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**CUSAX: Combined Use of the Session Initiation Protocol (SIP) and the  
Extensible Messaging and Presence Protocol (XMPP)  
draft-iovov-xmpp-cusax-06**

**Abstract**

This document describes suggested practices for combined use of the Session Initiation Protocol (SIP) and the Extensible Messaging and Presence Protocol (XMPP). Such practices aim to provide a single fully featured real-time communication service by using complementary subsets of features from each of the protocols. Typically such subsets would include telephony capabilities from SIP and instant messaging and presence capabilities from XMPP. This specification does not define any new protocols or syntax for either SIP or XMPP. However, implementing the practices outlined in this document may require modifying or at least reconfiguring existing client and server-side software. Also, it is not the purpose of this document to make recommendations as to whether or not such combined use should be preferred to the mechanisms provided natively by each protocol (for example, SIP's SIMPLE or XMPP's Jingle). It merely aims to provide guidance to those who are interested in such a combined use.

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## [1.](#) Introduction

Historically SIP [[RFC3261](#)] and XMPP [[RFC6120](#)] have often been implemented and deployed with different purposes: from its very start SIP's primary goal has been to provide a means of conducting "Internet telephone calls". XMPP on the other hand, has, from its Jabber days, been mostly used for instant messaging and presence [[RFC6121](#)], as well as related services such as groupchat rooms [[XEP-0045](#)].

For various reasons, these trends have continued through the years even after each of the protocols had been equipped to provide the features it was initially lacking:



- o Today, in the context of the SIMPLE working group, the IETF has defined a number of protocols and protocol extensions that not only allow for SIP to be used for regular instant messaging and presence but that also provide mechanisms for elaborated features such as multi-user chats, server-stored contact lists, file transfer and others.
- o Similarly, the XMPP community and the XMPP Standards Foundation have worked on defining a number of XMPP Extension Protocols (XEPs) that provide XMPP implementations with the means of establishing end-to-end sessions. These extensions are often jointly referred to as Jingle and arguably their most popular use case is audio and video calling.

Despite these advances, SIP remains the protocol of choice for telephony-like services, especially in enterprises where users are accustomed to features such as voice mail, call park, call queues, conference bridges and many others that are rarely (if at all) available in Jingle-based software. XMPP implementations, on the other hand, greatly outnumber and outperform those available for instant messaging and presence extensions developed in the SIMPLE WG, such as MSRP [[RFC4975](#)] and XCAP [[RFC4825](#)].

As a result, a number of adopters have found themselves needing features that are not offered by any single-protocol solution, but that separately exist in SIP and XMPP implementations. The idea of seamlessly using both protocols together would hence often appeal to service providers. Most often, such a service would employ SIP exclusively for audio, video, and telephony services and rely on XMPP for anything else varying from chat, contact list management, and presence to whiteboarding and exchanging files. Because these services and clients involve the combined use of SIP and XMPP, we label them "CUSAX" for short.

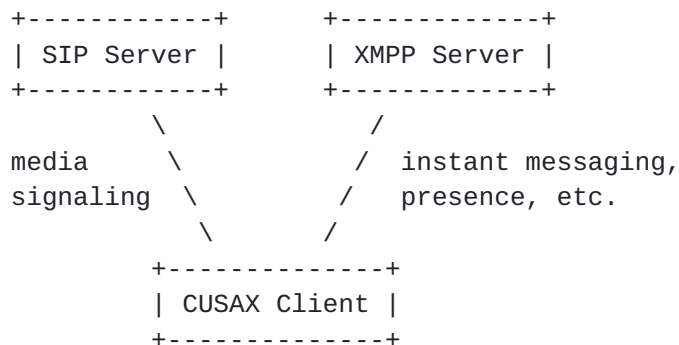


Figure 1: Division of Responsibilities



This document explains how such hybrid offerings can be achieved with a minimum of modifications to existing software while providing an optimal user experience. It covers server discovery, determining a SIP Address of Record (AOR) while using XMPP, and determining an XMPP Jabber Identifier ("JID") from incoming SIP requests. Most of the text here pertains to client behavior but it also recommends certain server-side configurations.

Note that this document is focused on coexistence of SIP and XMPP functionality in end-user-oriented clients. By intent it does not define methods for protocol-level mapping between SIP and XMPP, as might be used within a server-side gateway between a SIP network and an XMPP network (a separate series of documents has been produced that defines such mappings). More generally, this document does not describe service policies for inter-domain communication (often called "federation") between service providers (e.g., how a service provider that offers a combined SIP-XMPP service might communicate with a SIP-only or XMPP-only service), nor does it describe the reasons why a service provider might choose SIP or XMPP for various features.

This document concentrates on use cases where the SIP services and XMPP services are controlled by one and the same provider, since that assumption greatly simplifies both client implementation and server-side deployment (e.g., a single service provider can enforce common or coordinated policies across both the SIP and XMPP aspects of a CUSAX service, which is not possible if a SIP service is offered by one provider and an XMPP service is offered by another). Since this document is of an informational nature, it is not unreasonable for clients to apply some of the guidelines here even in cases where there is no established relationship between the SIP and the XMPP services (for example, it is reasonable for a client to provide a way for its users to easily start a call to a phone number recorded in a vCard or obtained from a user directory). However, the exact set of rules to follow in such cases is left to application developers.

Finally, this document makes a further simplifying assumption by discussing only the use of a single client, not use of and coordination among multiple endpoints controlled by the same user (e.g., user agents running simultaneously on a laptop computer, tablet, and mobile phone).

## **2. Client Bootstrap**

One of the main problems of using two distinct protocols when providing one service is the impact on usability. Email services, for example, have long been affected by the mixed use of SMTP for outgoing mail and POP3 or IMAP for incoming mail. Although standard



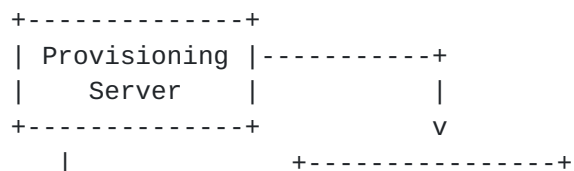
service discovery methods (such as the proper DNS records) make it possible for a user agent to locate the right host(s) at which to connect, they do not provide the kind of detailed information that is needed to actually configure the user agent for use with the service. As a result, it is rather complicated for inexperienced users to configure a mail client and start using it with a new service, and Internet service providers often need to provide configuration instructions for various mail clients. Client developers and communication device manufacturers on the other hand often ship with a number of wizards that enable users to easily set up a new account for a number of popular email services. While this may improve the situation to some extent, the user experience is still clearly sub-optimal.

While it should be possible for CUSAX users to manually configure their separate SIP and XMPP accounts, service providers offering CUSAX services to users of dual-stack SIP/XMPP clients ought to provide means of online provisioning, typically by means of a web-based service at an HTTP URI (naturally single-purpose SIP services or XMPP services could offer online provisioning as well, but they can be especially helpful where the two aspects of the CUSAX service need to have several configuration options in common). While the specifics of such mechanisms are outside the scope of this specification, they should make it possible for a service provider to remotely configure the clients based on minimal user input (e.g., only a user ID and password).

Because many of the features that a CUSAX client would prefer in one protocol would also be available in the other, clients should make it possible for such features to be disabled for a specific account. In particular, it is suggested that clients allow for audio and video calling features to be disabled for XMPP accounts, and that instant messaging and presence features should also be made optional for SIP accounts.

The main advantage of this approach is that clients would be able to continue to function properly and use the complete feature set of standalone SIP and XMPP accounts.

Once clients have been provisioned, they need to independently log into the SIP and XMPP accounts that make up the CUSAX "service" and then maintain both these connections as displayed in Figure 2.







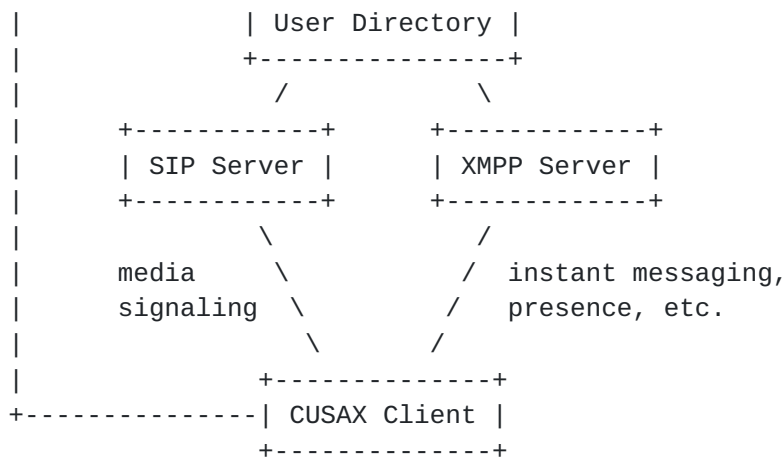


Figure 2: Example Deployment

In order to improve the user experience, when reporting connection status clients may also wish to present the XMPP connection as an "instant messaging" or a "chat" account. Similarly they could also depict the SIP connection as a "Voice and Video" or a "Telephony" connection. The exact naming is of course entirely up to implementers. The point is that, in cases where SIP and XMPP are components of a service offered by a single provider, such presentation could help users better understand why they are being shown two different connections for what they perceive as a single service. It could alleviate especially situations where one of these connections is disrupted while the other one is still active. Naturally, the developers of a CUSAX client or the providers of a CUSAX service might decide not to accept such situations and force a client to completely disconnect unless both aspects are successfully connected.

Clients may also choose to delay their XMPP connection until they have been successfully registered on SIP. This would help avoid the situation where a user appears online to its contacts but calling it would fail because their clients is still connecting to the SIP aspect of their CUSAX service.

### 3. Operation

Once a CUSAX client has been provisioned and authorized to connect to the corresponding SIP and XMPP services it would proceed by retrieving its XMPP roster.

The client should use XMPP for all forms of communication with the contacts from this roster, which will occur naturally because they were retrieved through XMPP. Audio/video features however, are disabled in the XMPP stack, so any form of communication based on



these features (e.g. direct calls, conferences, desktop streaming, etc.) will happen over SIP. The rest of this section describes deployment, discovery, usability and linking semantics that allow CUSAX clients to fall back and seamlessly use SIP for these features.

### **3.1. Server-Side Setup**

In order for CUSAX to function properly, XMPP service administrators should make sure that at least one of the vCard [RFC6350] "tel" fields for each contact is properly populated with a SIP URI or a phone number when an XMPP protocol for vCard storage is used (e.g., [XEP-0054] or [XEP-0292]). There are no limitations as to the form of that number. For example while it is desirable to maintain a certain consistency between SIP AORs and XMPP JIDs, that is by no means required. It is quite important however that the phone number or SIP AOR stored in the vCard be reachable through the SIP aspect of this CUSAX service. (The same considerations apply even if the directory storage format is not vCard.)

Administrators may also choose to include the "video" tel type defined in [RFC6350] for accounts that would be capable of handling video communication.

To ensure that the foregoing approach is always respected, service providers might consider (1) preventing clients (and hence users) from modifying the vCard "tel" fields or (2) applying some form of validation before storing changes. Of course such validation would be feasible mostly in cases where a single provider controls both the XMPP and the SIP service since such providers would "know" (e.g., based on use of a common user database for both services) what SIP AOR corresponds to a given XMPP user (as indicated in Figure 2).

### **3.2. Client-Side Discovery and Usability**

When rendering the roster for a particular XMPP account CUSAX clients should make sure that users are presented with a "Call" option for each roster entry that has a properly set "tel" field. This is the case even if calling features have been disabled for that particular XMPP account, as advised by this document. The usefulness of such a feature is not limited to CUSAX. After all, numbers are entered in vCards or stored in directories in order to be dialed and called. Hence, as long as an XMPP client has any means of conducting a call it may wish to make it possible for the user to easily dial any numbers that it learned through whatever means.

Clients that have separate triggers (e.g., buttons) for audio calls and video calls may choose to use the presence or absence of the "video" tel type defined in [RFC6350] as the basis for choosing



whether to enable or disable the possibility for starting video calls (i.e., if there is no "video" tel type for a particular contact, do not provide a way for the user to start a video call with that contact).

In addition to discovering phone numbers from vCards or user directories, clients may also check for alternative communication methods as advertised in XMPP presence broadcasts and Personal Eventing Protocol nodes as described in XEP-0152: Reachability Addresses [[XEP-0152](#)]. However, these indications are merely hints, and a receiving client ought not associate a SIP address and an XMPP address unless it has some way to verify the association (e.g., the vCard of the XMPP account lists the SIP address and the vCard of the SIP account lists the XMPP address, or the association is made explicit in a record provided by a trusted directory). Alternatively or in cases where vCard or directory data is not available, a CUSAX client could take the user's own address book as the canonical source for contact addresses.

### **3.3. Indicating a Relation Between SIP and XMPP Accounts**

In order to improve usability, in cases where clients are provisioned with only a single telephony-capable account they ought to initiate calls immediately upon user request without asking users to indicate an account that the call should go through. This way CUSAX users (whose only account with calling capabilities is usually the SIP part of their service) would have a better experience, since from the user's perspective calls "just work at the click of a button".

In some cases however, clients will be configured with more than the two XMPP and SIP accounts provisioned by the CUSAX provider. Users are likely to add additional stand-alone XMPP or SIP accounts (or accounts for other communications protocols), any of which might have both telephony and instant messaging capabilities. Such situations can introduce additional ambiguity since all of the telephony-capable accounts could be used for calling the numbers the client has learned from vCards or directories.

To avoid such confusion, client implementers and CUSAX service providers may choose to indicate the existence of a special relationship between the SIP and XMPP accounts of a CUSAX service. For example, let's say that Alice's service provider has opened both an XMPP account and a SIP account for her. During or after provisioning, her client could indicate that `alice@xmpp.example.com` has a CUSAX relation to `alice@sip.example.com` (i.e., that they are two aspects of the same service). This way whenever Alice triggers a call to a contact in her XMPP roster, the client would preferentially initiate this call through her `example.com` SIP account even if other



possibilities exist (such as the XMPP account where the vCard was obtained or a SIP account with another provider).

If, on the other hand, no relationship has been configured or discovered between a SIP account and an XMPP account, and the client is aware of multiple telephony-capable accounts, it ought to present the user with the choice of reaching the contact through any of those accounts. This includes the source XMPP account where the vCard was obtained (in case its telephony capabilities are not disabled through configuration or provisioning), in order to guarantee proper operation for XMPP accounts that are not part of a CUSAX deployment.

### **3.4. Matching Incoming SIP Calls to XMPP JIDs**

When receiving SIP calls, clients may wish to determine the identity of the caller and a corresponding XMPP roster entry so that users could revert to chatting or other forms of communication that require XMPP. To do so clients could search their roster for an entry whose vCard has a "tel" field matching the originator of the call. In addition, in order to avoid the effort of iterating over an entire roster and retrieving all vCards, CUSAX clients may use a SIP Call-Info header whose 'purpose' header field parameter has a value of "impp" as described in [[I-D.saintandre-impp-call-info](#)]. An example follows.

Call-Info: <xmpp:alice@xmpp.example.com> ;purpose=impp

Note that the information from the Call-Info header should only be used as a cue: the actual AOR-to-JID binding would still need to be confirmed by a vCard entry or through some other trusted means (such as an enterprise directory). If this confirmation succeeds the client would not need to search the entire roster and retrieve all vCards. Not performing the check might enable any caller (including malicious ones) to employ someone else's identity and perform various scams or Man-in-the-Middle attacks.

However, although an AOR-to-JID binding can be a helpful hint to the user, nothing in the foregoing paragraph ought to be construed as necessarily discouraging users, clients, or service providers from accepting calls originated by entities that are not established contacts of the user (e.g., as reflected in the user's roster); that is a policy matter for the user, client, or service provider.

## **4. Multi-Party Interactions**

CUSAX clients that support the SIP conferencing framework [[RFC4353](#)] can detect when a call they are participating in is actually a





conference and can then subscribe for conference state updates as per [RFC4575]. A regular SIP user agent would also use the same conference URI for text communication with the Message Session Relay Protocol (MSRP). However, given that SIP's instant messaging capabilities would normally be disabled (or simply not supported) in CUSAX deployments, an XMPP Multi-User Chat (MUC) [XEP-0045] associated with the conference can be announced/discovered through <service-uris> bearing the "grouptextchat" purpose [I-D.ivoiv-grouptextchat-purpose]. Similarly, an XMPP MUC can advertise the SIP URI of an associated service for audio/video interactions using the 'audio-video-uri' field of the "muc#roominfo" data form [XEP-0004] to include extended information [XEP-0128] about the MUC room within XMPP service discovery [XEP-0030]; see [XEP-0045] for an example.

Once a CUSAX client joins the MUC associated with a particular call it should not rely on any synchronization between the two. Both the SIP conference and the XMPP MUC would function independently, each issuing and delivering its own state updates. It is hence possible that that certain peers would temporarily or permanently be reachable in only one of the two conferences. This would typically be the case with single-stack clients that have only joined the SIP call or the XMPP MUC. It is therefore important for CUSAX clients to provide a clear indication to users as to the level of participation of the various participants. In other words, a user needs to be able to easily understand whether a certain participant can receive text messages, audio/video, or both.

Of course, tighter integration between the XMPP MUC and the SIP conference is also possible. Permissions, roles, kicks and bans that are granted and performed in the MUC can easily be imitated by the conference focus/mixer into the SIP call. If for example, a certain MUC member is muted, the conference mixer can choose to also apply the mute on the media stream corresponding to that participant. The details and exact level of such integration is of course entirely up to implementers and service providers.

The approach above describes one relatively lightweight possibility of combining SIP and XMPP multi-party interaction semantics without requiring tight integration between the two. As with the rest of this specification, this approach is by no means normative. Implementation and future specifications may define other methods or provide other suggestions for improving the Unified Communications user experience in cases of multi-user chats in conference calling.

## **5. Federation**



In theory there are no technical reasons why federation would require special behavior from CUSAX clients. However, it is worth noting that differences in administration policies may sometimes lead to potentially confusing user experiences.

For example, let's say atlanta.example.com observes the CUSAX policies described in this specification. All XMPP users at atlanta.example.com are hence configured to have vCards that match their SIP identities. Alice is therefore used to making free, high-quality SIP calls to all the people in her roster. Alice can also make calls to the PSTN by simply dialing numbers. She may even be used to these calls being billed to her online account so she would be careful about how long they last. This is not a problem for her since she can easily distinguish between a free SIP call (one that she made by calling one her roster entries) from a paid PSTN call that she dialed as a number.

Then Alice adds xmpp:bob@biloxi.example.com. The Biloxi domain only has an XMPP service. There is no SIP server and Bob uses a regular, XMPP-only client. Bob has however added his mobile number to his vCard in order to make it easily accessible to his contacts. Alice's client would pick up this number and make it possible for Alice to start a call to Bob's mobile phone number.

This could be a problem because, other than the fact that Bob's address is from a different domain, Alice would have no obvious and straightforward cues telling her that this is in fact a call to the PSTN. In addition to the potentially lower audio quality, Alice may also end up incurring unexpected charges for such calls.

In order to avoid such issues, providers maintaining a CUSAX service for the users in their domain may choose to provide additional cues (e.g., a user interface warning or an audio tone or message) indicating that a call would incur charges.

A slightly less disturbing scenario, where a SIP service might only allow communication with intra-domain numbers, would simply prevent Alice from establishing a call with Bob's mobile. Providers should hence make sure that calls to inter-domain numbers are flagged with an appropriate audio or textual warning.

## **6. Summary of Suggested Practices**

The following practices are suggested for CUSAX user agents:

1. By default, prefer SIP for audio and video, and XMPP for messaging and presence.



2. Use XMPP for all forms of communication with the contacts from the XMPP roster, with the exception of features that are based on establishing real-time sessions (e.g. audio/video calls) in which case use SIP.
3. Provide on-line provisioning options for providers to remotely setup SIP and XMPP accounts so that users wouldn't need to go through a multi-step configuration process.
4. Provide on-line provisioning options for providers to completely disable features for an account associated with a given protocol (SIP or XMPP) if the features are preferred in another protocol (XMPP or SIP).
5. Present a "Call" option for each roster entry that has a properly set "tel" field.
6. If the client is provisioned with only a single telephony-capable account, initiate calls immediately upon user request without asking users to indicate an account that the call should go through.
7. If no relationship has been configured or discovered between a SIP account and an XMPP account, and the client is aware of multiple telephony-capable accounts, present the user with the choice of reaching the contact through any of those accounts.
8. Optionally, indicate the existence of a special relationship between the SIP and XMPP accounts of a CUSAX service.
9. Optionally, present the XMPP connection as an "instant messaging" or a "chat" account and the SIP connection as a "Voice and Video" or a "Telephony" account.
10. Optionally, determine the identity of the audio/video caller and a corresponding XMPP roster entry so that the user could revert to textual chatting or other forms of communication that require XMPP.
11. Optionally, delay the XMPP connection until after a SIP connection has been successfully registered.
12. Optionally, check for alternative communication methods (SIP addresses advertised over XMPP, and XMPP addresses advertised over SIP).

The following practices are suggested for CUSAX services:



1. Use online provisioning and configuration of accounts so that users won't need to setup two separate accounts for your service.
2. Use online provisioning so that calling features are disabled for all XMPP accounts.
3. Ensure that at least one of the vCard "tel" fields for each XMPP user is properly populated with a SIP URI or a phone number that are reachable through your SIP service.
4. Optionally, include the "video" tel type for accounts that are capable of handling video communication.
5. Optionally, provision clients with information indicating that specific SIP and XMPP accounts are related in a CUSAX service.
6. Optionally, attach a "Call-Info" header with an "impp" purpose to all your SIP INVITE messages, so that clients can more rapidly associate a caller with a roster entry and display a "Caller ID".

## **7. Security Considerations**

Use of the same user agent with two different accounts providing complementary features introduces the possibility of mismatches between the security profiles of those accounts or features. For example, the SIP aspect and XMPP aspect of the CUSAX service might offer different authentication options (e.g., digest authentication for SIP as specified in [[RFC3261](#)] and SCRAM authentication [[RFC5802](#)] for XMPP as specified in [[RFC6120](#)]). Similarly, a CUSAX client might successfully negotiate Transport Layer Security (TLS) [[RFC5246](#)] when connecting to the XMPP aspect of the service but not when connecting to the SIP aspect. Such mismatches could introduce the possibility of downgrade attacks. User agent developers and service providers ought to ensure that such mismatches are avoided as much as possible.

Refer to the specifications for the relevant SIP and XMPP features for detailed security considerations applying to each "stack" in a CUSAX client.

## **8. IANA Considerations**

This document has no actions for the IANA.





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## [Appendix A](#). Acknowledgements

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