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STEPS OF OPTIMISING SIP FOR NARROW BAND SIGNALLING CHANNELS

Technical Report

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Abstract:

This report presents the process of optimizing the Ascii based Session Initiation Protocol that has been selected as the future signaling protocol for cellular systems such as 3G. We use the Mobile Originated call signaling flow from the 3G specification 24.228v5-12 (latest at the time of writing the TR for 3G release 5) to demonstrate the process. The initial steps of optimization are carried out in the ascii domain. They show what is achievable if we stick to ascii representation. One branch of optimization is carried out by introducing a binary encoding using Type-Length-Value and Type-Value –pair encoding styles. After each step we calculate the signaling efficiency. In an alternative branch of optimization we demonstrate the efficiency of using a fixed compression algorithm on the original and the optimized ascii. The motivation for the optimization is that in a 3G context ascii based SIP is a clearly inefficient signaling method for narrow band services. Moreover, the approach of using signaling compression (SigComp) does not seem to be able to achieve the kind of optimization in signaling efficiency as is needed. It also leads to a huge waste of cpu power and memory. Signaling compression using the dictionary approach in fact means that the devices engaged in signaling use a lot of resources for the futile purpose of translating signaling from ascii to binary, from the dictionary based binary back to ascii, from the resulting ascii to binary that is understandable for the actual signaling processor. This process can only be characterized as being similar or worse than what signaling processors need to do to process an analogue signaling such as R2. For motivation purposes we also show an ISDN signaling flow for comparison. Our solution preserves most of the SIP principles and the SIP logic and achieves a compression ratio of better than 10 to 1.

Keywords: Signaling, SIP, optimization, signaling compression.

Contents

1 Introduction

The motivation for the optimization is that in a 3G context ascii based SIP is a clearly inefficient signaling method for narrow band services. By the inefficiency we mean that signaling uses a lot of bits compared to the service itself. We will calculate the size of the MO flow and show that it uses as many bits as a voice call of 18.2 second if an AMR codec is used with 4.75 kbps encoding. A few years ago many cellular operators paid huge amounts of money for the 3G radio licenses. It is difficult to see why they would now be happy to fill those expensive radio frequencies with signaling overhead that is not chargeable. It is also difficult to see what kind of services could be so valuable for the end users that it would make economic sense to pay for such signaling overhead.

Moreover, the approach of using signaling compression (SigComp as specified in RFCs 3320, 3321 and 3485), as suggested by the 3GPP, can not achieve the kind of optimization in signaling efficiency as is needed. It leads to a huge waste of cpu power and memory. Signaling compression using the dictionary approach in fact means that the devices engaged in signaling use a lot of resources for the futile purpose of translating signaling from ascii to binary using a dynamically created dictionary, from the dictionary based binary back to ascii, from the resulting ascii to binary that is understandable for the actual signaling processor. This process can only be characterized as being similar or worse than what signaling processors need to do to process an analogue signaling such as R2.

In [2] a rather comprehensive SigComp was implemented without Huffman coding. That achieved a compression ratio of less than 4:1. In order to estimate the potential of Huffman coding we demonstrate the use of a fixed compression algorithm that uses the idea of replacing redundant strings with distance-length pair referring to the first occurrence of the string (LZ77) and Huffman coding. These are the principles that are used in the well known tools for compression (gzip, zlib and WinZip).

One could argue that because of the huge amounts of cpu power and cheap memory even a modern mobile phone has due to the Moore's law, all this waste does not matter. However, mobile devices are powered by batteries and energy density is developing only at a very modest pace. For their newest models mobile phone vendors constantly have a long list of candidate features that they can not fit into their phones. Surely, by freeing some space from futile SIP functions some of those hopefully useful new features can be provided economically to the end users.

The next generation networks provide broadband connectivity to end users. SIP and the 3G IMS architecture are meant for such networks be they cellular, wireless or wireline. In some of these networks with mains powered devices and Megabits available to each user the described signaling inefficiency does not really matter much – this is easy to demonstrate even based on our analysis. However, we argue that in case of battery powered end user devices and shared access networks such as cellular, for the purpose of essentially *narrow band services* we can not ignore the efficiency of signaling.

If SIP and IMS intend to be successful, the system must be able to do well something that is easily understood by the end users. 3G supports circuit switched call signaling and operators are forced to deploy 3G with Circuit switched capabilities due to Emergency call capability that is initially supported only by the CS subsystem. Therefore, all 3G mobile phones have the CS signaling. Since the CS call system is in place, IMS can only hope to be deployed by bringing in services that can not be provided by the CS signaling. Such services must be very valuable to counter the signaling overhead introduced by 3G SIP. It makes all the economic sense in the world to lower the entry barrier to IMS deployment by getting rid of the inefficiency in 3G SIP signaling.

Since making calls is among the most valuable services an operator can offer, 3G IMS should provide them efficiently.

For motivation purposes we also show an ISDN signaling flow for comparison.

Our solution preserves most of the SIP principles and the SIP logic and achieves a compression ratio of better than 10 to 1. Our comparison with ISDN Q.931/Q.921 shows that SIP uses about 100 times more bits than ISDN signaling. Optimizing SIP further than our solution is possible and may even be desirable. However, this would most likely mean going to a model of communicating finite state machines. Such protocols are being created for packet based cellular networks by transferring GSM signaling protocols to work over IP. For this reason, we argue that going as far as abandoning SIP principles and introducing a communicating finite state machine model for access signaling with a starting point in SIP does not make sense.

The rest of the Report is structured as follows. First we show an ISDN signaling flow for comparison. In Section 3, we reconstruct the Mobile Originated signaling flow from 24.228, version 5-12 that is the latest signaling flow specification at the time of writing this TR for 3G, release 5. In Section 4, we present the process of optimizing the ascii based Session Initiation Protocol in the ascii domain. We use the reconstructed MO call signaling flow to demonstrate the process. We show what is achievable if we stick to ascii representation. In Section 5 we introduce a binary encoding using Type-Length-Value and Type-Value –pair encoding styles. After each step we calculate the signaling efficiency. In Section 6 we explore an alternative optimization approach by doing all our logical optimizations first in the ascii domain and then in Section 7 applying WinZip compression on the resulting best ascii and the original messages. Finally, in Section 8, we conclude.

2 An ISDN Signaling Flow

The flow has been captured in the TKK Netlab by Vesa Kosonen. Layer two flags and the LAPD checksums are not shown in the captured flow.

From one ISDN phone (29) to another ISDN phone (32) ENBLOCK SENDING- HEXA representation

Conn:1 Card:1 Channel:D 52 11:50:09.565
RESYNCRONIZATION
Conn:1 Card:1 Channel:D 53 11:50:09.567
ACTIVATION REQ
Conn:1 Card:1 Channel:D 54 11:50:09.570
ACTIVATION IND
Conn:1 Card:1 Channel:D 55 11:50:09.580
SABME
Conn:1 Card:1 Channel:D 56 11:50:09.601
UNNUMBERED ACK
Conn:1 Card:1 Channel:D 57 11:50:09.644
L2: INFO 00990000080113050403809 0A31801836C02008070038033327D0291817E0104
L3: **SETUP** 0801130504038090A31801836C02008070038033327D0291817E0104
Conn:1 Card:1 Channel:D 58 11:50:09.692
RECEIVER READY
Conn:1 Card:1 Channel:D 59 11:50:09.817
L2 INFO : 029900020801930218018A
L3 **CALL PROCEEDING** : 0801930218018A
Conn:1 Card:1 Channel:D 60 11:50:09.826
RECEIVER READY
Conn:1 Card:1 Channel:D 61 11:50:10.370
L2 INFO : 0299020208019301
L3 **ALERTING** : 08019301
Conn:1 Card:1 Channel:D 62 11:50:10.378
RECEIVER READY
Conn:1 Card:1 Channel:D 63 11:50:14.019
L2 INFO 02990402080193072905690107 14247C038090A3
L3 **CONNECT** : 0801930729056901071 4247C038090A3
Conn:1 Card:1 Channel:D 64 11:50:14.027
RECEIVER READY
Conn:1 Card:1 Channel:D 65 11:50:14.866
L2 INFO : 029906020801934508028190 1E028188
L3 **DISCONNECT** : 08019345080281901E028188
Conn:1 Card:1 Channel:D 66 11:50:14.874
RECEIVER READY
Conn:1 Card:1 Channel:D 67 11:50:15.117
L2: INFO : 009902080801134D08028090
L3: **RELEASE** : 0801134D08028090
Conn:1 Card:1 Channel:D 68 11:50:15.166
RECEIVER READY
Conn:1 Card:1 Channel:D 69 11:50:15.282

L2 INFO : 029908040801935A
L3 **RELEASE COMPLETE** : 0801935A
Conn:1 Card:1 Channel:D 70 11:50:15.290
RECEIVER READY
Conn:1 Card:1 Channel:D 71 11:50:20.414
DISCONNECT

By cleaning up we get Table 1. LAPD (layer 2 content) is shown in bold. The reminder is Q.931. The used telephone numbers are just two digits. In our comparison, we need to add a reasonable amount of digits to this flow to account for some average number length.

Table 1: An ISDN Signaling Flow.

SETUP	**00990000**0801130504038090A31801836C02008070038033327D0291817E0104
CALL PROC	**02990002**0801930218018A
Alerting	**02990202**08019301
Connect	**02990402**080193072905690107142447C038090A3
Disconnect	**02990602**08019345080281901E028188
Release	**00990208**0801134D08028090
Release Complete	**02990804**0801935A

Counting the bits message by message and accumulating the bits over the whole flow we get Table 2.

Table 2: Summary of the ISDN flow

A Message	B Hexa content	C LEN(B)	D Bits	E N-len =8	F Cumu-lative of E	G ms of F	H Add CK+Deli-miters
SETUP	009900000801130504038090A31801836C02008070038033327D0291817E0104	64	256	352	352	22	24
CALL PROC	029900020801930218018A	22	88	88	440	28	30
Alerting	0299020208019301	16	64	64	504	32	34
Connect	0299040208019307290569010714247C038090A3	40	160	160	664	42	44
Discon-nect	0299060208019345080281901E028188	32	128	128	792	50	52
Release	009902080801134D08028090	24	96	96	888	56	58
Release Complete	029908040801935A	16	64	64	952	60	62

The cumulative milliseconds of the ISDN signaling flow, assuming a 16 kbps signaling channel in the 2B+D interface, en-block sending, 8 digits for both the Called Party Number and the Calling Party Number and taking into account both the Q.931 bits and the LAPD bits in the frames that carry the signaling information are in column H of Table 2. The result is also presented graphically in Figure 1.

Figure 1: Summary of an ISDN call setup and Release

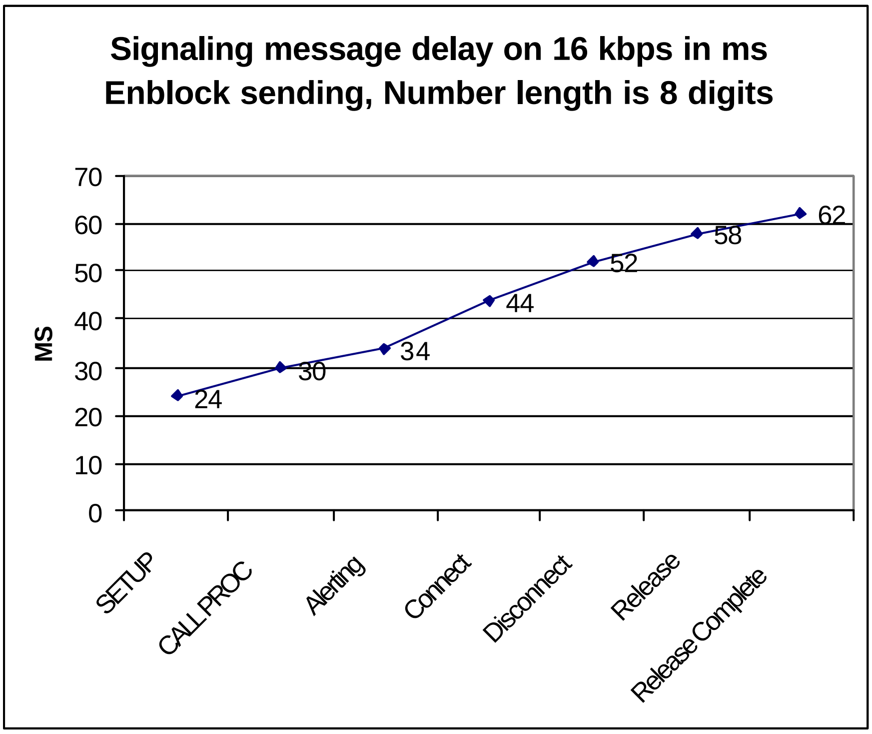


Table 3 presents some bitcount metrics on the ISDN signaling flow.

ISDN signaling **bitcount** summary	Number length = 8		
	L3	L2+L3	L2 overhead %
Call setup signaling (enblock)	536	664	24 %
Avg message in call setup	134	166	
Call setup and release total	728	952	31 %
Avg message in total	104	136	

Conclusion on ISDN signaling

The signaling transfer delay occurs due to the need to transfer the signaling bits on a signaling channel of finite speed. The signaling transfer delay is calculated according to formula 1.

$$\text{Transfer delay} = \text{size/signaling channel speed} \qquad (1)$$

The total signaling transfer delay for call setup on a 2B+D access is less than 50 ms and total delay including also call release is slightly above 60 ms.

3 3G Signaling Flows Reconstructed and Metrics on them

A lot of the repeated ascii in signaling flows is omitted in 24.228 according to certain rules. In order to have a clear case example for a 3G SIP signaling flow, we select the Mobile Originated call signaling in 24.228. This choice is arbitrary but representative among the flows.

An alternative to the MO flow would be the MT flow. However, it is not much different from the MO flow. It is actually a bit more verbose than the MO flow.

In the following sections, we will present the MO flow, some metrics on the corresponding MT flow and a release flow. Section 3.1 gives the MO flow.

3.1 The 3G MO signaling flow based on 24.228 v5.12.

```
INVITE tel:+1-212-555-2222 SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>
P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: precondition, sec-agree
Proxy-Require: sec-agree
Supported: 100rel
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
                 port-c=8642; port-s=7531
Content-Type: application/sdp
Content-Length: (571)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=rtpmap:99 MP4V-ES
a=fmtp:98 profile-level-id=0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event


SIP/2.0 100 Trying
```

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Content-Length: 0
```

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>,
 <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>
P-Media-Authorization: 0020000100100101706466312e686f6d65312e6e6574000c
                       02013942563330373200
P-Asserted-Identity: "John Smith" <tel:+1-212-555-2222>
Privacy: none
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 INVITE
Require: 100rel
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
RSeq: 9021
Content-Type: application/sdp
Content-Length: (636)

v=0
o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=video 10001 RTP/AVP 98 99
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:98 H263
a=rtpmap:99 MP4V-ES
a=fmtp:98 profile-level-id=0
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event
```

```
PRACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
        <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 128 PRACK
Require: precondition, sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
                 port-c=8642; port-s=7531
RAck: 9021 127 INVITE
Content-Type: application/sdp
```

```
Content-Length: (555)

v=0
o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event


SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 128 PRACK
Content-Type: application/sdp
Content-Length: (612)

v=0
o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::eee:fff:aaa:bbb
t=0 0
m=video 10001 RTP/AVP 98
b=AS:75
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 6544 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event


UPDATE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
```

```
Cseq: 129 UPDATE
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
                 port-c=8642; port-s=7531
Content-Type: application/sdp
Content-Length: (563)

v=0
o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event


SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 129 UPDATE
Content-Type: application/sdp
Content-Length: (563)

v=0
o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=0 0
m=video 3400 RTP/AVP 98
b=AS:75
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:98 H263
a=fmtp:98 profile-level-id=0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event


SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>,
     <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>
From: <sip:user1_public1@home1.net>;tag=171828
```

```
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 129 INVITE
Require: 100rel
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
RSeq: 9022
Content-Length: 0
```

```
PRACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
        <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 PRACK
Require: sec-agree
Proxy-Require: sec-agree
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531
RAck: 9022 127 INVITE
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 PRACK
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Record-Route: sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>,
       <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 INVITE
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE
Content-Length: 0
```

```
ACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
Max-Forwards: 70
Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
              <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>
From: <sip:user1_public1@home1.net>;tag=171828
To: <tel:+1-212-555-2222>;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 127 ACK
Content-Length: 0
```

3.2 Metrics on the MO and the MT flows

Table 4 shows the number of visible ascii characters in the flow per message in column two. The next column sums the number of all ascii characters to be transferred in each message. Column four shows the accumulation of the ascii in the flow. Two last columns show the size of the messages the flow in bits (assuming that UTF-8 encoding is used as specified for SIP).

Table 4: Summary of the MO session establishment flow in 24.228

	Visible char	Including CRLF	Cumulative ascii	Cumulative MO bits	Cumulative MT Bits
INVITE	1341	1427	1427	11416	13496
100 Trying	235	249	1676	13408	17768
183 Session Progress	1306	1390	3066	24528	30608
PRACK	1234	1310	4376	35008	40088
200 OK	836	902	5278	42224	49736
UPDATE	1209	1283	6561	52488	58904
200 OK	792	854	7415	59320	68128
180 Ringing	534	558	7973	63784	75312
PRACK	646	674	8647	69176	79880
200 OK	241	255	8902	71216	84344
200 OK	503	523	9425	75400	91352
ACK	429	447	9872	78976	95728

For comparison the last column shows the size of the MT flow calculated from 24.228. We do not provide the MT flow itself, but with a little effort one can reconstruct it from 24.228. Table 5 provides more details on the MT flow in 24.228.

Table 5: Metrics on the MT session setup.

Message	Visible ascii	All Ascii	Cumulative	Cum Bits
INVITE	1603	1687	1687	13496
100 Trying	520	534	2221	17768
183 Session Progress	1523	1605	3826	30608
PRACK	1117	1185	5011	40088
200 OK	1138	1206	6217	49736
UPDATE	1082	1146	7363	58904
200 OK	1089	1153	8516	68128
180 Ringing	872	898	9414	75312
PRACK	553	571	9985	79880
200 OK	542	558	10543	84344
200 OK	854	876	11419	91352
ACK	531	547	11966	95728

Table 5 shows that the MT flow is a bit more verbose than the MO flow.

3.3 Release of a session

Our source provides a release flow session in Chapter 8. We can pick either the UE that initiates the release or the one that reacts to the release. Here we present the release on the interface that initiates the release.

Next we present the release flow.

MT session release, Mobile in a visited network

Mobile UE initiates session release to P-CSCF

```
  BYE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0
  Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
  Max-Forwards: 70
  Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>
  P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11
  From: <sip:user1_public1@home1.net>;tag=171828
  To: <tel:+1-212-555-2222>;tag=314159
  Call-ID: cb03a0s09a2sdfglkj490333
  Require: sec-agree
  Proxy-Require: sec-agree
  Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321;
port-c=8642; port-s=7531
  CSeq: 153 BYE
  Content-Length: 0
```

OK from P-CSCF to UE on the release originating side

```
  SIP/2.0 200 OK
  Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7
  From: <sip:user1_public1@home1.net>;tag=171828
  To: <tel:+1-212-555-2222>;tag=314159
  Call-ID: cb03a0s09a2sdfglkj490333
  CSeq: 153 BYE
  Content-Length: 0
```

Metrics on the release are presented in Table 6.

Table 6: Release flow metrics

Message	Visible Ascii	With CRLF
Bye	659	685
200 OK	239	253
Total		938

3.4 Summary of the MT and MO flow signal transfer delay

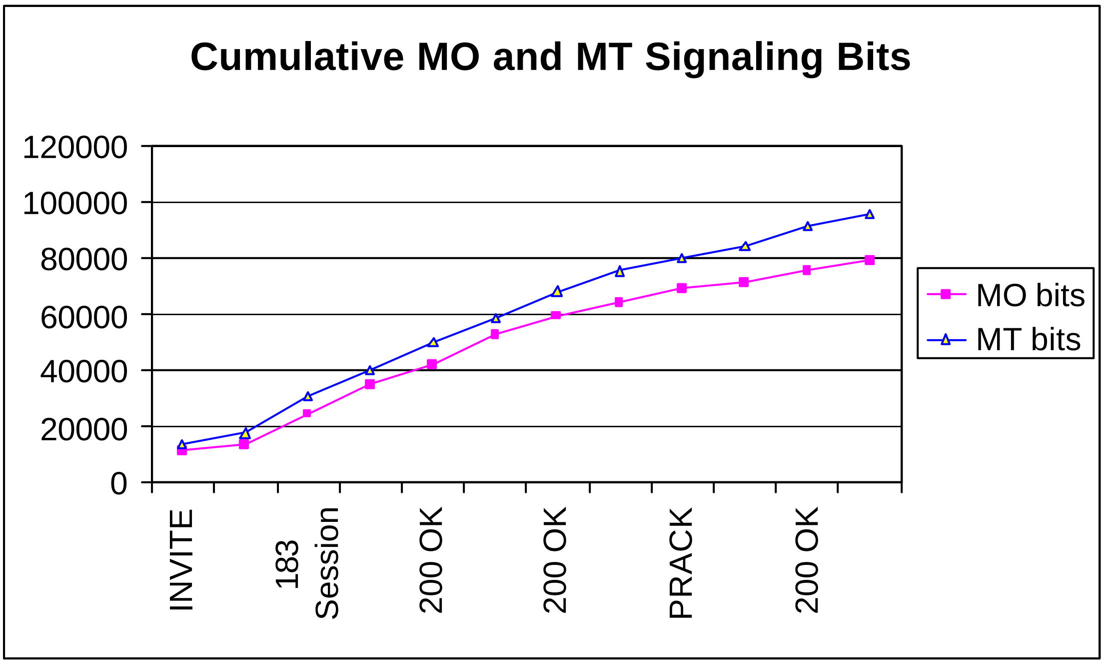
Figure 2 presents the size of the MO and MT flows in bits. Using formula one, we can convert the values on the vertical axis into signaling transfer delay. For example, let us assume that we spend 10 kbps for the signaling channel. As the result, 100 000 bits equals to 10 seconds in signaling transfer delay.

In a Mobile to Mobile both the MO and MT delays are summed provided there is no parallelism in the two flows. One can argue that such parallelism is likely to occur with one message only when OK of the Update message on the MO interface and the MT interface is transferring the 180 Ringing message.

Even discounting for such parallelism, if 10 kbps is allocated for signaling on both the MO and the MT interfaces, the total signaling transfer delay for the call setup is 17.5 seconds on signaling channels of 10 kbps.

To avoid such delays, the operators have the choice of allocating more than 10 kbps for the signaling channel and they are forced to do that in order not to create another WAP. It seems reasonable to suggest that a signaling channel of 128 kbps is needed for the IMS.

Figure 2: Bitscounts on the MO and MT SIP flows.



Another way of looking the efficiency of SIP signaling in 3G is compare the number of signaling bits to voice bits. With an AMR codec reasonable voice quality may be achieved even with the 4.75 kbps mode. (One reason of using such a low bit rate codec is the packet header overhead).

If we add the release flow in Section 3.3 at the end of the MO flow, we get that with 4.75 kbps AMR codec, it takes 18.2 seconds of continuous voice to generate as many voice bits as are needed for call signaling on the MO interface.

4 Optimizing the MO SIP Flow in the Ascii Domain

We will present the optimization process step by step so that it will be easy to see what can be achieved by different optimization methods. We start the optimization process by placing the MO flow of Section 3.1 in a spreadsheet, numbering the lines and marking the lines that can not be removed. Obviously, the start lines can not be removed because they imply that a certain event has occurred. It is also clear that both parties in the MO signaling i.e. the Mobile device and the P-CSCF can number the lines of the flow exactly in the same way as we are doing in the spreadsheet.

4.1 Replacing the redundant lines by line references

The optimization starts by sorting the lines so that line redundancy is easy to see. A redundant line in the ascii based SIP is easy to replace by a line reference that would always be pointing to a line that has occurred previously in the flow. The sender of a signal would just replace a redundant line with a short line reference of the form "=´line number´" or "=´line-number´ ´line-number´" where one, two or three digits are needed for the line number. Since the receiver is able to number the lines in exactly the same way as the sender and it has stored all the previous lines of the flow, it can easily replace the reference with the stored line. (In this document we use a slightly different format for the line number references for compatibility reasons with our spreadsheet program).

Since one can not make much sense of a flow sorted in an alphabetical order, we sort it back in to the order of appearance in the original flow but with the redundant lines replaced by the line references. The result is presented in Table 7. Column one shows the line number of the flow. Column two shows the visible characters of the SIP line itself. Column three shows the number of all characters in the line.

Table 7: The MO flow with redundant lines replaced by line references.

Line Nr	SIP	Nrof char
1	**INVITE tel:+1-212-555-2222 SIP/2.0**	36
2	Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp;branch=z9hG4bKnashds7	82
3	Max-Forwards: 70	18
4	Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>	79
5	P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56
6	P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73
7	Privacy: none	15
8	From: <sip:user1_public1@home1.net>;tag=171828	48
9	To: <tel:+1-212-555-2222>	27
10	Call-ID: cb03a0s09a2sdfglkj490333	35
11	Cseq: 127 INVITE	18
12	Require: precondition, sec-agree	34

13	Proxy-Require: sec-agree	26
14	Supported: 100rel	19
15	Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357;comp=sigcomp>	58
16	Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64
17	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113
18	Content-Type: application/sdp	31
19	Content-Length: (570)	23
20		2
21	v=0	5
22	o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd	56
23	s=-	5
24	c=IN IP6 5555::aaa:bbb:ccc:ddd	32
25	t=0 0	7
26	m=video 3400 RTP/AVP 98 99	28
27	b=AS:75	9
28	a=curr:qos local none	23
29	a=curr:qos remote none	24
30	a=des:qos mandatory local sendrecv	36
31	a=des:qos none remote sendrecv	32
32	a=rtpmap:98 H263	18
33	a=rtpmap:99 MP4V-ES	21
34	a=fmtp:98 profile-level-id=0	30
35	m=audio 3456 RTP/AVP 97 96	28
36	b=AS:25.4	11
37	#L=28 29 30 31 90	17
42	a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41
43	a=rtpmap:96 telephone-event	29
44	**SIP/2.0 100 Trying**	20
45	#L=2 8 9 10 11	14
50	Content-Length: 0	19
51	**SIP/2.0 183 Session Progress**	30
52	#L=2	4
53	Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>, <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>	146
54	P-Media-Authorization: 0020000100100101706466312e686f6d65312e6e6574000c02013942563330373200	93
55	P-Asserted-Identity: "John Smith" <tel:+1-212-555-2222>	57
56	#L=7 8 9 10 11	14
61	Require: 100rel	17
62	Contact: <sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp>	58
63	#L=16	5
64	RSeq: 9021	12
65	#L=18	5
66	Content-Length: (636)	23
67		2
68	#L=21	5
69	o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb	56

70	#L=23	5
71	c=IN IP6 5555::eee:fff:aaa:bbb	32
72	#L=25	5
73	m=video 10001 RTP/AVP 98 99	29
74	#L=27 28 29 30	14
78	a=des:qos mandatory remote sendrecv	37
79	a=conf:qos remote sendrecv	28
80	#L=32 33 34	11
83	m=audio 6544 RTP/AVP 97 96	28
84	#L=36 28 29 30 78 79	20
90	a=rtpmap:97 AMR	17
91	#L=42 43	8
93	**PRACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0**	61
94	#L=2 3 6	9
97	Route: <sip:pcscf1.home1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>	133
98	#L=8	4
99	To: <tel:+1-212-555-2222>;tag=314159	38
100	#L=10	5
101	Cseq: 128 PRACK	17
102	#L=12 13 17	11
105	RAck: 9021 127 INVITE	23
106	#L=18	5
107	Content-Length: (555)	23
108		2
109	#L=21	5
110	o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd	56
111	#L=23 24 25	11
114	m=video 3400 RTP/AVP 98	25
115	#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50
131	**SIP/2.0 200 OK**	16
132	#L=2 8 99 10 101 18	19
138	Content-Length: (612)	23
139		2
140	#L=21	5
141	o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb	56
142	#L=23 71 25	11
145	m=video 10001 RTP/AVP 98	26
146	#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56
164	**UPDATE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0**	62
165	#L=2 3 6 97 8 99 10	19
172	Cseq: 129 UPDATE	18
173	Require: sec-agree	20
174	#L=13 17 18	11
177	Content-Length: (563)	23
178		2
179	#L=21	5
180	o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd	56

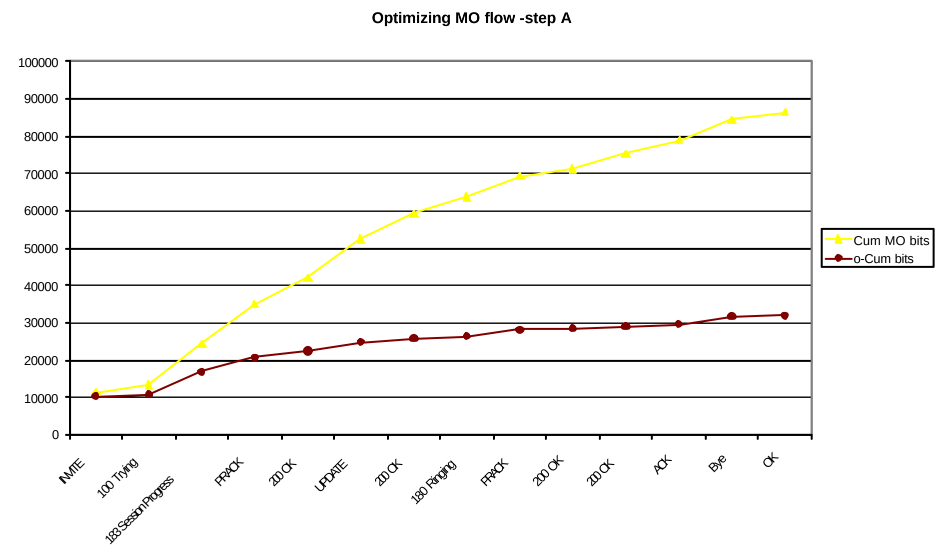
181	#L=23 24 25 114 27	18
186	a=curr:qos local sendrecv	27
187	#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45
201	**SIP/2.0 200 OK**	16
202	#L=2 8 99 10 172 18 177	23
209		2
210	#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72
232	**SIP/2.0 180 Ringing**	21
233	#L=2 53 8 99 10	15
238	Cseq: 129 INVITE	18
239	#L=61 62 16	11
242	RSeq: 9022	12
243	#L=50	5
244	**PRACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0**	61
245	#L=2 3 6 97 8 99 10	19
252	Cseq: 130 PRACK	17
253	#L=173 13	9
255	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81
256	RAck: 9022 127 INVITE	23
257	#L=50	5
258	**SIP/2.0 200 OK**	16
259	#L=2 8 99 10 252 50	19
265	**SIP/2.0 200 OK**	16
266	#L=2 53 8 99 10	15
271	Cseq: 130 INVITE	18
272	#L=62 16 50	11
275	**ACK sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0**	59
276	#L=2 3 97 8 99 10	17
282	Cseq: 127 ACK	15
283	#L=50	5
284	**BYE sip:[5555::eee:fff:aaa:bbb]:8805;comp=sigcomp SIP/2.0**	59
285	#L=2 3	6
287	Route: <sip:pcscf1.visited1.net:7531;lr;comp=sigcomp>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>	139
288	#L=6 8 99 10 173 13 17	22
295	CSeq: 153 BYE	15
296	#L=50	5
297	**SIP/2.0 200 OK**	16
298	#L= 2 8 99 10 295 50	20
	Total chars after replacement	**4004**

	Original Total for flow incl release	**10 810**
	Original Total bits for flow incl release	**71 435**
	Compression ratio	**2,7**

By replacing the redundant lines with line references we have achieved a compression ratio of 2.7. In the process we have for example removed the session identification headers from all the messages except the INVITE message.

We present our result graphically in Figure 3.

Figure 3: MO flow optimization – redundant lines removed.



The Figure demonstrates that the method is rather effective towards the end of the flow but has a very limited impact on the first three messages.

Converting back to milliseconds – we still are creating a signaling transfer delay of 3.2 seconds on a 10 kbps signaling channel, down from about 10 seconds with the original SIP flow. The result generates as many signaling bits for call signaling as the voice stream of 6.7 seconds when the 4.75 kbps AMR codec is used.

It also seems that removing session identification from messages does not necessarily lead to a robust signaling method.

4.2 Assuming a stateful proxy in the network

Let us now assume that in a cellular network the signaling will always flow through a proxy that can have some state and that the UE and the proxy (P-CSCF) are connected in a point to point manner to each other. We can remove the headers from SIP that are there because of the stateless proxy option.

Tempting headers to be removed are the To –header because the information in it is not used by the network, the Call-ID –header just because it seems pointless to carry a global call identifier in every message, the Max-Forwards and Via –headers because our communicating parties are connected in a point to point manner over a PDP context, all references to SigComp because we are not going to use SigComp anyway. Note also that the above modifications lead to removing lines from the protocol. As a result, we also need to remove the line references in the flow to the removed lines.

For robustness we need to replace the session identification used in the current SIP with short virtual identifiers that a significant only on our interface. In this paper we assume that the UE will identify the session using its own virtual reference value for the session and the proxy will generate its own reference value. The combination of the two reference values, the IP addresses and the port values of the UE and the proxy identify a session on the MO interface unambiguously.

Table 8 presents the result after these modifications.

Table 8: Removing extra headers and parameters.

Line	SIP	Chars	Msg
1	**INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12**	44	
4	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>	66	
5	P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56	
6	P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73	
7	Privacy: none	15	
8	From: <sip:user1_public1@home1.net>;tag=171828	48	
11	Cseq: 127 INVITE	18	
12	Require: precondition, sec-agree	34	
13	Proxy-Require: sec-agree	26	
14	Supported: 100rel	19	
15	Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>	45	
16	Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64	
17	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113	
18	Content-Type: application/sdp	31	
19	Content-Length: (570)	23	
20		2	
21	v=0	5	
22	o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd	56	
23	s=-	5	
24	c=IN IP6 5555::aaa:bbb:ccc:ddd	32	
25	t=0 0	7	
26	m=video 3400 RTP/AVP 98 99	28	
27	b=AS:75	9	
28	a=curr:qos local none	23	
29	a=curr:qos remote none	24	
30	a=des:qos mandatory local sendrecv	36	
31	a=des:qos none remote sendrecv	32	
32	a=rtpmap:98 H263	18	
33	a=rtpmap:99 MP4V-ES	21	
34	a=fmtp:98 profile-level-id=0	30	
35	m=audio 3456 RTP/AVP 97 96	28	
36	b=AS:25.4	11	
37	#L=28 29 30 31 90	17	
42	a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41	

43	a=rtpmap:96 telephone-event	29	1129
44	**SIP/2.0 100 Trying Sf:U12 N12345**	34	
45	#L=8 11	7	
50	Content-Length: 0	19	60
51	**SIP/2.0 183 Session Progress Sf:U12**	37	
53	Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>, <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr>	133	
54	P-Media-Authorization: 00200001001001017064663 12e686f6d65312e6e657400 0c02013942563330373200	93	
55	P-Asserted-Identity: "John Smith" <tel:+1-212-555-2222>	57	
56	#L=7 8 11	9	
61	Require: 100rel	17	
62	Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>	45	
63	#L=16	5	
64	RSeq: 9021	12	
65	#L=18	5	
66	Content-Length: (636)	23	
67		2	
68	#L=21	5	
69	o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb	56	
70	#L=23	5	
71	c=IN IP6 5555::eee:fff:aaa:bbb	32	
72	#L=25	5	
73	m=video 10001 RTP/AVP 98 99	29	
74	#L=27 28 29 30	14	
78	a=des:qos mandatory remote sendrecv	37	
79	a=conf:qos remote sendrecv	28	
80	#L=32 33 34	11	
83	m=audio 6544 RTP/AVP 97 96	28	
84	#L=36 28 29 30 78 79	20	
90	a=rtpmap:97 AMR	17	
91	#L=42 43	8	733
93	**PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	59	
94	#L=6	5	
97	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>	120	
98	#L=8	4	
101	Cseq: 128 PRACK	17	
102	#L=12 13 17	11	
105	RAck: 9021 127 INVITE	23	
106	#L=18	5	
107	Content-Length: (555)	23	
108		2	
109	#L=21	5	
110	o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd	56	
111	#L=23 24 25	11	

114	m=video 3400 RTP/AVP 98	25	
115	#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50	416
131	**SIP/2.0 200 OK Sf:U12**	23	
132	#L=8 101 18	11	
138	Content-Length: (612)	23	
139		2	
140	#L=21	5	
141	o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb	56	
142	#L=23 71 25	11	
145	m=video 10001 RTP/AVP 98	26	
146	#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56	213
164	**UPDATE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	60	
165	#L=6 97 8	9	
172	Cseq: 129 UPDATE	18	
173	Require: sec-agree	20	
174	#L=13 17 18	11	
177	Content-Length: (563)	23	
178		2	
179	#L=21	5	
180	o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd	56	
181	#L=23 24 25 114 27	18	
186	a=curr:qos local sendrecv	27	
187	#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45	294
201	**SIP/2.0 200 OK Sf:U12**	23	
202	#L=8 172 18 177	15	
209		2	
210	#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72	112
232	**SIP/2.0 180 Ringing Sf:U12**	28	
233	#L=53 8	7	
238	Cseq: 129 INVITE	18	
239	#L=61 62 16	11	
242	RSeq: 9022	12	
243	#L=50	5	81
244	**PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	59	
245	#L=6 97 8	9	
252	Cseq: 130 PRACK	17	
253	#L=173 13	9	
255	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81	
256	RAck: 9022 127 INVITE	23	
257	#L=50	5	203
258	**SIP/2.0 200 OK Sf:U12**	23	
259	#L=8 252 50	11	34
265	**SIP/2.0 200 OK Sf:U12**	23	
266	#L=53 8	7	
271	Cseq: 130 INVITE	18	

272	#L=62 16 50	11	59
275	**ACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	57	
276	#L=97 8	7	
282	Cseq: 127 ACK	15	
283	#L=50	5	84
284	**BYE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	57	
287	Route: <sip:pcscf1.visited1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>	126	
288	#L=6 8 173 13 17	16	
295	CSeq: 153 BYE	15	
296	#L=50	5	219
297	**SIP/2.0 200 OK Sf:U12**	23	
298	#L= 8 295 50	12	35
	Total chars after replacement	**3672**	

After the modifications the compression ratio is 2.9.

We can see from what we have left in Table 8 that by identifying repeated elements on the parameter level, we could still compress the result somewhat in the ascii domain. One can also argue that the point when we would need to start deviating from the ideal of clarity and human readability is close if we were to continue in this direction. We will nevertheless pursue this approach in Section 6.

For now, our next step is to move to the binary domain in order to obtain a significant improvement in compressing SIP. While taking this step, let's keep in mind that we are not in the business of defining a new protocol but the idea is to introduce a new representation for SIP.

5 Optimizing the MO SIP Flow in the Binary Domain

In this Chapter we will first discuss the common header. Next we will perform a basic replacement of Ascii representation with TLV encoding line by line. In order to achieve a higher compression ratio we will introduce TLV –element references for elements that have long values and finally we will discuss routing delegation as a means to optimize SIP signaling in a cellular context.

5.1 A possible common header for all messages

Let us first define a common header for all messages. This is presented in Figure 4.

<table>
<tr><td colspan="2">0 1 2 3 4 5 6 7</td><td>0 1 2 3 4 5 6 7</td><td>0 1 2 3 4 5 6 7</td><td>0 1 2 3 4 5 6 7</td></tr>
<tr><td>V</td><td>xxxA</td><td colspan="3">Session Reference Value</td></tr>
<tr><td colspan="3">R/M Msg Type or Response Code</td><td colspan="2">Cseq</td></tr>
</table>

Figure 4: A common Header for all TLV encoded SIP messages

The common header has fields for session and transaction identification, for SIP version, for Message types that distinguish request messages and for response codes that appear in responses. For the reference value we also need a flag to indicate whether it was allocated by the UE or the proxy or alternatively by the sender or the receiver.

For network allocated session references at least 3 bytes are needed because it is easy to imagine a system that can handle more than 65 000 simultaneous sessions. On the other hand the need to be able to identify more than 16 Million simultaneous sessions locally is not clear. Therefore 3 bytes would seem to be just right. For sequence numbers for ensuring the order of messages in a session it would seem that 16 bits would be quite enough in a cellular context.

Response codes can be encoded with 10 bits (response codes are between 100 and 699). By using one bit to distinguish between response codes and message types, we can encode both the message type and the response code in the same field since they do not appear in the same messages.

Alternatively, one could argue that since the common header appears in all messages, it should be as short as possible. With some compromises it might be possible to put all those same fields in 6 bytes. We can take these two common header sizes into account in our further analysis.

5.2 Basic TLV encoding for SIP

We are now ready to assign types and values for all the remaining information in our SIP flow. The result is in table 9.

Table 9: A possible TLV/TV-pair encoding for the MO flow.

MO FLOW BASE TLV

		Chars	Nrof New Types	Length Bytes	Value Bytes	Sum Bytes	Sum msg Hder=8
1	**INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12**	44	1	1	12	14	
4	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>	66	2	4	45	53	
5	P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56	2	3	20	26	
6	P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73	3	3	14	20	
7	Privacy: none	15	1	1	1	3	
8	From: <sip:user1_public1@home1.net>;tag=171828	48	1	1	27	29	
11	Cseq: 127 INVITE	18	0	0	0	0	
12	Require: precondition, sec-agree	34	1	1	2	4	
13	Proxy-Require: sec-agree	26	1	1	1	3	
14	Supported: 100rel	19	1	1	1	3	
15	Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>	45	1	3	29	35	
16	Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64	1	1	8	10	
17	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113	5	7	16	30	
18	Content-Type: application/sdp	31	1	1	1	3	
19	Content-Length: (570)	23	1	2		3	
20		2				0	
21	v=0	5	1	1	1	3	
22	o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd	56	4	5	16	26	
23	s=-	5				0	
24	c=IN IP6 5555::aaa:bbb:ccc:ddd	32	1	3	17	23	
25	t=0 0	7				0	
26	M=video 3400 RTP/AVP 98 99	28	5	6	6	18	
27	b=AS:75	9	1	1	2	4	
28	a=curr:qos local none	23	1	1	2	4	
29	a=curr:qos remote none	24				0	
30	a=des:qos mandatory local sendrecv	36	1	1	2	4	
31	a=des:qos none remote sendrecv	32				0	
32	a=rtpmap:98 H263	18	1	1	2	4	
33	a=rtpmap:99 MP4V-ES	21		1	2	4	
34	a=fmtp:98 profile-level-id=0	30	2	3	2	8	
35	M=audio 3456 RTP/AVP 97 96	28		6	6	18	
36	b=AS:25.4	11		1	2	4	

37	#L=28 29 30 31 90	17	1	1	5	7	
42	a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41	2	3	6	12	
43	a=rtpmap:96 telephone-event	29		1	2	4	387
44	**SIP/2.0 100 Trying Sf:U12 N12345**	34	1	1	3	5	
45	#L=8 11	7		1	2	4	
50	Content-Length: 0	19		1	1	3	20
51	**SIP/2.0 183 Session Progress Sf:U12**	37				0	
53	Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>, <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr>	133	1	6	90	102	
54	P-Media-Authorization: 0020000100100101706466312e686f6 d65312e6e6574000c02013942563330 373200	93	1	1	34	36	
55	P-Asserted-Identity: "John Smith" <tel:+1-212-555-2222>	57	1	3	22	28	
56	#L=7 8 11	9		1	3	5	
61	Require: 100rel	17		1	1	3	
62	Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>	45		3	29	35	
63	#L=16	5		1	1	3	
64	RSeq: 9021	12	1	1	2	4	
65	#L=18	5		1	1	3	
66	Content-Length: (636)	23		1	1	3	
67		2				0	
68	#L=21	5		1	1	3	
69	o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb	56		5	16	26	
70	#L=23	5		1	1	3	
71	c=IN IP6 5555::eee:fff:aaa:bbb	32		3	17	23	
72	#L=25	5		1	1	3	
73	M=video 10001 RTP/AVP 98 99	29		6	6	18	
74	#L=27 28 29 30	14		1	4	6	
78	a=des:qos mandatory remote sendrecv	37		1	1	3	
79	a=conf:qos remote sendrec v	28	1	1	1	3	
80	#L=32 33 34	11		1	3	5	
83	M=audio 6544 RTP/AVP 97 96	28		6	6	18	
84	#L=36 28 29 30 78 79	20		1	6	8	
90	a=rtpmap:97 AMR	17		1	2	4	
91	#L=42 43	8		1	2	4	357
93	**PRACK** **sip:[5555::eee:fff:aaa:bbb]:8805;** **SIP/2.0 Sf:N12345**	59		2	29	33	
94	#L=6	5		1	1	3	
97	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>	120		5	84	94	
98	#L=8	4		1	1	3	

101	Cseq: 128 PRACK	17		0	0	0	
102	#L=12 13 17	11		1	3	5	
105	RAck: 9021 127 INVITE	23	4	4	4	12	
106	#L=18	5		1	1	3	
107	Content-Length: (555)	23		1	1	3	
108		2				0	
109	#L=21	5		1	1	3	
110	o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd	56		5	16	26	
111	#L=23 24 25	11		1	3	5	
114	M=video 3400 RTP/AVP 98	25		5	5	15	
115	#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50		1	16	18	231
131	**SIP/2.0 200 OK Sf:U12**	23				0	
132	#L=8 101 18	11		1	3	5	
138	Content-Length: (612)	23		1	1	3	
139		2				0	
140	#L=21	5		1	1	3	
141	o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb	56		5	16	26	
142	#L=23 71 25	11		1	3	5	
145	m=video 10001 RTP/AVP 98	26		5	5	15	
146	#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56		1	18	20	85
164	**UPDATE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	60		2	29	33	
165	#L=6 97 8	9		1	3	5	
172	Cseq: 129 UPDATE	18				0	
173	Require: sec-agree	20		1	1	3	
174	#L=13 17 18	11		1	3	5	
177	Content-Length: (563)	23		1	1	3	
178		2				0	
179	#L=21	5		1	1	3	
180	o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd	56		5	16	26	
181	#L=23 24 25 114 27	18		1	5	7	
186	a=curr:qos local sendrecv	27		1	1	3	
187	#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45		1	14	16	112
201	**SIP/2.0 200 OK Sf:U12**	23				0	
202	#L=8 172 18 177	15		1	4	6	
209		2				0	
210	#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72		1	22	24	38
232	**SIP/2.0 180 Ringing Sf:U12**	28				0	
233	#L=53 8	7		1	2	4	
238	Cseq: 129 INVITE	18				0	
239	#L=61 62 16	11		1	3	5	
242	RSeq: 9022	12		1	2	4	
243	#L=50	5		1	1	3	24
244	**PRACK**	59		2	29	33	

	sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345						
245	#L=6 97 8	9		1	3	5	
252	Cseq: 130 PRACK	17				0	
253	#L=173 13	9		1	2	4	
255	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81		6	10	22	
256	RAck: 9022 127 INV ITE	23		4	4	12	
257	#L=50	5		1	1	3	87
258	**SIP/2.0 200 OK Sf:U12**	23				0	
259	#L=8 252 50	11		1	3	5	13
265	**SIP/2.0 200 OK Sf:U12**	23				0	
266	#L=53 8	7		1	2	4	
271	Cseq: 130 INVITE	18				0	
272	#L=62 16 50	11		1	3	5	17
275	**ACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	57		2	29	33	
276	#L=97 8	7		1	2	4	
282	Cseq: 127 ACK	15				0	
283	#L=50	5		1	1	3	48
284	**BYE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345**	57		2	29	33	
287	Route: <sip:pcscf1.visited1.net:7531;lr>, <sip:scscf1.home1. net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>	126		6	90	102	
288	#L=6 8 173 13 17	16		1	5	7	
295	CSeq: 153 BYE	15				0	
296	#L=50	5		1	1	3	153
297	**SIP/2.0 200 OK Sf:U12**	23				0	
298	#L= 8 295 50	12	1	6	6	13	21
	Total chars after re placement	**3672**	53	220	1047	**1481**	**1593**
	Original Total for flow incl release	10810					
	Original Total bits for flow incl release						
	Compression ratio	**2.9**					**6,8**

The basic line by line replacement of ascii representation by TLV encoding requires 53 types each of one byte, 1047 bytes for values and the total of 1593 bytes including the 8 byte headers for each message. For some complex headers we use nested TLV encoding. The result is a compression ratio of 6.8. If we manage to squeeze the header to 6 bytes, the compression ratio grows to 6.9.

5.3 Replacing repeated long values with TLV references

Like lines, both the signaling sender and the receiver can number TLV elements and therefore they will understand references to TLV elements equally. We can easily identify repeated parameters in our flow by taking long values from the ascii MO flow onto their own lines in a spreadsheet, ordering the lines in an alphabetical order, replacing the repeated elements with the references pointing to the first element in a sequence of identical elements and reordering the lines of the table into their original order.

The result is presented in Table 10.

Table 10: The Impact of TLV references

SIP Content	New TLV ref	TLV ref Types	Removed Rem Lngth	Msg SUM
INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12				
Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>				
P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>				
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11				
Privacy: none				
From: <sip:user1_public1@home1.net>;tag=171828				
Cseq: 127 INVITE				
Require: precondition, sec-agree				
Proxy-Require: sec-agree				
Supported: 100rel				
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>				
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE				
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531				
Content-Type: application/sdp				
Content-Length: (570)				
v=0				
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd				
2987933615	:22:36	1	4	
s=-				
c=IN IP6 5555::aaa:bbb:ccc:ddd				
aaa:bbb:ccc:ddd	:22:38	1	15	
t=0 0				
m=video 3400 RTP/AVP 98 99				
b=AS:75				
a=curr:qos local none				
a=curr:qos remote none				
a=des:qos mandatory local sendrecv				

a=des:qos none remote sendrecv
a=rtpmap:98 H263
a=rtpmap:99 MP4V-ES
a=fmtp:98 profile-level-id=0
m=audio 3456 RTP/AVP 97 96
b=AS:25.4
#L=28 29 30 31 90
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 telephone-event 368
SIP/2.0 100 Trying Sf:U12 N12345
#L=8 11
Content-Length: 0 20
SIP/2.0 183 Session Progress Sf:U12
Record-Route: <sip:pcscf2.visited2.net;lr>,
<sip:scscf2.home2.net;lr>, <sip:scscf1.home1.net;lr>,
<sip:pcscf1.visited1.net:7531;lr>
sip:scscf1.home1.net :4:6 1 20
P-Media-Authorization:
0020000100100101706466312e686f6d65312e6e6574000c
02013942563330373200
P-Asserted-Identity: "John Smith" <tel:+1-212-555-2222>
tel:+1-212-555-2222 :1:2 1 15
#L=7 8 11
Require: 100rel
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>
#L=16
RSeq: 9021
#L=18
Content-Length: (636)

#L=21
o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb
2987933623 :69:83 1 4
#L=23
c=IN IP6 5555::eee:fff:aaa:bbb
eee:fff:aaa:bbb :69:85 1 15
#L=25
m=video 10001 RTP/AVP 98 99
#L=27 28 29 30
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv
#L=32 33 34
m=audio 6544 RTP/AVP 97 96
#L=36 28 29 30 78 79
a=rtpmap:97 AMR
#L=42 43 303
PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345
sip:[5555::eee:fff:aaa:bbb] :62:74 1 27
#L=6
Route: <sip:pcscf1.home1.net:7531;lr>,

<sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>				
sip:pcscf1.home1.net	:4:4	1	20	
sip:scscf1.home1.net	:4:6	1	20	
sip:scscf2.home2.net	:53:63	1	20	
#L=8				
Cseq: 128 PRACK				
#L=12 13 17				
RAck: 9021 127 INVITE				
#L=18				
Content-Length: (555)				
#L=21				
o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd				
aaa:bbb:ccc:ddd	:22:38	1	15	
#L=23 24 25				
m=video 3400 RTP/AVP 98				
#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43				129
SIP/2.0 200 OK Sf:U12				
#L=8 101 18				
Content-Length: (612)				
#L=21				
o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb				
2987933623	:69:83	1	4	
eee:fff:aaa:bbb	:69:85	1	15	
#L=23 71 25				
m=video 10001 RTP/AVP 98				
#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43				66
UPDATE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345				
sip:[5555::eee:fff:aaa:bbb]	:62:74	1	27	
#L=6 97 8				
Cseq: 129 UPDATE				
Require: sec-agree				
#L=13 17 18				
Content-Length: (563)				
#L=21				
o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd				
2987933615	:22:36	1	4	
aaa:bbb:ccc:ddd	:22:38	1	15	
#L=23 24 25 114 27				
a=curr:qos local sendrecv				
#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43				66
SIP/2.0 200 OK Sf:U12				
#L=8 172 18 177				
#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186				38

29 30 78 90 42 43				
SIP/2.0 180 Ringing Sf:U12				
#L=53 8				
Cseq: 129 INVITE				
#L=61 62 16				
RSeq: 9022				
#L=50				24
PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345				
sip:[5555::eee:fff:aaa:bbb]	:62:74	1	27	
#L=6 97 8				
Cseq: 130 PRACK				
#L=173 13				
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531				
RAck: 9022 127 INVITE				
#L=50				60
SIP/2.0 200 OK Sf:U12				
#L=8 252 50				13
SIP/2.0 200 OK Sf:U12				
#L=53 8				
Cseq: 130 INVITE				
#L=62 16 50				17
ACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345				
sip:[5555::eee:fff:aaa:bbb]	:62:74	1	27	
#L=97 8				
Cseq: 127 ACK				
#L=50				21
BYE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345				
sip:[5555::eee:fff:aaa:bbb]	:62:74	1	27	
Route: <sip:pcscf1.visited1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>				
sip:pcscf1.visited1.net	:53:65	1	23	
sip:scscf1.home1.net	:4:6	1	20	
sip:scscf2.home2.net	:53:63	1	20	
sip:pcscf2.visited2.net	:53:62	1	23	
#L=6 8 173 13 17				
CSeq: 153 BYE				
#L=50				40
SIP/2.0 200 OK Sf:U12				
#L= 8 295 50				21
Total chars after replacement		**23**	**407**	**1186**
Original Total for flow incl release	10 810			
Compression ratio				**9,1**

The second column shows the reference itself, the first number indicates the line where the parameter appeared first, the second refers to the line numbering in a spreadsheet that has long parameter values on their own lines.

The third column shows that we are able to replace 23 repeated long parameter values with references that in our table are value pair encoded for brevity. The total number of bytes that will be replaced by references is 407 (i.e. 26% from the 1593 bytes remaining after the previous step.) The resulting flow is 1186 bytes long giving a compression ratio of 9.1 compared to the original MO flow.

5.4 Routing delegation

IMS as presented in 24.228 conforms to the SIP feature that source routing of SIP messages is available for the terminals. For this purpose SIP has the ROUTE and ROUTE-RECORD headers. This opens a door for devices to request routing that does not conform to the business relationships between operators. Since IMS is about operator provided services, such a door should be considered an option that is most often not useful in a cellular context.

In our model the P-CSCF always keeps state and the user device has a point to point connection with the P-CSCF. Therefore, it would seem reasonable to provide an option where the user devices can delegate routing responsibility completely to the network. Some operators may even see that this option is the only option over the cellular air interface.

A user could request for routing delegation in the first ROUTE header in the INVITE message. In the remaining messages no ROUTE or ROUTE RECORD headers will appear.

Based on this option, Table 11 shows the final optimized MO flow.

Table 11: Routing delegation option

	Chars	Char-Msg	Routing Deleg Removed Bytes	Final Msg Sums
INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12	44			
Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>	66			
P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56			
P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73			
Privacy: none	15			
From: <sip:user1_public1@home1.net>;tag=171828	48			
Cseq: 127 INVITE	18			
Require: precondition, sec-agree	34			
Proxy-Require: sec-agree	26			
Supported: 100rel	19			
Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>	45			
Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64			
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113			
Content-Type: application/sdp	31			
Content-Length: (570)	23			

	2			
v=0	5			
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd	56			
2987933615				
s=-	5			
c=IN IP6 5555::aaa:bbb:ccc:ddd	32			
aaa:bbb:ccc:ddd				
t=0 0	7			
m=video 3400 RTP/AVP 98 99	28			
b=AS:75	9			
a=curr:qos local none	23			
a=curr:qos remote none	24			
a=des:qos mandatory local sendrecv	36			
a=des:qos none remote sendrecv	32			
a=rtpmap:98 H263	18			
a=rtpmap:99 MP4V-ES	21			
a=fmtp:98 profile-level-id=0	30			
m=audio 3456 RTP/AVP 97 96	28			
b=AS:25.4	11			
#L=28 29 30 31 90	17			
a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41			
a=rtpmap:96 telephone-event	29	1129		370
SIP/2.0 100 Trying Sf:U12 N12345	34			
#L=8 11	7			
Content-Length: 0	19	53		20
SIP/2.0 183 Session Progress Sf:U12	37			
Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>, <sip:scscf1.home1.net;lr>, <sip:pcscf1.visited1.net:7531;lr>	133		82	
sip:scscf1.home1.net				
P-Media-Authorization: 0020000100100101706466312e686f6d65312e6e657 4000c02013942563330373200	93			
P-Asserted-Identity: "John Smith" <tel:+1-212-555- 2222>	57			
tel:+1-212-555-2222				
#L=7 8 11	9			
Require: 100rel	17			
Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>	45			
#L=16	5			
RSeq: 9021	12			
#L=18	5			
Content-Length: (636)	23			
	2			
#L=21	5			
o=- 2987933623 2987933623 IN IP6 5555::eee:fff:aaa:bbb	56			
2987933623				
#L=23	5			
c=IN IP6 5555::eee:fff:aaa:bbb	32			
eee:fff:aaa:bbb				

#L=25	5			
m=video 10001 RTP/AVP 98 99	29			
#L=27 28 29 30	14			
a=des:qos mandatory remote sendrecv	37			
a=conf:qos remote sendrecv	28			
#L=32 33 34	11			
m=audio 6544 RTP/AVP 97 96	28			
#L=36 28 29 30 78 79	20			
a=rtpmap:97 AMR	17			
#L=42 43	8	733		221
PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345	59			
sip:[5555::eee:fff:aaa:bbb]				
#L=6	5			
Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.home2.net;lr>	120		34	
sip:pcscf1.home1.net				
sip:scscf1.home1.net				
sip:scscf2.home2.net				
#L=8	4			
Cseq: 128 PRACK	17			
#L=12 13 17	11			
RAck: 9021 127 INVITE	23			
#L=18	5			
Content-Length: (555)	23			
	2			
#L=21	5			
o=- 2987933615 2987933616 IN IP6 5555::aaa:bbb:ccc:ddd	56			
aaa:bbb:ccc:ddd				
#L=23 24 25	11			
m=video 3400 RTP/AVP 98	25			
#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50	416		95
SIP/2.0 200 OK Sf:U12	23			
#L=8 101 18	11			
Content-Length: (612)	23			
	2			
#L=21	5			
o=- 2987933623 2987933624 IN IP6 5555::eee:fff:aaa:bbb	56			
2987933623				
eee:fff:aaa:bbb				
#L=23 71 25	11			
m=video 10001 RTP/AVP 98	26			
#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56	213		66
UPDATE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345	60			
sip:[5555::eee:fff:aaa:bbb]				
#L=6 97 8	9			
Cseq: 129 UPDATE	18			

Require: sec-agree	20		
#L=13 17 18	11		
Content-Length: (563)	23		
	2		
#L=21	5		
o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd	56		
2987933615			
aaa:bbb:ccc:ddd			
#L=23 24 25 114 27	18		
a=curr:qos local sendrecv	27		
#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45	294	66
SIP/2.0 200 OK Sf:U12	23		
#L=8 172 18 177	15		
	2		
#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72	112	38
SIP/2.0 180 Ringing Sf:U12	28		
#L=53 8	7		
Cseq: 129 INVITE	18		
#L=61 62 16	11		
RSeq: 9022	12		
#L=50	5	81	24
PRACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345	59		
sip:[5555::eee:fff:aaa:bbb]			
#L=6 97 8	9		
Cseq: 130 PRACK	17		
#L=173 13	9		
Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81		
RAck: 9022 127 INVITE	23		
#L=50	5	203	60
SIP/2.0 200 OK Sf:U12	23		
#L=8 252 50	11	34	13
SIP/2.0 200 OK Sf:U12	23		
#L=53 8	7		
Cseq: 130 INVITE	18		
#L=62 16 50	11	59	17
ACK sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345	57		
sip:[5555::eee:fff:aaa:bbb]			
#L=97 8	7		
Cseq: 127 ACK	15		
#L=50	5	84	21
BYE sip:[5555::eee:fff:aaa:bbb]:8805; SIP/2.0 Sf:N12345	57		
sip:[5555::eee:fff:aaa:bbb]			
Route: <sip:pcscf1.visited1.net:7531;lr>, <sip:scscf1.home1.net;lr>, <sip:scscf2.home2.net;lr>, <sip:pcscf2.visited2.net;lr>	126		16
sip:pcscf1.visited1.net			

sip:scscf1.home1.net				
sip:scscf2.home2.net				
sip:pcscf2.visited2.net				
#L=6 8 173 13 17	16			
CSeq: 153 BYE	15			
#L=50	5	219		24
SIP/2.0 200 OK Sf:U12	23			
#L= 8 295 50	12	35		21
Total chars after replacement with headers	**3672**		**132**	**1056**
Original Total for flow incl release	10810			
Original Total bits for flow incl release				
Compression ratio	2,9			**10,2**

Routing delegation allows us to remove 132 bytes from the flow. After signaling routing delegation the resulting MO flow is 1056 bytes and the compression ratio is 10.2 when compared to the original MO flow in release 5 based on 24.228. For comparison we show the metrics on the optimized ascii version of the flow in columns two and three. The request for the routing delegation option is embedded in the INVITE message in the last column.

6 Best ascii

Having replaced redundant lines by line references and prior to introducing TLV encoding, we could alternatively replace also long valued parameters with parameter references and perform routing delegation in the ascii domain. Let us introduce parameter references of the form: Pf:´line-nr´.´parameter number in line´ or Pf:´line-nr´.´parameter number in line´-´parameter number in line´.

The result after introducing parameter references in the ascii domain is in Table 12.

Table 12: MO flow with parameter references added to the result in Table 8.

	SIP/SDP Content	Chars	Char-Msg
1	**INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12**	44	
3	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>	66	
7	P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56	
10	P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73	
14	Privacy: none	15	
15	From: <sip:user1_public1@home1.net>;tag=171828	48	
17	Cseq: 127 INVITE	18	
18	Require: precondition, sec-agree	34	
19	Proxy-Require: sec-agree	26	
20	Supported: 100rel	19	
21	Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>	45	
24	Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64	

25	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113	
31	Content-Type: application/sdp	31	
32	Content-Length: (570)	23	
33		2	
34	v=0	5	
35	o=- 2987933615 Pf:35.1 IN IP6 5555::aaa:bbb:ccc:ddd	53	
39	s=-	5	
40	c=Pf:35.3-5	13	
42	t=0 0	7	
43	m=video 3400 RTP/AVP 98 99	28	
44	b=AS:75	9	
45	a=curr:qos local none	23	
46	a=curr:qos remote none	24	
47	a=des:qos mandatory local sendrecv	36	
48	a=des:qos none remote sendrecv	32	
49	a=rtpmap:98 H263	18	
50	a=rtpmap:99 MP4V-ES	21	
51	a=fmtp:98 profile-level-id=0	30	
52	m=audio 3456 RTP/AVP 97 96	28	
53	b=AS:25.4	11	
54	#L=28 29 30 31 90	17	
55	a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41	
56	a=rtpmap:96 telephone-event	29	1107
57	**SIP/2.0 100 Trying Sf:U12 N12345**	34	
58	#L=8 11	7	
59	Content-Length: 0	19	60
60	**SIP/2.0 183 Session Progress Sf:U12**	37	
61	Record-Route: <sip:pcscf2.visited2.net;lr>, <sip:scscf2.home2.net;lr>, <Pf:3.2>, <sip:pcscf1.visited1.net:7531;lr>	116	
67	P-Media-Authorization: 0020000100100101706466312e686f6d65312e6e6574000c 02013942563330373200	93	
68	P-Asserted-Identity: "John Smith" <Pf:1.1>	44	
71	#L=7 8 11	9	
72	Require: 100rel	17	
73	Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>	45	
76	#L=16	5	
77	RSeq: 9021	12	
78	#L=18	5	
79	Content-Length: (636)	23	
80		2	
81	#L=21	5	
82	o=- 2987933623 Pf:82.1 IN IP6 5555::eee:fff:aaa:bbb	53	
86	#L=23	5	
87	c=Pf:82.3-5	13	
89	#L=25	5	
90	m=video 10001 RTP/AVP 98 99	29	
91	#L=27 28 29 30	14	
92	a=des:qos mandatory remote sendrecv	37	

93	a=conf:qos remote sendrecv	28	
94	#L=32 33 34	11	
95	m=audio 6544 RTP/AVP 97 96	28	
96	#L=36 28 29 30 78 79	20	
97	a=rtpmap:97 AMR	17	
98	#L=42 43	8	681
99	**PRACK Pf:73.1:8805; SIP/2.0 Sf:N12345**	39	
102	#L=6	5	
103	Route: <Pf:3.1:7531;lr>, <Pf:3.2>, <Pf:61.2>, <sip:pcscf2.home2.net;lr>	73	
108	#L=8	4	
109	Cseq: 128 PRACK	17	
110	#L=12 13 17	11	
111	RAck: 9021 127 INVITE	23	
112	#L=18	5	
113	Content-Length: (555)	23	
114		2	
115	#L=21	5	
116	o=- 2987933615 2987933616 Pf:35.3-5	37	
120	#L=23 24 25	11	
121	m=video 3400 RTP/AVP 98	25	
122	#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50	330
123	**SIP/2.0 200 OK Sf:U12**	23	
124	#L=8 101 18	11	
125	Content-Length: (612)	23	
126		2	
127	#L=21	5	
128	o=- Pf:82.1 2987933624 Pf:82.3-5	34	
132	#L=23 71 25	11	
133	m=video 10001 RTP/AVP 98	26	
134	#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56	191
135	**UPDATE Pf:73.1-2; SIP/2.0 Sf:N12345**	37	
137	#L=6 97 8	9	
138	Cseq: 129 UPDATE	18	
139	Require: sec-agree	20	
140	#L=13 17 18	11	
141	Content-Length: (563)	23	
142		2	
143	#L=21	5	
144	o=- Pf:35.1 2987933617 Pf:35.3-5	34	
148	#L=23 24 25 114 27	18	
149	a=curr:qos local sendrecv	27	
150	#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45	249
151	**SIP/2.0 200 OK Sf:U12**	23	
152	#L=8 172 18 177	15	
153		2	
154	#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72	112
155	**SIP/2.0 180 Ringing Sf:U12**	28	
156	#L=53 8	7	
157	Cseq: 129 INVITE	18	

158	#L=61 62 16	11	
159	RSeq: 9022	12	
160	#L=50	5	81
161	**PRACK Pf:73.1-2; SIP/2.0 Sf:N12345**	36	
164	#L=6 97 8	9	
165	Cseq: 130 PRACK	17	
166	#L=173 13	9	
167	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81	
169	RAck: 9022 127 INVITE	23	
170	#L=50	5	180
171	**SIP/2.0 200 OK Sf:U12**	23	
172	#L=8 252 50	11	34
173	**SIP/2.0 200 OK Sf:U12**	23	
174	#L=53 8	7	
175	Cseq: 130 INVITE	18	
176	#L=62 16 50	11	59
177	**ACK Pf:73.1-2; SIP/2.0 Sf:N12345**	34	
180	#L=97 8	7	
181	Cseq: 127 ACK	15	
182	#L=50	5	61
183	**BYE Pf:73.1-2; SIP/2.0 Sf:N12345**	34	
186	Route: <Pf:61.4-5>, <Pf:3.3>, <Pf:61.2>, <Pf:61.1>	52	
191	#L=6 8 173 13 17	16	
192	CSeq: 153 BYE	15	
193	#L=50	5	122
194	**SIP/2.0 200 OK Sf:U12**	23	
195	#L= 8 295 50	12	35
	Total chars after replacement	**3302**	**3302**
	Original Total for flow incl release	10810	
	Compression ratio	3,3	

In Table 12 we took just the longest values and created parameter references for them. So, the the term "best ascii" should not be taken literally. We also omitted the possibility of replacing long labels with shorthand variants.

On the result we can apply Routing Delegation in the ascii domain. The result is in Table 13. Column one shows the line number, column two the optimized SIP/SDP Content, column three the number of bytes in each line of the message, column four the sum of bytes in the column two ascii message. Column 5 shows how many bytes are consumed for line and parameter references in each line and finally column six shows the length of the message without the references. Such message lengths would potentially be applicable in case of the communicating finite state machine approach for signaling. We show this only for comparison rather than suggesting that such an approach would be recommended.

Table 13: Best Ascii.

	Optimized SIP/SDP Content	Chars	Char-Msg	Ref Len	Per msg
1	**INVITE tel:+1-212-555-2222 SIP/2.0 Sf: U12**	44			

3	Route: <sip:pcscf1.home1.net:7531;lr>, <sip:scscf1.home1.net;lr>, RD	70			
7	P-Preferred-Identity: "John Doe" <tel:+1-212-555-1111>	56			
10	P-Access-Network-Info: 3GPP-UTRAN-TDD; utran-cell-id-3gpp=234151D0FCE11	73			
14	Privacy: none	15			
15	From: <sip:user1_public1@home1.net>;tag=171828	48			
17	Cseq: 127 INVITE	18			
18	Require: precondition, sec-agree	34			
19	Proxy-Require: sec-agree	26			
20	Supported: 100rel	19			
21	Contact: <sip:[5555::aaa:bbb:ccc:ddd]:1357>	45			
24	Allow: INVITE, ACK, CANCEL, BYE, PRACK, UPDATE, REFER, MESSAGE	64			
25	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c=98765432; spi-s=87654321; port-c=8642; port-s=7531	113			
31	Content-Type: application/sdp	31			
32	Content-Length: (570)	23			
33		2			
34	v=0	5			
35	o=- 2987933615 Pf:35.1 IN IP6 5555::aaa:bbb:ccc:ddd	53		7	
39	s=-	5			
40	c=Pf:35.3-5	13		13	
42	t=0 0	7			
43	m=video 3400 RTP/AVP 98 99	28			
44	b=AS:75	9			
45	a=curr:qos local none	23			
46	a=curr:qos remote none	24			
47	a=des:qos mandatory local sendrecv	36			
48	a=des:qos none remote sendrecv	32			
49	a=rtpmap:98 H263	18			
50	a=rtpmap:99 MP4V-ES	21			
51	a=fmtp:98 profile-level-id=0	30			
52	m=audio 3456 RTP/AVP 97 96	28			
53	b=AS:25.4	11			
54	#L=28 29 30 31 90	17		17	
55	a=fmtp:97 mode-set=0,2,5,7; maxframes=2	41			
56	a=rtpmap:96 telephone-event	29	1111		1074
57	**SIP/2.0 100 Trying Sf:U12 N12345**	34			
58	#L=8 11	7		7	
59	Content-Length: 0	19	60		53
60	**SIP/2.0 183 Session Progress Sf:U12**	37			
67	P-Media-Authorization: 0020000100100101706466312e686f6d65312e6e6574000c02013942563330373200	93			
68	P-Asserted-Identity: "John Smith" <Pf:1.1>	44		9	
71	#L=7 8 11	9		9	
72	Require: 100rel	17			

73	Contact: <sip:[5555::eee:fff:aaa:bbb]:8805>	45			
76	#L=16	5		5	
77	RSeq: 9021	12			
78	#L=18	5		5	
79	Content-Length: (636)	23			
80		2			
81	#L=21	5		5	
82	o=- 2987933623 Pf:82.1 IN IP6 5555::eee:fff:aaa:bbb	53		8	
86	#L=23	5		5	
87	c=Pf:82.3-5	13		13	
89	#L=25	5			
90	m=video 10001 RTP/AVP 98 99	29			
91	#L=27 28 29 30	14		14	
92	a=des:qos mandatory remote sendrecv	37			
93	a=conf:qos remote sendrecv	28			
94	#L=32 33 34	11		11	
95	m=audio 6544 RTP/AVP 97 96	28			
96	#L=36 28 29 30 78 79	20		20	
97	a=rtpmap:97 AMR	17			
98	#L=42 43	8	565	8	453
99	**PRACK Pf:73.1-2; SIP/2.0 Sf:N12345**	36		11	
102	#L=6	5		5	
108	#L=8	4		4	
109	Cseq: 128 PRACK	17			
110	#L=12 13 17	11		11	
111	RAck: 9021 127 INVITE	23			
112	#L=18	5		5	
113	Content-Length: (555)	23			
114		2			
115	#L=21	5		5	
116	o=- 2987933615 2987933616 Pf:35.3-5	37		10	
120	#L=23 24 25	11		11	
121	m=video 3400 RTP/AVP 98	25			
122	#L=27 28 29 30 78 32 34 35 36 28 29 30 78 90 42 43	50	254	50	142
123	**SIP/2.0 200 OK Sf:U12**	23			
124	#L=8 101 18	11		11	
125	Content-Length: (612)	23			
126		2			
127	#L=21	5		5	
128	o=- Pf:82.1 2987933624 Pf:82.3-5	34		18	
132	#L=23 71 25	11		11	
133	m=video 10001 RTP/AVP 98	26			
134	#L=27 28 29 30 78 79 32 34 83 36 28 29 30 78 79 90 42 43	56	191	56	90
135	**UPDATE Pf:73.1-2; SIP/2.0 Sf:N12345**	37		13	
137	#L=6 97 8	9		9	
138	Cseq: 129 UPDATE	18			
139	Require: sec-agree	20			
140	#L=13 17 18	11		11	

141	Content-Length: (563)	23			
142		2			
143	#L=21	5		5	
144	o=- Pf:35.1 2987933617 Pf:35.3-5	34		20	
148	#L=23 24 25 114 27	18		18	
149	a=curr:qos local sendrecv	27			
150	#L=29 30 78 32 34 35 36 186 29 30 78 90 42 43	45	249	45	128
151	**SIP/2.0 200 OK Sf:U12**	23			
152	#L=8 172 18 177	15		15	
153		2			
154	#L=21 180 23 24 25 114 27 186 29 30 78 32 34 35 36 186 29 30 78 90 42 43	72	112	72	25
155	**SIP/2.0 180 Ringing Sf:U12**	28			
156	#L=53 8	7		7	
157	Cseq: 129 INVITE	18			
158	#L=61 62 16	11		11	
159	RSeq: 9022	12			
160	#L=50	5	81	5	58
161	**PRACK Pf:73.1-2; SIP/2.0 Sf:N12345**	36		11	
164	#L=6 97 8	9		9	
165	Cseq: 130 PRACK	17			
166	#L=173 13	9		9	
167	Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi=87654321; port1=7531	81			
169	RAck: 9022 127 INVITE	23			
170	#L=50	5	180	5	146
171	**SIP/2.0 200 OK Sf:U12**	23			
172	#L=8 252 50	11	34	11	23
173	**SIP/2.0 200 OK Sf:U12**	23			
174	#L=53 8	7		7	
175	Cseq: 130 INVITE	18			
176	#L=62 16 50	11	59	11	41
177	**ACK Pf:73.1-2; SIP/2.0 Sf:N12345**	34		11	
180	#L=97 8	7		7	
181	Cseq: 127 ACK	15			
182	#L=50	5	61	5	38
183	**BYE Pf:73.1-2; SIP/2.0 Sf:N12345**	34		11	
191	#L=6 8 173 13 17	16		16	
192	CSeq: 153 BYE	15			
193	#L=50	5	70	5	38
194	**SIP/2.0 200 OK Sf:U12**	23			
195	#L= 8 295 50	12	35	12	23
	Total chars after replacement	**3062**		730	2332
	Original Total for flow incl release	10810			
	Compression ratio	**3,53**			**4,64**

For comparison we show the compression graphically using different methods in Figure 5.

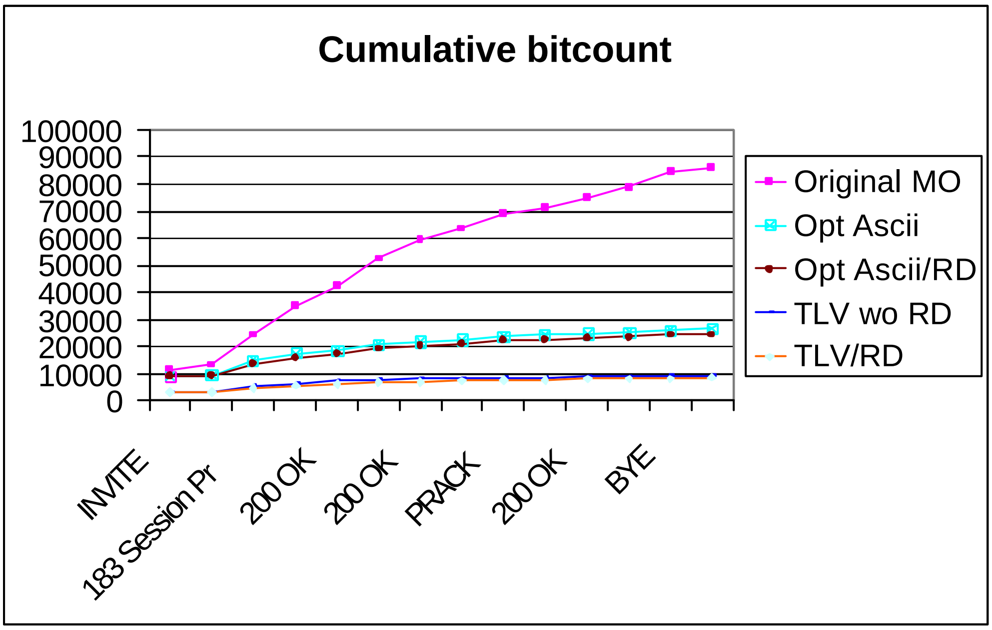


Figure 5: Comparison of different optimizations in accumulated bits

7 Compression with a fixed algorithm

The well-known programs such as gzip, zlib and Winzip are all based on first replacing redundant strings of bytes with distance-length pairs and then representing the result in a Huffman code. In Huffman code frequent symbols are presented by short codes and less frequent symbols with longer symbols. The Huffman tree is transferred with the coded file.

We can apply a zip program either on the original MO messages or on the best ascii version of SIP we have produced. Moreover, we can apply a zip program to individual messages or incrementally similar to what SigComp is doing. Incremental application means that a message *i* in a flow is compressed against the flow formed by messages 1….*i*-1. In the context of signaling this would work provided that session identification information is available to the receiver of a compressed message, so that it can find the history for decompressing the just received message. It would also be best to fix the Huffman code for the whole flow.

7.1 Zipping SIP messages one by one

Let us use WinZip for approximating this process. We break the flow in Table 13 and the original flow in Section 3.1 into ascii files per message and throw them into WinZip. The results are in Table 14. (Jörg Ott has done the same using gzip. The results are within 10% of the ones we present here. The result of gzip is a little longer than the result of WinZip).

Table 14: Results of Winzipping the original and the best ascii message by message.

SIP Message	Original Size	Winzipped individually	Best Ascii Size	Best Ascii Individually WinZipped
INVITE	1427	777	1111	698
100 Trying	251	216	60	60
183 Progress	1397	716	565	381
PRACK	1314	695	254	191
200 OK	904	469	191	146
UPDATE	1287	670	249	196
200 OK	856	457	112	86
180 Ringing	561	368	81	75
PRACK	677	443	180	160
200 OK	257	218	34	34
200 OK	526	345	59	59
ACK	450	290	61	61
Bye	537	314	70	69
OK	549	340	35	35
	10993	6318	3062	2251
Compression ratio	**1**	**1,7**	**3,6**	**4,9**

The results show that the approach is not very efficient and achieves only the compression ratio of 1.7 on the original messages. This is likely to be due to the relatively small size of files that do not give much for the algorithm to work on. Naturally, if we apply WinZip to the best arcii of Table 13, the result is better but the contribution of the zipping function towards the result is only a factor of 1.33. The lower contribution is explained by the fact that the messages is the best ascii are shorter than in the original flow.

Also we note that in a working solution, we would probably need to add overhead for session identification and take care of negotiating the particular compression algorithm in use. Hopefully, the latter could be done in the registration procedure rather than in each session.

7.2 Incremental Zipping

Let us now see, what can be achieved if we apply WinZip incrementally. For this purpose we create ascii files of all subflows using the original flow of Section 3.1 and the best ascii flow of Table 13 as the two independent sources. We WinZip the files. The result is in Table 15.

The result of incremental zipping of the original flow gives an overall compression ratio of 8:1 for the whole flow. This is approximately two times better than SigComp without Huffman coding. The result of incremental zipping of the best ascii is only slightly more compact than direct application of the zipping algorithm onto the original flow. This is expected and shows that the zipping algorithm is pretty good at finding repeated lines and parameters. The small improvement can be explained by the simplifications of the protocol that we did to get to the best ascii in Table 13. Nevertheless, this way we achieve a compression ratio of 8.7:1 compared to the original ascii SIP of the MO flow. The gain of incremental zipping on the best ascii is still 2.5:1. On the best ascii, incremental WinZip is 1.8 times better than what the application of the same algorithm individually on every message can achieve.

Based on this analysis, we can suggest that there is a reasonable alternative of applying freeware algorithms of LZ77 and Huffman coding found in zlib, gzip and WinZip with a fixed Huffman code on SIP. This approach would need to add a minimal session identification on a zipped message taking may be 4 bytes per message so that the receiver would be able to identify the correct history of the session and use the zipping algorithm incrementally. The result would likely be more compact than what SigComp can do because there would be no need to transfer byte code.

We admit that the comparison appearing from our results may not be quite fair on SigComp for two reasons: (1) the SigComp implementation that we use as a reference did not do Huffman Coding and (2) WinZip on a reasonable personal computer does not have memory limitations for such small files as the SIP messages while a SigComp implementation for a P-CSCF must take into account that the system must be able to process thousands of sessions simultaneously. Memory may be cheap but for the amounts needed for compression algorithms, it is not free.

We expect that our incremental zipping exercise shows an upper boundary what SigComp with Huffman coding can possibly achieve.

Table 15: Incremental compression of SIP using WinZip

SIP Message	Original MO Flow – Incremental WinZip: Cumulative size	WinZip	Compression Ratio	Best Ascii – Incremental Compression with WinZip: Cumulative size	WinZip	Compression Ratio Cmp to Original MO Flow	Compression Ratio Cmp to Best Ascii	Compression Ratio cmp to individual WinZip of Best Ascii
INVITE	1427	777	1,8	1110	698	2,0	1,6	1,0
100 Trying	1678	800	2,1	1172	725	2,3	1,6	1,0
183 Progress	3075	1024	3,0	1739	951	3,2	1,8	1,2
PRACK	4389	1109	4,0	1995	1032	4,3	1,9	1,3
200 OK	5293	1147	4,6	2188	1079	4,9	2,0	1,4
UPDATE	6580	1188	5,5	2439	1130	5,8	2,2	1,5
200 OK	7436	1200	6,2	2553	1149	6,5	2,2	1,5
180 Ringing	7997	1225	6,5	2638	1180	6,8	2,2	1,6
PRACK	8674	1254	6,9	2818	1206	7,2	2,3	1,7
200 OK	8931	1263	7,1	2854	1213	7,4	2,4	1,7
200 OK	9457	1279	7,4	2915	1223	7,7	2,4	1,7
ACK	9907	1290	7,7	2978	1234	8,0	2,4	1,7
Bye	10444	1361	7,7	3050	1253	8,3	2,4	1,8
OK	10993	1374	8,0	3087	1258	8,7	2,5	1,8

7.3 *Efficiency of Zipping of SIP*

Figure 6 places all the data points of WinZipping SIP messages and flows of different size in one diagram.

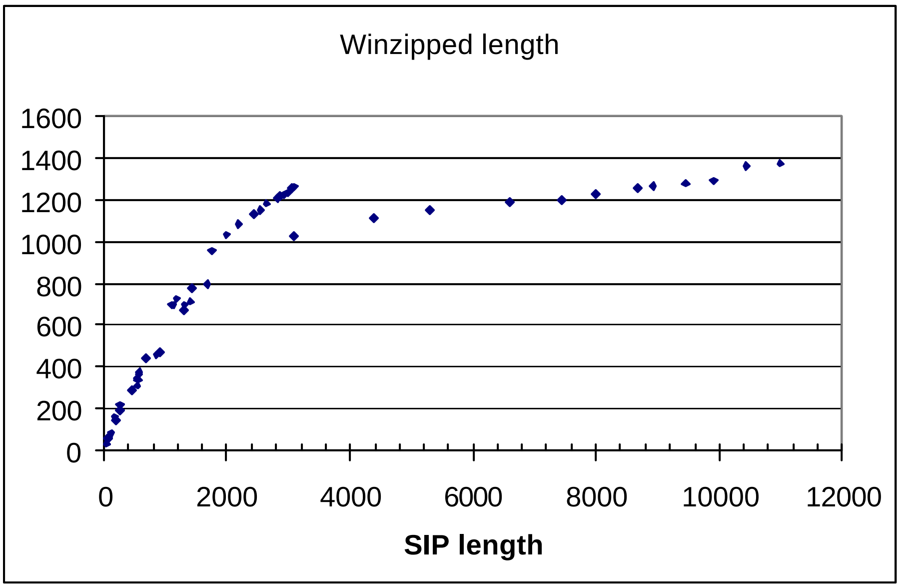


Figure 6: The relationship of SIP message/flow length to the WinZipped length.

The Figure tries to generalize our results on using a zipping algorithm for compressing SIP and SDP messages and flows of messages. It shows that roughly speaking until the size of about 2000 bytes WinZip gradually achieves a compression ratio of about 2:1 and the reminder of a flow the ratio is about 25:1, i.e. some 400 zipped bytes are generated from about 10 000 bytes of SIP/SDP material. Quite naturally, for message sizes of just a few hundred bytes, the compression ratio is significantly less than 2:1.

Let us now consider how efficient a real working solution based on LZ77 and Huffman coding could be for SIP and SDP in cellular networks provided that the solution allows agreeing on the Huffman code in the registration procedure. By looking at Figure 6, we can guess that due to the fact that no Huffman tree information needs to be transferred, it may be possible that for call sessions this approach could also achieve the compression ratio of 10:1 like we have demonstrated with TLV –encoding especially if we do some protocol simplifications such as we have suggested in this report. However, this remains to be proven with a prototype.

8 Conclusions

This report has demonstrated in detail that first by removing repetition on the line level and then on parameter level we can compress SIP signaling significantly. Moreover, we have shown that by replacing ascii (UTF-8) encoding and introducing a binary encoding of the mixed TLV and type-value pair style we can achieve a compression ratio of slightly better than 10 to 1 in SIP signaling for cellular networks. This is significantly better than the SigComp without Huffman Coding we use as reference that achieved a compression ratio of less than 4:1.

We have not used the type value pair encoding in place of full TLV consistently. There may be some bits to be saved by replacing some fixed length TLV elements (not many) with type-value pairs. It is also tempting to move the Preferred Identity Header from the session setup procedure to the registration sequence.

In Section 6, we also explored the approach of doing all our logical optimizations in the ascii domain and in Section 7 applying a fixed algorithm on the result. This revealed one reasonable alternative approach to compression. I.e use a fixed algorithm such as LZ77 and Huffman coding found in gzip, WinZip, zlib and the like incrementally on the flow. This would seem to be possible and effcient if a few bytes are added to each compressed message for session identification and if the Huffman code is fixed for the whole flow. For example the code could be agreed in registration. Alternatively, a code could be standardized. The latter may prove to be inefficient due to different frequencies of symbols appearing in names in different parts of the world.

To summarize the explored methods we present Figure 7.

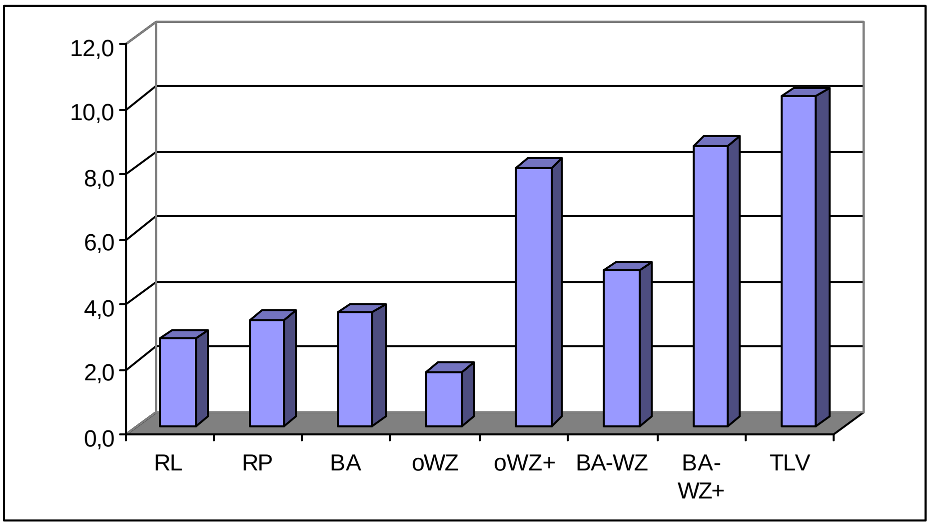


Figure 7: Summary of Compression ratios achieved by different compression methods
RL – Replace redundant lines with line references, RP – replace also redundant long parameters with parameter references, BA – Best Ascii with Routing Delegation and protocol simplifications, oWZ – Original MO Flow Winzipped message by message, oWZ+ - Original MO Flow Winzipped incrementally, BA-WZ – Best Ascii WinZipped message by message, BA-WZ+ - Best Ascii winzipped incrementally and TLV – TLV/Type-Value pair encoded protocol with routing delegation and other simplifications.

For our analysis we used the WinZip tool. Results with gzip seem to be a little less efficient than the ones we show with WinZip. The difference is however less than 10% compared to WinZip. Also this alternative seems to perform better in terms of signaling efficiency than SigComp without Huffman coding. It may even be possible to achieve a compression ratio of better than 10:1 for a session flow with this approach as well. This remains to be proven with a prototype. As is, we expect that it gives an upper boundary on what SigComp with Huffman coding could achieve considering that SigComp will need quite a bit of overhead to use the Huffman coding. Naturally, applying Huffman coding would also mean investing quite a lot of memory and cpu cycles for the effort. This downside applies to SigComp and Zipping alike.

Figure 7 shows that SigComp without Huffman coding achieves approximately the same level of compression as we could demonstrate with minor protocol simplifications and by replacing redundant lines and long valued parameters with references to the first occurrence of the line or the long parameter value respectively still keeping the presentation in ascii and thus human readable.

One should mind that the zipping solutions shown in the Figure are not real signaling solutions but approximations created using WinZip on files containing SIP/SDP messages from the MO flow or sub flows of the MO flow.

Compared to ISDN signaling our optimized TLV SIP still uses some 8 450 bits for a Mobile Originated call setup and release while Q.931/Q921 need only about 1000 bits. On a signaling channel of 16kbps the signaling transfer delay of our optimized TLV SIP is about 530 milliseconds, 8.5 times more than the 62 milliseconds needed for Q.931/Q.921. Considering that our point of comparison does not do justice to mobile telephony and that it would be more fair to compare with GSM signaling, we argue that the result is good enough for implementing efficient communications services in the Packet Switched domain in future cellular networks.

We have every reason to believe that a TLV encoded SIP would consume several orders of magnitude less memory and computing cycles than a combination of ascii SIP and SigComp. This remains to be shown by a prototype. We expect that SIP with a fixed zipping algorithm may also consume a considerable amount of memory and computing cycles on a system with thousands of SIP sessions running in parallel.

Moreover, the resulting SIP after our optimizations is somewhat simplified and better suited to a model in which services are provided by the network. Following this model we have assumed that the P-CSCF always keeps state and that the user device is point-to-point connected with the closest IMS server i.e. the P-CSCF.

The result is still SIP and we leave it for further study to show that such a TLV encoded SIP can be introduced as an extension to the overall SIP architecture in a reasonable manner. For now we show in Figure 8 a possible architecture of SIP extended with the option of processing also TLV/type-value pair encoded SIP. New elements are shown in pink.

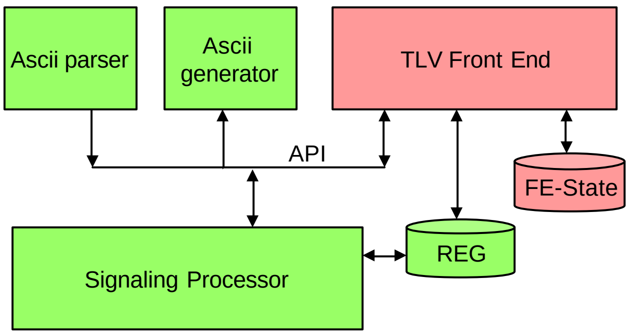


Figure 8: Extended SIP Architecture.

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