The Potential of Consolidating SIP and XMPP Based Communication for Telecommunication Carriers

Sebastian Schumann, Michael Maruschke, and Eugen Mikoczy

Slovak University of Technology (STU),
Ilkovicova 3, 81219 Bratislava, Slovak Republic
{schumann,mikoczy}@ktl.elf.stuba.sk
Deutsche Telekom AG, Hochschule für Telekommunikation Leipzig (HfTL),
Gustav-Freytag-Str. 43-45, 04277 Leipzig, Germany
maruschke@hft-leipzig.de

Abstract. This paper describes an approach for combining Session Initiation Protocol (SIP) based voice communication with Extensible Messaging and Presence Protocol (XMPP) based presence enhancements. The actual role of SIP and XMPP in the Internet Protocol (IP) based communication was analyzed, especially from the telecommunication carrier (Telco) point of view.

The proposed infrastructure extends a typical SIP infrastructure with XMPP for presence status integration. XMPP will be used as instant messaging and presence (IM/P) service infrastructure, the presence information will be extended with SIP phone status information of telecommunication endpoints. A first prototype has been developed and tested successfully.

Keywords: SIP, XMPP, Presence, Federation, Collaboration, Telco, Web 2.0, NGN.

1 Introduction

This paper discusses various approaches to overcome the current lack of sophisticated and field-tested presence-enabled communication infrastructures in Telco environments. The target is to provide a service environment for both multimedia sessions and IM/P.

The use of instant messaging has recently become popular not only in private, but also in the business segment. When people communicate through instant messaging systems, one major service enabler must be included: Presence. Without knowing the current state of the other endpoint (i.e., its presence information), the *instant* message would be nothing more than an ordinary E-mail. Knowing that the other party is actually online enables small ad-hoc text-based dialogs and other instant services.

So far, instant messaging and presence infrastructures are usually provided by Internet Service Providers (ISP). Presence states are exchanged between clients

T. Magedanz et al. (Eds.): TridentCom 2010, LNICST 46, pp. 713–726, 2011.

that users have running on their computers or notebooks. If a Telco would like to implement the presence service in its role as a service provider, its first major function would be enabling instant messaging. In order to not just be yet another IM/P provider, Telco's have to think of ways to provide a unique presence service. They can do this by enriching the presence states of their users with telephone state information. This makes their presence system unique amongst providers of the current ISP world[1].

2 Status Quo

Within this paper, the "telecommunication world" will be used as a summary of Telco's and smaller providers that have specialized in offering their customers VoIP based telephony, almost entirely using SIP for session signaling. To date, this telecommunication world itself has not been pushing forward instant messaging and presence.

In contrast to facts mentioned above, the internet community has been using instant messaging and presence for many years. Together, they became part of many Web 2.0 web sites to offer real-time communication to their users.

The "internet world" will be used as a synonym for companies that host web pages or web services and offer communication services – mainly through instant messages.

Some instant messaging clients may already provide some form of voice communication. The protocols used for those clients are either proprietary or they use another important protocol specified by the Internet Engineering Task Force (IETF) exactly for this purpose: XMPP. The use of instant messages and presence is very mature and well implemented already. However, the voice transmission has so far either been proprietary, or is not at a highly developed stage for voice service providers.

This paper summarizes both service protocols mentioned above – SIP and XMPP – and proposes a federated architecture for current Telco's to extend their multimedia infrastructure with mature instant messaging and presence.

While some existing research approaches (e.g., [1], [2]) address the interworking function for converged IP messaging services such as E-mail, Short Messaging Service (SMS) or XMPP instant messaging, the primarily focus of this paper is the integration of SIP telephony states in an XMPP framework and building a converged SIP/XMPP based communication infrastructure. It is not desired to have users from the SIP domain communicating with users from the XMPP domain through gateways. Those gateways have already been realized (e.g., [3]) and they are working fine already. In the context of this paper however, it is assumed instead that *a single user* is present in *both domains* (i.e., in the SIP domain for voice communication and in the XMPP domain for messaging and presence

---

[1] It is well known that Skype is an ISP client/system that combines presence and telephony. In this paper, the term "Telco" refers to fixed-line telecommunication providers with a Voice over IP (VoIP) infrastructure. Customers use the Telco service in the classic way through hardphones.

exchange). The target of our research is a mature communication infrastructure that can be implemented *today* in the telecommunication world.

3 Protocols and Their Context

3.1 SIP and Presence

SIP is a well-established protocol for the telephony over IP based networks, which has been standardized by the IETF. Many Telco's have already implemented it in their network in order to provide VoIP telephony. The market knows a variety of mature server implementations (proprietary and open-source) end devices and many softphones. [4] defines SIP as "an application-layer control protocol for creating, modifying, and terminating sessions with one or more participants".

The basic SIP methods only enable registration and session establishment. Several extensions exist, which enable the protocol to handle presence. The following possibilities are required for using SIP as presence protocol:

- Means for subscribing to presence information of users
- Means for being notified with presence information of users
- Means for publishing own presence information
- Means for managing watcher authorization

With the CPP (Common Profile for Presence) specified in [5], an abstract model for delivering presence information is available. This model describes the messages between the client presence application and the presence management server. SIP based presence uses the general event notification framework from [6], which defines `SUBSCRIBE` and `NOTIFY`. The particular SIP presence functions for subscriptions and notifications are defined in [7]. This extension of the event notification framework describes basic means to retrieve presence information.

[7] mentions three possible methods showing how the presence service can obtain presence states:

- Analyzing the SIP `REGISTER` messages[2]
- Co-location of the Presence User Agent (PUA) with the presence server
- The client uploads the presence information to the presence server

For the third method, [8] extends SIP with another method: SIP `PUBLISH`. This completes the SIP methods necessary for both sending and retrieving presence information.

An event template package for management of watchers (acc. [9]), and even more importantly for the handling of authorization with the Extensible Markup Language (XML) Configuration Access Protocol (XCAP) (acc. [10,11]), complete the complex framework for SIP based presence.

[2] Here the gathering of advanced important information is already mentioned: The indication of instant messaging support by the client (e.g., SIP `Allow` header for `MESSAGE`). As in our concept, instant messaging as a very important service to be enabled by presence and should be available at the endpoint.

For quite some time, SIP servers have implemented SIP extensions mentioned to handle the exchange of presence information between users. A few of these have even started to integrate or offer interfaces for some XML Document Management Servers (XDMS). However, currently there are only a few – if any – SIP clients that support SIP and XCAP at a satisfactory level. Interoperability between clients is usually not provided yet. One reason for this is that the IETF has defined SIP and XCAP, but does not consider an overall view regarding a presence architecture with all its functions and tasks.

An integrated SIP/XCAP based infrastructure that provides a framework to deliver the presence service with major required functionalities as mentioned within this paper has been standardized by the Open Mobile Alliance (OMA) in [12]. The approach will also be used in the IP Multimedia Subsystem (IMS). For the extensions of a currently SIP based architecture, the extension with presence requires far more than only adding a presence server and an XDMS. The internal interfaces (some of which are even out of scope of the current standard) have to be implemented, services such as resource lists require a tight integration of resource list server, presence server and XDMS and finally clients have to fully support the standards to guarantee a proper handling and satisfying all expectations the users have on privacy.

In summary, Telco's that want to implement the presence services now, and also remain end device vendor independent and close to standard compliant in their implementations, will immediately be confronted with many tasks, if they decide building the service on SIP. Both the scope and complexity of the problems have been shown in this section.

3.2 XMPP and Presence with Messaging

XMPP is standardized by the IETF and mainly designed "for the purpose of building instant messaging and presence applications" (from [13], [14]). While SIP is similar to the Hypertext Transport Protocol (HTTP)[3], XMPP is based on XML.

Knowing that SIP was extended with IM/P, it is not surprising that XMPP has been extended towards session establishment. The extension is called Jingle, standardized in [15], [16]. It is currently implemented in some clients, but not considerable for any larger Telco environment[4]. Despite this fact, all IM/P related advanced functions like

- Authorization handling
- Group chat/multi-user chat
- Server-side storage of the buddy list
- Remote control of clients

are really well implemented in XMPP.

[3] It has for example header/body or header fields on single lines with value and attribute separated by colon.

[4] No hardware endpoints exist, SIP is already well established within the Telco network.

The server as well as client software is very mature and many large companies[5] are successfully using XMPP as their choice for offering IM/P based services or products. Implementing XMPP would leave many customers the free choice in their client and the ability to re-use their currently installed application.

For a Telco operators network, XMPP would be the number one choice at this time to

- provide instant messaging and presence service
- offer a variety of existing clients (the users do not have to get used to yet another client)
- go for standardized protocols, also to decrease the internal implementation efforts necessary.

As discussed within this section, there is no first choice in only one protocol for telephony and presence management. At the moment, the best option would be to keep SIP for session signaling and extend the network with XMPP for instant messaging. Users could use multi-protocol clients to access both services and use the benefits from both. This would however not be without problems.

From the user point of view, multi-protocol clients are currently used because their contacts use different instant messaging systems that do not share a common standard. The multi-protocol clients help the users to unite their contacts in one program. From the operator point of view, this merging at the client-side cannot be controlled. If the operator provides SIP based presence for telephony status information and an XMPP network for their customers to immediately start to chat[6], it is up to the customer and their software to merge the states of their contacts.

An alternative approach is to provide a single infrastructure and to federate the presence states server-side. By federating the existing SIP infrastructure (that exists and inherits all mentioned benefits for multimedia communication) with an XMPP environment (to take advantage of the mature XMPP implementations and large internet community using it), the operator can easily extend its current infrastructure with IM/P. The customers would only need an XMPP client for desktop messaging in addition to the SIP softphones or hardphones already in use. The XMPP states would include the presence states of their VoIP communication devices, hence providing extended presence information that would help the Telco to separate from the current ISP presence.

4 Consolidated SIP and XMPP Architecture

4.1 Focus

The main concept that this paper proposes can be derived by taking all considerations and observations from the previous sections into account. The proposal

---

[5] Cisco, Google, HP or Sun.

[6] Providing an XMPP access allows the customers also to use their account for integration with the large XMPP clouds in the existing internet world.

assumes that a Telco wants to extend its SIP based telephony infrastructure with presence. The Telco wants to offer a mature infrastructure to its customers and allow them an easy and fast use of its new service. The customers should not necessarily require any new software. They should, however, benefit from call state information the Telco can provide.

The operator has generally two choices, if standardized protocols[7] are mandatory:

- SIP based IM/P
- XMPP based IM/P

The main advantage of SIP based IM/P is that only one protocol will be used within the Telco's signaling network. In the end, it follows existing next-generation network (OMA/IMS) approaches using only SIP for all parts of communication. As mentioned in section 3.1, the OMA concept provides further concepts of inter-domain and inter-service communication. The SIP based concept is however still considered immature (regarding exact specifications that cover *all* required parts for presence) and not yet sufficiently interoperable (regarding available clients). It still takes time for vendors to catch up.

As the target presence service should be mature as well as a solution for offering the service immediately, the paper proposes to base it on XMPP. A positive side-effect is that the solution is *at this very moment* already interoperable with existing IM/P infrastructures and well established in client desktop messaging.

4.2 Concept

The typical user that is discussed within this paper uses primarily a SIP hardware phone for voice communication. A small number of users is also using software based SIP phones. The user is most likely using XMPP instant messaging clients to connect to an IM/P service infrastructure.

The concept will be based on a typical communications set-up as depicted in Fig. 1. The existing SIP communication infrastructure contains all components in a simplified way required to provide the customer the basic telephony service (e.g., proxy, registrar, media resources, gateway). The existing IM/P service infrastructure consists of XMPP clients and servers. Both infrastructures are currently isolated.

To extend this SIP infrastructure depicted in Fig. 1, a transparent proxy has been placed between the signaling endpoints (User Equipment (UE), SIP server). This proxy is aware of all signaling traffic of the UE. From the session related exchanged SIP messages, a co-located PUA can generate SIP `PUBLISH` messages. The generation of publications makes general sense for the following events:

- A SIP UE is registering (i.e., the user can be reached for voice communication)

[7] Both the SIP extensions for presence and XMPP, follow the general model for presence and instant messaging standardized by the IETF (acc. [17], [18]).

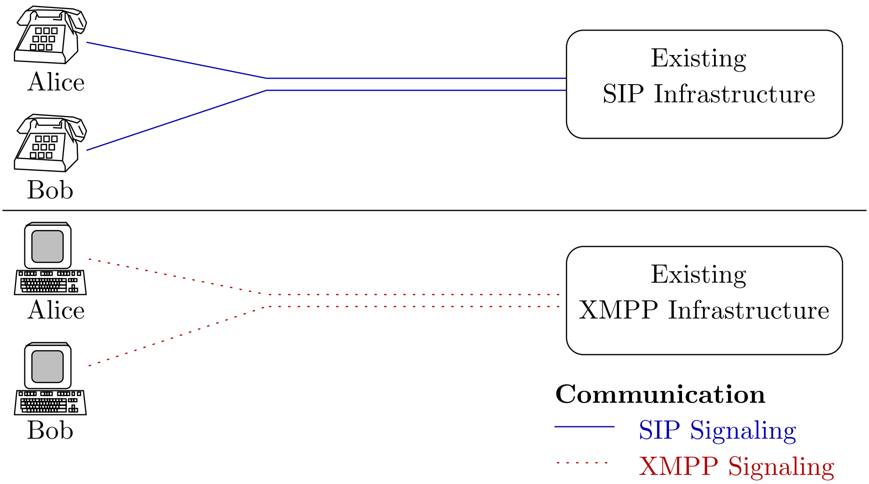

Fig. 1. Current SIP and XMPP isolated communication infrastructures

- A SIP UE is establishing a dialog (in-bound or out-bound call, i.e., the user is busy)
- Further events accessible[8]

In order to represent the SIP UE state in a user's XMPP presence state, it does not make too much sense to show that it is registered. In a typical environment, hardphones are registered all the time. This presence information is not helpful, as it is not conductive to a potential improvement in communication. The most relevant event is call state information. The off-hook event (knowing that a user is in a telephone call) is a typical busy state. The user is busy and does not want to instantly respond to messages.

The generated SIP `PUBLISH` events for a UE with an open dialog will be transferred into the XMPP world as presence state. This process will be explained in detail in the following subsections. For integrating the published information into XMPP, a gateway is required. This gateway will be referred to as XMPP Publisher in the following.

The tasks of this XMPP Publisher are:

- Creating XMPP `<presence>` stanzas from received SIP `PUBLISH` messages
- Map the SIP user identification to an XMPP user identification
- Provide authentication for that user towards his XMPP server and publish his presence state

[8] There are no means in SIP to signal certain changes like setting Do Not Disturb (DND) at a hardware endpoint. However, if the user is performing those changes with telephony call "star codes", they are signalled to the SIP system and the proxy could evaluate this to generate events. As this is usually not the only way to perform those settings, generating presence events from them is not considered within this paper.

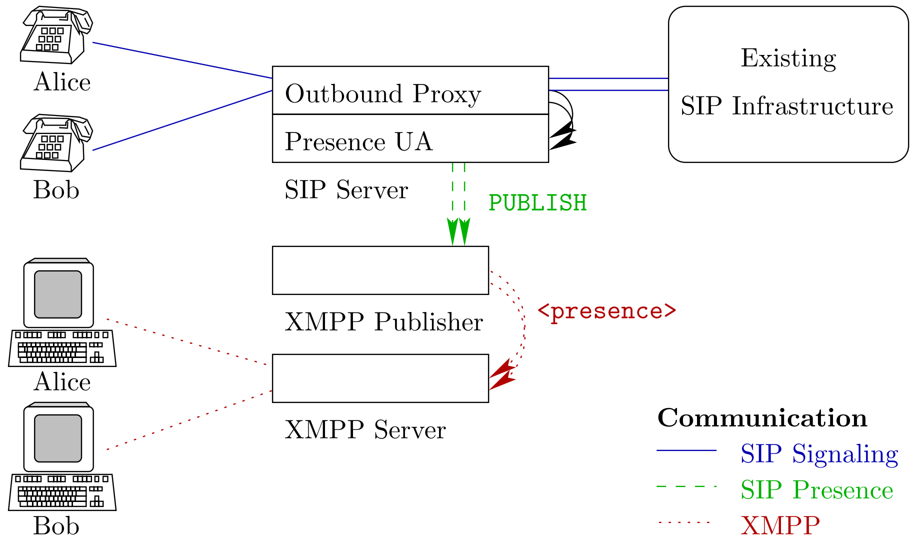

Fig. 2. Implemented federated SIP/XMPP communications infrastructure

This paper will focus on an XMPP based presence system as discussed. The proposed design can be either simplified (e.g., by co-locating the XMPP Publisher with the SIP proxy and skipping the PUA) or made more complex (e.g., in forwarding SIP `PUBLISH` messages from clients to the XMPP Publisher).

Fig. 2 shows the implemented extension of the previously shown isolated SIP and XMPP infrastructures as a prototype. It contains all components that have been introduced within this subsection. The proposal of this paper is based on the assumption that the SIP and XMPP server infrastructures are both managed by the Telco.

At this point, all SIP related presence information (call state information transformed to presence events and user generated presence events) is available within the XMPP network. The XMPP clients, which are used by the user for the exchange of messages in the "internet world", are enhanced with the information from the "telecommunication world" and change their states accordingly.

4.3 XMPP Publishing

Publishing the state for XMPP only makes sense, if a user has an account in this system. The presented concept can be used for both users that have accounts in the Telco's own XMPP network as well as users with an existing account in another domain.

For the publication of the presence state and merging it with the existing XMPP resources of the user, two basic use cases have been elaborated:

a) Publish a separate resource (e.g., "sip-phone") with the telephony state of the user
b) Modify the presence state of an existing resource

Case a) has been the most obvious case in the beginning of the research. The SIP presence state is presented as a separate, additional resource of the user. This resource can be either online (i.e., the user is registered), busy (i.e., the user is in a SIP dialog) or offline (i.e., user not registered). Publishing this resource can generally happen independent of whether the user is logged in with another resource or not. The technical realization is rather obvious: The XMPP Publisher must open an instance with a certain resource (e.g., "sip-phone") on behalf of the user to change the state accordingly.

Case b) is not so obvious in the beginning, but has several advantages in comparison with a). Although this case makes only sense when the user is already online with one or more resources, the approach supports the concept more than approach a). The primary instant messaging happens within the XMPP world; the SIP state is only supporting and refining. Hence, a single SIP state of a hardware endpoint is not necessarily required, as no IM conversation can happen with this endpoint anyway. On the other hand, when an XMPP resource is online, it makes perfect sense of refining its state and letting potential contacts know that the user is in a SIP conversation.

The second case has another advantage: XMPP users can only reach resources they can actually communicate with. In case a), the resource "sip-phone" is addressable; hence, users might send messages to it – only receiving an error message as response. By modifying the state of an existing resource as b) indicates, the user keeps control over it and has this resource available for messaging as well. The realization of case b) requires the used client to support [19]. During SIP registration, the XMPP Publisher will log on with a resource that has a negative priority on behalf of the user. This online resource will know the other resources. If the user makes a phone call, it will change the presence state of the client and set it to something pre-defined (e.g., "do-not-disturb" with the status "I am currently on the phone").

For both cases, resources are very important. Either a new resource or an existing resource will represent the users SIP state. Another not less important point to make the concept work is the priority that is assigned to a certain resource[9]. XMPP handles priorities as follows:

Resources with positive priority can receive messages. XMPP servers send messages to the resource with the highest priority. Some clients only show the state of the resource with the highest priority as user state, some show all.

Resources with negative priority must not receive messages. If only one resource is online, and this resource has a negative priority, XMPP servers handle the message as if the user was offline.

Case a) would conceptually work fine, if a resource would get a negative priority: The XMPP server would never deliver messages to it. If the user would not be online with another client, no messages would be lost. The major drawback here

[9] Although this concept would work regardless of the priorities, many clients will only present the overall state from the resource with the highest priority.

is that now client would indicate at first sight, that the user is in an actual conversation. All other states would have a higher priority and not be reflected in almost any client at all. Case b) on the opposite would perfectly work. The XMPP Publisher would modify the state of an existing resource with the highest priority and the state would be reflected immediately. A positive side effect is that the user could immediately change the state back, if desired[10].

4.4 Addressing

Several drafts (e.g., [20], [21]) discuss the general possibility to signal the XMPP Address of Record (AoR) in SIP (or vice versa), so the servers can make a correct assignment. The purpose of those drafts however differs from the approaches of this paper. While those drafts focus on an SIP/XMPP gateway for users from one world communicating with users from the other (as in [20]) or on converged SIP/XMPP clients and the signaling of the AoR for mappings between SIP and XMPP "communication streams" (as in [21]), this paper focusses on a single user in both worlds that wants to use solely XMPP for instant messaging and presence information exchange.

In the presented concept, an actual XMPP client instance is running on the XMPP Publisher on behalf of each user. This concept requires the provisioning of the Jabber ID (JID) *and* the password; hence, this additional signaling for assigning publications is not required. For each SIP user that has assigned an identity in the XMPP Publisher, the XMPP JID and password are provisioned separately. This allows the system to work with the Telco's XMPP infrastructure (pre-provisioned system), but also with separate XMPP servers.

In a provider pre-provisioned system, the SIP provider has also complete supervision over the XMPP server. The XMPP user account is provisioned at the same time as the SIP account. The XMPP Publisher has access to the XMPP user database to successfully authenticate against the XMPP system to modify the resource as a user.

For a separate XMPP server, the user has to provide his credentials to the XMPP Publisher. In both cases, the signaling of the XMPP AoR is not required as the assignment between SIP user and XMPP user AoR cannot be automatized.

4.5 Implemented Prototype

The currently implemented prototype uses several simplifications:

- The proxy is not using a PUA. The open-source SIP application server Open-SIPS [22] has been configured to recognize events in SIP that signal off-hook and on-hook. For both cases, the server calls the XMPP Publisher directly.
- The prototype is not using [19] yet, but a separate resource (as in case a) sec. 4.3). The resource increases its priority if the user is off-hook to be sure it is signaled in the client. The mentioned problems appear, but the concept is visible.

[10] If the XMPP resource that represents the SIP state in case a) supports [19], a client could change its state back as well. It could also increase the priority of its active resource. The authors are aware of this but consider it as too complex for a "quick change-back".

- The XMPP Publisher knows certain events (online, still online, on the phone, still on the phone, offline):
  - Online if the user first registers
  - Still online if the user re-registers
  - On the phone if the user initiates or receives a call
  - Still on the phone for re-`INVITE` messages
  - Offline if the user de-registers or the registration expires.
- The publishing towards XMPP is done using Bidirectional-streams over Synchronous HTTP (BOSH). The persistent Transport Control Protocol (TCP) connections (each published user towards the server) were not implemented, BOSH has rather been chosen with a certain timer (450 seconds). This allows the sending of "events" over non-persistent HTTP connections. The registrations and SIP session timer are forced to 300 seconds. This is the reason for the "still"-events (to allow a "refreshing" the timer).

A prototype according to the concept presented within this paper is currently being implemented and will be tested in practical scenarios.

5 Future Scope and Use Cases

The presented XMPP Publisher can even be of use when the Telco decides in the future to migrate to IMS or centralized SIP based presence management (IMS includes this). While the customer might be able to use sophisticated hardware or software for connecting his co-located SIP PUA, he will still most likely use the XMPP infrastructure. Fig. 3 shows that the XMPP Publisher can still be used to bring the centralized SIP presence information (that is stored in the presence server) easily into the XMPP network, by simply relaying the SIP publications to the XMPP Publisher.

The IMS user that might be using an IMS SIP compliant hardware endpoint still benefits from the proxy and connected PUA (to generate presence events from the SIP communication). Those events might be stored in the presence server of the IMS in the future. SIP/XCAP compliant IMS clients publish their presence states directly to the presence server. The presence server collects all those events and the co-located XDMS infrastructure (omitted in Fig. 3) handles privacy and XCAP documents the client provides. As clients in this IMS-Telco infrastructure will still have *and actively use* XMPP clients and their existing infrastructure, the XMPP Publisher will enrich all of these with IMS presence information.

The previously described scenario can further be improved by developing the XMPP Publisher into an "SIP/XMPP Publisher". The integrated XMPP client side, which is connected on behalf of the SIP user, is active as one of its resources. This also means that it is aware of the presence states of all its XMPP contacts. If some mapping can be implemented in the future (e.g., if SIP and XMPP belong to one operator, the username-part of SIP and XMPP AoR is unique and belongs to a single customer) this XMPP presence states can be "back-published" towards the SIP system as well. Both systems would then be aware

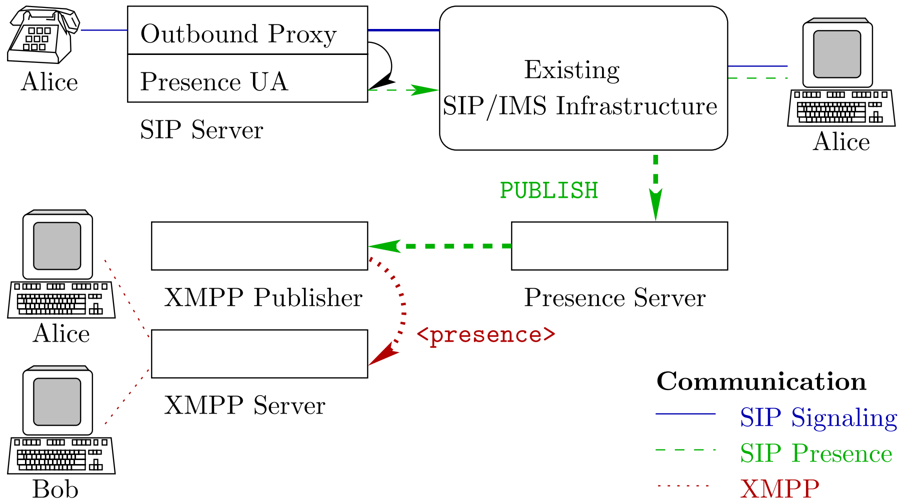

Fig. 3. Future integration of the XMPP Publisher

of the presence states of all users – at least where a proper mapping could be made. Converged messaging would furthermore even allow the communication throughout the "protocol borders".

Another future target is to improve the identity mapping and the related authorization (pre-provisioned user names and passwords). The authors are currently analyzing whether XMPP provides means and can be extended with a useful mechanism respectively to connect the XMPP Publisher similar to components with the XMPP server. A "trusted component" could then modify presence states of the served users. Moreover, the provisioning efforts would be minimized and the simplicity of the concept improved.

Additional use cases have been identified by the authors for the proposed federation of SIP and XMPP architectures. The interworking can be extended from presence-based services also to messaging/notification or location-based services. The information just needs to be mapped in a meaningful way. The near future may include also the initiation of voice/video communication or content streaming initiation. Some drafts are working on that as well.

The main advantage of both protocols is their extensibility and capability to transport XML encapsulated information in the protocol body. Finally, the same XML body should be used in the same manner and enable convergence of both worlds and easier interworking. The most relevant scenario is using this convergence to integrate some of the Web 2.0 technologies based on XML or XMPP for extension of integrated concepts with SIP based architecture. The goal is not replace SIP protocol technology with XMPP protocols but used each of their strengths and benefit from their interworking for Telco and Internet convergence. This wider scope of proposed federation should also extend the applicability and Web 2.0 service accessibility in the future. This is also valid for

NGN/IMS based architectures, as it would make them more attractive for application developers and internet users. In these cases the same services will be available on NGN networks, but also on the internet and most importantly providing the same user experience and large user community.

6 Conclusion

The presented concept is a good compromise between immediately starting to introduce presence in the telecommunication world. XMPP as widely used and adapted concept has been chosen.

The authors are aware and support the moves from the standardization committees to push SIP based presence management forward. IMS is from the author's perspective clearly the future; its presence management concepts included. The Telco's are however reminded that integration with the existing Web 2.0 infrastructure is important and the presented concept can highly increase the user experience. Customers will choose the Telco that does not only offer new converged services on their platform, but also respect the existing communication infrastructure of the clients and integrate it as good as possible.

Acknowledgments. The STU and the HfTL are actively cooperating since 2006. The universities are working together on common topics in the area of Next Generation Networks and participate in ongoing projects involving their core network interconnection.

The authors Maruschke and Schumann are members of the 'Institut für Telekommunikationsinformatik' at the HfTL. Working mainly in the area of Next Generation Networks, their focus is on migration of core networks of global telecommunication operators towards packet oriented networks as well as their interconnection.

This paper also presents some of the results and acquired experience from various research project such as NGNlab project [23], European Celtic-EURECA project Netlab[24], Leonardo da Vinci projects InCert [25] and Train2Cert [26], AV project: Converged technologies for next generation networks (NGN), Slovak National basic research projects VEGA No. 1/3094/06 and VEGA 1/4084/07.

References

1. Blum, N., Lampe, S., Magedanz, T.: Design of a Message Interworking Function for Converged IP Messaging in Next Generation Networks. In: Computers and Communications, ISCC 2009, pp. 80–85. IEEE Press, Sousse (2009)
2. OMA Converged IP Messaging Architecture, System Description System Description-V1 0-20091024-D, Open Mobile Alliance (2008)
3. OpenSIPS XMPP and PUA_XMPP module, `http://www.opensips.org/Resources/PuaExtensions#pu_xmpp`
4. Rosenberg, J., et al.: RFC 3261: SIP: Session Initiation Protocol. IETF (2002)
5. Peterson, J.: RFC 3859: Common Profile for Presence (CPP). IETF (2004)

6. Roach, A.B.: RFC 3265: Session Initiation Protocol (SIP)-Specific Event Notification. IETF (2002)
7. Rosenberg, J.: RFC 3856: A Presence Event Package for the Session Initiation Protocol (SIP). IETF (2004)
8. Niemi, A.: RFC 3903: Session Initiation Protocol (SIP) Extension for Event State Publication. IETF (2004)
9. Rosenberg, J.: RFC 3857: A Watcher Information Event Template-Package for the Session Initiation Protocol (SIP). IETF (2004)
10. Rosenberg, J.: RFC 4825: The Extensible Markup Language (XML) Configuration Access Protocol (XCAP). IETF (2007)
11. Rosenberg, J.: RFC 5025: Presence Authorization Rules. IETF (2007)
12. OMA Presence SIMPLE, Architecture Document OMA-AD-Presence_ SIMPLEV1_1-20080627-A, Open Mobile Alliance (2008)
13. Saint-Andre, P.: RFC 3920: Extensible Messaging and Presence Protocol (XMPP):Core. IETF (2004)
14. Saint-Andre, P.: RFC 3921: Extensible Messaging and Presence Protocol (XMPP): Instant Messaging and Presence. IETF (2004)
15. Ludwig, S., et al.: XEP-0166: Jingle. XMPP Standards Foundation (2009)
16. Ludwig, S., et al.: XEP-0167: Jingle Audio Media Description Format. XMPP Standards Foundation (2009)
17. Day, M., et al.: RFC 2778: A Model for Presence and Instant Messaging. IETF (2000)
18. Day, M., et al.: RFC 2779: Instant Messaging/Presence Protocol. IETF (2000)
19. Troncon, R., et al.: XEP-0146: Remote Controlling Clients. XMPP Standards Foundation (2006)
20. Sparks, R.: Internet Draft (expired): Establishing jabber Messaging Sessions with the Session Initiation Protocol. IETF (2002)
21. Veikkolainen, S., et al.: Internet Draft: Guidelines and Protocol Extensions for Combining SIP Based Real-time Media Sessions With XMPP Based Instant Messaging and Presence Service. IETF (2009)
22. OpenSIPS (Open SIP Server), a mature Open Source implementation of a SIP server, `http://www.opensips.org`
23. NGNlab - NGN laboratory at STU in Bratislava, `http://www.ngnlab.eu`
24. NetLab - Use Cases for Interconnected Testbeds and Living Labs, `http://www.celtic-initiative.org/Projects/NETLAB/`
25. InCert Next Generation Network Protocols Professionals certification in InCert, International Certificates of Excellence in Selected Areas of ICT, `http://incert.eu`
26. Train2Cert, Vocational Training for Certification in ICT, `http://train2cert.eu`