

External Alert-Info

sipXproxy is able to add an Alert-Info Header to SIP INVITE Messages to tell phones to play a different ring tone for internal and external calls. By default this feature is enabled and your Proxy should look like this:

Alert-Info

The Alert-Info header field can trigger alternate ringer sounds on many telephones, it is used to mark external phonecalls as such and trigger a different ringer sound if the phone is configured for it.

Enable for internal calls	<input type="checkbox"/>	(Vorgabe: Nicht selektiert)
Internal call field value	<input type="text" value="<http://internal.call>;in"/>	(Vorgabe: <http://internal.call>;info=alert-internal;x-line-id=0)
The default value was chosen to satisfy the requirements of Snom and Polycom phones and RFC 3281.		
Enable for external calls	<input checked="" type="checkbox"/>	(Vorgabe: Selektiert)
External call field value	<input type="text" value="<http://external.call>;in"/>	(Vorgabe: <http://external.call>;info=alert-external;x-line-id=0)
The default value was chosen to satisfy the requirements of Snom and Polycom phones and RFC 3281.		
When the field already exists:	<input type="text" value="Wert beibehalten"/>	(Vorgabe: Wert beibehalten)
Phones can be configured to send their own Alert-Info field or other redundant proxies could already have inserted the field.		

This proxy plugin looks at the FROM-Header and checks if the from domain is equal to the SIP Domain of the sipXcom system. If calls are not from the SIP domain, calls from a Gateway or Session Border Controller are thus detected as external calls.

Some phones use the information of the FROM-Header to fill their call history (e.g. Yealink T4x-Series). In this special case the phones (on re-dial from the phone) may try to call directly via the gateway and fail due to security settings.

To get this working it is necessary to set the SIP Domain inside the FROM-Header, which would otherwise break the Alert-Info detection.

An alternative way to mark this calls as external is to set an extra parameter in the SBC or Gateway in the FROM-Header which is called "x-sipx-alert-info=external".

You can add this tag or the Alert-Info Header directly in your gateway. For example, in a Patton Gateway you can't add a customer header but you can modify the FROM:

CONFIGURATION MENU

Home
Import/Export

Network
IP/DNS
NAT/NAPT
ACL
QoS
BGP
Virtual Router
DynDNS
DHCP Server
DHCP Relay
PPP Profiles

Telephony
Call-Router
SIP
VoIP Profiles
Tone Profiles
PSTN Profiles

Ports
Ethernet
E1/T1

Various
System
AAA
Time
Reports
Syslog

Save
Reload

About
License

Configuration Incoming Call Address Translation **Outgoing Call Address Translation** Status

INVITE Headers	From Where To Take The User Part	From Where To Take The Host/Port Part
To	<input type="radio"/> <input type="text"/> <input checked="" type="radio"/> Use called party URI or E.164 number (Default)	<input type="radio"/> <input type="text"/> <input type="radio"/> Use configured local host name <input type="radio"/> Use called party URI <input checked="" type="radio"/> Use configured remote host and port
From	<input type="radio"/> <input type="text"/> <input checked="" type="radio"/> Use calling party URI or E.164 number (Default)	<input checked="" type="radio"/> <!--sipx-alert-info=external--> <input type="text"/> <input type="radio"/> Use calling party URI (Default) <input type="radio"/> Use configured local host name <input type="radio"/> Use local IP address
P-Asserted-/P-Preferred-Identity	<input type="radio"/> <input type="text"/> <input checked="" type="radio"/> Use calling party URI or E.164 number (Default) <input type="radio"/> Use calling party URI (Default) <input type="radio"/> Use calling party redirecting E.164 number <input checked="" type="radio"/> Use calling party E.164 number <input type="text" value="single-primary"/>	<input checked="" type="radio"/> bm.test.iant.de <input type="text"/> <input type="radio"/> Use called party URI <input type="radio"/> Use configured local host name <input type="radio"/> Use local IP address
Diversion	Calling Redirecting Number	<input type="radio"/> <input type="text"/> <input type="radio"/> Don't send diversion header (Default) <input type="radio"/> Use configured remote host and port <input checked="" type="radio"/> Use called party URI <input type="radio"/> Use configured local host name

Apply ✓

If your devices are configured this way, the INVITE Message that your Phone will receive will look more or less like the following and the phone will play a different ring tone for external vs. internal calls.

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Session Initiation Protocol (SIP as raw text)
INVITE sip: [redacted];transport=TCP;x-sipX-nonat SIP/2.0\r\n
Record-Route: [redacted];5060;lr;sipXecs-CallDest=INT;sipXecs-rs=%2Aauth%7E.%2Afrom%7E0DQ5Y2ExMjA2Nw%60%60%21fe5194024e6cf2c8198fe1f018:
Via: SIP/2.0 [redacted];branch=z9hG4bK-XX-59c8RrERqqtmrzFEPafG7lWI3A\r\n
Via: SIP/2.0 [redacted];branch=z9hG4bK-XX-59b9xJ_39MTRAPm1WG0HK0Sr_g~hDIJUkAAQ`Ntk_6TLXJOLg\r\n
Via: SIP/2.0 [redacted];branch=z9hG4bK477deee6fe5e09b07\r\n
Max-Forwards: 18\r\n
From: [redacted]<[redacted]@iant.de;x-sipx-alert-info=external>;tag=849ca12067\r\n
To: [redacted]\r\n
Call-Id: 306b345679986838\r\n
Cseq: 5411 INVITE\r\n
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, REFER, REGISTER\r\n
Contact: [redacted];ob;transport=tcp;x-sipX-nonat>\r\n
Supported: replaces, outbound, path\r\n
User-Agent: Patton SM4960 1E30V 00A0BA05C50E R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18\r\n
Content-Type: application/sdp\r\n
Content-Length: 253\r\n
Date: Tue, 17 Jan 2017 15:11:52 GMT\r\n
Expires: 30\r\n
Alert-Info: <http://external.call>;info=alert-external;x-line-id=0\r\n
\r\n
v=0\r\n
o=MxSIP 0 151 IN IP4 [redacted]\r\n
s=SIP Call\r\n
c=IN IP4 [redacted]\r\n
t=0 0\r\n
m=audio 5050 RTP/AVP 8 0 91 101\r\n
a=rtpmap:8 PCMA/8000\r\n
a=rtpmap:0 PCMU/8000\r\n
a=rtpmap:91 X-CLEAR-CHANNEL/8000\r\n
a=rtpmap:101 telephone-event/8000\r\n
a=fmtp:101 0-16\r\n
a=sendrecv\r\n

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