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Proxy-to-Proxy Extensions

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AT&T, CableLabs, 3Com, Cisco, Com21, General Instrument, Lucent Cable, NetSpeak, Telcordia

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IETF Presentation

User Field

- LNP look ups
  - LNP looks ups should be done once per call and not by every proxy hop
    - » Signaling should indicate when a LNP look up has already been done
- Packet telephone network must support special services such as conference bridges and 3-way calling
  - 3-way calling requires signaling of flash hook or some other method to convey flash hook
    - » Signaling must support a variety of special services including conference bridging, 3-way calling, call-return, call-trace

User Field

- Network private information stored at the end-point
  - Scalability and reliability of packet telephone system can be enhanced by maintaining call state in the end-point. However this information is considered network private and stored at end-point in an encrypted format.
    - » Signaling must be able to retrieve private network information such as calling party number from end-points.
    - » Support of privacy and state at end-point is covered by other IETF drafts

Gate Header

- QoS must be done on a call by call basis
  - Telephone users should not be alerted if QoS can not be provided
- QoS requires authorization and authentication
  - End-points are not trusted and theft of service attempts are likely to occur

Billing Information Header

- Revenue is an incentive for QoS assurances
  - Telephone service providers incentive for providing QoS assurances is revenue
    - » Signaling protocol should be versatile to support billing for those who desire it
    - » Billing information needs to be communicated to various trusted entities
    - » Billing information communication should be on secure links
- Packet telephone signaling should support different billing models
    - » Flexible billing information signaling is key to supporting different billing model

Header Changes Requested

- Modification to Existing Header Syntax
  - » Request-URI user field
- New Headers
  - » Gate
  - » Billing-Info
  - » Billing-Id

User Field

- Basic Syntax

User = telephone-subscriber *(“,” dcs-user-parameter)
| dcs-user-parameter *(“,” dcs-user-parameter)
| *(unreserved | escaped | “&” | “=“ | “+” | “$” | “,”)

telephone-subscriber = global-phone-number | local-phone-number
| special-service-name
dcs-user-param = lnp-param | private-param | other-param
lnp-param = “lnp=” token
private-param = “private=“ token
Special-service-name = “return-call” | “call-trace” | “bridge”

- Example

INVITE sip: +1-212-555-2222,lnp=212-234@Host(DP-t); user=phone SIP/2.0

Gate

◆ Basic Syntax

```
Gate               = "Gate" ":" [hostport "/"] Gate-Id
                        [";" Gate-Key ";" Gate-CipherSuite]
Gate-Id            = 1*alphanum
Gate-Key           = 1*alphanum
Gate-CipherSuite   = token
```

◆ Example

Gate: cmts-o.provider/12S345;ABCDEF;Clear

cmts-o.provider	= Hostport of originating CMTS
12S345	= Gate Id between MTA-o and CMTS-o
ABCDEF	= Key for gate co-ordination messages cipher
Clear	= Indicates gate co-ordination messages are not ciphered

Billing-Info

- Basic Syntax

Billing-Info = “Billing-Info” “:” hostport “<“Acct-Data “>”

Acct-Data = 1*unreserved | (1*unreserved “,” Acct-Data)

- Example

Billing-Info: Record-Keeper-o. Provider <332-44-5199/847-262-2393/847-262-2154/ABCDEF>

Record-Keeper.Provider-o	= Hostport of originating record keeping server
332-44-5199	= Account number
847-262-2393	= Originating phone number
847-262-2154	= Terminating phone number
ABCDEF	= An encryption key for messages to the record keeping server

Billing-Id

- Basic Syntax

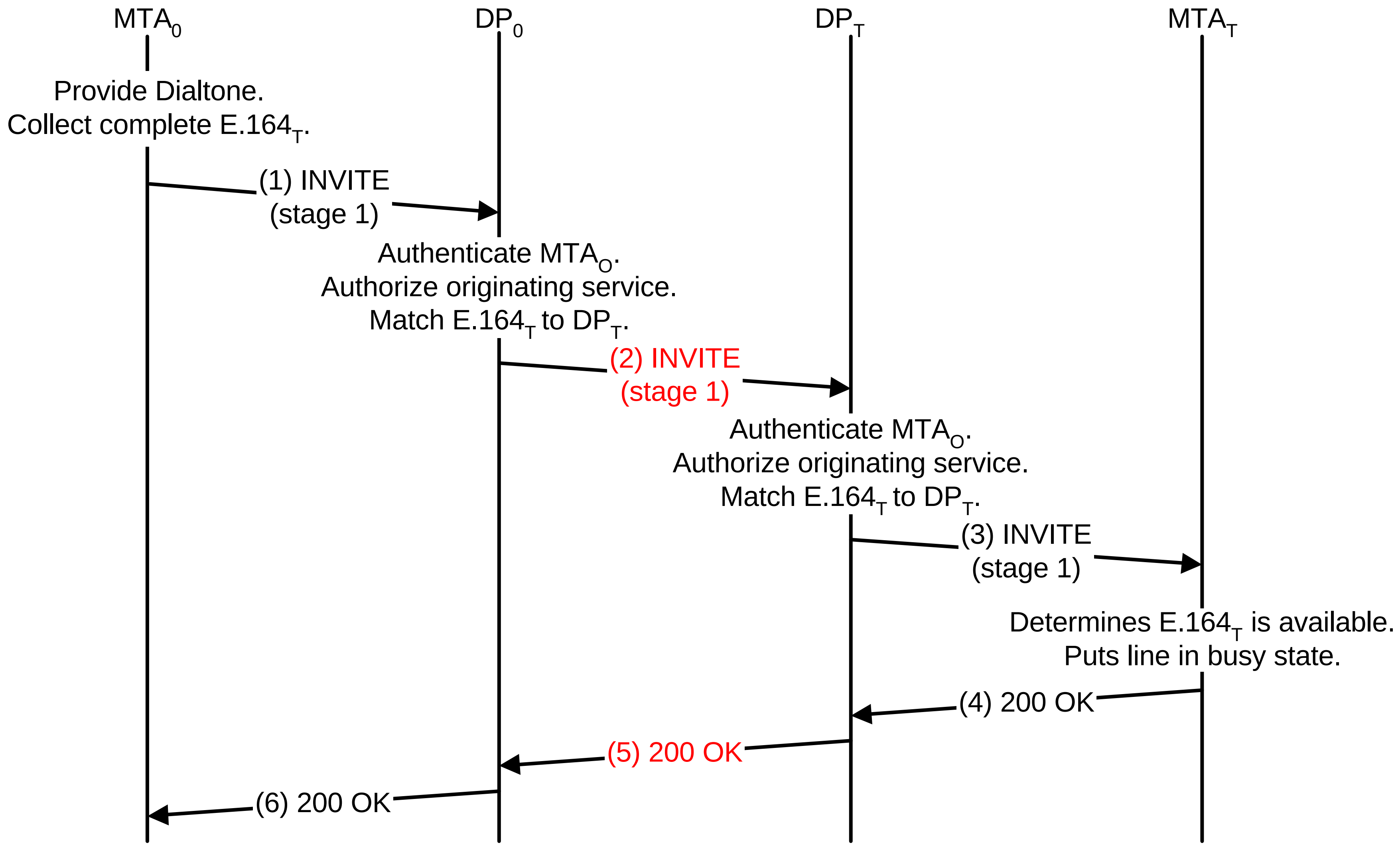
Billing-Id = "Billing-Id" ":" 1*unreserved

- Example

Billing-Id: ABC123

ABC123= function of (time stamp, IP address, event count)

Basic Call



MTA0
DP0
DPT
MTAT
Provide Dialtone.
Collect complete E.164T.
(1) INVITE
(stage 1)
Authenticate MTAO.
Authorize originating service.
Match E.164T to DPT.
(2) INVITE
(stage 1)
Authenticate MTAO.
Authorize originating service.
Match E.164T to DPT.
(3) INVITE
(stage 1)
Determines E.164T is available.
Puts line in busy state.
(4) 200 OK
(5) 200 OK
(6) 200 OK

Proxy-to-Proxy
Invite(stage 1) for Basic Call

(2) INVITE (stage1):

INVITE sip: +1-212-555-2222,lnp=212-234@Host(DP-t); user=phone SIP/2.0
Via: SIP/2.0/UDP Host(DP-o.provider);branch=1
Via: SIP/2.0/UDP Host(mta-o.provider)

Caller: John Doe; +1-212-555-1111

Anonymity: Off

Gate: Host(cmts-o.provider):3612/17S30124/37FA1948

Billing-Info: Host(rks-o.provider)<5123-0123-4567-8900/212-555-1111/212-555-2222>

Billing-ID: Host(dp-o.provider):36123E5C:0152

Require: DCS

Proxy-Require: DCS
From: “Alien Blaster” <sip:BASE64(SHA-1(555-1111; time=36123E5B;seq=72))>
To: sip:BASE64(SHA-1(555-2222; time=36123E5B; seq=73))
Call-ID: BASE64(SHA-1(555-1111;time=36123E5B;seq=72))
CSeq: 127 INVITE

Proxy-To-Proxy
200 OK Response to Invite (stage 1) for Basic Call

(5) 200-OK:

SIP/2.0 200 OK

Via: SIP/2.0/UDP Host(dp-o.provider);branch=1

Via: SIP/2.0/UDP Host(mta-o.provider)

Record-Route: Host(dp-t.provider), Host(mta-t.provider)

Gate: Host(cmts-t.provider):4321/31S14621/37FA1948

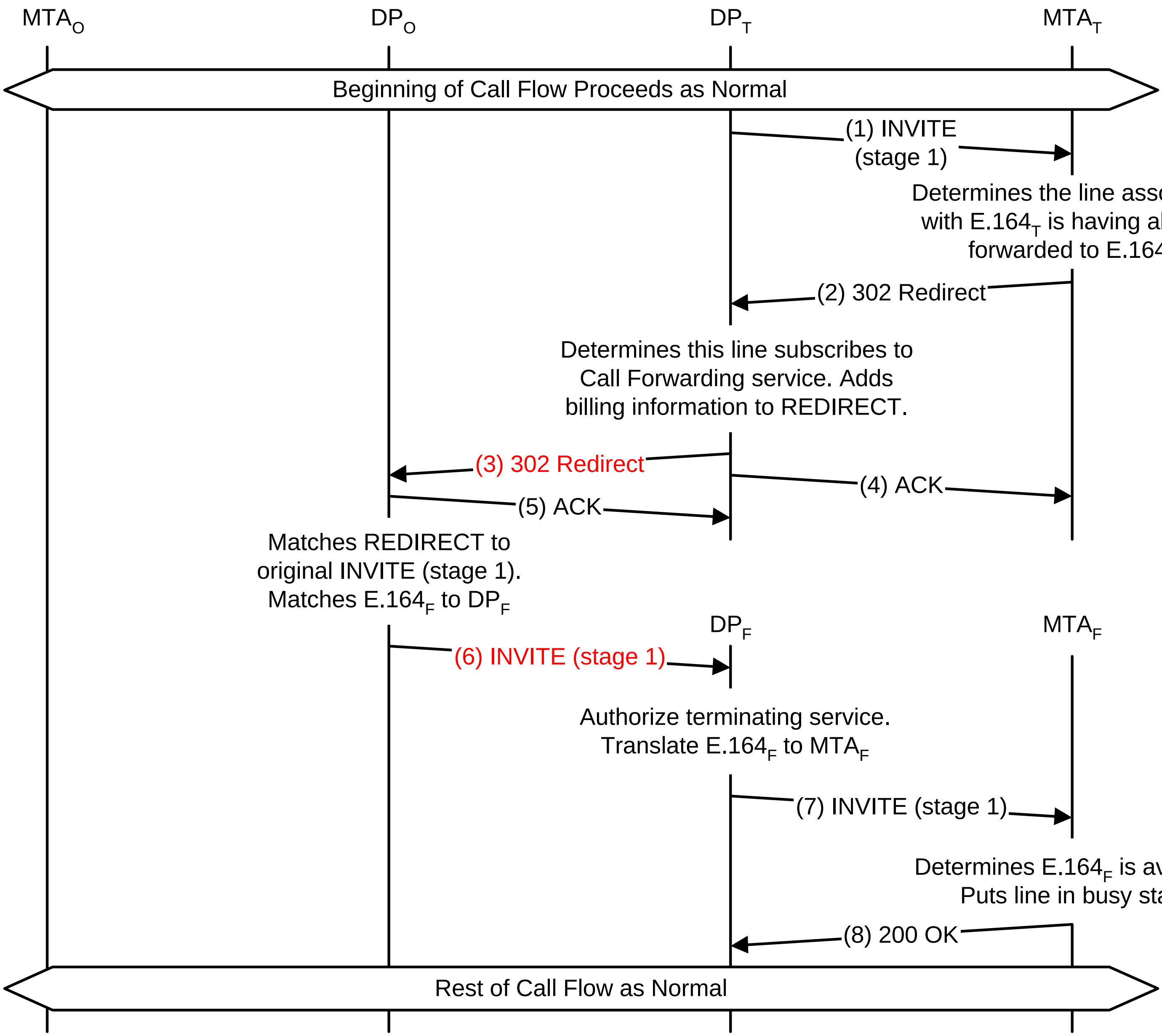
From: “Alien Blaster” <sip:BASE64(SHA-1(555-1111; time=36123E5B;seq=72)) >

To: sip:BASE64(SHA-1(555-2222; time=36123E5B; seq=73))

Call-ID: BASE64(SHA-1(555-1111;time=36123E5B;seq=72))

CSeq: 127 INVITE

Call Forwarding Unconditional



Call Forwarding Unconditional
302 Redirect

(3) 302-Redirect

SIP/2.0 302 Moved Temporarily

Via: SIP/2.0/UDP Host(dp-o.provider); branch = 1

Via: SIP/2.0/UDP Host(mta-o.provider)

Billing-Info: Host(rks-o.provider)<5123-0123-4567-8900/212-555-1111/212-555-2222>

Billing-Info: Host(rks-t.provider)<4278-9865-8765-9000/212-555-2222/212-555-3333>

Billing-ID: Host(dp-o.provider):36123E5C:0152

From: “Alien Blaster” <sip:BASE64(SHA-1(555-1111; time=36123E5B;seq=72)) >

To: sip:BASE64(SHA-1(555-2222; time=36123E5B; seq=73))

Call-ID: BASE64(SHA-1(555-1111;time=36123E5B;seq=72))

Cseq: 127 INVITE

Contact: tel:+1-212-555-3333

Call Forwarding Unconditional
Invite (stage 1)

(6) INVITE (stage1):

INVITE sip:+1-212-555-3333,lnp=212-265@Host(dp-f.provider) SIP/2.0

Via: SIP/2.0/UDP Host(dp-o.provider); branch = 2

Via: SIP/2.0/UDP Host(mta-o.provider);

Caller: John Doe; 212-555-1111

Anonymity: Off

Gate: Host(cmts-o.provider):3612/17S30124/37FA1948

Billing-Info: Host(rks-o.provider)<5123-0123-4567-8900/212-555-1111/212-555-2222>

Billing-Info: Host(rks-t.provider)<4278-9865-8765-9000/212-555-2222/212-555-3333>

Billing-ID: Host(dp-o.provider):36123E5C:0152

Require: DCS

Proxy-Require: DCS

From: “Alien Blaster” <sip:BASE64(SHA-1(555-1111; time=36123E5B;seq=72)) >

To: sip:BASE64(SHA-1(555-2222; time=36123E5B; seq=73))

Call-ID: BASE64(SHA-1(555-1111;time=36123E5B;seq=72))

CSeq: 127 INVITE