Interoperation with the Circuit-Switched Telephone System

- interoperation scenarios
- mapping E.164 phone numbers to IP addresses
- finding hop-off gateways: TRIP

Interoperation assumptions

- each IP phone needs to be reachable from the PSTN, without "dialing" an IP address $\longrightarrow$ possible IETF working group, ETSI Tiphon
  - LDAP
  - DNS
- multiple gateways from IP to PSTN
  - (almost) any gateway can call any phone $\longrightarrow$ locate "best"/"closest"/"cheapest"
  - attributes
  - distributing and merging attributes $\longrightarrow$ TRIP

DNS for number mapping

- SCP may treat DNS as yet another database
- use DNS NAPTR rewriting for local number portability (LNP)
- find out which signaling services are supported

Issues for any scheme:

- rapid, secure, globally visible updates (e.g., for LNP)
- ownership of addresses or neutral third party?

e164.arpa

1. convert to digits: +46-8-56264000 $\longrightarrow$ 46856264000
2. insert dots: 4.6.8.5.6.4.0.0.0
3. reverse: 0.0.0.4.6.5.8.6.4
4. append TLD: 0.0.0.4.6.5.8.6.4.e164.arpa

Phone number rewriting

- NAPTR (RFC 2168, being updated): originally for URN rewriting
- flag: “a” for A RR, “s” for SRV RR

```
2.8.0.4.6.2.6.5.8.6.4.e164.int.
IN NAPTR  10 10 "a" "sip+N2R"       "" "sip:paf@swip.net".
IN NAPTR 102 10 "s" "potscall+N2R" "" _potscall._tcp.paf.swip
IN NAPTR 102 10 "a" "smtp+N2R"      "" "mailto:paf@swip.net".
```

Local number portability

- old number: +46-346-23232, but has moved to a 'telcoy'.

```
2.3.2.3.2.6.4.3.6.4.e164.int. IN NS ns.telcoy.net.
```

- Telco Y allocates +46-8-919191:

```
$ORIGIN 6.4.e164.int.
 _potscall._tcp.2.3.2.3.2.6.4.3 IN SRV 10 10 1.9.1.9.1.9.8
```

Translation from E.164 to SIP

+46-76-11223344 gets translated via Global Title Translation by the SCP finding DNS record:

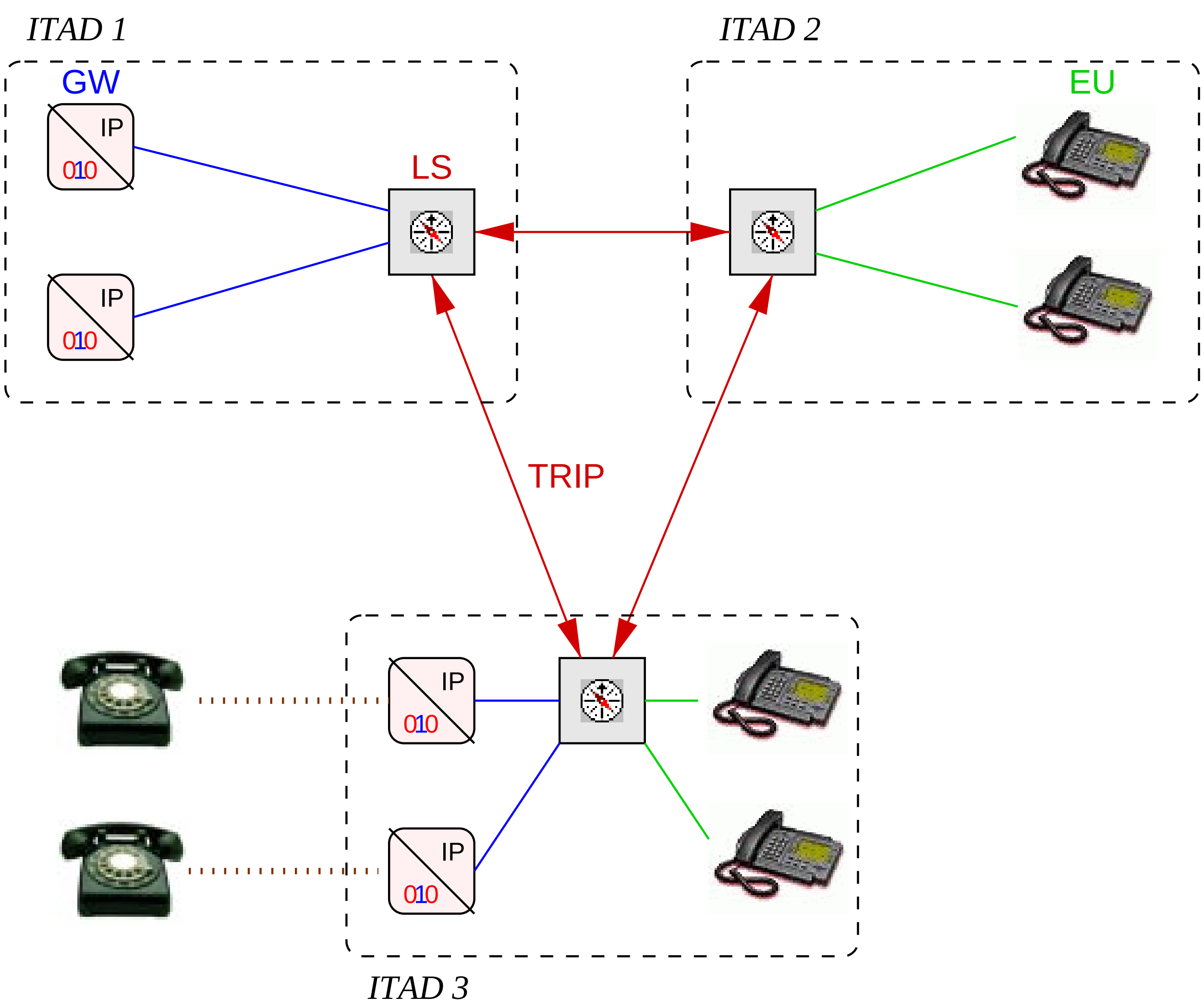
```
4.4.3.3.2.2.1.1.6.7.6.4.e164.int. IN SOA ....
    IN NS ....
    IN NAPTR 100 10 "a" "sip+N2R" "" "sip:foobar@x.example.ne
```

Then, call can only be made by SIP. The SCP routes the call to the correct gateway, using a mechanism currently not specified.

PSTN-to-VoIP gateway location

- currently, not specified
- use hop-off closest to IP destination or where doing number translation?
- maybe get IP address via SIP OPTIONS and then determine “best” hop-off point?

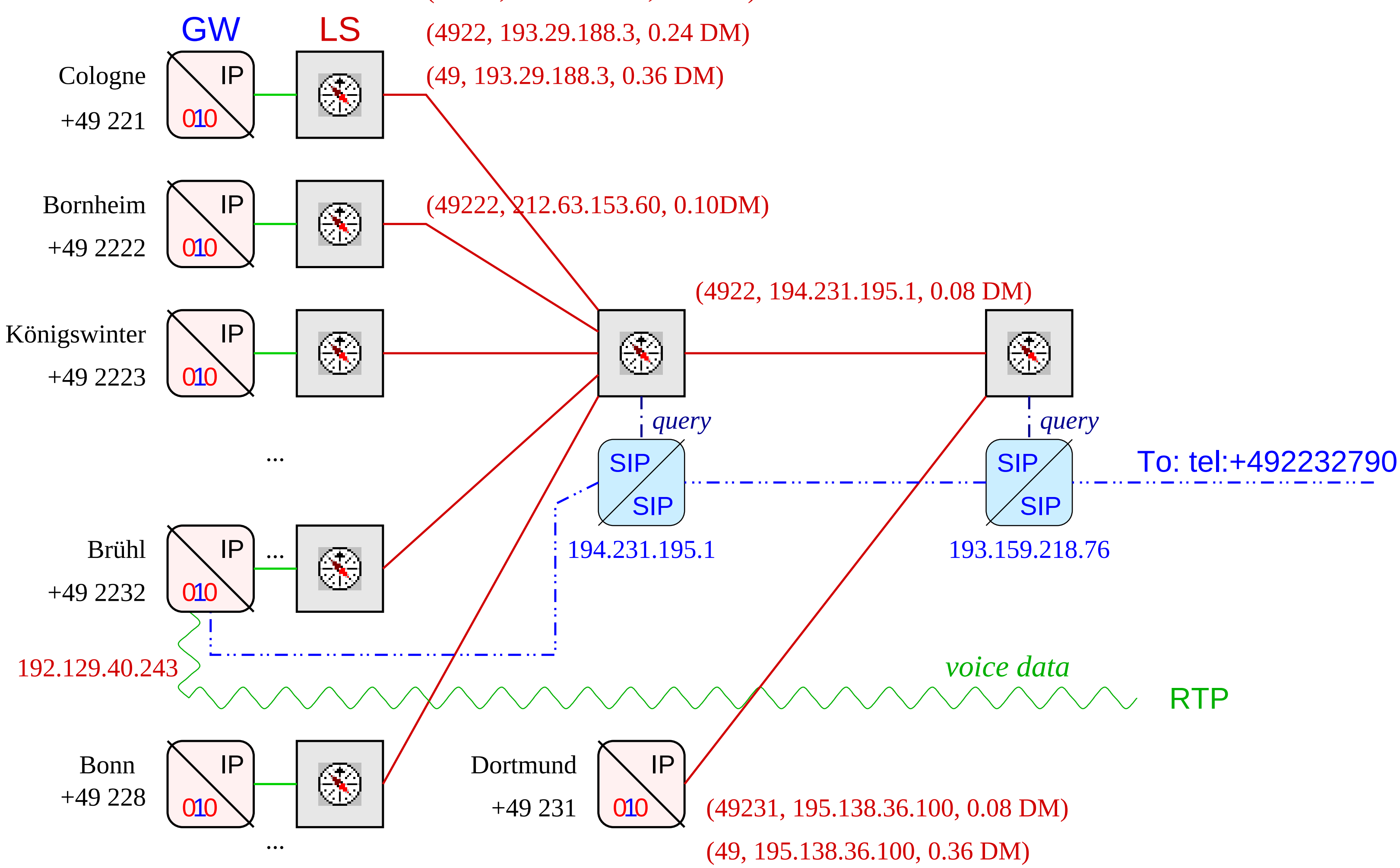
Gateway location protocol architecture



Gateway location protocol

- inter-domain distribution of gateway properties
- multiple signaling hops - aggregation $\rightarrow$ additional server
- aggregation of entries (e.g., Cologne/Bonn area +49221, +49228, +492222, $\ldots \longrightarrow$ +4922)
- but difficult in NANP: NJ has 201, 609, 732, 908, 973.
- local calling areas may have a hundred exchange codes!
- generally, use most specific match
- policy filtering to restrict advertising prefixes

Gateway aggregation



Attributes

DestinationPhoneNumbers	phone number prefix (+1201)
NextHopSignalingServer	direct H.323, SIP, … here
AdvertisementPath	≈ BGP AS_PATH, object path
GatewayCapacity	load balancing?
SignalingProtocols	H.323, SIP, …
Pricing	$/minute, setup charges, increment, …
LastModifiedBy	verifiable source of attribute
NumSignalingHops	min/max *signaling* (not media) hops
AtomicAggregate	aggregation indicator
MultiExitDisc	preference between multiple LS–LS links

Attribute transitivity

Optional: if not set, must be understood to process routing object;

IndependentTransitive: propagate even if not understood;

DependentTransitive: propagate if not understood, but don't touch next-hop;

LastModifiedByFlag: covered by last-modified-by parameter.

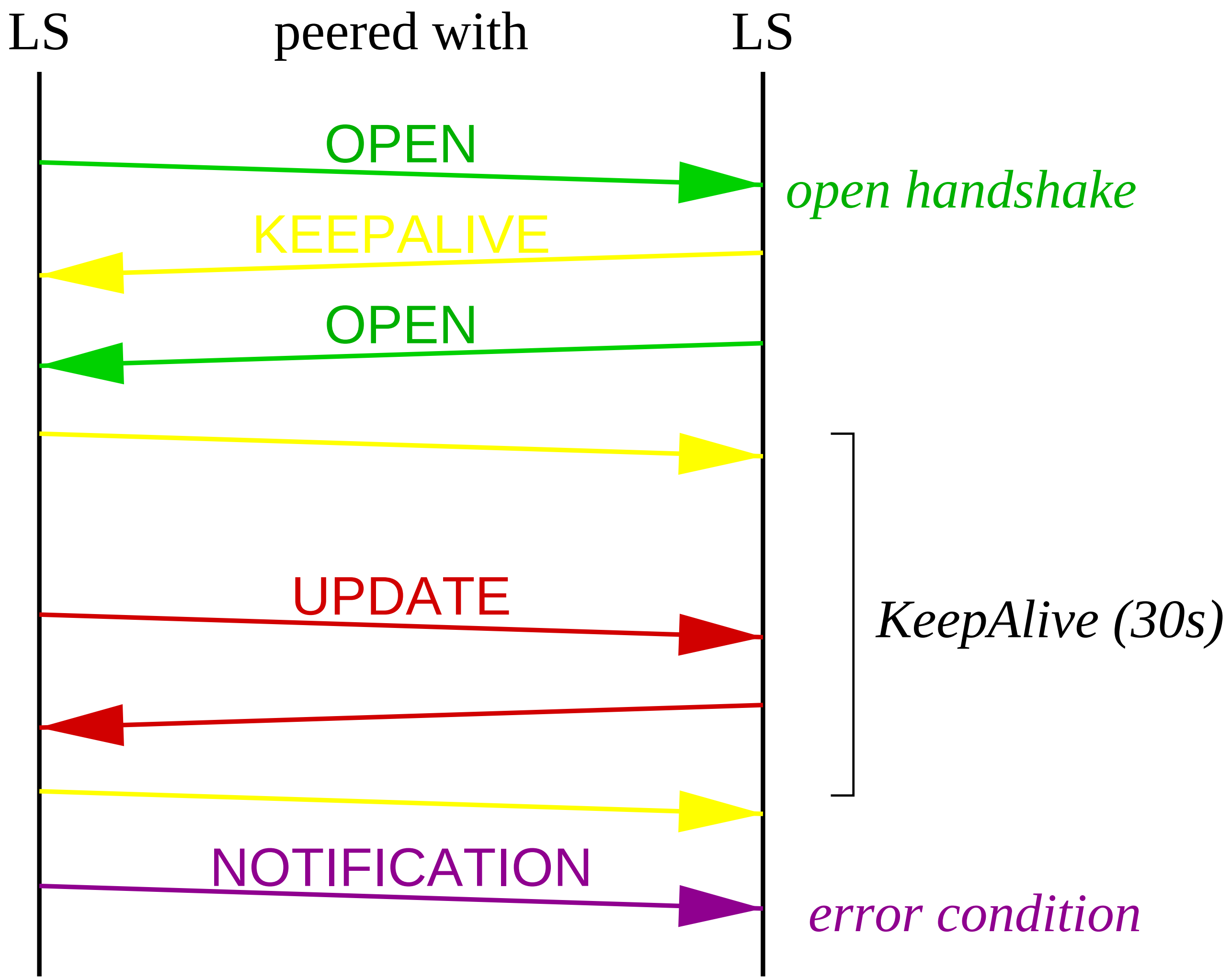
route objects always contain (DestinationPhoneNumbers, NextHopSignalingServer), where server is in ITAD of originating LS

Telephony Routing Information Protocol (TRIP)

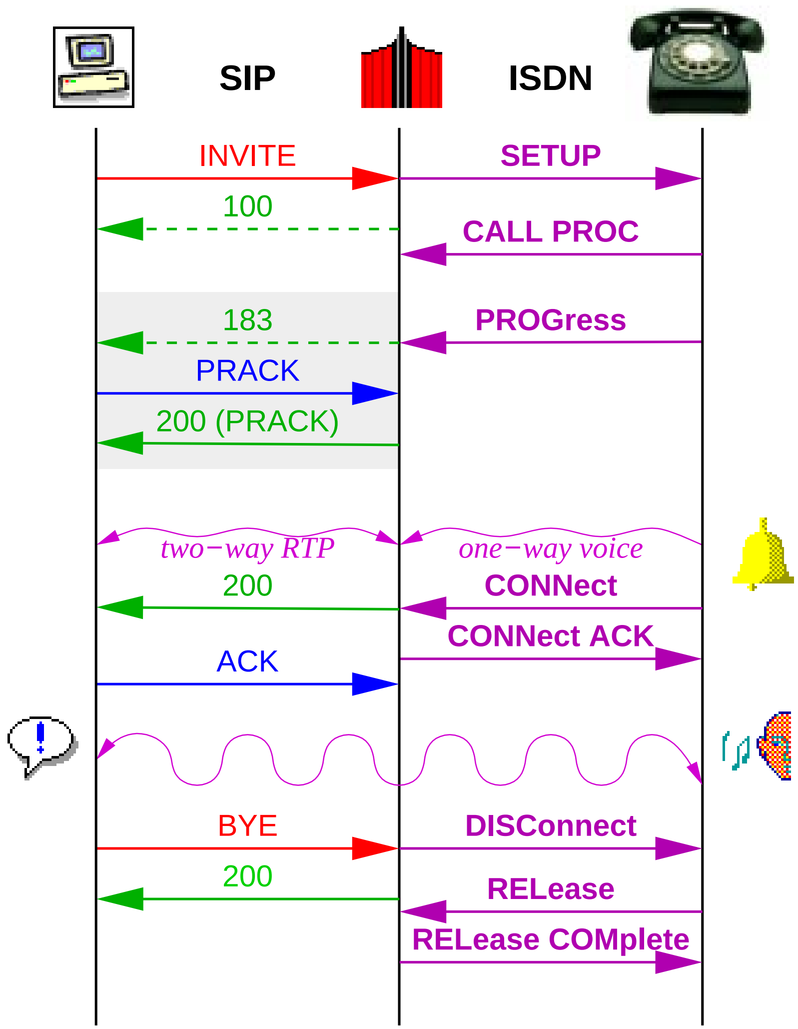
- from BGP-4: transport (TCP), message format (TLV)
- from OSPF and SCSP: flooding
- initially, exchange routing database when establishing connection
- incremental updates (no refreshes)

```
UPDATE
 advertised route
 signaling server
 price
 withdrawn route
 withdrawn route
```

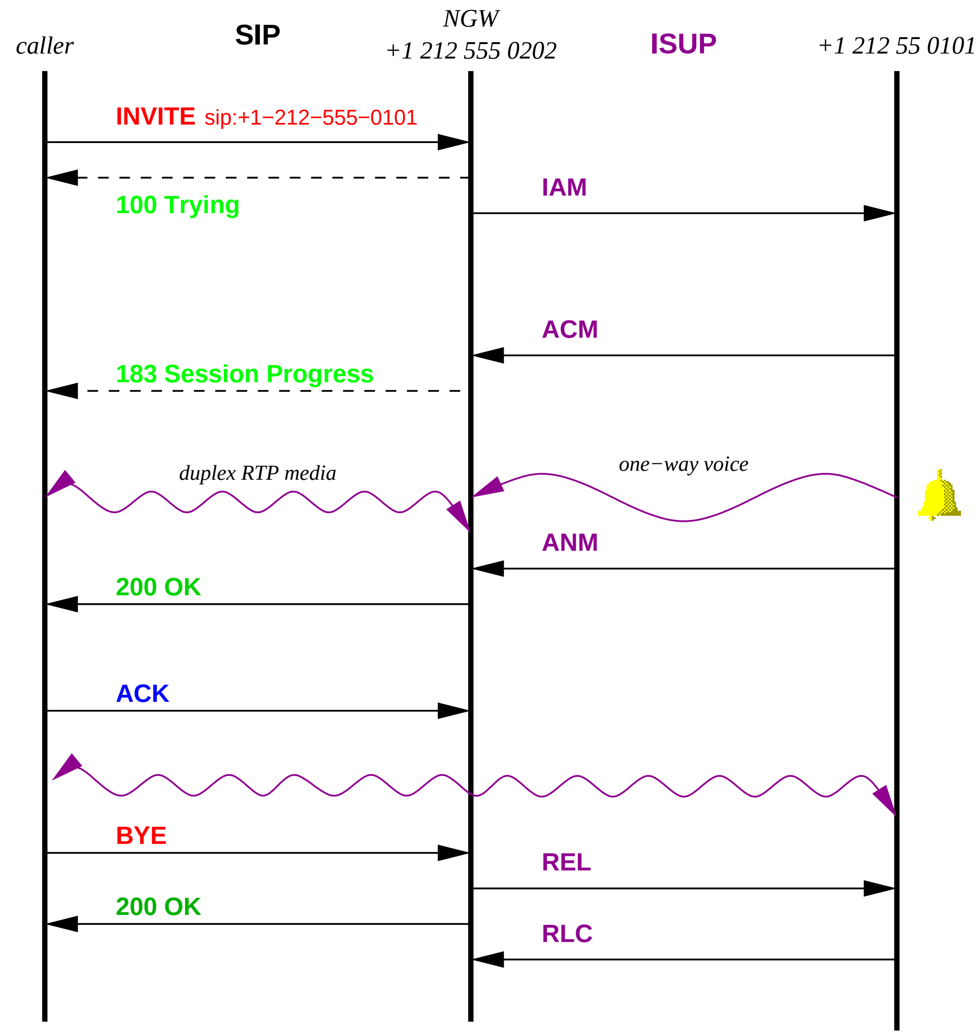
TRIP


LS
peered with
LS
OPEN
open handshake
KEEPALIVE
OPEN
UPDATE
KeepAlive (30s)
NOTIFICATION
error condition

Signaling: SIP to ISDN



Signaling: SIP to ISUP



References

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