NovaTel Systems
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NRS SIP REDIRECT MESSAGING.

SIP Messaging

Introduction

NRS is NovaTel's state of the art, SIP Redirect Server. Based on RFC 3261 it can be used as a routing server for any SIP Network. The NRS Server can be easily integrated with any switch supporting RFC 3261 30X based Redirect. The following shows example SIP messaging used between the NRS SIP Redirect Server and a vendor Switch.

Example

In this example the following applies;

The Vendor Switch is working on IP: 77.88.145.214

The NRS Sip Redirect Sever is working on IP: 111.88.145.214

The customer prefix is 970029 and the destination number is 8801768131111. Hence the combined number is: 970029#8801768131111

The following is the full dump of the IP packets involved in the messaging. Colouring has been used to more easily identify the components in the messages. The actual SIP Invite, SIP Redirect and SIP Acknowledgment messages can be seen at the third indentation as laid out below.

SIP Invite Message

```
11:10:55.780256 IP (tos 0x0, ttl 255, id 37918, offset 0,
flags [DF], proto UDP (17), length 727)
    77.88.145.214.5060 > 111.88.145.214.5060: SIP, length: 699
     INVITE sip:970029#8801768131111@111.88.145.214 SIP/2.0
     Via: SIP/2.0/UDP
77.88.145.214:5060;branch=z9hG4bK54944a41cfb86556
     From:
<sip:442074813400@77.88.145.214>;tag=27d54471afa712c1
     To: <sip:970029#8801768131111@111.88.145.214>
     Call-ID: 2f4071870994d47a@77.88.145.214
     CSeq: 20 INVITE
     Contact: <sip:NOVATEL@77.88.145.214>
     Max-Forwards: 70
     Allow: INVITE, ACK, BYE, CANCEL, INFO, UPDATE, REGISTER,
PRACK, OPTIONS, REFER, SUBSCRIBE, NOTIFY
     Content-Type: application/sdp
     Content-Length:   196

     v=0
     o=NOVATEL 123456 654321 IN IP4 77.88.145.214
     s=-
     c=IN IP4 77.88.145.214
     t=0 0
     m=audio 31870 RTP/AVP 18 101
     a=rtpmap:18 G729/8000/1
     a=rtpmap:101 telephone-event/8000/1
     a=fmtp:101 0-15
```

SIP Redirect Message

```
11:10:55.782942 IP (tos 0x0, ttl 64, id 0, offset 0, flags
[DF], proto UDP (17), length 532)
    111.88.145.214.5060 > 77.88.145.214.5060: SIP, length: 504
      SIP/2.0 302 Moved Temporarily
      Via: SIP/2.0/UDP
77.88.145.214:5060;branch=z9hG4bK54944a41cfb86556
      To:  <sip:970029#8801768131111@111.88.145.214>
      From:
<sip:442074813400@77.88.145.214>;tag=27d54471afa712c1
      call-id: 2f4071870994d47a@77.88.145.214
      CSeq: 20 INVITE
      Contact:  <sip:181#8801768131111@77.88.145.214:5060>,
<sip:182#8801768131111@77.88.145.214:5060>,
<sip:505#8801768131111@77.88.145.214:5060>,
<sip:501#8801768131111@77.88.145.214:5060>,
<sip:236#8801768131111@77.88.145.214:5060>
```

SIP Acknowledgement

```
11:10:55.783233 IP (tos 0x0, ttl 255, id 37919, offset 0,
flags [DF], proto UDP (17), length 335)
    77.88.145.214.5060 > 111.88.145.214.5060: SIP, length: 307
      ACK sip:970029#8801768131111@111.88.145.214 SIP/2.0
      Via: SIP/2.0/UDP
77.88.145.214:5060;branch=z9hG4bK54944a41cfb86556
      From:
<sip:442074813400@77.88.145.214>;tag=27d54471afa712c1
      To: <sip:970029#8801768131111@111.88.145.214>
      Call-ID: 2f4071870994d47a@77.88.145.214
      CSeq: 20 ACK
      Content-Length: 0
```

Explanation

The vendor switch uses the Contact field from the SIP 302 Redirect message to work out routing for this phone call.

```
Contact:  <sip:181#8801768131111@77.88.145.214:5060>,
 <sip:182#8801768131111@77.88.145.214:5060>,
 <sip:505#8801768131111@77.88.145.214:5060>,
 <sip:501#8801768131111@77.88.145.214:5060>,
 <sip:236#8801768131111@77.88.145.214:5060>
```

In this example the routing hunting list will look like:

181# - first choice carrier
182# - second choice carrier
505# - third choice carrier
501# - forth choice carrier
236# - fifth choice carrier

In other words, after receiving the SIP Redirect message, it will try to connect the call to 181# the first choice carrier, then if it fails, the switch tries the next carrier, 182# ... etc. If none of the carriers can connect the call, then a call is failed.