SIP Voice Configlet

November 2022

Version 8.0

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1 Common Settings

1.1 Audio (RTP)

The audio codecs permitted by eir SIP trunks are G.722, G.711a and G.729. Calls placed over eir SIP trunks will have unsupported codecs removed from the SDP section.

RFC2833 support for sending DTMF is mandatory.

Note: G.729 codec support is a mandatory minimum requirement for calling over eir SIP trunks.

Codec	Bandwidth (inc ethernet headers)*	EF Bandwidth	Recommended Packetisation window
G.722	97 kbps	80 kbps	20 ms
G.711a	97 kbps	80 kbps	20 ms
G.729	30 kbps	24 kbps	30 ms

* assuming Ethernet 802.3 with 802.1q header, as used by eir's IPVPN service

Calls sourced from eir will use media ports in the range 20000 to 39999, but eir will accept media ports outside this range if requested by customer equipment.

1.2 Signalling

- Equipment connecting to eir SIP trunks requires SIP Provisional Message Reliability RFC3262 (PRACK).
- Equipment connecting to eir SIP trunks must support the SIP UPDATE (RFC 3311), and the SIP REFER method (RFC 3515).
- SIP messages should be sent using the transport agreed at the time the order was raised. Both UDP and TCP will be accepted but responses will always be sent using the transport used for the original order.
- SIP messages should be sent using UDP. TCP will be accepted but responses will always be sent using UDP on port 5060.
- For service keep-alives and failover monitoring SIP OPTIONS ping must be supported. The eir SBC will not respond to ICMP ping requests.
- For outbound calls to PSTN, that is numbers not in the private numbering plan on SIP trunk, outbound INVITEs must follow this format:
  Calling number must be a valid number for the site, as provided in number range in supporting configlet email. E.g. '+35317001234'
  Called number in SIP 'To:' header must be in national or international format, with preceding escape code digit (default 0), e.g.: '0016006001' or '00035316006001'.

  INVITE sip:0016006001@159.134.113.84;user=phone SIP/2.0
  Allow: PRACK,ACK,CANCEL,BYE,SUBSCRIBE,NOTIFY,INVITE,REFER,OPTIONS,PUBLISH,INFO,UPDATE,REGISTER
  Allow-Events: hold,talk
  Call-ID: OA2BC04E68353768910399 13DA8088
  Contact: <sip:+35317001234@192.168.254.10;user=phone>
  Content-Type: application/sdp
  CSeq: 5644 INVITE
  From: "John Doe" <sip:+35317001234@192.168.254.10;user=phone>;tag=1C3B
  Max-Forwards: 70
  P-Asserted-Identity: "John Doe" <sip:+35317001234@192.168.254.10;user=phone>
  Privacy: none
  Supported: replaces, 100rel
  To: <sip:0016006001@159.134.113.84;user=phone>
  Via: SIP/2.0/UDP 192.168.254.10;branch=z9hG4bK-5794-3CDD
  Content-Length: 187

v=0
o=User1 3742198277 3742198277 IN IP4 192.168.254.10
s=Session SDP
c=IN IP4 192.168.254.10
t=0 0
m=audio 16432 RTP/AVP 18
a=rtpmap:9 G722/8000
a=ptime:20
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:30

- For inbound calls from PSTN, an example of a valid inbound SIP INVITE is:

INVITE sip:+35317071234;tgrp=HSQ;trunk-context=hsqvoip.ngv.eircom.net@sbc.ngv.eircom.net;user=phone SIP/2.0
Via: SIP/2.0/UDP 159.134.113.212:5060;branch=z9hG4bK37dmm530109gfcbgi4h0.1
To: "35317071234"<sip:+35317071234;tgrp=HSQ;trunk-context=hsqvoip.ngv.eircom.net@sbc.ngv.eircom.net;user=phone>
From: <sip:+35315007070@dbnb.ngv.eircom.net;user=phone>;tag=SDdccbe01-1447795723-1418033462428-
Call-ID: SDdccbe01-b1f08a12f80a07b779427f9405bb445e-a04bm71
CSeq: 347127375 INVITE
Contact: <sip:+35315007070@159.134.113.212:5060;transport=udp>
Content-Type: application/sdp
Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Supported: 100rel

- Where users have telephones allocated to an existing PSTN line for inbound calls, but outbound calls will be sent over SIP trunk then these numbers are configured as 'mirrored' numbers on the SIP trunk. A range of 076 numbers will be allocated and these have to be sent as the calling number in order to display the number on the PSTN line to the called party.

2 SIP over Private VPN Connection (unregistered)

2.1 SIP over Private VPN Connection

- Registration is not required; service is over a private VPN and is tied to specific IP addresses given at time or order. This is SIP Connect 'Static Mode' compliant.
- SIP gateway addresses: 159.134.113.84:5060 & 159.134.113.212:5060
- SIP requests are only accepted from the IP address(es) given on order, as provided in supporting SIP service configlet email.
- When testing connectivity SIP gateways will not respond to ICMP PING, IP connectivity may be verified by pinging network IP gateways at 159.134.113.83 and 159.134.113.211. eir can enable SIP OPTIONS ping toward customer equipment to allow
- Packets should be marked with DSCP QoS as follows.
  EF used for voice traffic
  AF31 used for call signalling

3 SIP over public Internet Connection (registered)

3.1 SIP over public Internet Connection

- SIP proxy discovery using RFC3263 is recommended using domain **sipv-bt-pub.ngv.eircom.net**.
- Registration is required over internet connections before Calls may be received or made. The service follows the SIP Connect 'Registration Mode' standard.
- Registration can only be made to one eir SBC, once successfully registered all SIP messaging must be sent to the registered gateway only. Any SIP requests sent to other gateways will be rejected with 403 Forbidden responses.
- SIP messages should be sent using UDP using standard port 5060. TCP signalling will be accepted.
- For service keep-alives, and monitoring connectivity SIP OPTIONS ping is supported from the eir SIP gateways.
- As QoS is not supported over internet connections, it is important to verify that adequate bandwidth is available to calls at all times.
- Encryptions supported are TLSv1.2 or TLSv1.1. Support for SSLv3 and TLSv1 has been deprecated.

4 Registered SIP Trunk over Cisco

4.1 Config

```
!
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 mode border-element
 allow-connections sip to sip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 modem passthrough nse codec g711alaw
 sip
  asserted-id pai
  asymmetric payload dtmf
  sip-profiles 100
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g729r8 bytes 30
 codec preference 3 g722-64
!
!
voice class sip-profiles 100
 request INVITE sip-header SIP-Req-URI modify "; SIP/2.0" ";user=phone SIP/2.0"
!
!
!
!
voice class tenant 220
  registrar dns:sipv-bt-pub.ngv.eircom.net:5060 expires 60
  credentials username pVIMxxxxxx_TG1 password 6 xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx realm ngv.eircom.net
  authentication username pVIMxxxxx_TG1 password 6 xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
  sip-server dns:sipv-bt-pub.ngv.eircom.net
  outbound-proxy dns:sipv-bt-pub.ngv.eircom.net
!
dial-peer voice 100 voip
```

```
translation-profile outgoing outbound-to-eir-SIP
preference 1
destination-pattern .T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip tenant 220
dtmf-relay rtp-nte
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
!
```

5 SIP Diversion

5.1 SIP Diversion

For SIP diverts the following rules should be borne in mind:

- A diverted or deflected call is the same as any other originating call on the trunk, so there should not be any need to modify the values of the From, To, PAI or Contact headers compared to a normal outgoing call attempt
- The diversion information should only be conveyed in History-Info or Diversion headers, nowhere else.

eir's SIP trunking service requires a valid caller to be identified in order to route the call. For a diverted call the original PAI and From headers identify the original caller, so a Diversion or History-info header is added to identify the diverting party as the source of this call leg, e.g.:

```
Diversion: <"Bob" sip:+35312345678@ngv.eircom.net>;privacy=off;reason=unconditional;screen=no
```

The user part of the diversion header must follow the E164 format and be a valid number on the SIP trunk in order to route. The most common mistakes involve changing the From/PAI/Contact values to that of the original calling party. This results in the network not recognising the calling number as a valid SIP trunk customer (404 User Not Found) or triggers the network loop prevention protection (Forward Loop detected), resulting in a 603 Declined error response. Diversion headers when present must contain a valid User part in the URI with an E164 number assigned to the SIP trunk. This format for example will fail:

```
Diversion: <"Bob" sip:1234@ngv.eircom.net>;privacy=off;reason=unconditional;screen=no
```

There is an order of priority in which the network examines and validates the caller, with diversion related headers at the highest priority:

1. History-Info:
2. Diversion:
3. P-Asserted-Identity:
4. Remote-Party-ID:
5. From:

The two key requirements from the network perspective are that the To: header contains an identifiable call recipient and that the calling party is identifiable from the headers listed above and entitled to make such a call, including when CLI is restricted by diverting user.

The information presented in the History-Info or Diversion header takes precedence over other headers, so no further manipulation of values is necessary or desirable in order to complete a call divert.

5.1.1 Common Mistakes:

- Same value used in Diversion and PAI/From headers (forward loop)
- PAI and From headers contain original calling number which is not a valid SIP trunk user
- PAI contains anonymous as the caller for CLI presentation restricted calls
- Diversion header is present and contains an invalid user for the SIP trunk.
- Diversion contains 'anonymous' as user part when CLI is restricted.

5.1.2 Example INVITE of a successful diverted call with CLI presented

```
INVITE sip:008cccccccc@ngv.eircom.net:5060;user=phone;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.144.101.72:5060;branch=z9hG4bK2dmkq01090uuil7fb6f0.1
Via: SIP/2.0/TCP 10.xxx.yyy.zzz:5060;branch=z9hG4bK312BC12333
From: "Alice" <sip:+353bbbbbbbb@ngv.eircom.net>;tag=SDlqmc601-52F84424-403
To: "Bob" <sip:008cccccccc@ngv.eircom.net;user=phone>
Date: Tue, 28 May 2019 11:11:38 GMT
Call-ID: SDlqmc601-c3390d009ea8c73ef126dc1cf7b4fc0d-a04bm71
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 1534924416-0000065536-0000260035-0518392074
User-Agent: Cisco-SIPGateway/IOS-15.4.3.M3
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1559041898
Contact: <sip:+353bbbbbbbb;tgrp=VIMnnnnn-TG1;trunk-
context=xXXXXX.ngv.eircom.net@10.144.101.72:50
60;transport=tcp>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
P-Asserted-Identity: "B name" <sip:+353bbbbbbbb@ngv.eircom.net>
Diversion: "A
name"<sip:+353aaaaaaaa@ngv.eircom.net>;privacy=off;reason=unconditional;screen=
yes
Session-Expires: 1800
```

5.1.3 Example INVITE of a successful diverted call with CLI restricted

```
INVITE sip:008cccccccc@ngv.eircom.net:5060;user=phone;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.144.69.72:5060;branch=z9hG4bK4menu7006on0ov5n0ac0.1
Via: SIP/2.0/UDP 10.xxx.yyy.zzz:5060;branch=z9hG4bK131C2CBE9
From: <sip:anonymous@ngv.eircom.net>;tag=SD9p6p701-975BD64C-347
To: <sip:008cccccccc@ngv.eircom.net;user=phone>
Date: Tue, 28 May 2019 12:07:02 IRL
Call-ID: SD9p6p701-6a9a73615e8188bc9d7a5e73d197a4c6-a04bk71
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3069891712-0000065536-0000506832-0098770954
User-Agent: Cisco-SIPGateway/IOS-16.6.3
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1559041622
Contact: <sip:anonymous;tgrp=VIM88nnnnn-TG2;trunk-
context=xXXXXX.ngv.eircom.net@10.144.69.72:5060;t
ransport=tcp>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 65
P-Asserted-Identity: <sip:+353bbbbbbbb@ngv.eircom.net>
Privacy: id
Diversion: <sip:+353aaaaaaaa@ngv.eircom.net>;privacy=off;reason=unconditional;screen=no
Session-ID: 3b76088af1c55d19bc4ac2dea423fd13;remote=00000000000000000000000000000000
```