         A Taxonomy of Session Initiation Protocol (SIP) Back-to-Back User Agents

Abstract

   In many SIP deployments, SIP entities exist in the SIP signaling path
   between the originating and final terminating endpoints, which go
   beyond the definition of a SIP proxy, performing functions not
   defined in Standards Track RFCs. The only term for such devices
   provided in RFC 3261 is for a Back-to-Back User Agent (B2BUA), which
   is defined as the logical concatenation of a SIP User Agent Server
   (UAS) and User Agent Client (UAC).

   There are numerous types of SIP B2BUAs performing different roles in
   different ways; for example, IP Private Branch Exchanges (IPBXs),
   Session Border Controllers (SBCs), and Application Servers (ASs).
   This document identifies several common B2BUA roles in order to
   provide taxonomy other documents can use and reference.

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1. Introduction

   In current SIP deployments, there are numerous forms of Back-to-Back
   User Agents (B2BUAs), operating at various levels of transparency and
   for various purposes, with widely varying behavior from a SIP
   perspective. Some act as pure SIP proxies and only change to the
   role of B2BUA in order to generate BYEs to terminate dead sessions.
   Some are full User Agent stacks with only high-level event and
   application logic binding the User Agent Server (UAS) and User Agent
   Client (UAC) sides. Some B2BUAs operate only in the SIP signaling
   plane, while others participate in the media plane as well.

   As more SIP domains are deployed and interconnected, the probability
   of a single SIP session crossing multiple B2BUAs at both the
   signaling and media planes increases significantly.

   This document provides a taxonomy of several common B2BUA roles so
   that other documents may refer to them using their given names
   without redefining them in each document.

2. Terminology

   The following terms are defined in [RFC3261], Section 6.

   B2BUA: a SIP Back-to-Back User Agent, which is the logical
             combination of a User Agent Server (UAS) and User Agent
             Client (UAC).

   UAS: a SIP User Agent Server.

   UAC: a SIP User Agent Client.

3. B2BUA Role Types

   Within the context of this document, the classification refers to a
   B2BUA role, not a particular system type. A given system type may
   change its role in the middle of a SIP session, for example, when a
   stateful proxy tears down a session by generating BYEs or when an SBC
   [RFC5853] performs transcoding or REFER termination.

   Furthermore, this document defines "B2BUA" following the definition
   provided in [RFC3261], which is the logical concatenation of a UAS
   and UAC. A typical centralized conference server, for example, is
   not a B2BUA because it is the target UAS of multiple UACs, whereby
   the UACs individually and independently initiate separate SIP
   sessions to the central conference server. Likewise, a third-party
   call control transcoder, as described in Section 3.1 of [RFC5369], is

not a B2BUA, whereas an inline (conference bridge) transcoder based
on [RFC5370] is a B2BUA.

3.1. Signaling Plane B2BUA Roles

A signaling plane B2BUA is one that operates only on the SIP message
protocol layer and only with SIP messages and headers but not with
the media itself in any way. This implies that it does not modify
the Session Description Protocol (SDP) bodies, although it may save
them and/or operate on other MIME body types. This category is
further subdivided into specific roles as described in this section.

It should be noted that there is a large variety of modifications
made by "signaling plane B2BUAs".

3.1.1. Proxy-B2BUA

A Proxy-B2BUA is one that appears, from a SIP perspective, to be a
SIP proxy based on [RFC3261] and its extensions, except that it
maintains a sufficient dialog state to generate in-dialog SIP
messages on its own and does so in specific cases. The most common
example of this is a SIP proxy that can generate BYE requests to tear
down a dead session.

A Proxy-B2BUA does not modify the received header fields such as To,
From, or Contact, and it only modifies the Via and Record-Route
header fields following the rules in [RFC3261] and its extensions.
If a Proxy-B2BUA can generate in-dialog messages, then it will also
need to modify the CSeq header after it has generated its own. A
Proxy-B2BUA neither modifies nor inspects MIME bodies (including
SDP), does not have any awareness of media, will forward any method
type, etc.

3.1.2. Signaling-only

A Signaling-only B2BUA is one that operates at the SIP layer but in
ways beyond those of [RFC3261] proxies, even for normally forwarded
requests. For example, such a B2BUA might replace the Contact URI,
modify or remove all Via and Record-Route headers, modify the To and
From header fields, modify or inspect specific MIME bodies, etc. No
SIP header field is guaranteed to be copied from the received request
on the UAS side to the generated request on the UAC side.

An example of such a B2BUA would be some form of an Application
Server and a PBX, such as a server that locally processes REFER
requests and generates new INVITEs on behalf of the REFER's target.
Another example would be a privacy service proxy [RFC3323] performing
the 'header' privacy function.

3.1.3. SDP-Modifying Signaling-only

   An SDP-Modifying Signaling-only B2BUA is one that operates in the
   signaling plane only and is not in the media path, but it does modify
   SDP bodies and is thus aware of and understands SDP syntax and
   semantics. In some cases, some Application Servers and PBXs act in
   this role, for example, to remove certain codec choices or merge two
   media endpoints into one SDP offer.

   These B2BUAs don't do anything that changes the path that the media
   takes (in particular, they don't insert themselves on the media
   path), but they may make SDP changes that affect what is sent on the
   media plane.

3.2. Signaling/Media Plane B2BUA Roles

   A signaling/media plane B2BUA is one that operates at both the SIP
   and media planes and not only on SIP messages but also on SDP and
   potentially the Real-time Transport Protocol (RTP) / the Real-Time
   Control Protocol (RTCP) [RFC3550] or other media. Such a B2BUA may
   or may not replace the Contact URI, modify or remove all Via and
   Record-Route headers, modify the To and From header fields, etc. No
   SIP header field is guaranteed to be copied from the received request
   on the UAS side to the generated request on the UAC side, and SDP
   will also be modified.

   An example of such a B2BUA would be a Session Border Controller (SBC)
   performing the functions defined in [RFC5853], a B2BUA transcoder as
   defined in [RFC5370], a rich-ringtone Application Server, or a
   recording system. Another example would be a privacy service proxy
   [RFC3323] performing the 'session' privacy function.

   Note that a signaling/media plane B2BUA need not be instantiated in a
   single physical system, but it may be decomposed into separate
   signaling and media functions.

   The signaling/media plane B2BUA category is further subdivided into
   specific roles as described in this section.

3.2.1. Media Relay

   A B2BUA that performs a media-relay role is one that terminates the
   media plane at the IP and transport (UDP/TCP) layers on its UAS and
   UAC sides, but neither modifies nor restricts which forms of payload
   are carried within the packets. Rather, the payload is transparently
   copied from one side to the other. Such a role may or may not
   support only UDP, only TCP, both UDP and TCP, as well as other
   transport types. It may also involve policing the IP packets to fit

within a bandwidth limit or converting from IPv4 to IPv6, or vice
versa. This is typically similar to NAT behavior, except a NAT
operating in both directions on both the source and destination
information; it is often found as the default behavior in SBCs.

3.2.2. Media Aware

A B2BUA that performs a media-aware role is similar to a media relay
except that it inspects and potentially modifies the payload carried
in UDP or TCP (as it could be RTP or RTCP [RFC3550]), but it does not
at a codec or higher layer. An example of such a role is a Secure
Real-time Transport Protocol (SRTP) [RFC3711] terminator, which does
not need to care about the RTP payload but does care about the RTP
header; or, a device that monitors RTCP for QoS information; or, a
device that multiplexes/demultiplexes RTP and RTCP packets on the
same 5-tuple.

3.2.3. Media Termination

A B2BUA that performs a media-termination role is one that operates
at the media payload layer, such as RTP/RTCP codec or the Message
Session Relay Protocol (MSRP) message layer and higher. Such a role
may only terminate/generate specific RTP media, such as dual-tone
multi-frequency (DTMF) packets [RFC4733], or it may convert between
media codecs or act as a Back-to-Back MSRP [RFC4975] agent. This is
the role performed by transcoders, conference servers based on
[RFC5366], etc.

4. Mapping SIP Device Types to B2BUA Roles

Although the B2BUA roles defined previously do not define system
types, as discussed in Section 3, some discussion of what common
system types perform which defined roles may be appropriate. This
section provides such a 'mapping' for general cases to aid in
understanding of the roles.

4.1. SIP PBXs and Softswitches

SIP-enabled Private Branch Exchanges (SIP PBXs) and softswitches are
market category terms and are not specified in any standard. In
general, they can perform every role described in this document at
any given time based on their architecture or local policy. Some are
based on architectures that make them the equivalent of a SIP UAS and
UAC connected with a logical Primary Rate Interface (PRI) loopback;
others are built as a SIP proxy core with optional modules to "do
more".

4.2. Application Servers

   Application Servers are a broad marketing term and are not specified
   in any standard in general, although the Third Generation Partnership
   Project (3GPP) and other organizations specify some specific
   Application Server functions and behaviors. Common examples of
   Application Server functions are message-waiting indicators (MWIs),
   Find Me/Follow Me services, privacy services, call center Automatic
   Call Distributor (ACD) services, call screening, and Voice Call
   Continuity (VCC) services. Some only operate in the signaling plane
   in either Proxy-B2BUA or Signaling-only B2BUA roles; others operate
   as full Media-termination B2BUAs, such as when providing Interactive
   Voice Recognition (IVR), rich ringtones, or integrated voicemail
   services.

4.3. Session Border Controllers

   Session Border Controllers (SBCs) are a market category term and are
   not specified in any standard. Some of the common functions
   performed by SBCs are described in [RFC5853], but in general, they
   can perform every role described in this document at any given time
   based on local policy. By default, most SBCs are either Media-relay
   or Media-aware B2BUAs and replace the Contact URI; remove the Via and
   Record-Route headers; modify Call-ID, To, From, and various other
   headers; and modify SDP. Some SBCs remove all headers, all bodies,
   and reject all method types unless explicitly allowed by local
   policy; other SBCs pass all such elements through unless explicitly
   forbidden by local policy.

4.4. Transcoders

   Transcoders perform the function of transcoding one audio or video
   media codec type to another, such as G.711 to G.729. As such, they
   perform the Media-termination role, although they may only terminate
   media in specific cases of codec mismatch between the two ends.
   Although [RFC5369] and [RFC5370] define two types of SIP transcoders,
   in practice, a popular variant of the inline conference bridge model
   [RFC5370] is to behave as a SIP B2BUA without using the resource-list
   mechanism but rather simply routing a normal INVITE request through a
   B2BUA with a built-in transcoder. SIP transcoder architectures are
   based on everything from SIP media servers and SBCs to looped-back
   Time Division Multiplexing (TDM) gateways, and thus run the gamut
   from replacing only specific headers/bodies and SDP content needed to
   perform their function to replacing almost all SIP headers and SDP
   content. Some transcoders save and remove SDP offers in INVITEs from
   the UAC, and wait for an offer in the response from the UAS, similar

   to a Third Party Call Control (3PCC) model; others just insert
   additional codecs in SDP offers and only transcode if the inserted
   codec(s) is selected in the answer.

4.5. Conference Servers

   In general, conference servers do not fall under the term "B2BUA" as
   defined by this document, since typically they involve multiple SIP
   UACs initiating independent SIP sessions to the single conference
   UAS. However, a conference server supporting [RFC5366], whereby the
   received INVITE triggers the conference focus UAS to initiate
   multiple INVITEs as a UAC, would be in a Media-termination B2BUA role
   when performing that function.

4.6. P-CSCF and IBCF Functions

   The Proxy-Call Session Control Function (P-CSCF) and the
   Interconnection Border Control Function (IBCF) are defined by 3GPP
   [IMS] standards, and when coupled with the IP Multimedia Subsystems
   (IMS) media plane gateways (IMS Access Gateway (AGW), Transition
   Gateway (TrGW), etc.), they typically form a logical Media-relay or
   Media-aware B2BUA role.

4.7. S-CSCF Function

   The Serving-Call Session Control Function (S-CSCF) is defined by 3GPP
   [IMS] standards and typically follows a Proxy-B2BUA role.

5. Security Considerations

   Security risks are specific to each type of B2BUA, so little can be
   said in general. Of course, adding extra systems in the
   communication path creates extra points of attack, reduces or
   eliminates the ability to perform end-to-end encryption, decreases
   the privacy of SIP communications, and complicates trust models.
   Thus, every B2BUA design requires particular attention to security
   analysis.

   A few general points can be made:

   1. The B2BUA processing of SDP and media packets is an impediment to
       the deployment of end-to-end SRTP and reduces the ability to
       deploy new, stronger forms of SRTP key exchange.

   2. The ability for B2BUAs to modify any SIP header field value and
       body impacts the ability to deploy SIP identity and message
       integrity.

   3. The management and configuration mechanisms of B2BUAs are a
       tempting point of attack and must be strongly defended.

   Further security considerations related to the functionality
   described in this document are addressed in the relevant references.

6. Informative References

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