

asterisk_nortel-1140.pcap

SIP REGISTER

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4227 REGISTER
User-Agent	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;+sip.instance="<urn:uuid:00000000-0000-0000-...
Expires	120

[Frame 1] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

SIP 401 Unauthorized

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4227 REGISTER
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"5fb59ad5"

Frame 2 | 2010-10-10T09:20:23.632041Z

SIP REGISTER

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4228 REGISTER
User-Agent	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;+sip.instance="<urn:uuid:00000000-...
Auth-Scheme	Digest
Auth-User	"5001"
Realm	"asterisk"
Auth-URI	"sip:10.21.0.131"
Digest	"c6e9f472c8a2921a8ef-f6e00bbd08238"
Nonce	"5fb59ad5"
Algorithm	MD5
Expires	120

[Frame 3] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

SIP 200 OK

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4228 REGISTER
Expires	120
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;expires=120

[Frame 4] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

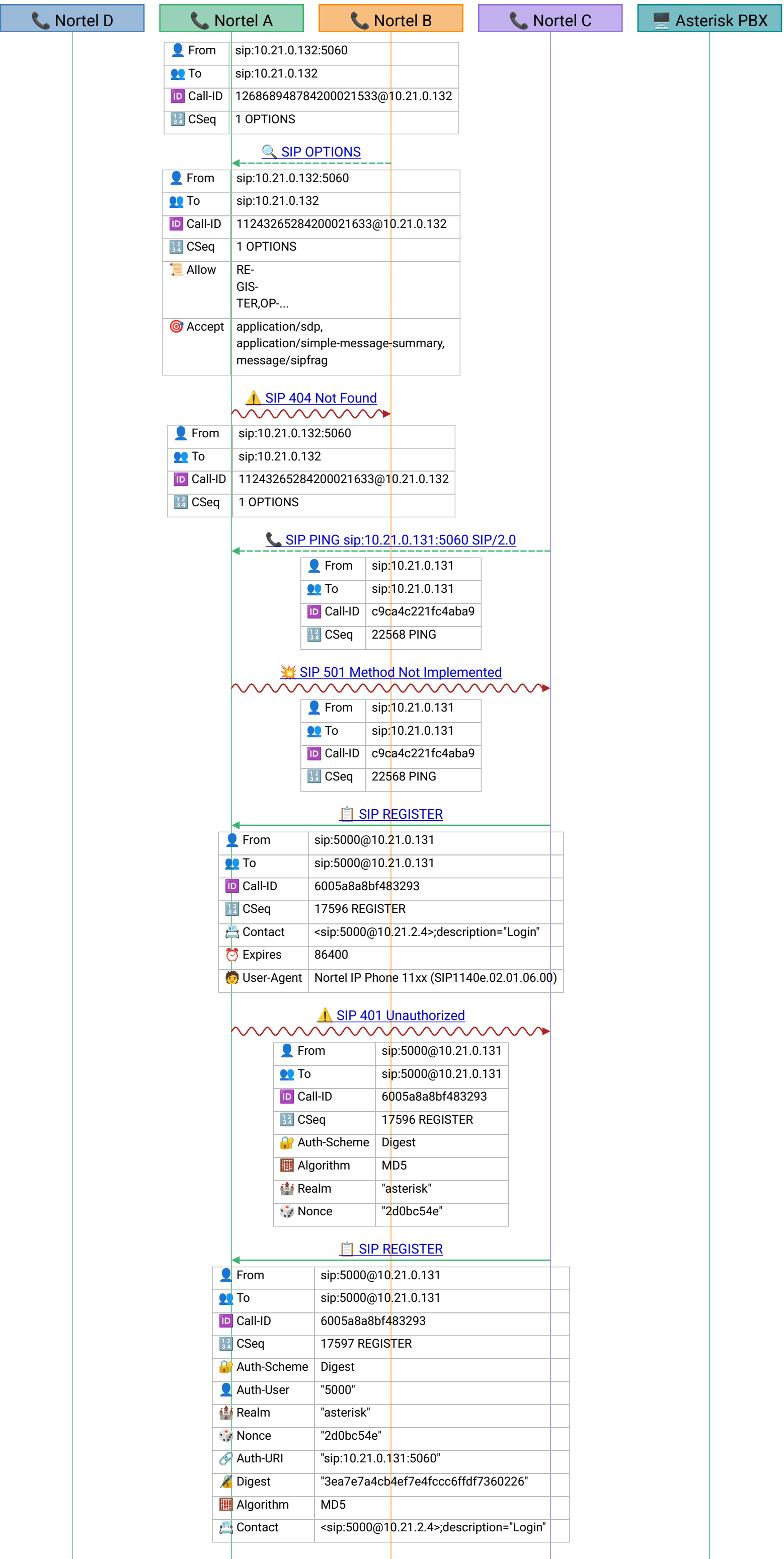
SIP OPTIONS

From	sip:10.21.0.132:5060
To	sip:10.21.0.132
Call-ID	126868948784200021533@10.21.0.132
CSeq	1 OPTIONS
Allow	RE-GIS-TER,OP...
Accept	application/sdp, application/simple-message-summary, message/sipfrag

Frame 5 | 2010-10-10T09:20:28.607226Z

SIP 404 Not Found

Frame 6 | 2010-10-10T09:20:28.615788Z



Frame 7 | 2010-10-10T09:21:28.683623Z

Frame 8 | 2010-10-10T09:21:28.693594Z

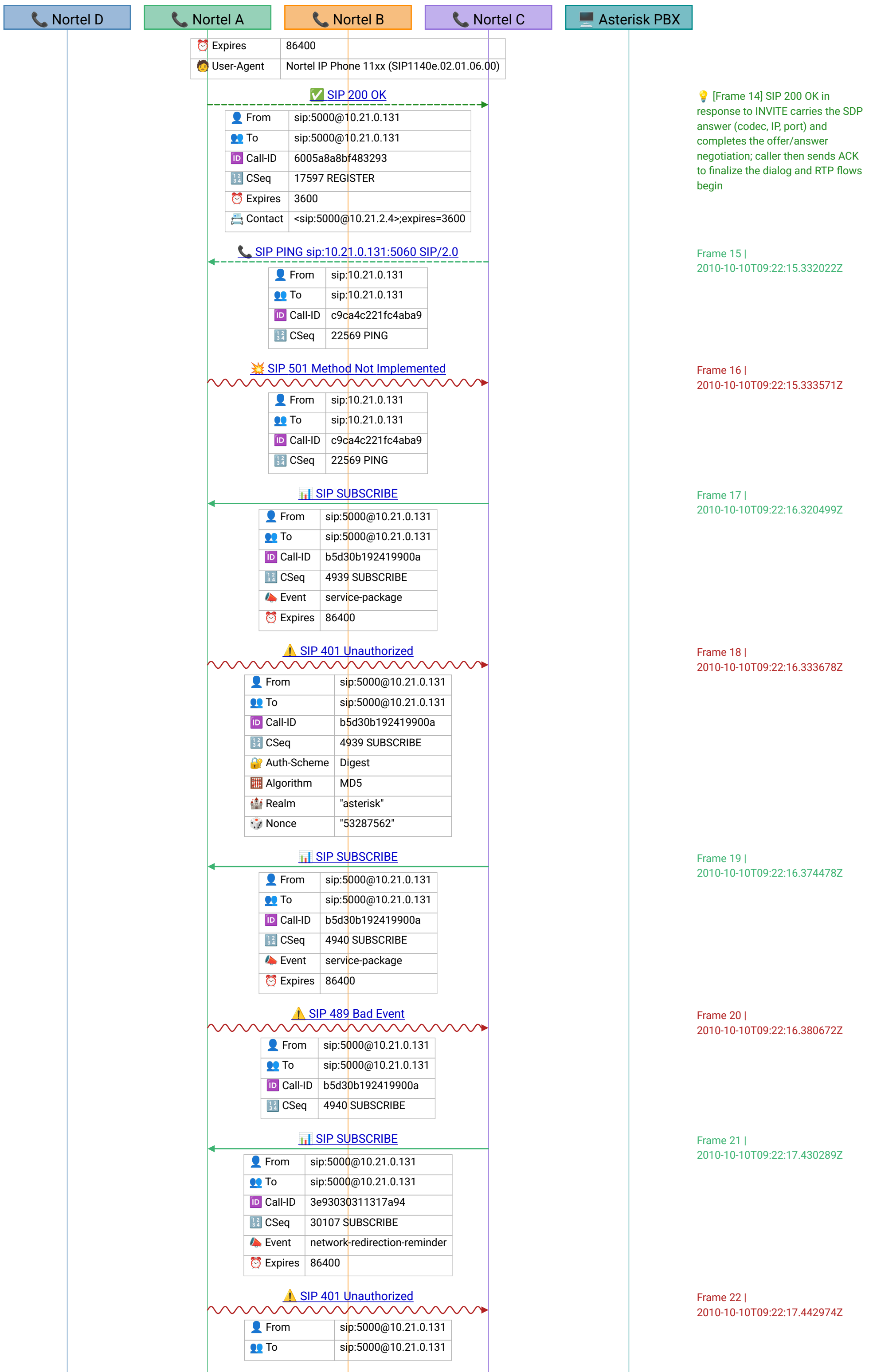
Frame 9 | 2010-10-10T09:22:14.663806Z

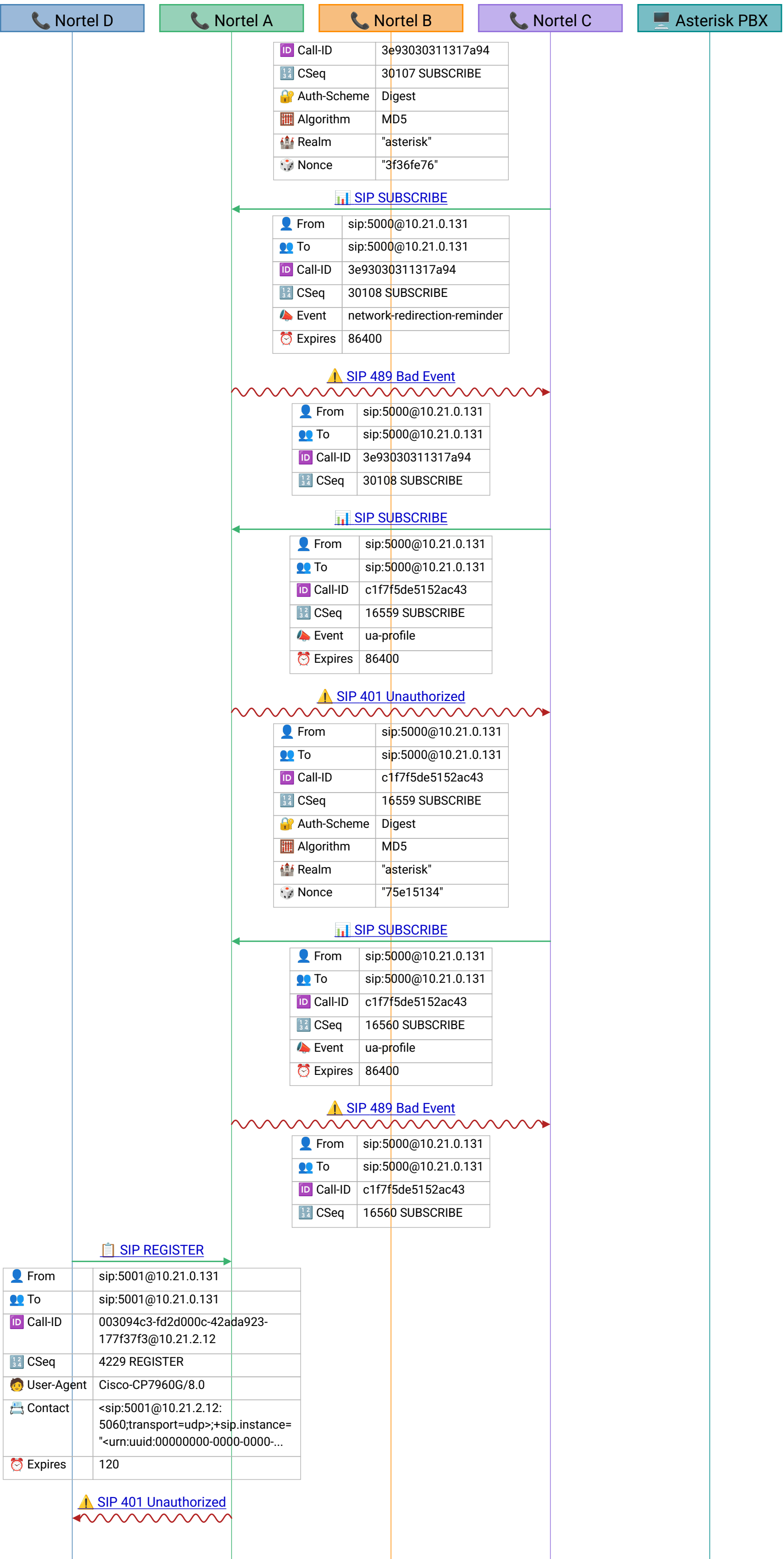
Frame 10 | 2010-10-10T09:22:14.677349Z

💡 [Frame 11] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

Frame 12 | 2010-10-10T09:22:15.083689Z

💡 [Frame 13] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval





Frame 23 |
2010-10-10T09:22:17.484622Z

Frame 24 |
2010-10-10T09:22:17.489782Z

Frame 25 |
2010-10-10T09:22:18.541648Z

Frame 26 |
2010-10-10T09:22:18.552358Z

Frame 27 |
2010-10-10T09:22:18.593911Z

Frame 28 |
2010-10-10T09:22:18.599122Z

💡 [Frame 29] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

Frame 30 |
2010-10-10T09:22:18.865Z

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖨 Asterisk PBX

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4229 REGISTER
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"7aa58970"

SIP REGISTER

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4230 REGISTER
User-Agent	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;+sip.instance="urn:uuid:00000000-...
Auth-Scheme	Digest
Auth-User	"5001"
Realm	"asterisk"
Auth-URI	"sip:10.21.0.131"
Digest	"505b-fe82346500-de5c2b1a53fbc7c737"
Nonce	"7aa58970"
Algorithm	MD5
Expires	120

SIP 200 OK

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4230 REGISTER
Expires	120
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;expires=120

SIP SUBSCRIBE

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	8e59ba99c98ada2f
CSeq	6861 SUBSCRIBE
Event	message-summary
Expires	86400

SIP 401 Unauthorized

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	8e59ba99c98ada2f
CSeq	6861 SUBSCRIBE
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"3b590cf2"

SIP SUBSCRIBE

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	8e59ba99c98ada2f
CSeq	6862 SUBSCRIBE
Event	message-summary
Expires	86400

SIP 404 Not found (no mailbox)

[Frame 31] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

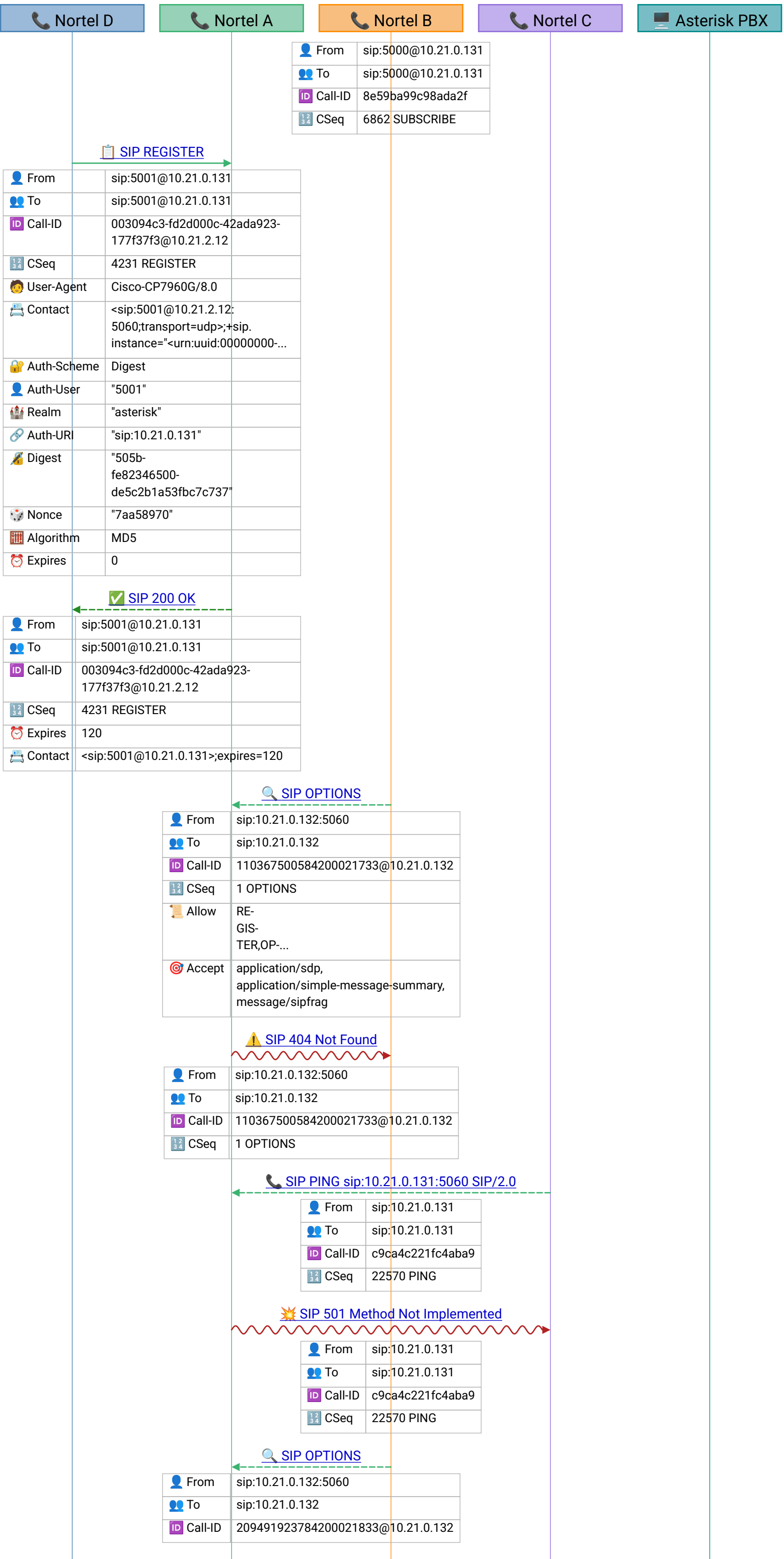
[Frame 32] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 33 | 2010-10-10T09:22:19.649879Z

Frame 34 | 2010-10-10T09:22:19.662237Z

Frame 35 | 2010-10-10T09:22:19.705835Z

Frame 36 | 2010-10-10T09:22:19.70851Z



From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	8e59ba99c98ada2f
CSeq	6862 SUBSCRIBE

SIP REGISTER

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4231 REGISTER
User-Agent	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>;+sip.instance="<urn:uuid:00000000-...
Auth-Scheme	Digest
Auth-User	"5001"
Realm	"asterisk"
Auth-URI	"sip:10.21.0.131"
Digest	"505b-fe82346500-de5c2b1a53fbc7c737"
Nonce	"7aa58970"
Algorithm	MD5
Expires	0

SIP 200 OK

From	sip:5001@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d000c-42ada923-177f37f3@10.21.2.12
CSeq	4231 REGISTER
Expires	120
Contact	<sip:5001@10.21.0.131>;expires=120

SIP OPTIONS

From	sip:10.21.0.132:5060
To	sip:10.21.0.132
Call-ID	110367500584200021733@10.21.0.132
CSeq	1 OPTIONS
Allow	RE-GIS-TER,OP...
Accept	application/sdp, application/simple-message-summary, message/sipfrag

SIP 404 Not Found

From	sip:10.21.0.132:5060
To	sip:10.21.0.132
Call-ID	110367500584200021733@10.21.0.132
CSeq	1 OPTIONS

SIP PING sip:10.21.0.131:5060 SIP/2.0

From	sip:10.21.0.131
To	sip:10.21.0.131
Call-ID	c9ca4c221fc4aba9
CSeq	22570 PING

SIP 501 Method Not Implemented

From	sip:10.21.0.131
To	sip:10.21.0.131
Call-ID	c9ca4c221fc4aba9
CSeq	22570 PING

SIP OPTIONS

From	sip:10.21.0.132:5060
To	sip:10.21.0.132
Call-ID	209491923784200021833@10.21.0.132

[Frame 37] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

[Frame 38] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

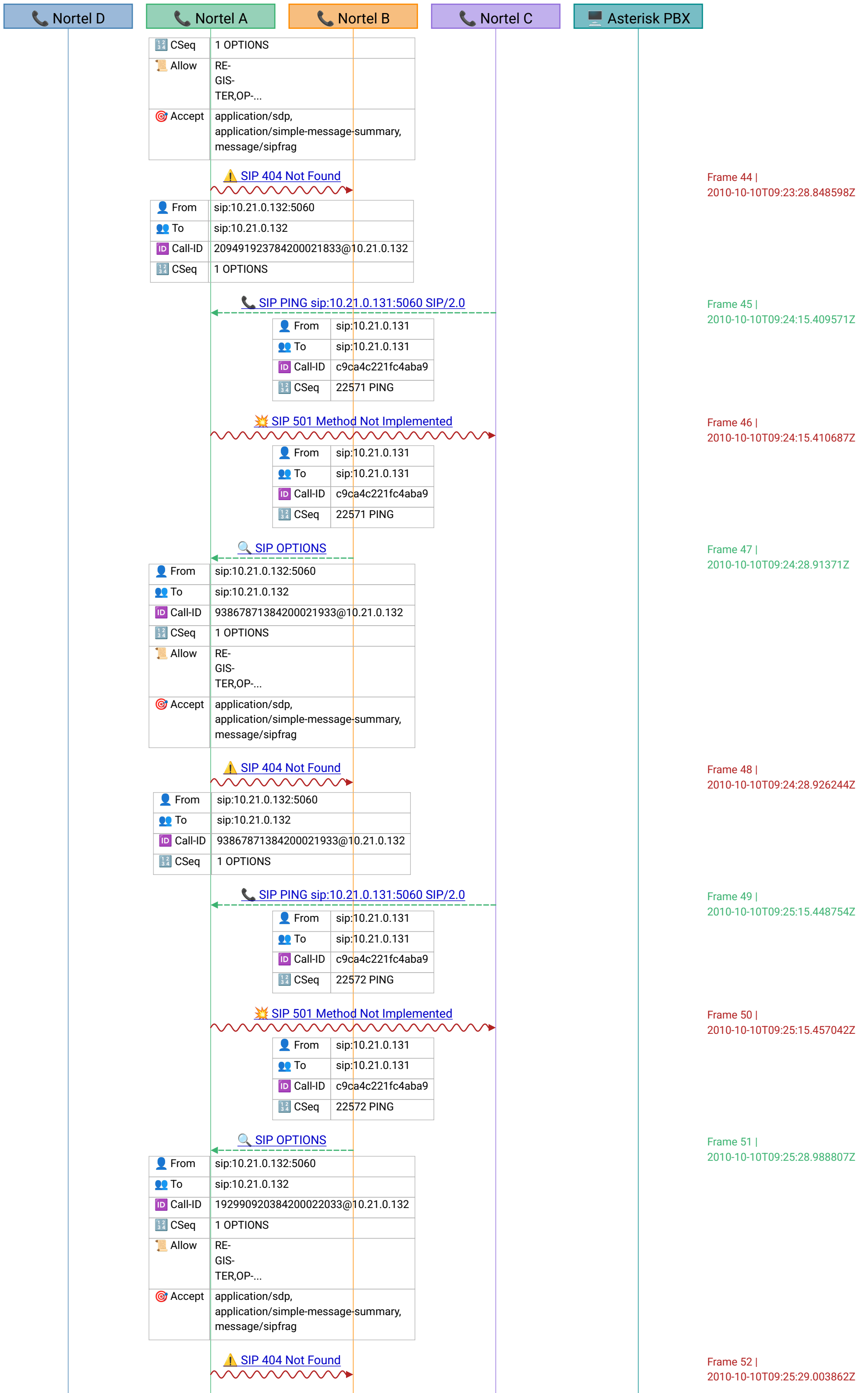
Frame 39 | 2010-10-10T09:22:28.759518Z

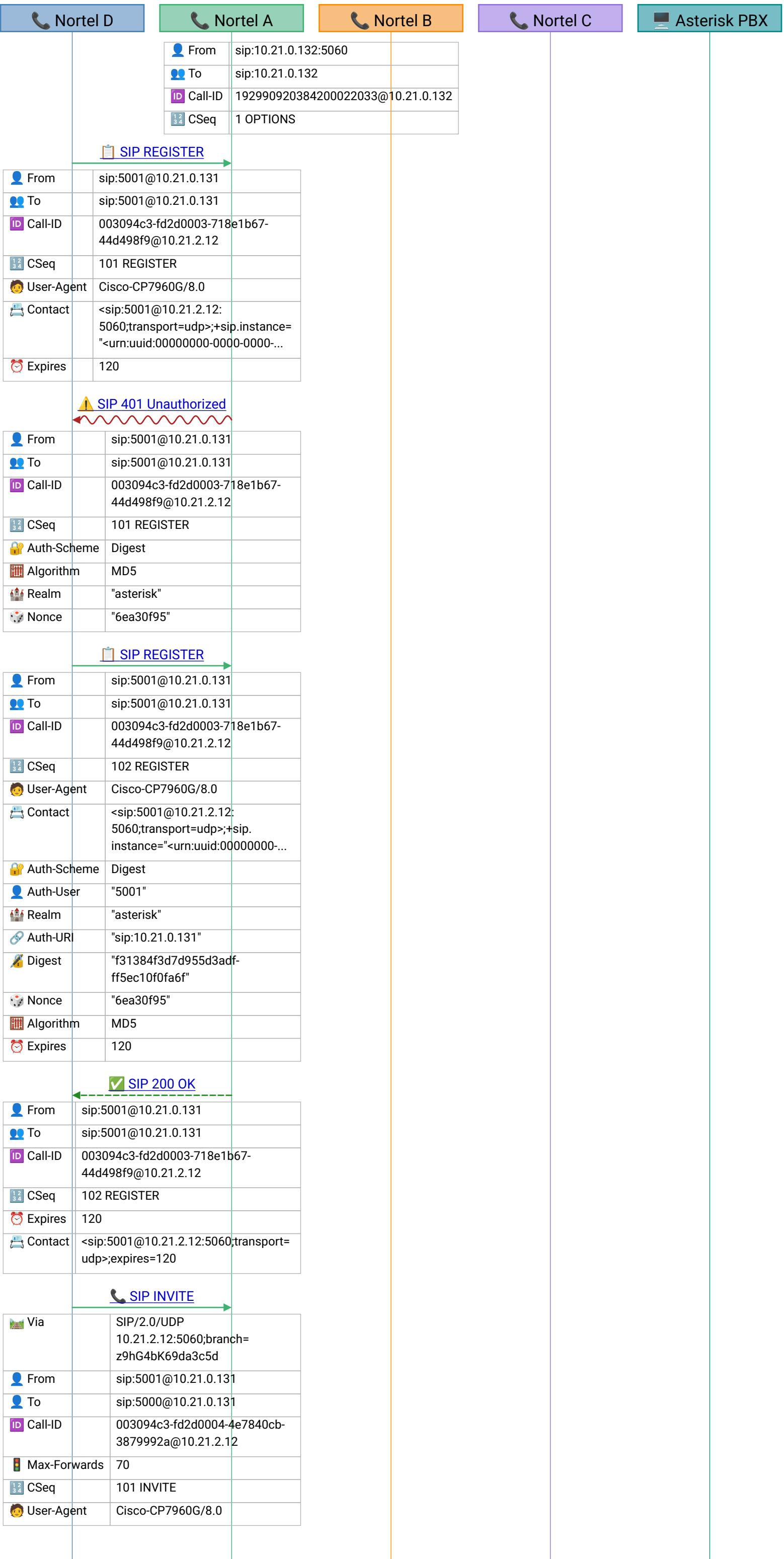
Frame 40 | 2010-10-10T09:22:28.771037Z

Frame 41 | 2010-10-10T09:23:15.369271Z

Frame 42 | 2010-10-10T09:23:15.380031Z

Frame 43 | 2010-10-10T09:23:28.836255Z





💡 [Frame 53] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

Frame 54 | 2010-10-10T09:25:51.941172Z

💡 [Frame 55] SIP REGISTER binds a user's SIP URI (AOR) to the contact address where they can be reached; typically uses HTTP Digest authentication (401/407 challenge-response) and refreshes via Expires interval

💡 [Frame 56] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

💡 [Frame 57] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖨️ Asterisk PBX

Contact	<sip:5001@10.21.2.12:5060;transport=udp>
SDP-Owner	Cisco-SIPUA 21409 0 IN IP4 10.21.2.12
SDP-Session	SIP Call
Media	audio 17164 RTP/AVP 0 8 18 101
SDP-Conn	IN IP4 10.21.2.12
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:18 G729/8000...
	4 more

⚠️ SIP 401 Unauthorized

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	101 INVITE
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"7081c3f8"

✅ SIP ACK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	101 ACK

📞 SIP INVITE

Via	SIP/2.0/UDP 10.21.2.12:5060;branch=z9hG4bK6b6597ee
From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
Max-Forwards	70
CSeq	102 INVITE
User-Agent	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>
SDP-Owner	Cisco-SIPUA 21409 0 IN IP4 10.21.2.12
SDP-Session	SIP Call
Media	audio 17164 RTP/AVP 0 8 18 101
SDP-Conn	IN IP4 10.21.2.12
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:18 G729/8000...
	4 more

🕒 SIP 100 Trying

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	102 INVITE

📞 SIP INVITE

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK58bbd2da
Max-Forwards	70
From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Contact	<sip:5001@10.21.0.131>
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131

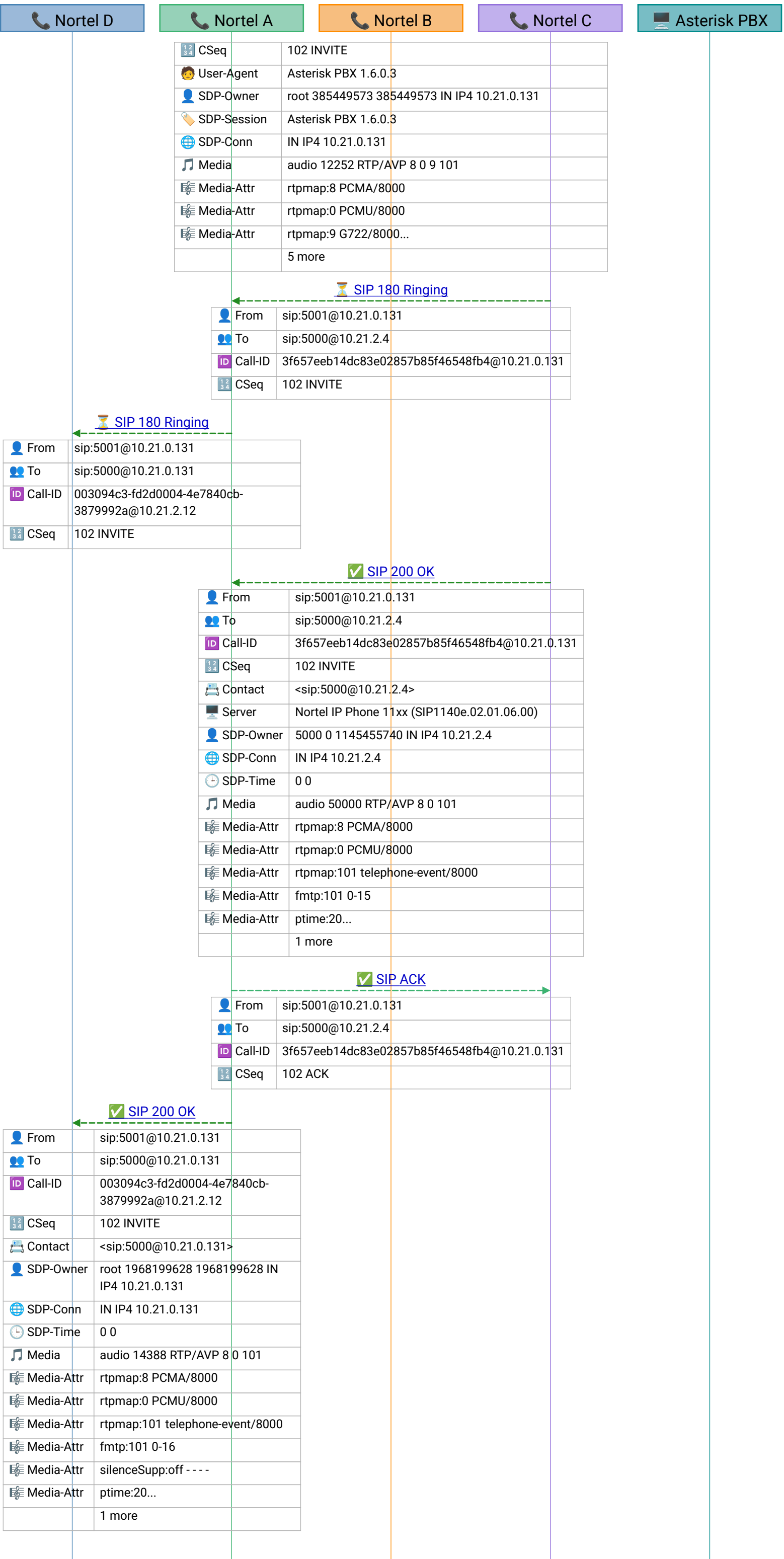
Frame 58 | 2010-10-10T09:26:04.273019Z

Frame 59 | 2010-10-10T09:26:04.345392Z

💡 [Frame 60] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

Frame 61 | 2010-10-10T09:26:04.410148Z

💡 [Frame 62] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams



Frame 63 |
2010-10-10T09:26:04.642419Z

Frame 64 |
2010-10-10T09:26:04.650355Z

💡 [Frame 65] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 66 |
2010-10-10T09:26:06.847348Z

💡 [Frame 67] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖥️ Asterisk PBX

Frame 68 |
2010-10-10T09:26:07.001105Z

SIP ACK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	102 ACK

SIP INVITE

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK67c7f51c
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Contact	<sip:5000@10.21.0.131>
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	102 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 1968199628 1968199629 IN IP4 10.21.2.4
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.2.4
Media	audio 50000 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-16...
	3 more

[Frame 69] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP INVITE

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK1017cd38
Max-Forwards	70
From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Contact	<sip:5001@10.21.0.131>
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131
CSeq	103 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 385449573 385449574 IN IP4 10.21.2.12
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.2.12
Media	audio 17164 RTP/AVP 8
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	silenceSupp:off - - - -
Media-Attr	ptime:20...
	1 more

[Frame 70] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	102 INVITE
Server	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>
SDP-Owner	Cisco-SIPUA 21409 1 IN IP4 10.21.2.12
SDP-Time	0 0
Media	audio 17164 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.2.12
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-15
Media-Attr	sendrecv

[Frame 71] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

SIP ACK

Frame 72 |
2010-10-10T09:26:07.175695Z



From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	102 ACK

SIP 100 Trying

From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131
CSeq	103 INVITE

Frame 73 |
2010-10-10T09:26:07.200116Z

SIP 200 OK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131
CSeq	103 INVITE
Contact	<sip:5000@10.21.2.4>
Server	Nortel IP Phone 11xx (SIP1140e.02.01.06.00)
SDP-Owner	5000 0 1145455742 IN IP4 10.21.2.4
SDP-Conn	IN IP4 10.21.2.4
SDP-Time	0 0
Media	audio 50000 RTP/AVP 8
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	ptime:20
Media-Attr	sendrecv

[Frame 74] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

SIP ACK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131
CSeq	103 ACK

Frame 75 |
2010-10-10T09:26:07.237896Z

SIP BYE

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	103 BYE

[Frame 76] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

SIP 200 OK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	003094c3-fd2d0004-4e7840cb-3879992a@10.21.2.12
CSeq	103 BYE
Contact	<sip:5000@10.21.0.131>

[Frame 77] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

SIP INVITE

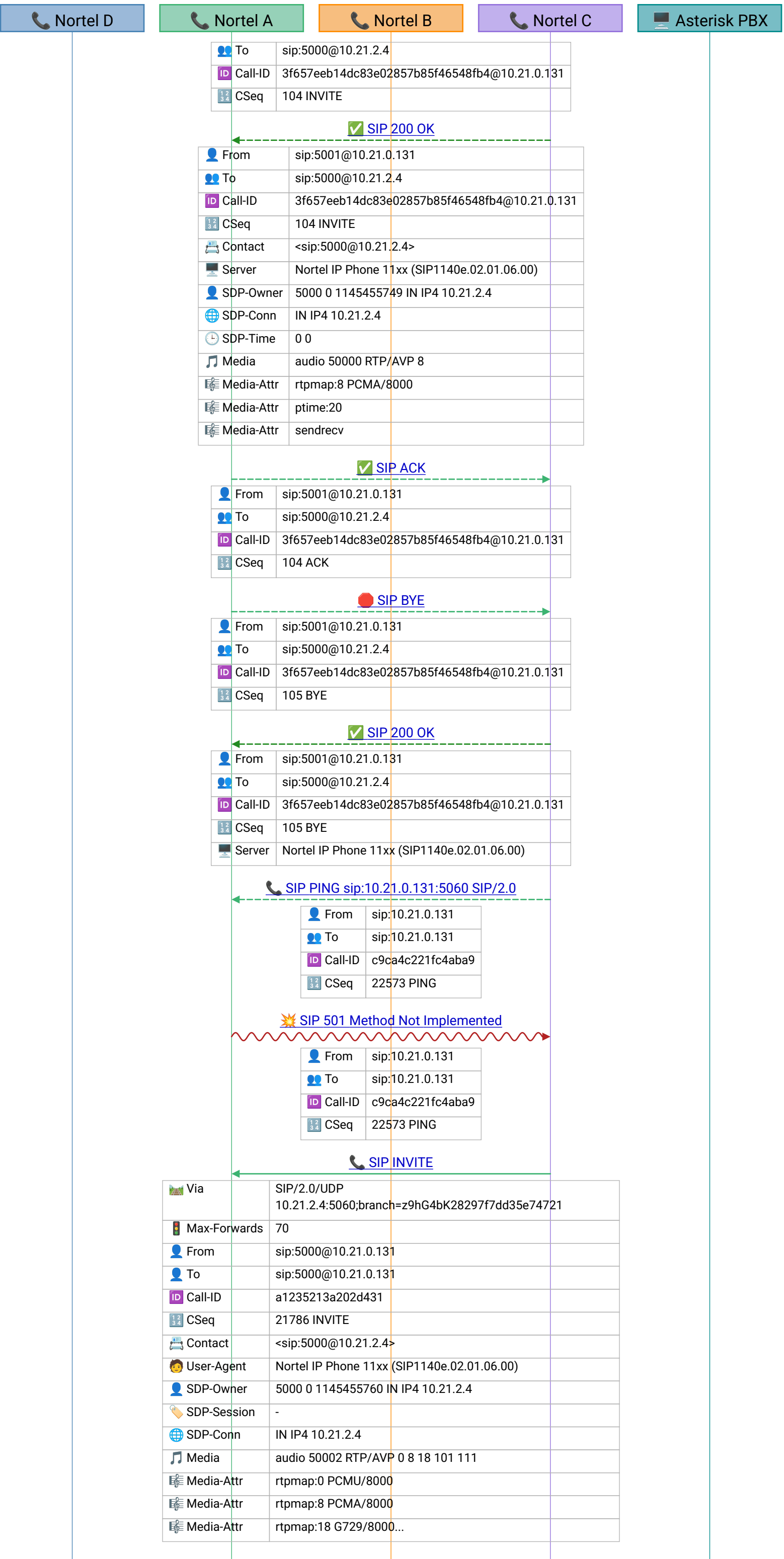
Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK6b4ec606
Max-Forwards	70
From	sip:5001@10.21.0.131
To	sip:5000@10.21.2.4
Contact	<sip:5001@10.21.0.131>
Call-ID	3f657eeb14dc83e02857b85f46548fb4@10.21.0.131
CSeq	104 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 385449573 385449575 IN IP4 10.21.0.131
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.0.131
Media	audio 12252 RTP/AVP 8
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	silenceSupp:off - - - -
Media-Attr	ptime:20...
	1 more

[Frame 78] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

From	sip:5001@10.21.0.131
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Frame 79 |
2010-10-10T09:26:13.942543Z



💡 [Frame 80] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 81 | 2010-10-10T09:26:13.972184Z

💡 [Frame 82] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

💡 [Frame 83] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 84 | 2010-10-10T09:26:15.489081Z

Frame 85 | 2010-10-10T09:26:15.503371Z

💡 [Frame 86] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams



3 more

SIP 401 Unauthorized

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21786 INVITE
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"4f008bd8"

Frame 87 | 2010-10-10T09:26:24.691039Z

SIP ACK

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21786 ACK

Frame 88 | 2010-10-10T09:26:24.749251Z

SIP INVITE

Via	SIP/2.0/UDP 10.21.2.4:5060;branch=z9hG4bKdab47dc077bbb5e8d
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21787 INVITE
Contact	<sip:5000@10.21.2.4>
User-Agent	Nortel IP Phone 11xx (SIP1140e.02.01.06.00)
SDP-Owner	5000 0 1145455760 IN IP4 10.21.2.4
SDP-Session	-
SDP-Conn	IN IP4 10.21.2.4
Media	audio 50002 RTP/AVP 0 8 18 101 111
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:18 G729/8000...
3 more	

[Frame 89] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21787 INVITE

Frame 90 | 2010-10-10T09:26:24.770207Z

SIP INVITE

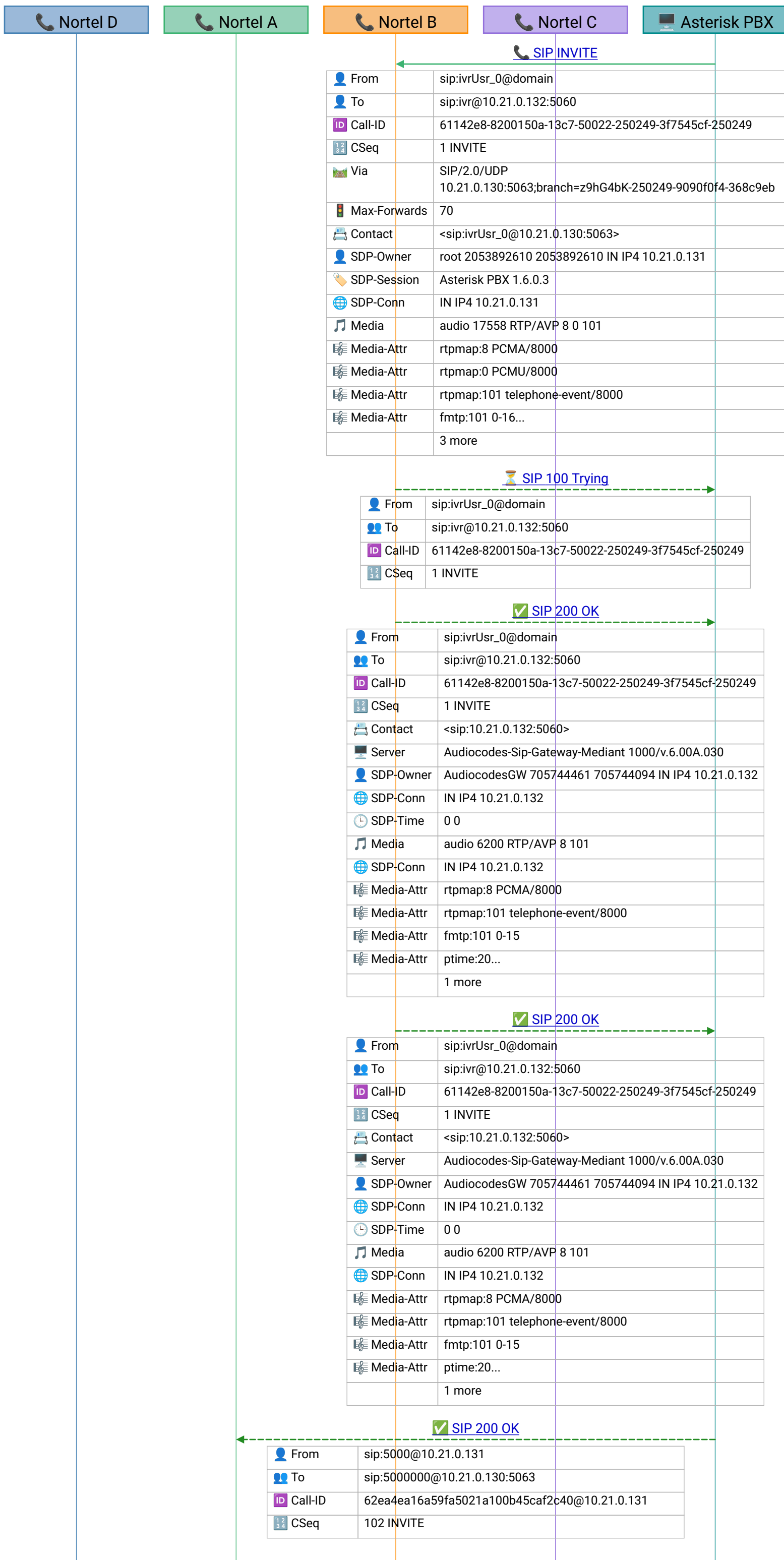
Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK4231233e
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Contact	<sip:5000@10.21.0.131>
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	102 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 2053892610 2053892610 IN IP4 10.21.0.131
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.0.131
Media	audio 17558 RTP/AVP 8 0 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:101 telephone-event/8000...
4 more	

[Frame 91] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	102 INVITE

Frame 92 | 2010-10-10T09:26:24.902167Z



💡 [Frame 93] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

Frame 94 | 2010-10-10T09:26:24.949758Z

💡 [Frame 95] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

💡 [Frame 96] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

💡 [Frame 97] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin



Contact	<sip:5000000@10.21.0.130:5063>
SDP-Owner	AudiocodesGW 705744461 705744094 IN IP4 10.21.0.132
SDP-Conn	IN IP4 10.21.0.132
SDP-Time	0 0
Media	audio 6200 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.0.132
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-15
Media-Attr	ptime:20
Media-Attr	sendrecv

✓ SIP ACK

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	102 ACK

Frame 98 | 2010-10-10T09:26:25.784657Z

✓ SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21787 INVITE
Contact	<sip:5000@10.21.0.131>
SDP-Owner	root 851456694 851456694 IN IP4 10.21.0.131
SDP-Conn	IN IP4 10.21.0.131
SDP-Time	0 0
Media	audio 13074 RTP/AVP 8 0 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-16
Media-Attr	silenceSupp:off - - - -
Media-Attr	ptime:20...
	1 more

💡 [Frame 99] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

✓ SIP ACK

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	1 ACK

Frame 100 | 2010-10-10T09:26:25.838223Z

✓ SIP ACK

From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	a1235213a202d431
CSeq	21787 ACK

Frame 101 | 2010-10-10T09:26:25.879703Z

📞 SIP INVITE

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK561247f5
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5000@10.21.0.131
Contact	<sip:5000@10.21.0.131>
Call-ID	a1235213a202d431
CSeq	102 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 851456694 851456695 IN IP4 10.21.0.132
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.0.132
Media	audio 6200 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-16...
	3 more

💡 [Frame 102] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

📞 SIP INVITE

💡 [Frame 103] SIP INVITE carries an SDP offer describing media

Nortel D

Nortel A

Nortel B

Nortel C

Asterisk PBX

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK28c2ba6d
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Contact	<sip:5000@10.21.0.131>
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	103 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 2053892610 2053892611 IN IP4 10.21.2.4
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.2.4
Media	audio 50002 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-16...
	3 more

capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP INVITE

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	2 INVITE
Via	SIP/2.0/UDP 10.21.0.130:5063;branch=z9hG4bK-25024b-9090f50a-2c4c6bb4
Max-Forwards	70
Contact	<sip:ivrUsr_0@10.21.0.130:5063>
SDP-Owner	root 2053892610 2053892611 IN IP4 10.21.2.4
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.2.4
Media	audio 50002 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-16
Media-Attr	silenceSupp:off - - - - -
	2 more

[Frame 104] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 200 OK

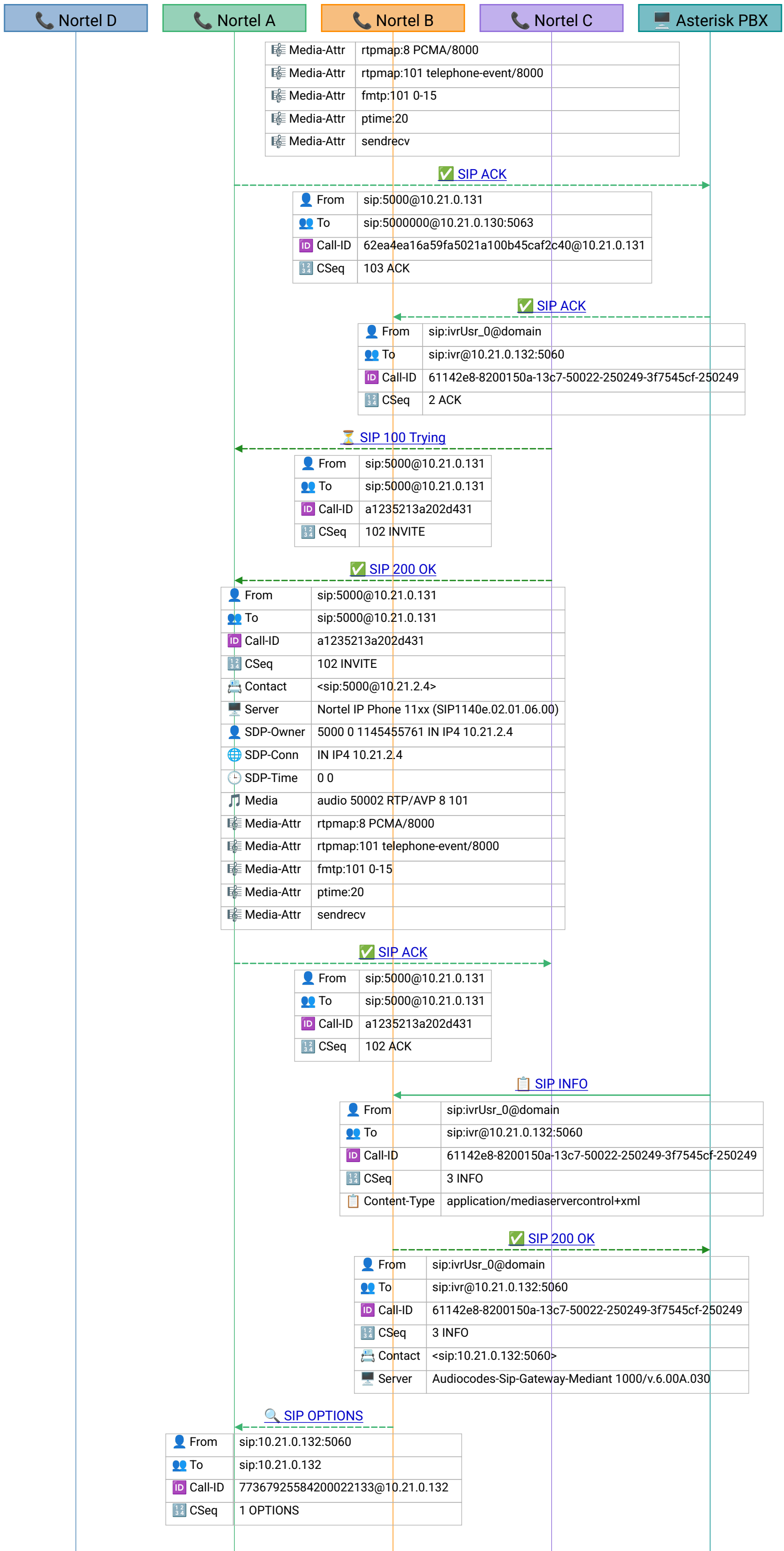
From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	2 INVITE
Contact	<sip:10.21.0.132:5060>
Server	Audiocodes-Sip-Gateway-Mediant 1000/v.6.00A.030
SDP-Owner	AudiocodesGW 705744461 705744095 IN IP4 10.21.0.132
SDP-Conn	IN IP4 10.21.0.132
SDP-Time	0 0
Media	audio 6200 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.0.132
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-15
Media-Attr	ptime:20...
	1 more

[Frame 105] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	103 INVITE
Contact	<sip:5000000@10.21.0.130:5063>
SDP-Owner	AudiocodesGW 705744461 705744095 IN IP4 10.21.0.132
SDP-Conn	IN IP4 10.21.0.132
SDP-Time	0 0
Media	audio 6200 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.0.132

[Frame 106] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin



Frame 107 | 2010-10-10T09:26:25.987827Z

Frame 108 | 2010-10-10T09:26:25.996061Z

Frame 109 | 2010-10-10T09:26:26.195123Z

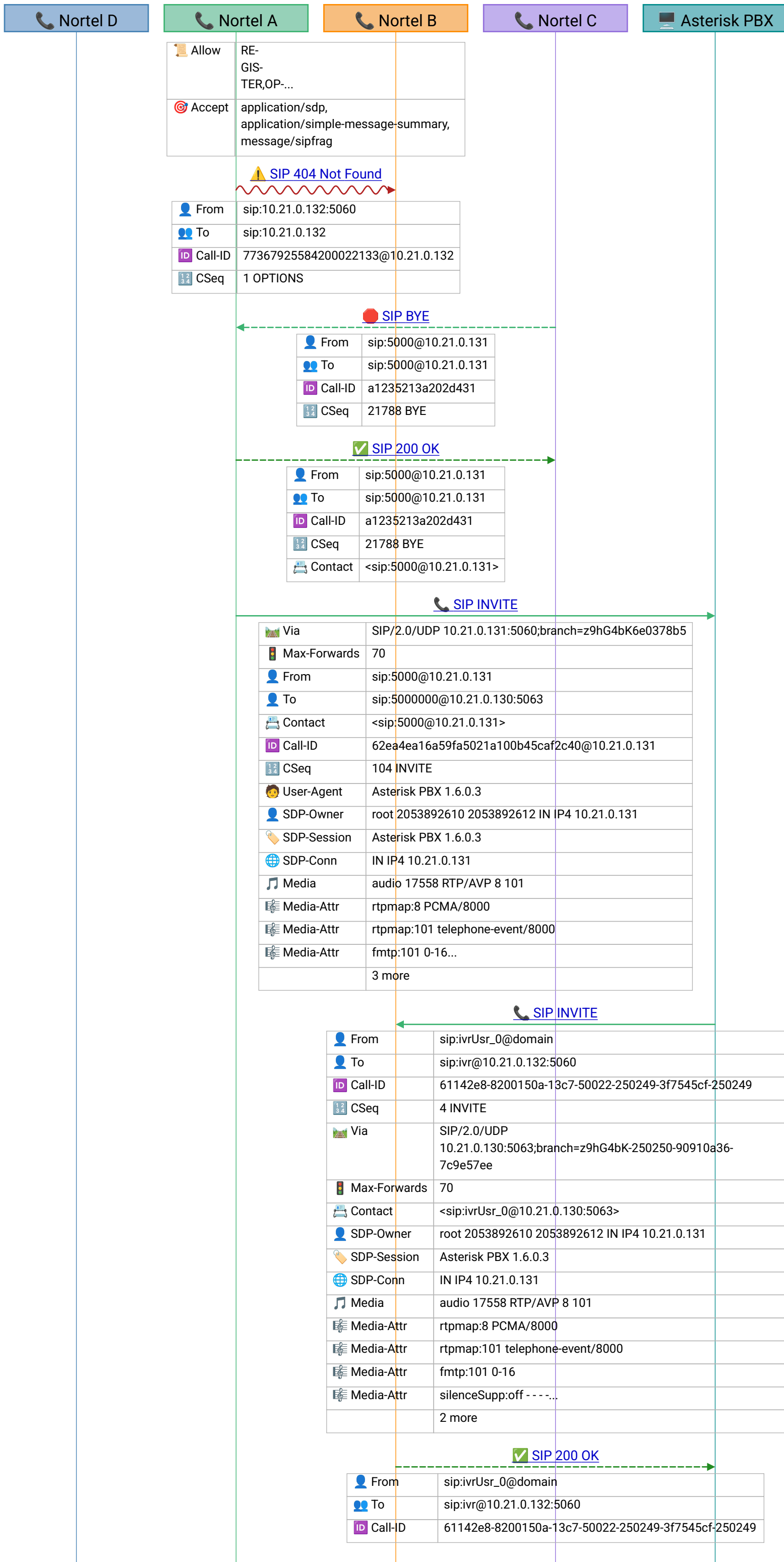
💡 [Frame 110] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 111 | 2010-10-10T09:26:26.222042Z

Frame 112 | 2010-10-10T09:26:27.14969Z

💡 [Frame 113] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 114 | 2010-10-10T09:26:29.066113Z



Frame 115 | 2010-10-10T09:26:29.081463Z

[Frame 116] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

[Frame 117] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

[Frame 118] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

[Frame 119] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

[Frame 120] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows

Nortel D
Nortel A
Nortel B
Nortel C
Asterisk PBX

CSeq	4 INVITE
Contact	<sip:10.21.0.132:5060>
Server	Audiocodes-Sip-Gateway-Mediant 1000/v.6.00A.030
SDP-Owner	AudiocodesGW 705744461 705744096 IN IP4 10.21.0.132
SDP-Conn	IN IP4 10.21.0.132
SDP-Time	0 0
Media	audio 6200 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.0.132
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-15
Media-Attr	ptime:20...
	1 more

begin

✓ SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	104 INVITE
Contact	<sip:5000000@10.21.0.130:5063>
SDP-Owner	AudiocodesGW 705744461 705744096 IN IP4 10.21.0.132
SDP-Conn	IN IP4 10.21.0.132
SDP-Time	0 0
Media	audio 6200 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.0.132
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmtp:101 0-15
Media-Attr	ptime:20
Media-Attr	sendrecv

💡 [Frame 121] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

✓ SIP ACK

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	104 ACK

Frame 122 | 2010-10-10T09:26:31.45668Z

● SIP BYE

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	105 BYE

💡 [Frame 123] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

✓ SIP ACK

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	4 ACK

Frame 124 | 2010-10-10T09:26:31.461888Z

✓ SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5000000@10.21.0.130:5063
Call-ID	62ea4ea16a59fa5021a100b45caf2c40@10.21.0.131
CSeq	105 BYE

💡 [Frame 125] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

📄 SIP INFO

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	5 INFO
Content-Type	application/mediaservercontrol+xml

Frame 126 | 2010-10-10T09:26:31.462792Z

✓ SIP 200 OK

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	5 INFO

💡 [Frame 127] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin



Contact	<sip:10.21.0.132:5060>
Server	Audiocodes-Sip-Gateway-Mediant 1000/v.6.00A.030

SIP INFO

From	sip:ivr@10.21.0.132:5060
To	sip:ivrUsr_0@domain
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	1 INFO
Content-Type	application/mediaservercontrol+xml

SIP 200 OK

From	sip:ivr@10.21.0.132:5060
To	sip:ivrUsr_0@domain
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	1 INFO
Contact	<sip:ivrUsr_0@10.21.0.130:5063>

SIP BYE

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	6 BYE

SIP 200 OK

From	sip:ivrUsr_0@domain
To	sip:ivr@10.21.0.132:5060
Call-ID	61142e8-8200150a-13c7-50022-250249-3f7545cf-250249
CSeq	6 BYE
Contact	<sip:10.21.0.132:5060>
Server	Audiocodes-Sip-Gateway-Mediant 1000/v.6.00A.030

SIP INVITE

Via	SIP/2.0/UDP 10.21.2.4:5060;branch=z9hG4bK9a9a12444d2d861f8
Max-Forwards	70
From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	12954 INVITE
Contact	<sip:5000@10.21.2.4>
User-Agent	Nortel IP Phone 11xx (SIP1140e.02.01.06.00)
SDP-Owner	5000 0 1145455775 IN IP4 10.21.2.4
SDP-Session	-
SDP-Conn	IN IP4 10.21.2.4
Media	audio 50004 RTP/AVP 0 8 18 101 111
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:18 G729/8000...
	3 more

SIP 401 Unauthorized

From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	12954 INVITE
Auth-Scheme	Digest
Algorithm	MD5
Realm	"asterisk"
Nonce	"7f90786a"

SIP ACK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	12954 ACK

SIP INVITE

Frame 128 |
2010-10-10T09:26:31.476541Z

[Frame 129] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

[Frame 130] SIP BYE terminates an established dialog; either party can send it, and the receiving side responds with 200 OK to confirm teardown of the RTP media streams and release resources

[Frame 131] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

[Frame 132] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

Frame 133 |
2010-10-10T09:26:40.065893Z

Frame 134 |
2010-10-10T09:26:40.119413Z

[Frame 135] SIP INVITE carries an SDP offer describing media

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖨️ Asterisk PBX

📡 Via	SIP/2.0/UDP 10.21.2.4:5060;branch=z9hG4bK92cd3c58534260ed2
📡 Max-Forwards	70
👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.0.131
🆔 Call-ID	2b3c3b4b8ccf6020
📄 CSeq	12955 INVITE
📄 Contact	<sip:5000@10.21.2.4>
👤 User-Agent	Nortel IP Phone 11xx (SIP1140e.02.01.06.00)
👤 SDP-Owner	5000 0 1145455775 IN IP4 10.21.2.4
👤 SDP-Session	-
🌐 SDP-Conn	IN IP4 10.21.2.4
🎵 Media	audio 50004 RTP/AVP 0 8 18 101 111
🎵 Media-Attr	rtpmap:0 PCMU/8000
🎵 Media-Attr	rtpmap:8 PCMA/8000
🎵 Media-Attr	rtpmap:18 G729/8000...
	3 more

🕒 SIP 100 Trying

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.0.131
🆔 Call-ID	2b3c3b4b8ccf6020
📄 CSeq	12955 INVITE

📞 SIP INVITE

📡 Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK282f4fa6
📡 Max-Forwards	70
👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
📄 Contact	<sip:5000@10.21.0.131>
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	102 INVITE
👤 User-Agent	Asterisk PBX 1.6.0.3
👤 SDP-Owner	root 853275738 853275738 IN IP4 10.21.0.131
👤 SDP-Session	Asterisk PBX 1.6.0.3
🌐 SDP-Conn	IN IP4 10.21.0.131
🎵 Media	audio 17400 RTP/AVP 8 0 9 101
🎵 Media-Attr	rtpmap:8 PCMA/8000
🎵 Media-Attr	rtpmap:0 PCMU/8000
🎵 Media-Attr	rtpmap:9 G722/8000...
	5 more

🕒 SIP 100 Trying

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	102 INVITE

🕒 SIP 180 Ringing

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	102 INVITE

🕒 SIP 180 Ringing

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.0.131
🆔 Call-ID	2b3c3b4b8ccf6020
📄 CSeq	12955 INVITE

capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

Frame 136 |
2010-10-10T09:26:40.128765Z

💡 [Frame 137] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

Frame 138 |
2010-10-10T09:26:40.275887Z

Frame 139 |
2010-10-10T09:26:40.383306Z

Frame 140 |
2010-10-10T09:26:40.393859Z

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖨️ Asterisk PBX

✓ SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.2.12:5060;transport=udp
Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
CSeq	102 INVITE
Server	Cisco-CP7960G/8.0
Contact	<sip:5001@10.21.2.12:5060;transport=udp>
SDP-Owner	Cisco-SIPUA 26441 0 IN IP4 10.21.2.12
SDP-Time	0 0
Media	audio 18502 RTP/AVP 8 101
SDP-Conn	IN IP4 10.21.2.12
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-15
Media-Attr	sendrecv

✓ SIP ACK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.2.12:5060;transport=udp
Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
CSeq	102 ACK

✓ SIP 200 OK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	12955 INVITE
Contact	<sip:5001@10.21.0.131>
SDP-Owner	root 288447963 288447963 IN IP4 10.21.0.131
SDP-Conn	IN IP4 10.21.0.131
SDP-Time	0 0
Media	audio 19492 RTP/AVP 8 0 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:0 PCMU/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-16
Media-Attr	silenceSupp:off - - - -
Media-Attr	pptime:20...
	1 more

✓ SIP ACK

From	sip:5000@10.21.0.131
To	sip:5001@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	12955 ACK

📞 SIP INVITE

Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK454b0470
Max-Forwards	70
From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Contact	<sip:5001@10.21.0.131>
Call-ID	2b3c3b4b8ccf6020
CSeq	102 INVITE
User-Agent	Asterisk PBX 1.6.0.3
SDP-Owner	root 288447963 288447964 IN IP4 10.21.2.12
SDP-Session	Asterisk PBX 1.6.0.3
SDP-Conn	IN IP4 10.21.2.12
Media	audio 18502 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000

💡 [Frame 141] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 142 | 2010-10-10T09:26:43.363006Z

💡 [Frame 143] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Frame 144 | 2010-10-10T09:26:43.459613Z

💡 [Frame 145] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

📞 Nortel D
📞 Nortel A
📞 Nortel B
📞 Nortel C
🖥️ Asterisk PBX

🎵 Media-Attr	fmtmp:101 0-16...
	3 more

SIP INVITE

🌐 Via	SIP/2.0/UDP 10.21.0.131:5060;branch=z9hG4bK0d632651
🚦 Max-Forwards	70
👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
📠 Contact	<sip:5000@10.21.0.131>
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	103 INVITE
👤 User-Agent	Asterisk PBX 1.6.0.3
👤 SDP-Owner	root 853275738 853275739 IN IP4 10.21.2.4
🏷️ SDP-Session	Asterisk PBX 1.6.0.3
🌐 SDP-Conn	IN IP4 10.21.2.4
🎵 Media	audio 50004 RTP/AVP 8 101
🎵 Media-Attr	rtpmap:8 PCMA/8000
🎵 Media-Attr	rtpmap:101 telephone-event/8000
🎵 Media-Attr	fmtmp:101 0-16...
	3 more

💡 [Frame 146] SIP INVITE carries an SDP offer describing media capabilities; the callee's 200 OK contains the SDP answer, completing the offer/answer negotiation for RTP media streams

SIP 100 Trying

👤 From	sip:5001@10.21.0.131
👤 To	sip:5000@10.21.0.131
🆔 Call-ID	2b3c3b4b8ccf6020
📄 CSeq	102 INVITE

Frame 147 | 2010-10-10T09:26:43.6846Z

SIP 200 OK

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	103 INVITE
🖥️ Server	Cisco-CP7960G/8.0
📠 Contact	<sip:5001@10.21.2.12:5060;transport=udp>
👤 SDP-Owner	Cisco-SIPUA 26441 1 IN IP4 10.21.2.12
🕒 SDP-Time	0 0
🎵 Media	audio 18502 RTP/AVP 8 101
🌐 SDP-Conn	IN IP4 10.21.2.12
🎵 Media-Attr	rtpmap:8 PCMA/8000
🎵 Media-Attr	rtpmap:101 telephone-event/8000
🎵 Media-Attr	fmtmp:101 0-15
🎵 Media-Attr	sendrecv

💡 [Frame 148] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

SIP ACK

👤 From	sip:5000@10.21.0.131
👤 To	sip:5001@10.21.2.12:5060;transport=udp
🆔 Call-ID	3cdb73460a38c475e-ca7c1373591b91@10.21.0.131
📄 CSeq	103 ACK

Frame 149 | 2010-10-10T09:26:43.691095Z

SIP 200 OK

👤 From	sip:5001@10.21.0.131
👤 To	sip:5000@10.21.0.131
🆔 Call-ID	2b3c3b4b8ccf6020
📄 CSeq	102 INVITE
📠 Contact	<sip:5000@10.21.2.4>
🖥️ Server	Nortel IP Phone 11xx (SIP1140e.02.01.06.00)
👤 SDP-Owner	5000 0 1145455779 IN IP4 10.21.2.4
🌐 SDP-Conn	IN IP4 10.21.2.4

💡 [Frame 150] SIP 200 OK in response to INVITE carries the SDP answer (codec, IP, port) and completes the offer/answer negotiation; caller then sends ACK to finalize the dialog and RTP flows begin

Nortel D Nortel A Nortel B Nortel C Asterisk PBX

SDP-Time	0 0
Media	audio 50004 RTP/AVP 8 101
Media-Attr	rtpmap:8 PCMA/8000
Media-Attr	rtpmap:101 telephone-event/8000
Media-Attr	fmp:101 0-15
Media-Attr	ptime:20
Media-Attr	sendrecv

✓ SIP ACK

From	sip:5001@10.21.0.131
To	sip:5000@10.21.0.131
Call-ID	2b3c3b4b8ccf6020
CSeq	102 ACK

Frame 151 |
2010-10-10T09:26:43.800612Z