Implementing Intelligent Network Services with the Call Processing Language

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Abstract - **The increase of Internet telephony and the rise of the XML as a language standard have recently promoted proposals of XML-based scripting languages that can be used for telecommunications service creation. This paper shows how CPL and VoiceXML can be used to implement traditional intelligent network services.**

Keywords - **service creation, intelligent network, CPL, VoiceXML**

I. INTRODUCTION

The next generation networks (NGN) will provide voice, video, multimedia and data communications using packed-based transmission rather than switched connections. The NGN will converge fixed and mobile access using different technologies. In this heterogeneous environment service development platforms will provide open access to service building blocks. The idea behind service creation in NGN is to provide application programming interfaces (APIs) through which network capability features are accessible to third party application developers. An API is interface providing access to or programmability of software resources. Such as database applications or telecommunication protocol stacks. An API defines software resources in terms of objects and methods, data types and parameters that operate on these objects. Several industry efforts have emerged to develop such open APIs, including Parlay, JAIN and Open Service Access (OSA). Through APIs the service layer does not focus on individual resources, but accesses service capabilities residing in a core telecommunications network.

Several markup languages are proposed for call control, including SCML, XTML, CPL and VoiceXML. The most mature of these proposals are the Call Processing Language (CPL) and the Voice eXtensible Markup Language (VoiceXML). In this paper it is shown how services we are accustomed to use in the telephony network and in the future supported in the Internet can be implemented by CPL and VoiceXML.

II. INTELLIGENCE IN INTERNET

A large number of advanced telephony services are addressed in the standards for Intelligent Network (IN). Services are composed of features that are commonly used functions. A service feature is a specific aspect of a service and can be used in conjunction with other services/service feature as part of commercial offering.

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The IN concept was invented for voice telephony networks and unfortunately it cannot be applied to the Internet in a straightforward way..

The architectural model of Internet telephony is rather different than that of traditional telephony network, where the intelligence is centralized In the Internet all signaling and media flow over an IP-based network and the intelligence is completely distributed. In addition, the Internet model transforms the location at which many services are performed. In general, end systems are assumed to be much more intelligent than traditional phones; so many service features which traditionally had resided within the network can be moved out to the edges and thus reducing the required network support. Other service features, such as those for numbering, routing, charging, access, and restriction, can be performed by specialized servers. Most of the services and service features of ITU-T IN can be provided by IETF standards for Internet telephony, e.g. SIP (the Session Initiation Protocol).

According to IETF definition of SIP, a SIP server can perform any of IN features that can be regarded as Internet equivalents. For example, a SIP proxy can query a DNS server for a particular IP address which resembles the way IN service logic translates freephone numbers. Number blocking services also have equivalent in the Internet. A SIP server can play a role of a firewall filtering traffic coming in from and going out to the Internet. A SIP server can modify or redirect SIP requests, but the definition does not specify how the SIP server is programmed to do this. That is why a CPL is developed that tells a SIP server what to do with a call.

Although there are many features of IN that can be recognized in the Internet, there are also a few IN features that simply do not make sense in the Internet. One of them is charging features. In the Internet there is traditionally no notion of charging for a connection. So services such as freephone or calling card calls simply do not make sense in the Internet.

III. CALL PROCESSING LANGUAGE

CPL is a language to describe and control Internet telephony services. It can be implemented on either network servers or user agents. It is meant to be independent of operating system or signaling protocol; it is anticipated that it will be used with both SIP and H.323. It is based on the XML, so parsing it is easy and many parsers for it are publicly available. The structure of the language maps closely to its behavior, so an editor can understand any valid script, even ones written by hand.

Implementations of CPL are expected to take place both in Internet telephony servers and in advanced clients; both can usefully process and direct users' calls, but a mechanism will be needed to transport scripts between clients and servers.

Each CPL script installed in a server has an owner, and is always associated with the address of his owner. A CPL script concerned with the owner of the script. A subaction is any action that can be reused within other actions. Top-level actions and subactions can contain the following elements:

- Switches, which compare an element of a SIP message against a set value and can make choices based on these comparisons. A switch in CPL can make a choice on the basis of originating or destination address, time and date, priority, or even free-form text strings in the subject or body of the SIP message.
- Location modifiers change the location to which the call must be routed. CPL allows location modifiers to specify anexplicit new address, or it can order the new address to be looked up in an internal or external database.
- Signaling actions specify what to do next with the SIP message after the possible modification. The options are to proxy the SIP message, to redirect it, or to reject it. Proxy means sending the message on to the destination or another SIP server and redirect means sending it back to the originator with a modified address.
- Non signaling actions are additional actions that are not related to the SIP message. There are two nonsignaling actions defined: sending an e-mail to a specified address, or writing information in a log file.

There are striking resemblances between CPL scripts and IN service scripts. Both specify how to process incoming and outgoing calls. Both are decision tree structures without loops.

CPL is non-expressively complete language; it can not be used to create arbitrary complex scripts, thereby limiting the resources a script needs for successful execution. For example, CPL has less expressive power than IN service features. IN service features that forward or screen calls can be translated directly into CPL constructs, but there are no CPL counterparts for the more complex IN service features for charging, call queuing, user interaction, and service data management. Table1 shows how IN service features can be translated to CPL.

Typically, CPL scripts execute within the context of a user agent. A user agent is an entity that exists in SIP and H.323 networks. User agents do not exist in present day IN networks; therefore, CPL scripts are not protocol agnostic.

CPL is only activated through call events; it cannot be activated through call unrelated events such as timer expiring or registration in a mobile IP network. CPL assumes that a database can be queried to provide location information of a registered user. CPL script can interact with a special voice-response server that provides user interaction functions, but it does not define how to handle with information that can be provided by a user interaction system.

CPL is targeted more at the end user, and is based on the philosophy that the end user uploads his call-processing scripts on a SIP server. CPL therefore does not allow any actions that can be considered internal to the service provider, such as charging or modifying data in a subscriber database.

CPL still offers a few important advantages that make it interesting for use in the IN environment. It is simple and transparent, and has a well-defined syntax and semantics. But most importantly, CPL has the potential to provide a uniform consists of top-level actions and subactions. The top-level actions describe what to do with incoming and outgoing calls call-processing language for SIP-based Internet telephony and IN.

IV. VOICE EXTENSIBLE MARKUP LANGUAGE

VoiceXML has been defined as a technology that allows a user to interact with the Internet through voice-recognition technology. Using VoiceXML, the user interacts with voice browser by listening to audio output that is either pre-recorded or computer-synthesized and submitting audio input through the user's natural speaking voice or through a keypad, such as a telephone. VoiceXML can also be described as a phone markup language that can be used for voice applications that provide phone access to content and information. VoiceXML is a high-level abstraction language and this means that developers with little training can use it. VoiceXML makes it easy to rapidly create new applications and shields developers from low level programming issues. VoiceXML also executes logic: main components of a VoiceXML-based speech service include tags, forms, rules that define the content and a speech browser for interpreting and presenting audio content.

The IN service features that support user interactions can be translated to VoiceXML dialogues.

V. IN SERVICE FEATURES

IN standards describe the services into two broad categories: services which IN vendor would actually wish to provide to customers; and service features which are lower-level building blocks used to construct the services.

This section summarizes in the Table 1 the characteristics of each service feature listed in Q.1211, in the context of the Internet telephony. The "location" column indicates the place at which the service feature is supported; "end" means that the service can be provided at an end system, "proxy" means at a proxy server, and "redirect" at a redirect server.

The "CPL equivalent" column presents CPL constructs that can be used in implementing the service feature. The CPL scripts are executed on SIP servers; thus only CPL constructs that implement IN service features provided by SIP proxy server and SIP redirect server are shown.

VI. CONCLUSION

In this paper XML-based technologies for NGN service creation are presented. The capabilities of CPL are presented for implementing of service features we are accustomed to use in telephony networks. CPL can be used as call-processing language for SIP-based telephony, but it has restricted expressive power. CPL is only activated through call events and does not define how to interact with servers that provide mobility management information (e.g. provide information for facilities nearest to the user location).

Hence, CPL does not support all services that can be based on underlying network capabilities. Conjugating CPL and Voice XML might be done in order to obtain user interaction that is supported by most of IN services.

TABLE 1 MAPPING IN SERVICE FEATURES TO CPL

Service feature	*Description of the service feature*	*Location*	*CPL equivalent*
Abbreviated Dialing (ABC)	Allows the definition of abbreviate dialing	End/Proxy/ Redirect	Location modifier
Attendant (ATT)	Allows virtual private network (VPN) to access an attendant position within the VPN	End/Proxy/ Redirect	Location modifier
Authentication (AUT)	Verifies that a user is allowed to exercise certain options.	End/Proxy	Address-switch followed by reject
Authorization code (AUTZ)	Allows a VPN user to override calling restric-tions.	Proxy	Address-switch followed by reject
Automatic call back (ACB)	Allows the called user to automatically call back the last caller	End	-
Call distribution (CD)	Specifies the percentage of calls to be distribu-ted among two or more destinations.	Proxy/Redirect	No direct equivalent
Call forwarding (CF)	Forwards incoming calls to another number	Proxy/Redirect	Location modifier followed by redi-rect
Call forwarding on busy/don't answer (CFC)	Forwards particular incoming calls if the called user is busy or does not answer within a specified number of rings.	Proxy/Redirect	Location modifier followed by pro-xy
Call gapping (GAP)	Restricts the number of calls that can be routed to a particular destination.	Proxy	Switch followed by reject
Call hold with ann-ouncement (CHA)	Allows subscribers to place a call on hold and transfer the call to another location.	End	-
Call limiter (LIM)	Specifies a maximum number of calls that can be simultaneously routed to a particular destination.	End	-
Call logging (LOG)	Creates a log record for each call received at a specified number.	All	Nonsignaling action log
Call queuing (QUE)	Queues calls meeting busy or no answer within a predetermined time. An announcement is given to the calling party.	End/Proxy	Proxy to queuing server
Call transfer (TRA)	Allows a user to place a call on hold and trans-fer the call to another location.	End	-
Call waiting (CW)	Notifies the called party, when on a call, that another party is trying to reach him.	End	-
Closed user group (CUG)	Allows a user to be a member of a set of VPN users who are normally authorized to make and receive calls only within the group.	End/Proxy	Address-switch followed by reject
Consultation calling (COC)	Allows the user to place a call on hold, in order to initiate a new call for consultation.	End	-
Customer profile management (CPM)	Allows users to manage their customer profiles in real time.	End/Proxy/ Redirect	Location modifier with lookup
Customized recor-ded announcement (CRA)	Allows a call to be completed to a (customized) terminating announcement instead of a subscriber line.	End	-
Customized ringing (CRG)	Allocates a distinctive ringing to a list of calling parties.	End	-
Destination user prompting (DUP)	Prompts called parties with a specific announce-ment.	End	-
Follow-me diversion (FMD)	Routes incoming calls to a new location.	End	-
Mass calling (MAS)	Allows the processing of huge numbers of inco-ming calls.	Proxy	No direct equivalent
Meet-me conference (MMC)	Reserves conference resources for making a multiparty call, indicating the date, time, and duration of the conference.	Other (e.g. using a confe-rence bridge)	-

TABLE 1 MAPPING IN SERVICE FEATURES TO CPL (CONTINUETION)

Service feature	*Description of the service feature*	*Location*	*CPL equivalent*
Multiway calling (MWC)	Establishes multiple, simultaneous calls among parties.	End	-
Off net access (OFA)	Allows VPN users to access the VPN from any non-VPN exchange.	All	Address switch followed by reject
Off net calling (ONC)	Allows VPN users to call outside the VPN.	Proxy	Address switch followed by reject
One number (ONE)	Permits two or more terminating lines, also at different locations, to have a single telephone number.	Proxy/Redirect	Location modifier followed by proxy/Location modifier followed by redirect
Origin-dependant routing (ODR)	Enables users to accept or reject calls and, if they accept a call, route the call according to the calling party's geographical location.	Proxy/Redirect	Address-switch
Originating call screening (OCS)	Allows users to bar calls from certain areas.	End/Proxy	Address-switch followed by reject
Originating user prompter (OUP)	Allows users to provide an announcement requesting calling parties to enter a digit or series of digits.	All	Proxy calls to a VoiceXML server
Personal numbering (PN)	Supports a UPT number that uniquely identifies each UPT user and is used by calling parties to reach the UPT user.	Proxy/Redirect	Location modifier followed by proxy
Premium charging (PRMC)	Allows for the payback of part of the cost of a call to a value-added service provider.	-	-
Private numbering plane (PNP)	Allows subscribers to maintain a numbering plan within their private network that is separate from the public network.	End/Proxy/ Redirect	Location modifier followed by redirect/reject
Reverse charging (REVC)	Allows subscribers to be charged for the entire cost of calls they receive.	-	No direct equivalent
Split charging (SPLC)	Allows calling and called parties to share the cost of a call.	-	No direct equivalent
Terminating call screening (TCS)	Screens calls based on the terminating number called.	End/Proxy	Address-switch followed by reject
Time-dependent routing (TDR)	Enables users to accept or reject calls and, if they accept a call, route the call according to time and date.	Proxy/Redirect	Time-switch

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