always obtain an approximate message of what was spoken at the source. Consequently, evaluation methods are needed for determining the quality of the received message. Particularly, this article focuses on speech quality alone, that is, on the RTP flow. We will not account for other aspects that make up a satisfactory call, such as delay to get the dial tone, connection success, and service availability.

Methods for speech quality evaluation are traditionally divided into two groups.

(1) Subjective. They quantify speech quality based on the opinion of a panel of listeners. Their main representative is the absolute category rating (ACR), during which listeners are asked to rate the absolute quality of speech samples, without comparing with a reference sample. The average of the individual ratings gives the mean opinion score (MOS), whose scale ranges from 1 (bad) to 5 (excellent). MOS values above 3.6 are considered acceptable for toll quality [3]. Note that some authors wrongly take the MOS—the outcome of the ACR tests—as being the ACR method itself. The involvement of human listeners makes ACR tests expensive and time-consuming. Moreover, subjective tests are not applicable to real-time monitoring [5].

(2) Objective. They use algorithms to estimate quality degradation and are further divided into three categories.

(a) Double-ended perceptual compares an input (original) and an output (degraded) speech signal to estimate quality, mapping the difference into the MOS scale. These methods are suitable for quality benchmarking and intrusive monitoring [Takahashi et al. 2004], but not for real-time monitoring because of the diffi-culty of having access to both original and degraded voice signals.—

(b) Single-ended perceptual does not require access to the reference signal and commonly rely on models of normative speech behavior. The ITU-T Recommen-dation P.563 [ITU-T 2004] represents the current state of the art of single-ended standard algorithm [ITU-T 2004; Malfait et al. 2006]. Recent research, however, has suggested that P.563 performance is compromised for VoIP applications [Ding et al. 2007; Falk and Chan 2008].

2.3. Overview of QoS Management of VoIP

The term QoS is used throughout the literature with many meanings, ranging from user's perception of the service to a set of connection parameters necessary for achieving a particular service quality. Here, we use *QoS* to refer to "a set of service requirements to be met by the network while transporting a flow" [Crawley et al. 1998], and QoE to describe resulting service features as perceived by the customer, such as MOS values.

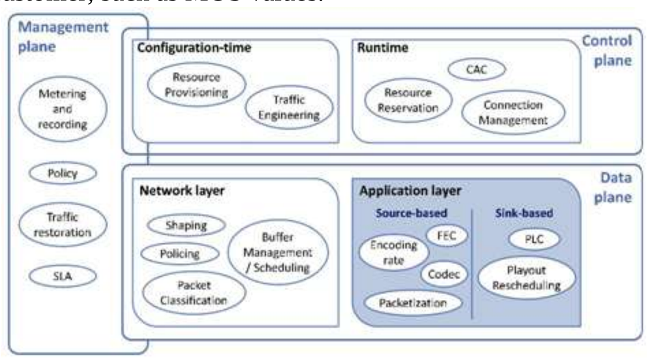

Figure 1: QoS management mechanisms for VoIP applications

Speech quality evaluation gives a snapshot of QoS and QoE problems in a VoIP system, but it does not offer the solution itself. In this sense, several mechanisms have been developed to manage the QoS and QoE of VoIP calls at runtime. Chen et al. [2003] classify these mechanisms into two planes: control and data. The ITU-T Recommendation Y.1291 [ITU-T 2006] adds a third plane: management. Bai and Ito [2004] subdivide the mechanisms supported by the application layer into source-based and sink-based. These three views are merged in Figure 1.

Application-layer mechanisms exploit the VoIP-specific characteristic to improve speech quality. They can be used to complement or substitute other QoS management mechanisms [Chen et al. 2003]. Some examples include codec switching, encoding rate control, and packetization adjustment, which adapt application's bandwidth demand over the network; forward error correction (FEC) and packet loss concealment (PLC), which adapt the robustness against network packet loss; and playout buffer reschedul-ing, which adapts the trade-off between end-to-end delay and packet discard.

Sink-based adaptation mechanisms have a quick response time, but they only react to network problems. In contrast, source-based ones can proactively change the bandwidth demand over the network, but they require a feedback message to trigger or stop their operation, which makes their reaction time slower.

In this article, we are especially interested in architectures that implement control mechanisms in the application layer and directly handle the RTP flow. The scope of this survey does not include control of signaling and billing information, which also belongs to the data plane of the QoS framework. Naturally, most of the surveyed works do not regard themselves as being part of this architectural QoS framework. Anyway, we have selected those works that place their solutions at the application layer, whether or not in conjunction with other control mechanisms.

2.4 Self-Adaptive Software

In the previous section, localization was the criterion used to classify QoS management mechanisms for VoIP into planes. Those mechanisms also could be classified according to their automation maturity, that is, how autonomously they may adapt the managed system's behavior in response to changes in the environment.

In this section, we present some basic concepts that will guide our further discussion on Section 3, where we review the literature about QoS management of VoIP from the self-adaptive software perspective.

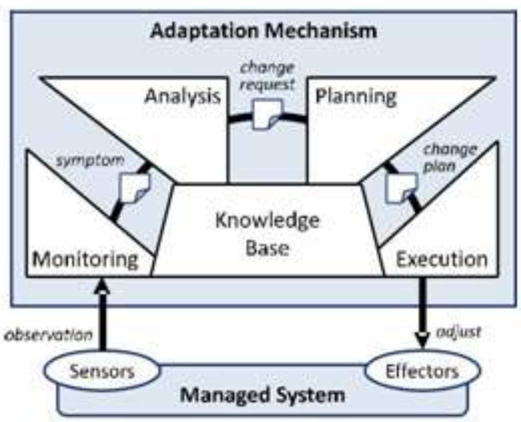

Figure 2: The feedback loop

2.4.1. The Feedback Loop: Adaptivity is not a Boolean property. It may be addressed by several components of a system and at different human-interference levels [Muller¨ et al. 2009]. Furthermore, it may appear in many guises, yet not explicitly. Anyway, what self-adaptive systems have in common is that design decisions are moved towards runtime to control dynamic behavior, so that they reason about their state (the self) and their environment (the context) [5]. This implies that a feedback loop lies at the heart of self-adaptive systems

A feedback loop, also known as an adaptation or autonomic loop, typically involves four key activities: monitoring, analysis, planning, and execution (MAPE). These activities are also referred to as collect, analyze, decide, and act, respectively [4]

As depicted in Figure 2, sensors collect data from the managed system. The feedback cycle starts with the monitoring of relevant data that reflect the current state of the system. Next, the system analyzes the collected data, structuring and reasoning about the raw data. Then, decisions must be planned about how to adapt the system to reach a desirable state. Finally, to implement the decision, the system must execute it by means of available effectors. Central to this loop, there is a knowledge base that keeps the necessary information about the managed entities and their operations. This reference model is also referred to as MAPE by the autonomic computing community [8].

In the context of a VoIP call, the monitoring activity is responsible for collecting relevant data that affects speech quality, such as delay, packet loss, and codec type. Next, the collected data is analyzed to identify unfavorable call conditions and their possible causes. Then, a decision action is planned depending on past actions, network conditions, and call configuration. Finally, the planned action is executed, which can entail changes in softphone configuration at sender and receiver endpoints, or cross-layer interactions among components in the voice path. A new control loop restarts, considering new conditions of the call and the results of past executions.

2.4.2. Adaptation Requirements: The requirements of self-adaptive software can be classified into four logical groups of questions [3].

1) Where (object of change) questions set out to locate the problem to be solved and the supporting mechanisms for recovery.
2) When (temporal properties) questions address temporal properties, such as when a change should be made, in which frequency that change should be taken, or whether change history should be controlled.
3) What (system properties) questions identify which attributes of the system can be changed through adaptive actions.
4) How (change support) questions address how the adjustable attributes can be changed; which adaptive action(s) should be applied; and how the order of changes, their costs, and after effects are considered for deciding the next action.

These groups of questions are used for eliciting adaptation requirements during developing and operation phases of the software life cycle. In this article, we use these questions to guide our literature review, identifying the self-adaptive characteristics hidden in the studied architectures. A more detailed discussion about self-adaptive software is provided by [9] and [3].

3. Classification of Mechanisms for Speech Quality Adaptation

In this section, we propose a classification system for VoIP architectures that somehow implements the four activities of the feedback loop (Figure 2), based on the considerations of Section 2.4 regarding self-adaptive software. It will serve as a guideline to the literature review about mechanisms of speech quality adaptation at the application layer. It is based on the where/what/when/how questions for eliciting the requirements for a self-adaptive system. With regard to the requirements about how and when adaptation should be applied to a managed voice flow, we can expand the three approaches identified by [11] to codec-adaptation only as follows.

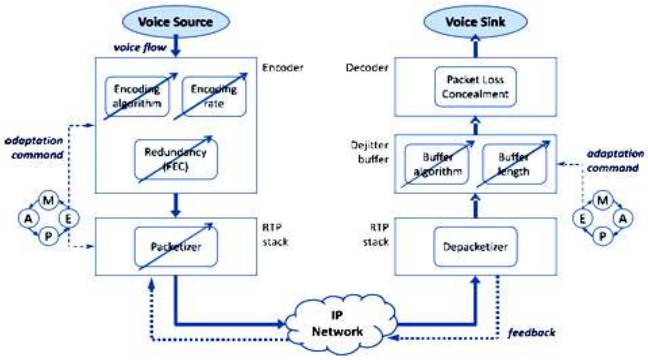

Figure 3: Placement of adjustable parameters used for executing the adaptation plan in a VoIP architecture

1) Nonadaptive. Performed by an intermediate node, it consists of dropping or blocking new calls that affect the quality of other ongoing calls. The RTP flow is not adapted [8] as an example.
2) Single-adaptive. Only one call is managed by the VoIP endpoints in such a manner that adjustable parameters (e.g., codec, packetization, FEC) are tuned during the ongoing call. It can be performed by sender, receiver, or intermediate node. It can be applied anytime during the call or silence periods between talkspurts.
3) Multi-adaptive. Performed by an intermediate node, more than one call is managed at once. It can be applied in two ways.
   a) Bulk traffic. The planning agent can adapt all ongoing calls managed by the intermediate node.[8] as an example.
   b) New calls. Adaptation is applied only when new calls are accepted by the intermediate node, without modifying current calls [11],Depending on which VoIP component executes the adaptation plan, we can classify the approaches to speech quality adaptation into two groups: source-based, where the adjustable parameters are available at the sender (Section 4), and sink-based, where the adjustable parameters are available at the receiver. Figure 3, inspired by [10], presents a conceptual diagram of where those adjustable parameters are located in the VoIP components of the RTP flow.

3.1 Adaptation-Related Variables

Usually, a system converts input signals into output signals by performing operations on the inputs and intermediate products. The values of measurable properties of a system's states are called variables [7]. A first step in designing an adaptive mechanism is to identify the key variables of the managed system. An adaptive mechanism usually deals with four kinds of variables,
1) Observation parameters are measurable variables from which the adaptive mech-anism can infer the status of the managed system.
2) Decision metrics characterize the system performance over a sampling period that the planning agent tries to optimize. They can be equivalent to a single observation parameter, such as delay and packet loss, or a synthesis of a set of observation parameters, such as MOS.
3) Performance references represent the desired system performance in terms of observation parameters.
4) Adjustable parameters correspond to the effectors in the feedback loop (Figure 2), an attribute of the managed system that can be manipulated to apply the necessary adaptations. Essentially, adaptive systems implement a transfer function that takes decision metrics as input and gives the amount of change (if needed) in the adjustable parameters as output.

As an illustration, let us consider an adaptive VoIP mechanism that uses packet loss for determining the MOS. If the MOS value is below 3.6, then some change in codec bitrate is triggered. In this example, packet loss is an observation parameter, since it cannot be controlled by sender and receiver, and it is not used for deciding about adjustments in the system. MOS is a decision metric, because it summarizes some observation parameters, and it is used for deliberating about changes. Codec bitrate is an adjustable parameter, since it can actually be controlled by the sender. Finally, the MOS threshold of 3.6 is a performance reference to the mechanism of the example.

Observation Parameters: The MOS, if taken alone, is not sufficient to diagnose the cause of problems in speech quality and to support recovery planning over the system [9]. Hence, it is recommended to collect more observation parameters to decide which adjustable parameters should be tuned.

Decision Metrics: In the adaptive VoIP architectures surveyed here, the most used decision metrics can be divided into three groups.
1) QoE metrics characterize the overall acceptability of the service as perceived by the end-user. The most used one is the MOS [11]. In most of the surveyed works, the MOS is determined by means of the E-model. The only exceptions are the works of [14], which uses a mix of PESQ and E-model, and Mohamed et al. [11], which uses an MOS value

determined by a neural network based on network conditions.

2) NQoS. Network QoS parameters comprise all metrics determined from measure-ments taken at the network layer, such as packet loss, network delay, jitter, band-width, throughput, and congestion level.
3) L2QoS. Layer-2 QoS parameters comprise all metrics determined at the underlying medium access technology, such as transmission rate and modulation scheme.

Performance References. Performance references should be decoupled from the source code that implements an adaptive VoIP system. A network administrator may update such values so that the MAPE agents can get them from a knowledge base.

Adjustable Parameters. Generally, VoIP applications may tune the parameters listed in the second column of Table I. These parameters should not be considered separately, because optimization with respect to a single parameter has harmful effects on the others [18]. This is expressed in the columns expected benefits and side effects, which list the factors of the E-model (Equations (1) to (3)) that are affected by changes in the adjustable parameters. These two columns evince the planning agent's challenge in improving some factor (benefits), constrained by the undesirable associated deterioration in other(s) factor(s) (side effects).
In this survey, we do not cover VoIP architectures that use PLC or header compression as adjustable parameters. Actual implementations of PLC techniques are normally intrinsic to the codec algorithm, such as G.711, G.729 and iLBC [14]. Moreover, header compression techniques are implemented on a hop-to-hop basis, rather than end-to-end [16].

3.2. Placement of the MAPE Agents

Basically, the MAPE agents in command of the VoIP feedback loop can be placed in three elements: sender endpoint, receiver endpoint, or an intermediate node. Most of the observation parameters (e.g., end-to-end delay, packet loss, and MOS) are deter-mined at the receiver side and sent back via a feedback channel (e.g., RTCP report). However, waiting for RTCP packets may cause further delay during reaction [17]. This is why the most important restriction when designing VoIP adaptive systems is to keep control and feedback information traffic at minimal.

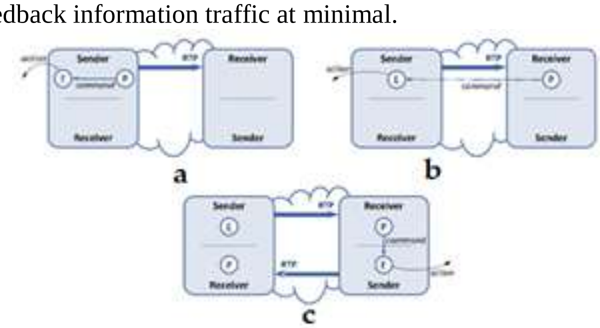

Figure 4: Three strategies for switching the current codec of an RTP flow: (a) both planning (P) and execution a) Agents at sender, (b) planning agent at the receiver and execution agent at sender, or (c) planning agent at the receiver and execution agent at the sender of the RTP counterflow.

In this section, we focus on the placement of the planning agent, as it determines the dynamics of both *estimation* messages, which gather observation parameters of the managed ongoing call, and *feedback* messages, which determine the adjustment parameters to be tuned accordingly.

Some mechanisms implement the planning agent in an intermediate node, which can be a media gateway [14], a wireless access point performing cross-layer QoS management [11], or a dedicated QoS management node [12].

When the planning agent is implemented in the endpoints, three main strategies are commonly adopted.

(1) *Adaptation decision and execution taken by the sender of the managed RTP flow.* The planning agent resides in the sender. When it decides to adapt some adjustable parameter, it sends an internal message to the execution agent for applying the changes. This is the most adopted strategy among the surveyed works.
(2) *Adaptation decision taken by the receiver, and execution taken by the sender of the managed RTP flow.* The planning agent resides in the receiver. When it decidesto adapt, it sends a feedback message to the execution agent implemented in the sender to apply the change. This strategy is adopted by [9].
(3) *Adaptation decision taken by the receiver, and execution taken by the sender of the managed RTP counterflow.* Insofar as a user can act as both speaker and listener,a VoIP call is made up of two RTP flows. If the planning agent at the receiver side of one flow decides to adapt, then it can ask the sender of the opposite flow to apply the change, by means of a re-INVITE SIP request, for example. Since they are physically implemented in the same softphone, only an internal message is needed to convey the change plan.
(4) Two other placement strategies were proposed by Escobar and Best [13], in which the caller analyzes and plans adaptation actions over the RTP flow; and by [10], in which the callee monitors speech quality and sends an alarm using the SIP instant message (IM) to the caller, which runs the adaptation scheme for codec switching. Note that *caller* and *callee* are roles performed by the endpoints during call establishment. After this, the RTP connection is set up, and the endpoints assume the roles of sender and receiver of the RTP flow.

4. Source-Based Adaptation

Source-based adaptation approaches are those executed by the sender endpoint to recover or optimize speech quality. The sender can be a VoIP terminal, implemented either in a soft phone or hard phone. It also can be a media gateway placed between the IP network and the access network, such as a PSTN or mobile telephone network.

Source-based QoS management is designed to improve long-term voice flow characteristics, since the sender endpoint has to wait for control messages from the planning agent telling it to change encoding configurations. In contrast, sink-based management is used to improve short-term quality. Its fast reaction does not change the encoding characteristics, but manages the delay-loss trade-off.

1) *Bandwidth control of media information.* Source-based adaptation can regulatethe amount of voice information per time unit it delivers to the network. This is done by switching the current codec, regulating the encoding rate (for VBR codecs), or adjusting the frame-per-packet ratio (packetization).
2) *Redundancy.* The sender can spread redundant information over several packetsso that frame information can be recovered at the receiver even if some packets are lost. The remainder of this section groups the sender-based control approaches to VoIP adaptation according to the adjustable parameter used by the planning and execution agents for changing the codec bitrate or for making the voice stream more robust to packet loss.

5. Playout Scheduling Adaptation

Dejitter (or playout) buffers remove jitter by temporarily storing the arriving voice packets and forwarding them to the decoder at regular time intervals. They are responsible also for reordering out-of-sequence packets. Dejitter buffers are broadly categorized as fixed (static) and adaptive (dynamic) size. [11] presented a comprehensive survey about playout scheduling in IP telephony and a taxonomy for dejitter buffer strategies, which is summarized in Figure 8 and briefly explained next.

(1) *Fixed (static) buffers.* End-to-end delay is kept constant for all voice packets, eitherat design time or during the call. Such a strategy is inefficient, since it is not resilient against the temporal variability of network behavior.

(2) *Adaptive (dynamic) buffers* try to find some optimal point in the trade-off betweenend-to-end delay and packet loss and dynamically adjust the buffer size accordingly. Depending on when the buffer size is adjusted, they can be further divided into two groups.
   - (a) *Intra-talkspurt*adjusts the end-to-end delay independently from silence periods, using waveform compression or extension.
   - (b) *Between-talkspurt*acts during periods of silence. They are used more often,because they do not require any signal-processing technique to change the length of the speech. They are further grouped depending on how they handle the trade-off between end-to-end delay and packet discard.
     - *Loss-intolerant* estimates network delay and sets the playout delay so thatonly a small fraction of packets are discarded. They do not take PLC into ac-count, resulting in an overestimation of the required playout delay. [15] further classify these strategies as *reactive*, which continuously estimate network delay and jitter to calculate playout deadlines, and *histogram-based*, which maintain a histogram of packet delay and choose theoptimal playout delay from it.
     - *Loss-tolerant* monitors the packet-loss ratio or buffer occupancy and adjustsplayout delay accordingly. An amount of packet loss is then allowed, and playout delay is set to reach this target value.
     - *Quality-based* seeks to maximize some metric linked to the end-user perceived quality, such as MOS.

6. Specification of Formal Models

Designing and verification techniques are necessary to assure that VoIP systems will exhibit the desired adaptive properties. During the design phase, the main interest is to describe and model both individual and social properties. Individual properties are observed on the micro scale (i.e., related to each single component, such as softphones, proxies, and gateways). Social properties are observed on the macro scale (i.e., emerged from cooperation between softphones, or negotiation between softphones and proxies). During the verification phase, the interest is to determine whether the VoIP system exhibits the designed properties.

Therefore, formal models are needed not only to assure the correctness of programs, but also to allow computer systems to reason about its internal states and environment in a really self-adaptive way. This includes representing and verifying some properties, such as functional correctness, reachability, safety, liveness, fairness, and timeliness, at different observation scales.

7. Conclusion

The area of self-adaptive software offers some principles for tackling the complexity of computer systems in general and the uncertainty of their environments. At the core of self-adaptive systems lies the feedback loop, composed of four activities that continuously run over the managed system: monitoring, analysis, planning, and execution.Particularly, self-adaptive software principles can be applied to manage voice over IP systems. Indeed, it has been applied to solve diverse problems in VoIP, yet not explicitly. This survey has highlighted the feedback cycle that lies behind several works in the literature about speech quality adaptation of VoIP systems. These works were organized into two groups, depending on which RTP endpoint executes the planned adaptation action: source-based and sink-based.Because VoIP is itself a broad research area, we have limited the scope of this article to QoS control mechanisms placed at the application layer. This does not mean that there is no adaptive approach for solving signaling and security problems in VoIP, but only that they were put aside for further investigation.

Finally, we identified a landscape of research challenges that should be addressed in future works to enable systematic and well-founded design and verification of adaptive VoIP systems. These challenges were divided into four groups: (1) improvement of existing work on adaptive VoIP, (2) extension of the scope of VoIP adaptation beyond QoS control, (3) management of knowledge information handled by MAPE agents, and (4) specification of formal models for designing and verifying autonomous VoIP systems.

8. Future Work

Some opportunities arise when we regard the problem of QoS control of VoIP under the self-adaptive software perspective. This vision poses new challenges for developing and validating VoIP systems with adaptation characteristics. These challenges are grouped into four basic keypoints, as follows.

Improvement of Existing VoIP Adaptation Mechanisms
Adaptation can be compared to a surgery, which aims to heal a patient, but should not kill her. Similarly, the side effects of adaptation should not worsen the problem that it tries to overcome. Therefore, current VoIP adaptation mechanisms should be aware of possible glitches during parameter transitions, which are not evaluated by the E-model or POLQA.

Furthermore, because of the widespread use of mobile devices, memory and power constraints should also be considered as decision metrics during the planning activity. Finally, VoIP adaptation mechanisms should be shifted from mere parameter tuning to strategy composition, in the sense of adopting new strategies to address concerns that were unforeseen during a system's development.

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