Design and Configuration of Context-aware VoIP Telephony Systems

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Keywords: Context, Context-aware, VoIP, Telephony, Design, Configuration.

Abstract: Voice over IP is a widely used concept with regard to a realization technology of different types of telephony systems, including those that are used in enterprises. Such systems consist of a call processing component and a set of desk endpoints that are pervasive from a user perspective. Those endpoints are usually not mobile, but in result can deliver a much greater set of functions needed in an everyday office work (e.g. secretary and attendant features). Those functions also make VoIP telephony systems well suited for using a context information to adapt to the situations of their users. The paper presents an approach to designing such systems from a context-aware point of view and organizing a configuration of their objects. A sample deployment has been prepared and evaluated.

1 INTRODUCTION

A context (Krawczyk and Nasiadka, 2011) with regard to a communication between people can be used in different ways (Schilit et al., 2007). However, they mostly regard cell phones and extension of their capabilities to interpret context data. Apart from those devices there is also a group of telephony systems (TS) used within enterprises and smaller companies. They consist of distributed or centralized call processing system and a set of desk endpoints (or softphones) that are pervasive from a user perspective. Those endpoints can deliver a much greater set of functions that are needed in an everyday office work (e.g. secretary and attendant features) than cellular phones. Such TS systems are often VoIP (Johnston, 2009) based solutions. As enterprises are global and hire great number of people located worldwide, those systems can become very large and complex. Nevertheless, they are supposed to offer their users a context-aware functionality. Delivering it can be challenging from a design and configuration point of view. The aim of this article is to propose a new approach to designing enterprise telephony systems and configuring telephony objects (e.g. endpoint devices) using information about their context. There has been prepared a sample deployment of a VoIP TS enhanced with the environment based on the system for dynamic service integration described in (Dziubich et al., 2012). Thanks to the presented approach the design process regarding context-awareness can be systemized and TS can dynamically adapt to changes in its and its users environment.

The reminder of the article is constructed as follows. Section 2 describes state of the art regarding context-aware behavior of telephony devices. In section 3 there is presented a model of a VoIP telephony system (VoIP TS). Section 4 describes how a context is used in VoIP TS across its layers. In Section 5 there is introduced a new approach to designing and configuring context-aware VoIP TS. This approach has been used to prepare a sample deployment which is described in Section 6. This section also briefly describes experimental results regarding introduction of that approach. Section 7 summarizes the paper.

2 RELATED WORK

The vast majority of publications regarding context-aware telephony refer to mobile interruptions. They address the problem of cell phones ringing at inappropriate moments (such as in a theater). In the paper (Schmidt et al., 2000) authors propose a mechanism that addresses this issue. Each mobile phone user can choose a state which expresses if he is willing to receive calls. Callers can then check what is the status of the person to whom the call is to be made and decide whether to make a call. Another approach is presented in the publication (Raento et al., 2005), where authors describe a framework that allows to automatically collect information about a context of a mobile phone user. This context can

Nasiadka S..
Design and Configuration of Context-aware VoIP Telephony Systems.
DOI: 10.5220/0004314100420048
In *Proceedings of the 3rd International Conference on Pervasive Embedded Computing and Communication Systems* (PECCS-2013), pages 42-48
ISBN: 978-989-8565-43-3

then be presented in various forms to the caller who can decide whether to continue with the call. However, such an approach means sharing certain private information (e.g. location) about people. The paper (Khalil and Connelly, 2006) draws attention to this aspect and describes what are people preferences regarding providing information about their situation. Described publications touch different aspects of VoIP telephony systems (including design, tools and frameworks), however they mostly regard mobile phones which greatly differ from enterprise VoIP systems. Another approach has been described in (Chihan et al., 2012), where authors concentrate on creating a framework for developing context-aware communication systems. That work focuses on communication services, but lacks the analysis of how a design and configuration of particular objects existing in a telephony system should be organized with regard to context-awareness.

Some commercial systems (like Cisco Unified Communications Manager) already use context data. For instance, Cisco softphones may be used by mobile users in different locations where the same distributed VoIP TS is deployed. Because of using VoIP technology the current location can be easily recognized (e.g. by the IP address) by the TS and during a registration of an endpoint and it can be assigned a different set of configuration entries than in its home location. That set of entries can influence mainly on its dialing capabilities (e.g. allowing dialing only internal calls) and on a codec it uses to encode a voice stream. A context can also be used during assigning different call routing rules. E.g. depending on a current time, an endpoint may have different dialling restrictions. During work hours it can be allowed to dial any type of numbers, but after work hours only internal and emergency calls are allowed. However, commercial products make use of only limited context data and only in some particular configuration aspects.

3 VoIP TELEPHONY SYSTEM MODEL

Telephony systems can be described using a layered model which is shown in Figure 1.

The lowest layer (endpoints layer) represents physical devices that are used by users to make calls (phones, softphones). That layer also includes gateways to Public Switched Telephone Network (PSTN) and other devices or software that are used as termination points of media streams (such as media resources - like transcoders, conference bridges, and so on). Those devices register to a TS through a register (registry layer), which includes information about values of parameters describing endpoints. Those parameters may represent a location of a device, its codec or video capabilities and so on. They are used in the process of call routing and establishment. That process is actually the most important one in every TS and is located in a separate layer - call routing layer. Based on the information from the registry the call routing process, routes and establishes calls between endpoints based on the input from a caller. The top layer - application layer - represents applications that utilize the call routing and establishment process, such as Automatic Call Distribution (ACD) or Auto Attendand (AA). Within each of the presented layers there have to be configuration, security and safety mechanisms available.

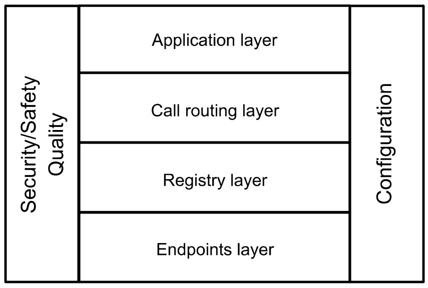

Figure 1: A telephony system layered model.

A VoIP Telephony System (VoIP TS) is a telephony system that uses the Internet Protocol (IP) as a transport protocol for voice and signalization packets. On the one hand it simplifies the management of the infrastruture, but on the other hand it causes difficulties in guaranteeing proper QoS characteristics - mainly regarding latency, jitter and queue management (Verma, 2011). That is why across all the layers of the ISO/OSI model (Tanenbaum, 2010) there have to be introduced advanced QoS mechanisms (like LLQ - Low Latency Queuing). Another feature that is available to users in VoIP TS systems is a mobility of endpoints. Wherever a particular endpoint is connected to the IP network, if only it can communicate with its VoIP TS, it can register and function with the same settings as in the home location. That gives users a consistent experience and allows administrators to easily manage a whole telephony infrastructure. Moreover, those endpoints also provide additional functionalities like access to web applications. As it can be observed the usage of the IP protocol mostly incluences on the endpoints and registry layers. Often the QoS mechanisms are controlled by the call routing layer.

Deploying a VoIP TS is a complex process which most important parts are a design and a configuration of particular TS mechanisms and objects across all the layers of the TS layered model (see Figure 1). Those parts are organized according to the user requirements which may regard a context-awareness. Because those requirements may be formulated late by the users during the deployment (e.g. based on the experience of using a partly deployed TS system) an agile-based approach is necessary. To introduce a new approach to designing and configuring a context-aware VoIP TS there will be analyzed a sample VoIP TS deployment. Its initial set of requirements is summarized in Table 1.

Table 1: Initial set of requirements for the sample VoIP TS deployment.

Id	Requirement
1	200 users in different locations (geographically distributed)
2	a redundancy (2 sites)
3	a local PSTN access and a toll bypass functionality
4	Direct Dial-In (DDI) numbers for users
5	calling to international numbers should be available only for a particular group of users
6	a Voice Mail account for each user
7	an ability to move users between locations and an automatic adjutsing of configuration parameters of their endpoints (e.g. codecs)
8	after work hours all incoming calls should be routed directly to cell phones or voice mail when no answer
9	an administration should only be allowed for users with appropriate skills
10	a line description, a presentation and a phone language should change after a user logs into a workstation

Some of the requirements regard functionalities that don't depend on context (like number of users and telephony numbers that they use). However, some of them are context-aware (with ids 7 - 10). How a context can be utilized in TS VoIP systems is described in the next section.

4 CONTEXT IN VoIP TS

With regard to context-aware systems there can be defined two types of input data (Krawczyk and Nasiadka, 2011). In case of a context-aware VoIP TS the first one is related to the user input which is a telephony number that the user enters to reach a called party (called DNIS number). The second one is the data that create context which can influence on different aspects of the TS behavior that are spread among layers from the model presented in Figure 1.

4.1 Endpoints

Endpoints can perform sensing, processing and acting functions. This is possible because in a VoIP TS they need to perform complex operations during each call (e.g. a TCP/IP connection establishment, encoding voice). Hence, they are equiped with a processor and memory which can also be utilized to perform additional operations, that are not directly related to the telephony. Those operations can be context-aware (e.g. switching a device to an idle mode after a particular amount of inactivity time). Thanks to this, the usage of a VoIP enterprise telephony makes the space where it is deployed intelligent (Intelligent Space - IS). That is further used in such concepts as intelligent buildings, where telephony endpoints are used as interfaces for users to non-telephony systems (like controlling an air condition or a room automation).

4.2 Registry

A registry that contains all the information about endpoints also needs to allow to enhance them with context data. Moreover, it should be possible to change a configuration of entries based on the current context. That is crucial e.g. for mobile users, who can connect an endpoint to the infrastructure anywhere they are. Depending on a location and other context data the information contained in the registry should be appropriately changed, so that e.g. a codec negotiation and call routing mechanisms can behave correctly.

4.3 Call Routing

Because of the use of the IP protocol for transporting media packets, different quality issues in a VoIP TS may arrise. Hence, depending on the situation in the network layer (which may also be included in a context) and information contained in the registry, the call routing mechanism has to establish connections differently to maximise a quality of calls (e.g. in case of a total Wide Area Network bandwidth saturation the VoIP TS system should use external PSTN circuits to establish long distance calls). Other type of context data that can be used during execution of processes from this layer is the current time.

4.4 Applications

A behavior of additional call handling applications (e.g. Interactive Voice Response, Voice Mail, Auto Attendant) may be different depending on an ANI number, a time, data, a profile of a caller and so on. A management and administration of the system and a login and logout functionality for users also creates context that can be used by applications.

Because endpoints provide an access to functionalities and information anytime and anywhere, they can be treated as pervasive devices. As a result, the layered model presented in Figure 1 can be divided into a pervasive part, which consists of the endpoints layer and a ubiquitous part, which consists of the registry, the call routing and the applications layer. That reflects the nature of enterprise telephony systems, where endpoints are distributed among users (they are physically close to them and each user has his own endpoint) and the rest of the system is logically centralized but accessible anywhere users need.

From the set of requirements presented in Table 1, last four of them regard context-awareness. To deliver a system which meets those requirements it is necessary do go through the process of its deployment which mainly includes design and configuration.

5 DESIGN AND CONFIGURATION APPROACH FOR CONTEXT-AWARE VoIP TS

A deployment of a context-aware VoIP TS can be systemized with the usage of the layered model presented in Figure 1. A resulting approach is organized according to the Agile method. Firstly, the requirements for a new VoIP TS deployment have to be assigned to the particular layers of the TS model. Next, each of the requirements needs to have assigned a priority expressing its importance from the user perspective. Based on the analysis, for each requirement there can be assessed a time that is needed to prepare a necessary configuration. The conducted analysis should include a necessary design of a part of the VoIP TS (which will satisfy the requirements) as well as interferences with the already deployed parts of the system. The design that regards context-aware requirements should include defining a situation (particular context) to which appearance in the space the VoIP TS will react, and also which configuration changes in the TS will take place. Based on the priorities, assessments and time constraints the deployment process can be divided into separate iterations. After each iteration is completed, there is delivered a useful from the user perspective functionality that can be tested and accepted. That also allows users to introduce new or correct previously specified requirements along with the progress of the deployment, which is a common situation during VoIP TS deployments. The organization of the sample deployment which resulted from implementing described approach is presented in Table 2 (context-aware requirements are marked with '(c)').

Comparing to the initial set of requirements (see Table 1) a one additional requirement has been formulated by users. After they used a partly deployed system (delivered in the first iteration) users decided that when someone logs out from a workstation, an office phone should switch to a stand-by (idle) mode and should require an authorization for making calls. That requirement (with id 11(c)) appeared during the second iteration and its realization has been planned for the third iteration. Within each iteration after the design is completed, a necessary configuration needs to be prepared.

VoIP TS systems may need to behave differently depending on their (and their objects) context. Hence, a particular object (apart from the initial configuration needed to satisfy context-unaware requirements) may need to be reconfigured depending on context data delivered by sensors (i.e. endpoints - see Section 4 or other sources). Nowadays, such context-aware reconfiguration can be based on a limited set of parameters (most often a time and a location). However, VoIP TS systems can be deployed in different environments and the definition of a context can change between deployments. That is why a VoIP TS should have a capability of defining a context and based on it - reconfiguring its particular objects. Context data can be sent by sensors using a SIP protocol (Pan et al., 2003) (in a form of additional headers) which is widely used in the telephony signalization.

Regarding an aspect of context-awareness, VoIP TS systems can be modelled using the CAA model introduced in (Krawczyk and Nasiadka, 2011). As such their architecture can be extended with components needed to execute CAA applications. Those components are responsible for context analysis, communication with sensors, knowlegde base management and changing configuration. The implementation of such extension for a particular VoIP TS deployment has been described in the next section.

Table 2: An organization of design and configuration tasks in the sample deployment.

Requirement id	Layer	Priority (1 - highest)	Time assessement in MD - ManDay	Iteration number
1, 2	application layer	1	25MD	1
3	call routing layer	1	5MD	1
4, 5	registry layer	1	7MD	1
6, 9(c)	application layer	2	11MD	1
7(c)	registry layer	1	5MD	2
8(c)	call routing layer	2	1MD	2
10(c)	registry layer	2	10MD	2
11(c)	endpoints layer and registry layer	2	5MD	3

6 IMPLEMENTATION

To see how the presented approach behaves in a real life scenario, there was prepared a sample deployment of a VoIP TS. It was organized according to Table 2. The deployment was based on the Cisco Unified Communications Manager (CUCM) 8.5 platform. That platform already has some context-aware mechanisms built into the registry and the call routing layers but they are too limited to fulfill all the requirements presented in Table 2 (i.e. requirements 9, 10 and 11 were not satisfied). Hence, those mechanisms were enhanced by the introduction of an execution environment for CAA applications (later called CAAEE) which was based on the system for dynamic service integration described in (Dziubich et al., 2012). The resulting structure of the deployed system is presented in Figure 2.

The CAAEE environment consists of a base services environment component, a context analysis component, an action execution component and a knowledge base component. The base services environment component communicates with custom .NET applications (installed on user workstations) which deliver context data. The data included a name of a user that logged in (or logged out) to the workstation associated with the endpoint (an association was configured in the CUCM) and a current time. It was then analyzed in the context analysis component. Based on the analysis, a particular configuration change was applied (through the action execution component of the CAAEE) to the telephony object using the AXL (Administration XML) interface. Based on the context-aware requirements there were designed the following rules which were further implemented in the CAAEE (each entry is in a form: a situation - a necessary change in the configuration):

- a user has logged out from the workstation - make a copy the CSS of the phone along with its partitions and patterns belonging to the partitions, enable the Forced Authorization Code setting on the patterns and set the new CSS on the phone; set the idle URL on the phone,
- a user has passed a training on the CUCM and gained a knowledge about the CUCM configuration - change the user profile in the CUCM so that the user can save configuration changes,
- a user has logged in to the workstation - change the CSS of the phone to the one that contains partitions that don't force an authorisation, (if exists) delete the CSS forcing an authorization, unset the idle URL, set a new line description, a presentation and a language settings.

The endpoints (Cisco 7965 IPPhones) sent their IP address directly to the CUCM during registration. Based on this information an appropriate Device Pool was chosen. That allowed to adjust configuration paramters for the endpoint according to its location which was determined based on the IP address. Additionally, a current time was also analyzed internally by the CUCM call routing process for each routed call. That analysis was performed on a partition level and enabled changes in the routing so that after work hours all incoming calls were routed to a voice mail or cell phones.

The use of CAAEE allowed to satisfy all the requirements that were identified in the Table 2. Some of them could be satisfied without that additional

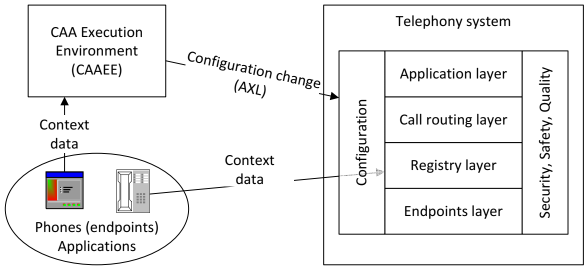

Figure 2: The structure of the sample context-aware VoIP TS system.

component - particularly context-unaware requirements (with ids 1 - 6) as well as context-aware ones for which the CUCM built-in mechanisms were sufficient (requirements with ids 7 - 8). It is worth to mention the latter regarded the call routing which is a real-time task. It can be expected that the use of a context during execution of those tasks can be limited only to built-in mechanisms as the additional analysis of context data can make them last unacceptably long. However, a configuration of those tasks (which doesn't have to be completed in a real-time) can successfuly be enhanced with a context-awareness. Hence, the CAAEE could be used for changing configuration of telephony objects across all the layers from the layered model presented in Figure 1. A particular change of the configuration needed to satisfy requirements with ids 9, 10 and 11 took about 3 seconds (measured from the time the situation appeared in the IS to the time that the new configuration was sent through the AXL interface). In case of the CUCM platform it is also important that after some configuration changes the endpoint has to be restarted. However, as those changes are done rarely (login and logout from the workstation) it wasn't annoying for users.

7 CONCLUSIONS

The paper presents a new approach to designing (and more generally deploying) and configuring context-aware enterprise VoIP telephony systems. Thanks to this approach a deployment process can be systemized and agile-oriented. To prove that the proposed approach is useful there has been prepared a deployment based on a commercial VoIP telephony solution. The implementation showed that a VoIP TS can be succesfully designed according to the presented method and that a proposed mechanism of changing a configuration based on the current context is useful and feasable for users. For the sample deployment there was used a currently available system for context analysis and adaptaion action execution. It allowed to dynamically change a configuration of particular telephony objects so that they behave more convenient from the end user perspective (they adapt to the current situation).

It is worth to mention that the proposed organization of a design and a configuration processes significantly differ from those that are used currently. This is because interactions between objects cannot be explicitly expressed within a configuration. They need to be taken into account by designing an appropriate contexts based on which a particular configuration is chosen. That is why it may be more difficult to deploy a VoIP system and test it before running in production. A requirement for new supporting tools arises in this area.

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