TIA IP Telephone Standard

Provisions for SIP

Version 0.1

18 June 1999

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1 Introduction

A business phone providing a SIP user agent allows the phone to have more intelligence and autonomy than is possible via master-slave call control protocols. This enables a number of features that are typically implemented in a business style PBX to be provided by the business phone itself or in conjunction with other SIP elements (i.e. registry, redirect, proxy servers). The objective for the SIP portion of the standard being developed by the TIA TR41.3.4 working group should be to standardize and ensure interoperability of feature sets (where SIP is involved) on SIP-enabled business phones.

The purpose of this document is to:

- Identify a set of mandatory and optional business phone features. The primary focus is on end user features, however operational features are also considered. Perhaps many features will come from those outlined in “Implementing Intelligent Network Services with the Session Initiation Protocol” [2]. It includes features such as transfer, conference, park, pick up, barge in, etc.

- Provide a mapping describing how each of these features should be accomplished using SIP. This mapping is intended to avoid feature interoperability problems where multiple business phone implementations use SIP differently to provide the same feature. The mapping description also identifies centralized SIP elements (i.e. registry, redirect and proxy servers) that are required to provide the feature. If deemed appropriate, a future version of this document will illustrate in detail the SIP messages exchanged between the elements. Note: Since the goal is to define interoperable business phones, the emphasis in these descriptions is on the phone’s perspective.

- Identify or define extensions to SIP that are needed to provide the features. Ideally no extensions to the SIP protocol will be necessary, however this is probably not the case. If extensions are required, the TIA group should first attempt to use existing “work in progress” that address these issues before it invents new solutions. Furthermore, such requirements and potential solutions should be submitted to the relevant IETF working group for their consideration in developing updates and extensions to SIP.

- Identify future considerations to be addressed (e.g. a standard means of configuration).

2 Overview

2.1 Scope

The SIP portion of the specification being produced by the TR41.3.4 working group should focus on identifying features that “MUST” or “SHOULD” be on SIP-enabled business phones.

User interface and higher level functions (i.e. above the call control protocol or which may be implemented “on top” of features) are explicitly out of scope as they do not impact interoperability and allow for vendor differentiation.

2.2 Design Approach

The intention is to provide an architecture independent solution for interoperable business phone features. The specification being produced by the TR41.3.4 working group should not dictate the location (i.e. phone, service network, etc.) of feature implementation if possible. Many business phone features may be implemented in service network (i.e. using SIP servers that do not terminate a call). Many features may be implemented in the phone. The objective is to specify how (by reference) SIP enables features to be implemented on the phone. The definition of how features may be implemented in a service network is beyond the scope of this document unless it requires SIP capabilities on the phone beyond the baseline identified in this document.

The baseline SIP-enabled business phone MUST provide the functionality defined in RFC 2453 for a Redirection Capable Client [1 §A.1] and, depending on the business phone features that are supported, SHOULD optionally support the following capabilities and extensions:

- Basic and Digest Authentication [1]
- Call Control Extensions [5]
- Caller Preference Extensions [6]

Some of the extensions proposed in “SIP Call Control Services” [5] are relevant for:

- Transfer – [need to define which transfer model]
- Conference – Bridged
- Conference – Fully Meshed

Similarly, Some of the extensions proposed in “SIP Caller Preferences and Callee Capabilities” [6] are relevant for:

- Camp On - uses the Request-Disposition field set to Queue

3 Features

The phone is a SIP user agent.

Unless otherwise stated all features are provided in the context of a phone.
This does not mean that these features may not be provided via other means. The other means are simply beyond the scope of this document.

For each feature, the following information is provided:

- Definition
- SIP extension(s) required
- Entities Involved
- Message/event trace
- Comments/Notes

3.1 Mandatory Features

3.1.1 Camp On

Mandatory
Extension(s): *org.ietf.sip.caller-preferences*
Entity(s): *none*

Definition

A caller attempting to call the callee receives a busy indication. The caller uses the camp on feature to get notification and have a call automatically set up when the callee is available.

Solution
The low tech. solution is for the caller to poll (i.e. send periodic INVITEs).

A more scalable solution is to have the callee's phone queue the call. The caller may indicate this using the Request-Disposition field set to Queue [6]. [It seems a shame to have to support the caller-preferences extension just for this one feature.]

3.1.2 Forwarding

Mandatory
Extension(s): *none*
Entity(s): *none*

Definition
Forward an incoming call on Busy, No Answer, Unconditional, Deflection

Solution
Supply the forwarding destination to the caller using the Contact field in the 302 (Moved Temporarily) response.

3.1.3 Hold – Near End

Mandatory
Extension(s): *none*
Entity(s): *none*

Definition
A user places a call on hold. The hold function is performed locally. The other parties to the call are not notified or signaled.

Solution
No SIP messaging is involved. The local phone keeps the RTP session alive but sends or indicates silence (should we require comfort noise packets at some interval to indicate that the call is still alive?).

Note
Hold w/ music or announcement is only an media issue. The media source may be stored locally or delivered via RTP or another mechanism. The local phone injects the media into its outbound RTP stream for the call.

[We should define a standard means of discovery of the RTP announcement stream.] Some options are:

- The phone is configured with SDP information.
- The phone is configured with the SIP URL for an announcement server to INVITE for a receive only call (there may be latency issues with this).
- The phone is configured with a(many) SIP From address from which it periodically receives INVITEs to receive only calls. The phone caches the SDP information until it is needed for hold.

3.1.4 Intercom/Paging

Mandatory (for phones with speaker phone)

Extension(s): *none*
Entity(s): *Caller Preferences*

Definition
An Announcement is played through the speaker on one or more (speaker-equipped) phones. The announcement may be sent to all phones in a company, a group of phones or to a specific user. There should be two categories of intercom usage: standard and emergency. Standard usage does not interrupt a busy phone.

Solution
The announcer's phone sends an INVITE to a well known SIP URL which represents the scope of the Intercom/Paging announcement (i.e. a specific phone - page_ceo_office@my.com, group of phones - page_building5@my.com, all phones in the company- page_all@my.com, etc.).

The INVITE message for a receive-only page will contain two tokens in the Request-Disposition field to indicate how the call should be handled. These token values indicate:

1. That the call should be auto-answered (the call will be answered without the phone ringing)
2. Whether to interrupt an existing call to deliver the page (for emergency pages) or whether to put the paging call through only if the phone is idle.

[6 (next revision)].

The phone must accept both multicast and unicast SIP INVITE messages. The entities involved in getting the INVITE from the announcer to the announcing phone(s) are implementation and installation specific. The mechanism (redirect servers, proxy servers, phone configuration) used to distribute paging announcements to the phone is an implementation detail that is out of scope for this document.

3.1.5 Message Waiting - Indication

Mandatory
Extension(s): *Subscription and Voice Mail Server*
Entity(s): *subscribe/notify*

Definition
The phone provides an indication that voice mail messages are available for a user.

Solution
The user interface is up to the implementation whether via LED indicator, stutter dial tone, etc. This feature assumes the existence of a SUBSCRIBE/NOTIFY mechanism [7]. The SUBSCRIBE/NOTIFY mechanism allows a SIP phone to make known its desire to receive notification of state changes associated with a service. [PINT draft reference.] When the phone subscribes to a service, it is sent a NOTIFY message containing the current state of that service. Additional NOTIFY messages are sent to subscribers whenever the state changes, until the phone unSUBSCRIBES from the service.

For the Message Waiting Indication this works as follows:
On initialization, the phone SUBSCRIBEs for the voice mail state for its user's SIP URL (e.g., the phone extension) with the relevant subscription server. The voice mail server and the subscription server may be distinct entities. The separation and communication between these two logical server entities is beyond the scope of this document. The server sends the current state of the user's voice mail account (i.e. n new messages, where n is 0 or more) in a NOTIFY message. The server sends additional NOTIFY messages as the state changes (e.g. 0 new messages after the user listens to voice mail).

3.1.6 Park

Mandatory
Extension(s): *call control, subscribe/notify*
Entity(s): *none*

Definition
A user chooses to put a call on hold such that another user may pick it up and take it off hold from another phone.

Solution
A call is already setup on a phone. The user wishes to park the call. The first step is to put the call on hold. Another entity (i.e. phone or server) which is interested in receiving parked call notifications for that phone, SUBSCRIBEs to the phone. As a result, that entity will receive notifications of current or future parked call states. As a call is currently parked at the phone the subscribing entity(s) is sent a NOTIFY message containing callId, To, From information and call state.

3.1.7 Pickup – Parked Call for Station, Group

Mandatory
Extension(s): *call control, subscribe/notify*
Entity(s): *subscription server*

Definition
A user wishes to answer a call parked from another phone. The user may indicate a call from a specific station (phone) or a group of phones.

Solution
When a user requests a call pickup for a specific SIP URL (e.g., extension), the user's phone SUBSCRIBEs for calls parked at the designated URL. The user's phone receives a NOTIFY message containing the data (callId, To, From, etc.) from the parked call. The user's phone then sends an INVITE to the phone with the parked call using the same callId, as the parked call. The call control extension field "Replaces" [SIP Call Control Services] is used to indicate that the user's phone should be added to the call and the phone with the parked call should be dropped from the call.

For the group case, the user requests a call pickup from a group of phones which contain the parked call. This case requires that an entity (park service) aggregate the NOTIFY messages for parked calls and works as follows:

1. On startup, the park service subscribes to all of the phones in the park group.
2. To pickup a call in a group, the user's phone SUBSCRIBES with the park service for the relevant group. The park service responds to the subscription request with the URLs of all of the calls that are currently parked for the group. The user's phone selects one of these calls and then uses the "Parked Call for Station" procedure (detailed above) to pickup that call.

3.1.8 Pickup – Ringing Station, Station in Group

Mandatory
Extension(s): *none*
Entity(s): *Registry and Proxy Server(s)*

Definition
A user wishes to answer a call ringing on another phone. The user may indicate a call from a specific station (phone) or a group of phones.

Solution
A phone is ringing. A user at another phone indicates the desire to pickup the ringing call at a specific phone. The user's phone REGISTERs using the ringing phone's SIP URL (e.g. extension). The registry server (if different from the proxy server) notifies the proxy server for the ringing phone of the registration. The proxy server forks (or cancels the INVITE to the ringing phone) and INVITEs the user's phone to the call. [1].

For the group case, the user's phone REGISTERs the group name (as opposed to the user/extension name for the call at the ringing phone). The registry and/or proxy server must understand the mapping (through configuration or other means) from group name to user name in the SIP URL. The rest of the call setup is the same as for the ringing station case.

3.1.9 Queued Calls

Mandatory
Extension(s): *? Caller Preference ?*
Entity(s): *none*

Definition

Solution
Can be accomplished by setting the Request-Disposition field to Queue or No-Queue

3.1.10 Transfer – Consultative

Mandatory
Extension(s): *Call Control*
Entity(s): *none*

Definition
In a consultative transfer, a party to the call puts the call on hold, creates a new call to the transfer destination, talks to the party at the transfer destination, transfers the call on hold to the transfer destination and hangs up.

Solution
[There are multiple ways to do this. We need to be specific and pick one.]
[*5 §nn*]

3.1.11 Transfer – Blind

Mandatory
Extension(s): *Call Control*
Entity(s): *none*

Definition
In a blind transfer, a party to the call transfers the other party(s) to the call to the transfer destination and drops out of the call, without talking (consulting) with the party at the transfer destination.

Solution
[There are multiple ways to do this. We need to be specific and pick one.]
[*5 §nn*]

3.2 Optional Features

3.2.1 Authentication

Optional
Extension(s): *none*
Entity(s): *none*

Definition
Ensure that an incoming call is from a trusted party.

Solution
WWW-Authenticate field [1] set to basic or digest

3.2.2 Barge In – Monitor, Talk

Optional
Extension(s): *call control*
Entity(s): *proxy server*

Definition
(AKA Direct Station Selection while busy) A privileged user wishes to monitor or join an existing call.

Solution
Bridged conference call.
The phone must be configured to automatically conference in "privileged" users to an existing call. A phone at either end of the call becomes the bridge. The privileged user's phone discovers the call through some means (i.e. via proxy server, ACD application etc.).

3.2.3 Conference – Bridged

Optional
Extension(s): *none*
Entity(s): *none*

Definition
A conference call is setup such that one phone acts as a conference bridge, mixing the audio media for the call. The other phones in the call are added directly or joined via consultative calls on the bridging phone.

Solution
The non-bridge phones set up normal two party calls with the bridging phone and are not "aware" that the calls are bridged. The bridging of the conference call becomes a higher level function on one of the phones.

Call control extensions are only needed if the bridging phone/user wishes to get out of the call. In this case the bridging phone transfers [*call control doc. Section nn*] the remaining parties to another bridging phone. Alternatively the conference call ends when the party on the bridging phone hangs up.

Note
Only the bridging phone needs to support this feature. The other parties need only support RFC 2543.

The bridged conference can also be created using a independent (non-phone) bridge. This will typically be the case for larger conference calls. However this is not very interesting from the phone's perspective as it merely look like a two party call between a phone and the bridge.

3.2.4 Conference –Fully Meshed – Add On, Join w/ Consultative Call

Optional
Extension(s): *Call Control*
Entity(s): *none*

Definition
A conference call is set up such that each phone in the call sends and receives audio media from all of the other phones in the call. . Phones are added in the call directly or joined via consultative calls on any one of the phones in the call.

Solution
[*call control doc. Section nn*]

Note
All parties to the call must support this feature (part of the call control extensions) and provide media mixing for the incoming streams.

3.2.5 Multiple Station Appearances – Ring

Optional
Extension(s): *none*
Entity(s): *forking proxy*

Definition
(AKA Shared Extension on Multiple Phones) Call rings at multiple stations (phones) for a single user

Solution
Forking proxy

3.2.6 Multiple Station Appearances – Indication

Optional
Extension(s): *subscribe/notify*
Entity(s): *Proxy and Subscription Server(s)*

Definition
(AKA Shared Extension on Multiple Phones) Call is indicated as active at multiple stations (phone), although only one phone need participate in the call.

Solution
On initialization, the phone SUBSCRIBEs for the call state of a particular SIP URL (e.g. extension) with the relevant subscription server. The proxy server and the subscription server may be distinct entities. The separation and communication between these two logical server entities is beyond the scope of this document. The server sends a NOTIFY for the current state (i.e. IDLE, RINGING, ESTABLISHED) and also sends NOTIFY message when the call state changes.

3.3 Out of Scope Features

This section identifies features that are beyond the scope of this document. This section will obviously not be comprehensive. The point is to identify why some common features are not candidates for inclusion in the specification. In general these features are excluded as they are more appropriately defined by market demand and vendor innovation.

Reasons for Exclusion

- UI – The feature is a user interface construct which does not require SIP constructs beyond those in the baseline.
- Funct – The feature is a higher level function that can be implemented using other SIP phone features.
- Config – The feature is related to providing configuration information for a phone and is a topic for future study.
- Net – The feature is a Network Service solution provided via SIP registration, redirect and/or proxy servers and only requires baseline SIP functionality on the SIP Phone.

Out of Scope Feature	**Reason(s) for Exclusion**
Caller ID	Funct, UI
Log-In, Log-Out	Funct, UI, Config
Phone Set Relocation	Config
System Speed Dial	Net
Station Speed Dial	Funct, UI
Redial	UI
Do Not Disturb	Funct (on top of forwarding)
Automatic Call Screening, Filtering, Selective Forwarding	Funct (on top of forwarding)
Time Dependent Forwarding/Screening	Funct (on top of forwarding)
Mute – near end	Funct, UI
Call Waiting	UI
911 Station Location Information	Config

4 For Future Study

4.1 Configuration Mechanism, Format and Content

A standard method for providing configuration information to a SIP Phone has not yet been defined. For the time being, we suggest noting that this area is for further study. Examples of configuration information that would be useful for a SIP phone include:

- Top-Level Dial Plan – allows the phone to determine when a user is done dialing such that a SIP URL can be constructed to send an INVITE. The dialing plan table might also identify which SIP server or address to use for certain digit combinations.
- Extension – to be used as a user ID for SIP registration.
- Identity – other user or device identification information which may be used for registration or retrieving additional configuration information.
- Station Speed Numbers – specific to the user
- Station Restrictions – relates to the top-level dial plan as to where a specific user of phone may call.
- Proxy Server(s) – which proxies to use for passing through a firewall, etc. This may relate to top-level dial plan.

5 Issues

How should we handle features which can be implemented either on the phone or in the service network?

- Make support of feature in service network mandatory, on phone optional?
- Should we describe both solutions or just the phone solution?

Does it make sense to create one or more IP Phone extensions containing only IP Phone capabilities, such that a SIP IP phone does not have to implement all of the capabilities in each of the extensions which implement some required IP Phone features? IP phones are very cost sensitive. Forcing a vendor to implement the complete capability set in all of the extensions referenced in this document could be costly.

Is a mechanism needed for remotely controlling or receiving event notifications related to:
Function Keys
Soft Keys
Display

6 References

[1] Session Initiation Protocol (RFC 2543), by M. Handley, et al., March 1999, IETF.

[2] Implementing Intelligent Network Services with the Session Initiation Protocol (CUCS-002-99), by J. Lennox, H. Schulzrinne, T. F. La Porta, January 1999, Columbia University

[3] Requirements fir SIP Servers and User Agents (V2.0), by Henning Schulzrinne, June 1, 1999, Columbia University

[4] Signaling for Internet Telephony (CUCS-005-98), by H. Schulzrinne and J. Rosenberg, January 1998, Columbia University.

[5] SIP Call Control Services (draft-ietf-mmusic-sip-cc-00.txt), by H. Schulzrinne, J. Rosenberg, March 13, 1998, IETF.

[6] SIP Caller Preferences and Callee Capabilities (draft-ietf-mmusic-sip-caller-00.txt), by H. Schulzrinne, J. Rosenberg, February 26, 1999, IETF

[7] The PINT Profile of SIP and SDP: aProtocol for IP Access to Telephone Call Services (draft-ietf-pint-profile-04.txt), by L. Conroy, S. Petrack, March 26, 1999

7 Revision History

Version	Date	Author	Comments
0.1	18 June 99	D. Petrie	Initial Version with comments and input from J. Rosenberg, H. Schulzrinne, R. Schaaf