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SIP Header Reduction to Support Delay Sensitive Applications

Haseeb Akhtar, Dave Brombal, Anthony Jones and Mohamed Khalil

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>THIS IS NORTEL

Requirements for Delay Sensitive Applications

- Short call setup time
  - » PDD (Post Dial Delay) < 4 sec for VoIP/VT applications
  - » Push-to-beep < 1 sec for PTT (Push to Talk) application
- Wireless bandwidth is restrictive
  - » Even for 3G/4G technologies the average throughput per user is in the 10s of Kbytes
    - ≈ Number of users/sector
    - ≈ Distance from the cell tower
    - ≈ Interference from neighboring sectors
  - » Use control channel to send/receive initial SIP messages
    - ≈ Removes traffic channel acquisition delay from the call setup time
    - ≈ Large text-based SIP messages can not be transmitted

Initial call setup messages (e.g. SIP Invite, 200 OK) must be reduced to ~200 bytes to support delay sensitive applications over wireless access

SIP Header Reduction - Drivers

IP based Applications and Access Technologies	Current Size (Bytes)	Reduced size using SIGCOMP (Bytes)	OTA Requirement (Bytes)
Standalone PTT – 1X with SDB and BAM enabled	400	200	< 252
PTT over Cellular (PoC) – IMS integrated over DOrA	1350	675	< 211 over ACH < 113 over CCH
VoIP – IMS integrated over DOrA	1260	630	

- SIGCOMP approach
  - » Use a finite dictionary to encode contents of each parameter
  - » Limited by the terms defined in the dictionary as per RFC 3485
    - » "The dictionary is not intended to evolve as SIP or SDP evolve. It is defined once, and stays as is forever."
- Nortel's proposed approach
  - » A new (dubbed '3G') dictionary - possibly inclusive of RFC 3485
  - » Send only the dynamic information OTA
  - » Store static data in the network (i.e. P-CSCF) during SIP Registration
  - » Client-Server encoding relationship
    - Introduce an EA (Encoding Assistant) Function at the UE and P-CSCF
    - A 'standard' method of encoding/decoding the SIP message at the UE and P-CSCF

- **SIP Headers shown here are for originating SIP Invite message**
- **Header reduction using SIGCOMP is assumed to be 50%**

SIP Header Reduction – Proposed Solution

Nortel's SIP header compression proposal is an Access independent solution

LEGEND

- SIGCOMP Dictionary
- 3G Dictionary
- Added by Encoding Assistant
- Dynamic parameters sent OTA

INVITE tel:+1-212-555-2222 SIP/2.0

Encode/decode buddy list by the EA function in UE/P-CSCF

Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]:1357; comp=sigcomp; branch=z9hG4bK nashds7

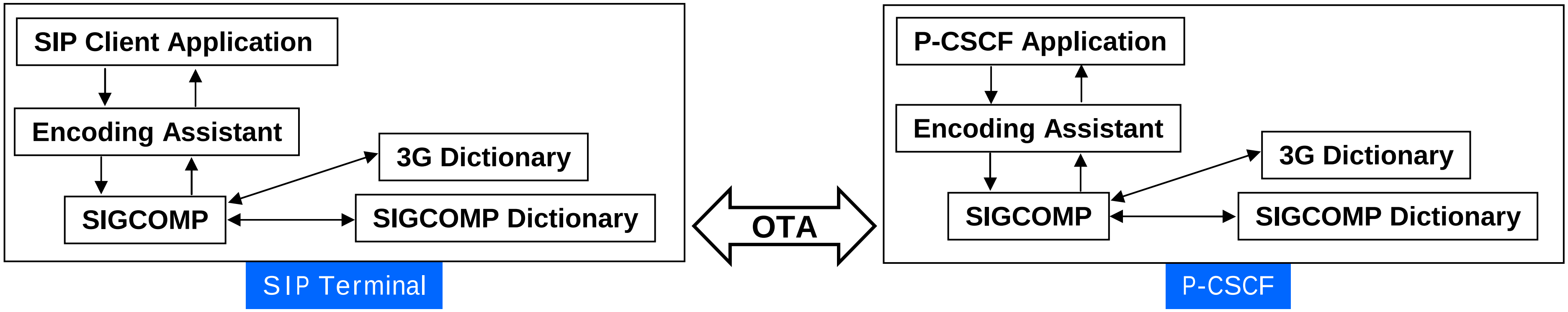
Privacy: none

Can also be inserted/retrieved by the EA function in UE/P-CSCF

From: <sip:user1_public1@home1.net>; tag= 171828

Security-Verify: ipsec-3gpp; q=0.1; alg=hmac-sha-1-96; spi-c= 98765432;

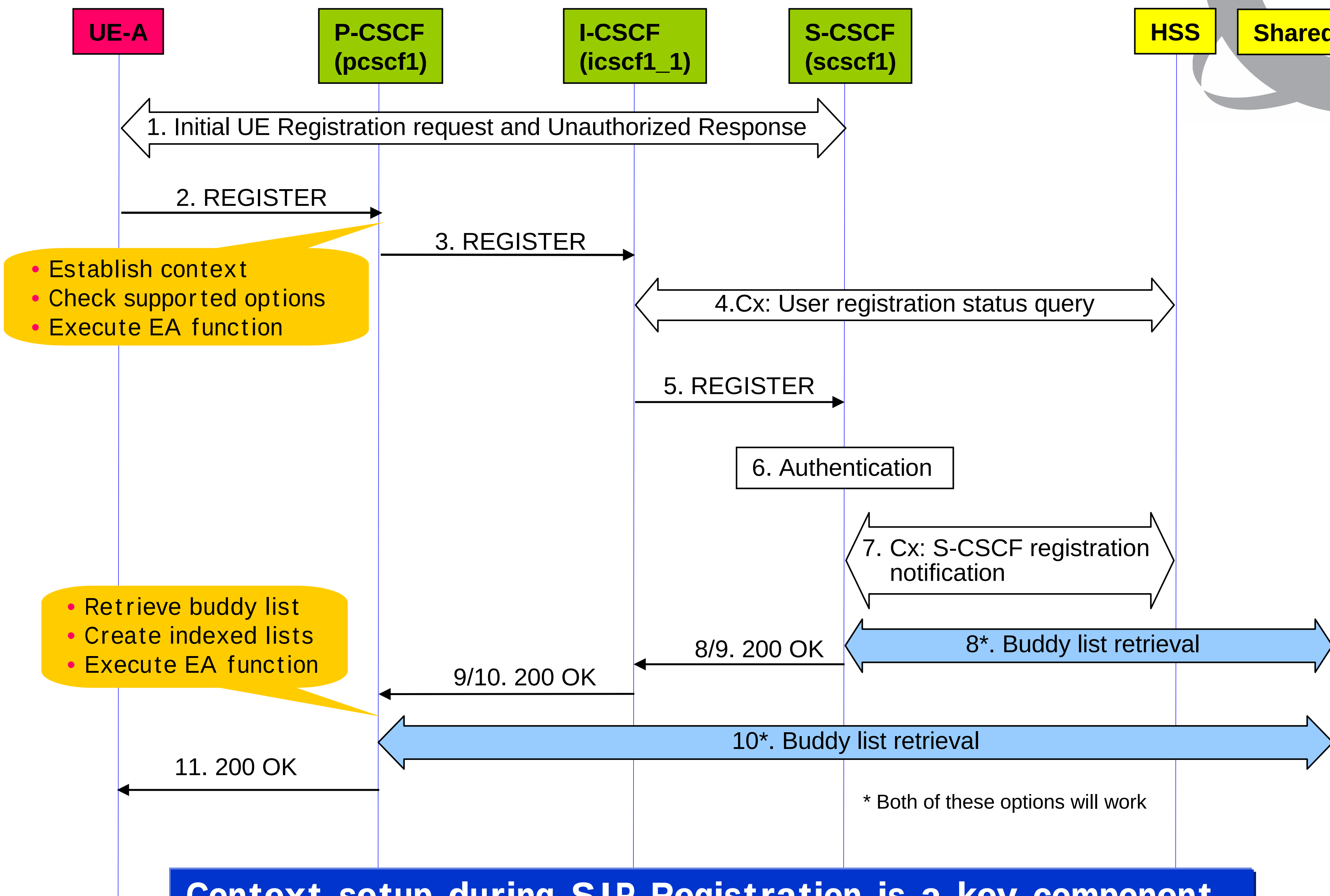
To: <tel:+1-212-555-2222>



Main Components of SIP Header Reduction Proposal

- Modification of IMS/MMD SIP Registration
  - » Establish context
    - ≈ Exchange indexed list of Identity components
      - IP addresses, URIs, buddy list etc.
      - To be used in SIP header fields: ‘Via’, ‘From’, ‘Contact’, ‘P-Preferred-Identity’ etc.
    - ≈ Exchange indexed list of access networks supported
      - To be used in ‘P-Access-Network-Info’ SIP Header field
    - ≈ Exchange indexed list of security protocols supported
      - To be used in ‘Security-Verify’ SIP Header field
  - » Identify supported functions
    - ≈ SIP Header Reduction algorithm
    - ≈ 3G dictionary
  - » Requires new or modified SIP Header Fields
- 3G Dictionary
  - » Introduce new mobility data elements not present in RFC 3485
  - » Avoid dynamically building the dictionary since initial SIP Invite needs to be reduced
- EA Function at the UE and P-CSCF
  - » Encode/decode SIP header fields
  - » Maintain SIP Header Reduction state information per SIP Registration session

Modification in IMS/MMD SIP Registration



Context setup during SIP Registration is a key component

New Option Tags and SIP Header Fields

- Option Tags for ‘Supported’ Header Field
  - » Option Tag ‘encode’
    - ≈ Indicates if SIP Header Reduction is supported
  - » Option Tag ‘3G-Dictionary’
    - ≈ Indicates the presence/absence of 3G Dictionary
- P-Encode-Identities
  - » Index of public IDs (IP addresses, URIs, E.164 etc.)
- P-Encode-Access-Networks
  - » Index of supported access networks such as CDMA, 802.11 etc.
- P-Encode-Security
  - » Index of security protocols supported such as IPSec, TLS etc.
- P-Contact-List
  - » Index of contact List
- P-Contact-List-Location
  - » Location of the database (such as shared XDM) for storing the contact list

Submitted as a draft to the IETF (March 2006)

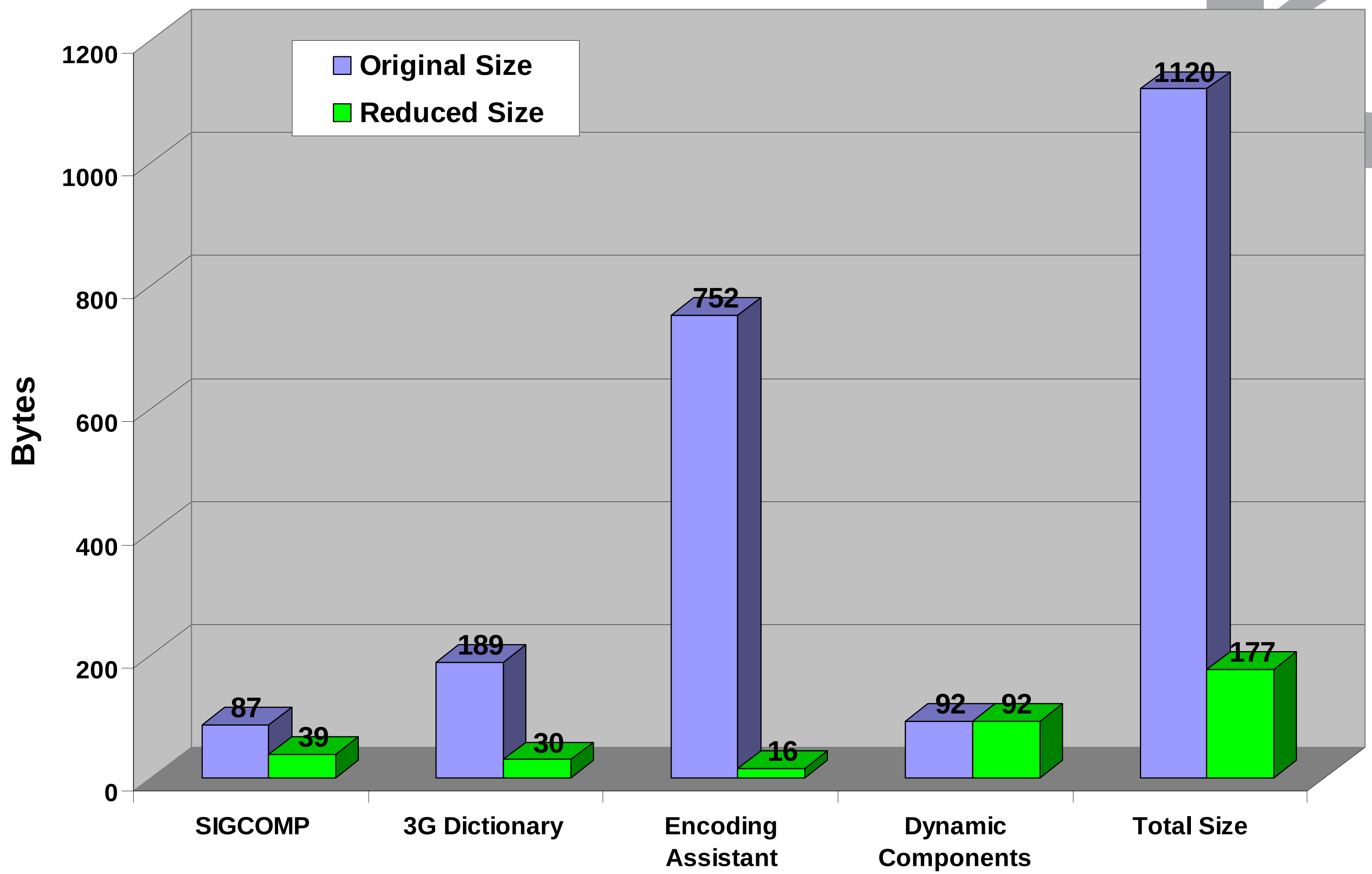
New Data Elements of 3G Dictionary

- SIP Header Field parameters
  - » 'Max-Forwards: 70'
  - » 'P-Preferred-Identity'
  - » 'P-Access-Network-Info'
  - » 'Require: sec-agree, precondition'
  - » 'Supported: 100 rel'
  - » 'Spi:s'
  - » 'Port:c='
  - » 'Port:s='
- SDP parameters
  - » 'Content-Type: application/SDP'
  - » 'a=des:qos mandatory, local sendrecv'
  - » 'a=des:qos none, local sendrecv'
  - » 'a=inactive'

• Not available in RFC 3485

• Submitted as a draft to the IETF (March 2006)

Savings from Reduced SIP Header Proposal



- Dictionary overhead is not included in the calculation
- Maximum length of URI (user@realm) is assumed to be 30 characters
- An 1120-long SIP Originating Invite message is used for this example

Summary

- Wireless operators have non-negotiable call setup requirements (95 percentile) for delay sensitive applications
  - » Push-to-beep < 750 ms
  - » PDD (Post Dial Delay) < 4 seconds
- Introduction of IMS will not allow to relax these requirements
- Success of both DOrA and UMTS access technologies depend on the QoE (Quality of Experience) of delay sensitive applications
- SIP has been decided as the protocol of choice for IMS
  - » The proposal attempts to reduce the standard SIP messages
- Current SIP message size (1000+ bytes) is too large to meet these call setup requirements for the delay sensitive applications
  - » Wireless access technologies have limited ACH/CC capacity
    - ≈ < 211 bytes in reverse direction for DOrA ACH
    - ≈ < 113 bytes in forward direction for DOrA CC
- The objective is to send the initial SIP messages (originating SIP Invite and terminating SIP 200 OK) over the control channels
- SIP Header Reduction proposal provides a solution to meet these non-negotiable call setup requirements
  - » Access independent IMS based solution
  - » Changes required in UE and P-CSCF components only
  - » Supports all delay sensitive SIP based applications such as PoC, VoIP, VT etc.

Backup chart

Backup

IMS/MMD Assumptions

- AT registers if P-CSCF is changed
  - User needs to re-register as soon as soon as a new P-CSCF is needed
- P-CSCF features
  - Presence of an User Agent
  - Adds any missing parameters in the SIP header so long the user is authenticated
  - Has enough memory to store both dictionaries and user specific profiles
    - Such as buddy list, public IDs etc.
  - Knows the physical address of the AT client (from source IP address)
- AT and the P-CSCF exchanges static information at SIP Registration
  - User's buddy list, IP address, URI, etc are to be stored by P-CSCF
- Buddy list allows up to $2^{10} = 1024$ entries
- During SIP Registration user identifies if the P-CSCF belongs to the home administrative domain
  - Use SIGCOMP only if the user is served by a P-CSCF that belongs to any foreign administrative domains
- IPSec/TLS between AT and P-CSCF
- Token for the dictionaries (SIGCOMP vs. Nortel) must be included OTA
- Maximum length of URI (users@realm) is assumed to be 30 characters
  - In case the called party is not in the buddy list
- Maximum length of the Call-ID is assumed to 30 characters

URLs for the IETF submissions

- draft-akhtar-sipping-header-reduction-00.txt
  - » Title: New SIP Headers for Reducing SIP Message Size
  - » URL: http://www.ietf.org/internet-drafts/draft-akhtar-sipping-header-reduction-00.txt.
- draft-akhtar-sipping-3g-static-dictionary-00.txt
  - » Title: 3G Wireless Support in the SIP/SDP Static Dictionary for Signaling Compression (SigComp)
  - » URL: http://www.ietf.org/internet-drafts/draft-akhtar-sipping-3g-static-dictionary-00.txt.

Glossary

AAA	Authentication, Authorization and Accounting
ACH	Access CHannel
AS	Application Server
BAM	Basic Access Mode
BSC	Base Station Controller
BTS	Base Transceiver Station
CC	Control Channel
CSCF	Call Session Control Function
DoS	Data over Signaling
DOrA	1xEV-DO rev A
GLMS	Group List Management Server
HLR	Home Location Register
HSS	Home Subscriber Server
I-CSCF	Interrogating CSCF
IMS	IP Multimedia Subsystem
ISC	IMS Service Control
MGCF	Media Gateway Control Function
MMD	Multimedia Domain
MRFC	Media Resource Function Controller

Glossary continued

MRFP	Media Resource Function Processor
MSC	Mobile Switching Center
OMA	Open Mobile Alliance
P-CSCF	Proxy CSCF
PCF	Packet Control Function
PDSN	Packet Data Serving Node
PoC	PTT over Cellular
PSTN	Public Switched Telephone Network
PTT	Push To Talk
RNC	Radio Network Controller
S-CSCF	Serving CSCF
SDB	Short Data Burst
SIP	Session Initiated Protocol
SS-7	Signaling System 7
VLR	Visitor Location Register
VoIP	Voice over IP