

A critical survey of protocols proposed by the IETF as enablers for customer interaction in an electronic customer relationship management system - Part II - Protocol Suite

1.0 Introduction

Today most e-businesses implement eCRM by using human-computer interaction and thereby reduce the need for human intermediaries. CRM entails four phases, 1) customer interaction, 2) data analysis and mining, 3) knowledge discovery, and 4) market planning [1]. Electronic customer interaction involves encouraging the customer to spend time electronically in order to obtain sufficient information regarding the customer's needs, preferences and requirements. This information is analyzed using a process such as data mining to extract knowledge about customer values. Customer values are, in turn, used as the guiding principle in the market-planning phase to customize and personalize the services/goods/sales offered by the enterprise. An integrated approach to eCRM is important because it can be effectively used to analyze information for continuous, online and real-time learning of customer values. The three stakeholders typically identified in an eCRM framework are the customer, the business enterprise and the provider of technology [1]. Part I – eCRM metrics of this paper, analysed the requirements of an effective eCRM customer interaction system and proposed a set of metrics from the perspective of the three stakeholders [2]. It is shown that the eCRM customer interaction metrics can be categorized as mutually exhaustive, mutually exclusive and non-overlapping. Good eCRM system design should aim at maximizing the mutually exhaustive requirements, minimizing the effects of mutually exclusive requirements and optimizing the overlapping requirement within cost objectives to enable effective electronic customer interaction.

Section 2 of this paper outlines the core objectives of an effective eCRM customer interaction system. Also, this section critically evaluates the various protocols as proposed by the IETF that may support one or more of these core objectives. Section 3 outlines an IETF based protocol suite that may be used for enabling real-time multi-media communication and non-real-time unified messaging in an eCRM system. It is shown that this protocol suite meets most of the core objectives of an effective customer interaction eCRM system.

2.0 Suitability of protocols proposed by IETF in meeting eCRM objectives

We will now evaluate the various protocols as proposed by the IETF that may support one or more of the following core objectives of an effective eCRM customer interaction system viz.,

1. Accessibility- a protocol that initiates and maintains multi-media sessions over IP and public switched television network (PSTN)
2. Responsiveness – protocol support for specifying quality of service (QoS), delay and latency, and bandwidth parameters of the communication channel,
3. Scalability – i.e., support for multiple simultaneous users,
4. Security and privacy – protocol provision for authentication, authorization and encryption of message,
5. Integrate Internet based services with legacy PSTN services,
6. Support in the protocol for seamless adoption from wireline to wireless media,
7. Integrate video, voice, fax, e-mail and instant messaging systems into one unified eCRM messaging system,

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Abstract

A modern enterprise needs to interact with customers anytime, anyhow and anywhere to be successful in the global marketplace. This level of customer interaction has become possible due to the advancements in network infrastructure and the simultaneous development of voice and multimedia protocols for seamless transport of information. Instant and unified messaging extends this capability to enable customer touch point integration. Part I of this paper provided a critical analysis of the metrics for customer interaction in an electronic customer relationship management (eCRM) system from customer, business and technology viewpoints. Based on this analysis, this paper will review the features and the services offered by some of the protocols as proposed by the Internet Engineering Task Force (IETF) with respect to their effectiveness in enabling effective customer interaction, and derive an IETF based protocol suite that may be used in an eCRM system.

Sommaire

Une entreprise moderne a besoin d'échanger avec ses clients peu importe le moment, la façon et l'endroit si elle veut obtenir du succès dans le marché économique mondial. Ce niveau d'interaction avec le client est devenu possible grâce au progrès réalisé dans l'architecture des réseaux et grâce au développement simultané de protocoles vocal et multimédia. Ces protocoles permettent un transport d'information sans faille. La messagerie instantanée et unifiée accroît l'interaction avec le client et permet l'intégration par points de contact. La première partie de cet article a proposé une analyse des métriques de l'interaction des consommateurs dans un système de gestion électronique des relations avec les consommateurs, dénoté eCRM, des points de vue des consommateurs, de l'entreprise et de la technologie. En se basant sur cette analyse, le présent article révisera les caractéristiques et les services offerts par certains des protocoles proposés par l'IETF relativement à leur efficacité à interagir avec les consommateurs. De plus, l'article présentera une suite de protocoles qui est basée sur l'IETF et qui peut être utilisée dans un système de gestion eCRM.

8. Support automatic call routing,
9. Support number portability, and
10. Service extensibility through modular construction of protocol. Support for interoperability of the various protocols that enable customer interaction in an eCRM system.

It should be noted that objectives 5-9 inclusive, will help implement customer touch point integration i.e., the customer is able to communicate with the enterprise through any communication channel, any device and any service.

The sections below discuss IETF proposed protocols that can be used for real-time multi-media communication session initiation, real-time instant messaging, unified internet mail messaging and, service and service provider portability.

2.1 Session Initiation Protocol

Session initiation protocol (SIP) is a user-to-user protocol developed by IETF [3] that uses text-based signalling for initiating, modifying, maintaining and terminating interactive communication sessions such as internet telephone calls, and multimedia conferences between one or more users. It is used to establish and maintain session level information such as bandwidth and media type (whether the message is a voice, video, fax, instant message or a combination of two or more of the above). SIP can be used to specify media characteristics, QoS parameters, security considerations, and to protect the identity and privacy of the caller. Specifying media characteristics such as bandwidth and acceptable latency is important in ensuring (near) real-time interaction with the customer. QoS parameters such as low packet delay and low packet loss help an enterprise reserve network resources before establishing a session. QoS parameters also determine the quality of the multimedia communication. The authentication feature of SIP provides a mechanism for access control, so that a SIP client may reject unauthorized or undesirable call attempts. Such security features are essential to prevent threats such as stalking, spamming and spoofing. SIP provides support for end-to-end encryption of the message, thus protecting the contents of the call from snoopers and stalkers to protect customer privacy. Further, SIP enhances the caller's privacy by giving the caller the right to not disclose the caller's identity, name or IP address, if so desired. SIP enables user devices to exchange information that enables customized services e.g., user location which finds the callee; user availability which indicates when a callee may become free to accept a call; call handling including call transfer, call forwarding and call termination; call management which screens incoming and outgoing calls; and call hold. Advanced services supported by SIP include support for 3-way conferences where the callee may invite a third-party into the call; single-line extension in which a call will ring several extensions in sequence; and customer specific call routing in which call forward and user location are used repeatedly to enable personal mobility systems. SIP services are integrated with legacy PSTN services by enabling the placement of calls between SIP devices, and PSTN devices by signalling call information and then routing the call through a SIP network gateway to the appropriate PSTN gateway and vice versa. Thus, it is seen that SIP meets the core eCRM objectives of enabling accessibility, responsiveness, secure communication, customer privacy, integrating Internet services with PSTN services, and automatic call routing for real time multi-media communications.

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2.2 Protocols for real time instant messaging

Typically, Instant Messaging (IM) consists of short text messages that are exchanged in real time over the Internet between the subscribers of an instant message service. IM gives customers, an easy way to communicate with the enterprise and is highly suitable for sending short text messages. Thus, IM is an important and scalable means of communication between retail customers and service personnel in e-business. If IM is extended to include real-time voice and video messages as well, then it can be used as a personal integrated communication touch point and can be used effectively by the enterprise to foster one-to-one customer-enterprise relationships. Currently, the developers of popular commercially available IM programs use proprietary protocols that may not be compatible with each other and therefore may not interoperate with one another. The business potential of instant messaging can be better realized, if IM applications use protocols that are compliant with [4, 5] as proposed by the IETF. In the instant message presence protocol (IMPP), as proposed by IETF, the user connects to a central presence service that verifies the user's identity and registers the user as being online. When another user registers and connects, the new user will know about the users already logged on because the presence service maintains information on who is online. When a user has a message to send, the sender delivers it to the instant message service, which then delivers the instant message to the recipient's inbox. An IM session compliant with RFC2778 and RFC2779 can be setup-using SIP. Since, IM allows a subscriber to track when another subscriber logs on and logs off (the presence information of other subscribers), security considerations are a

very important issue. RFC2778 and RFC2779 specify that presence information be distributed only to authenticated subscribers who are authorized to view it. This prevents stalking. IM protocols developed under IETF guidelines use end-to-end encryption of the message to prevent unauthorized access, insertion, modification and deletion of messages. Spoofing, i.e., replaying a genuine message is prevented using date stamps. IETF based IM protocols are required to be accessible in low-bandwidth, high latency environments to enable both wired and wireless messaging.

2.3 Protocols for non-real-time unified messaging

An enterprise may wish to receive all messages (voice messages, fax, text) in one mailbox and access all this information from any device (PC, terminal, mobile, hand held) of its choice. Thus, the characteristics of a good unified messaging system are device mobility, service mobility and terminal mobility [6]. Device mobility refers to the ability to redirect messages across various devices such as fax, pagers, phone, computer, handheld devices. Service mobility refers to ability to access various types of services such as e-mail, voice mail, and PSTN services from any user end-point i.e., all user end-points see the same raft of services. Terminal mobility refers to the ability of a user/endpoint to physically move from one physical location to another while still having the ability to redirect messages across devices and being able to access the same set of services [7]. Typically, a unified messaging system works by, using any standards-based email client-application on which subscribers can receive text e-mail, fax and voice messages in non real time; then play, view, store, delete, forward or share the message with others using a standard desktop client. A unified messaging system automatically converts voice and fax messages to digital formats such as WAV for voice, and tag image file format (TIFF-F) files for faxes. These data files are then attached to emails and delivered over the Internet, or virtually any corporate LAN, to the user's email system.

The Voice Profile Internet Mail (VPIM) workgroup of the IETF [7] intends to make VPIM the voice component of a unified messaging protocol suite. The user records a voice message then enters the VPIM address where the message should be delivered. VPIM uses 32K adaptive differential pulse code modulation (ADPCM) to encode the voice. Similarly, an image or a fax may be encoded in the TIFF-F format. VPIM converts the message to a Multipurpose Internet Mail Extension (MIME) or a simple mail transfer protocol (SMTP) or extended simple mail transfer protocol (ESMTP) attachment depending on the implementation. Since, VPIM uses open Internet standards (MIME or SMTP), it offers interoperability with other voice mail systems. The VPIM system locates the voice mailbox address for the intended recipient, and delivers the message over the Internet. The message can be retrieved like traditional e-mail using a POP or IMAP server. VPIMv2, therefore, leverages existing infrastructure including the enterprise intranet and legacy voice mail systems. These features will greatly enhance the appeal of VPIMv2 protocol to enterprises that would like to keep in touch with their customer pool (many of whom have Internet e-mail capable computers). VPIM supports multi-part voice messages that may be composed of more than one audio part. Similarly, multi-part mixed messages may be composed of audio, image, multi-media or text. VPIM systems may be either transport-conformant or content-conformant. Transport-conformant VPIM systems merely store and forward voice messages to a repository, while content-conformant systems have additional functionality that enables such systems to generate and interpret VPIM messages. On receipt of a message, the receiving servers send delivery status notification messages that indicate delivery, non-delivery or delay in delivering the message to the recipient. These features make VPIM attractive to an enterprise, which may want to deliver the same customized audio and fax messages to groups of customers informing them of the latest development in goods and services offered by the enterprise. In particular, this feature may be exploited by the enterprise for promotional purposes in targeting a subset of the customer pool that share common characteristics and interests. Thus, VPIM enables customization in customer interaction, in addition to offering unified messaging. However, spoofing is a concern in VPIM systems as the protocol does not provide a means of authenticating the sender. Like an Internet mail system spamming is a problem with VPIM. VPIM provides rudimentary privacy protection mechanisms that prevent a sender's voice message from being forwarded to anyone other than the intended recipient, if so desired by the sender.

It is imperative to integrate fax with Internet mail for a comprehensive unified messaging system, as it is appealing to the enterprise in terms of cost savings and opportunities for enhanced customer touch point integration. While, PSTN fax communication is real-time and session based, the Internet Fax (IFAX) protocol (as proposed by the IETF) is Internet messaging based and is currently not in real time. Internet Fax can operate in any one of four modes: simple (SIFAX) [8], extended (EIFAX), full (full mode fax profile for internet messaging FFPIM) and terminal (TMIFAX). Each mode of IFAX offers different levels of capabilities with respect to interoperability with legacy PSTN based fax machines and other IFAX devices, capacity to exchange information regarding their capability, negotiate connection parameters, fax processing status and confirmation of delivery. Capability negotiation includes sending information such as paper size, colour, image coding, image file structure, so that fax messages generated by transmitting end do not exceed the recipient's capabilities. Unlike legacy PSTN based fax devices, IFAX devices do not provide means for authenticating the sender. Hence spoofing and spamming are real concerns that the enterprise may address by providing virtual private networks, encrypted tunnels or transport layer security mechanisms.

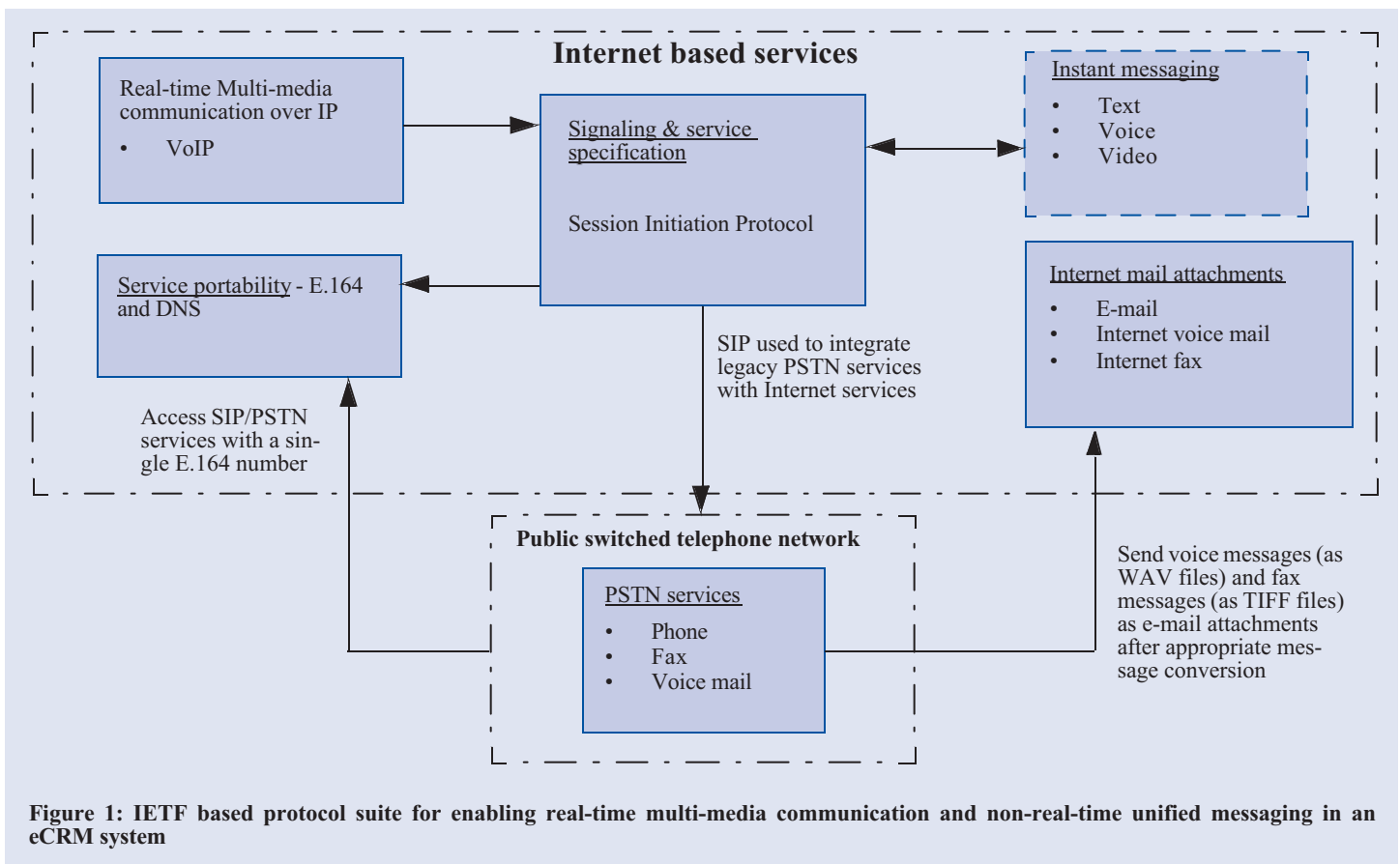
Thus, VPIM and IFAX support the core eCRM objective of integrating non real time video, voice, fax, and e-mail into one unified eCRM messaging system. Both VPIM and IFAX are scalable and can support multiple users simultaneously. Both protocols can also interoperate with legacy PSTN based voice mail and fax services

2.4 Protocols that support Number Portability

Number portability is a telecommunications network feature that refers to portability of dialled geographic numbers i.e., end users retain their dialled telephone numbers (E.164 number) irrespective of the subscribed services used, change in service provider and change in location. Accordingly, there are three different types of number portability viz., service portability, service provider portability and location portability [9]. Service provider portability is the ability of end users to retain, at the same location, existing dialled telephone numbers as they change from one service provider to another. Location portability is the ability of end users to retain the same dialled telephone number, even as they relocate from one geographic area to another. Service portability is the ability to retain the same dialled telephone number with the same

service provider as the subscriber changes service, e.g., from PSTN to ISDN.

The IETF is addressing the issue of service and service provider portability with the formulation of RFC2916 [9]. The service portability functionality is desirable as an enterprise/user typically subscribes to several services (each with its own name space) and is identified by a user identity in each name space. It would be advantageous for the enterprise, if a unique ID i.e., a single E.164 number can identify it. The naming service then maps the users unique ID to one of several user identities depending on the service desired. This feature can be used to enable customer touch point integration. Here, there are two scenarios, (i) an enterprise may wish to access any of several services from any user terminal in the enterprise with a single identified E.164 number (ii) a single E.164 number for customers to access several services to which they may subscribe, with seamless linking of services. The customer knows which services to use (SIP, e-mail, PSTN, fax) in order to reach the enterprise and vice versa. However, there are two requirements for such a naming service, viz., it is globally distributed and scalable. This scheme maps an E.164 number into a domain name system (DNS) entry. The DNS is a globally distributed mapping service that has an Internet based hierarchical directory structure and is therefore scalable. Also, DNS supports fast connectionless queries. The domain "e164.arpa" has been created to facilitate storage of E.164 numbers. The E.164 number is converted into a dotted string notation, which is then appended with the string ".e164.arpa" to obtain the DNS entry. Typically, each DNS entry is associated with a resource record that points to the set of uniform resource indicators (URI) associated with that DNS entry. This feature is now used to contain the list of services including phone numbers that exist for the specific DNS entry. The order of the list specifies the preference order of the service/phone numbers that can be used to contact the enterprise. New service records can be added to and old records deleted. Changes in service or service provider are now transparent to the customer and to the enterprise. The contents of the service registrar may change over time as more services evolve and are subscribed to by the enterprise. However, malicious/wrong entries can cause an incorrect URI/service/phone number to be associated with a given E.164 number. Intended or unintended removal of the URI/service/phone number from the resource record will cause a denial of service, as the enterprise is now not reachable through its E.164 number. Apart from these security issues, one must also consider that the



DNS is a public infrastructure and the E.164 to DNS service might overload it.

3.0 Protocol suite to enable customer interaction in eCRM using IETF protocols

Figure 1, shows the various protocols discussed in this paper and how they can be used to enable customer touch point integration. It is seen that SIP can be used to set up real-time multi-media communication sessions and also set up real-time text based instant messaging. Legacy real-time PSTN services such as phone calls, and fax transmissions to legacy PSTN devices can also be accessed from corporate internets through a SIP network gateway which would then route the calls to the appropriate PSTN gateway and vice-versa. When several services are subscribed to by the enterprise, then the enterprise may set up a single E.164 number to handle any of these services from any user terminal in the enterprise. Also a customer wishing to communicate with the enterprise on any communication channel may do so using this unique E.164 number. While the enterprise may subscribe to, and/or delete any of these services or change service providers, such changes are transparent to the customer. The enterprise can have access to a comprehensive unified messaging system that combines voice mail and images from, Internet based and PSTN systems, by transforming voice and images into appropriate file formats before transmitting them through the Internet e-mail system for non-real-time processing by customer service representatives at the enterprise or for providing information to customers.

4.0 Conclusion

This paper analyzed the signalling and service specification capability of SIP in initiating real-time multi-media communication. Specifically, SIP can be used to specify media characteristics, QoS parameters, security considerations, and protect the identity and privacy of the caller if so desired. To enable mobile eCRM, the capabilities of SIP may have to be extended to take into account the signalling, security, and other technical requirements in wireless systems. Currently, IM is an important and scalable means of real-time instant text communication between retail customers and service personnel in e-business. If IM is extended to include real-time voice and video messages as well, then an IM system can be an effective personal integrated communication touch point. The IETF IM workgroup is currently looking at ways to extend IM to wireless instant messaging platforms. VPIM and IFAX support the core eCRM objective of integrating non-real-time voice mail, fax, and e-mail into one unified eCRM messaging system. While both VPIM and IFAX are scalable, can support multiple users simultaneously and interoperate with legacy PSTN based voice mail and fax services, security and privacy issues are not adequately addressed in these protocols. Number portability is an important component of customer touch point integration as an enterprise can now be identified by a through a unique ID consisting of an E.164 number, irrespective of the services (such as PSTN or ISP) subscribed to and any change in service provider. The IETF proposed protocols consisting of SIP, IM, VPIM, IFAX and number portability together constitute a protocol suite that meet most of the core objectives of an effective customer interaction eCRM system. The IETF protocol suite considered in this paper can interoperate, with SIP being used to establish real-time multi-media and IM sessions and for accessing legacy PSTN services. The IETF protocol suite considered are continuing to evolve and are working towards extending the features of these protocols to enable mobile communication. In the near future, it may be possible to adapt this protocol suite to enable an enterprise to make the transition from wireline commerce to mobile-commerce.

5.0 References

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6.0 List of abbreviations used in the paper

DNS	- Domain Name System
eCRM	- Electronic Customer Relationship Management
IETF	- Internet Engineering Task Force
IFAX	- Internet Fax
IM	- Instant Messaging
IMPP	- Instant Messaging Presence Protocol
ISP	- Internet Service Provider
MIME	- Multipurpose Internet Mail Extension
PSTN	- Public Switched Telephone Network
QoS	- Quality Of Service
RFC	- Request For Comment
SIP	- Session Initiation Protocol
SMTP	- Simple Mail Transfer Protocol
TIFF	- Tag Image File Format
URI	- Uniform Resource Indicator
VPIM	- Voice Profile Internet Mail

This is the second of a two-part paper submission by the author. The first half of the article appeared in issue no.40 of the **IEEE Canadian Review** in the spring of 2002.

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