IETF Real-time Applications Emerging Standards

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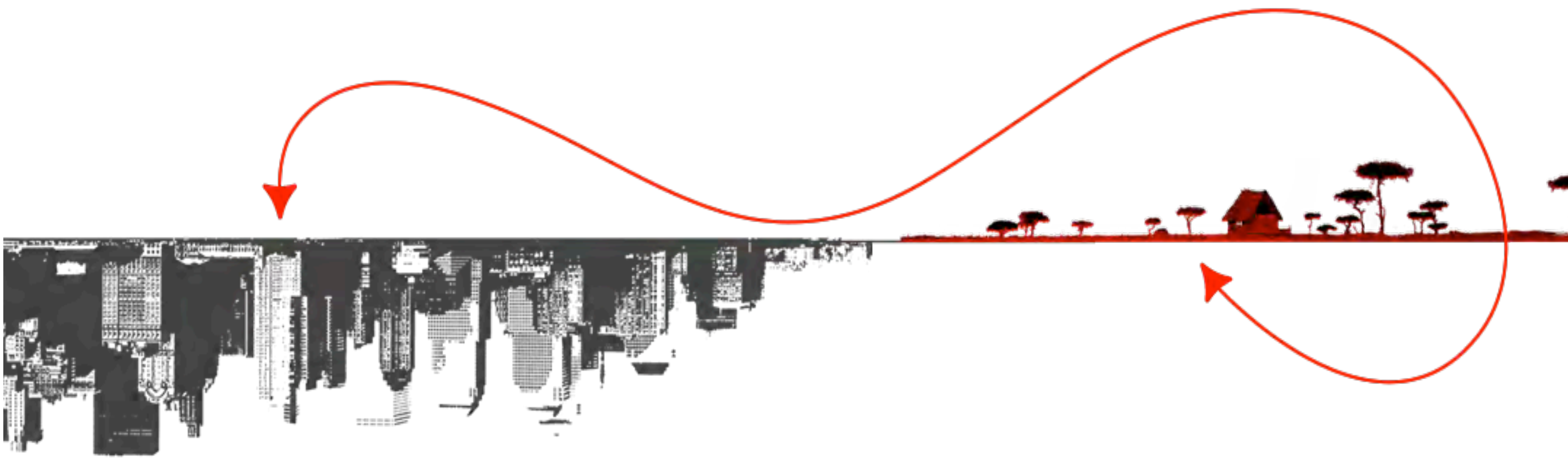
Overview

1. RAI WGs completing protocol deliverables
2. RAI WGs in progress
3. RAI model for new work and restructuring
4. New Protocol work underway



WGs near Completion/Closure

The following WGs have completed chartered deliverables (which have either already been published as RFCs or are undergoing AD review/in publication):

- ENUM: E.164 numbering to SIP/etc. URIs in DNS
- SPEERMINT: BCPs for SIP peering between operators
- MEDIACTRL: Defines the use of SIP as a control protocol to exchange media control packages (e.g., for IVR, conf control) between an Application Server and a Media Server
- XCON: centralized conferencing control (CCMP, BFCP)
- SPEECHSC: client control of IVR-type speech resources (MRCPv2)

WGs Completing Deliverables in 2011

- BLISS: Basic level of interoperability for specific features in SIP (e.g., Call Completion Busy Service)
- DRINKS: provisioning+exchange of the ENUM data
- SIMPLE: presence (PUBLISH/SUBSCRIBE) and instant messaging (MSRP)
- XMPP: a different protocol than SIP, following a client-server XML-based protocol for IM/Presence/VoIP sessions (a.k.a., Jabber/Jingle)

Ongoing WGs

- ATOCA: Authority-to-citizen alert (e.g., Tornado warnings, etc.)
- ECRIT: Protocols for emergency services including LoST protocol location stuff
- GEOPRIV: defines mechanisms for determining and delivering location information for applications (e.g., Emergency Services)
- P2PSIP: using DHT-based directories to find peers and relay packets for SIP sessions (RELAY)

Overview - Core RAI Area WGs

- AVT WG closed/reorganized in 2011.
  - AVTCORE: AVT core protocol maintenance.
  - AVTEXT: Extensions to AVT core protocols
  - PAYLOAD: Audio/video Transport New Payload types
  - XRBLOCK: Metric Blocks for use with RTCP's Extended Report Framework
- MMUSIC WG continues to handle ongoing updates/extensions to to SDP, RTSP and ICE.
- SIP and SIPPING WGs closed in 2009 due to issues such as:
  - High volume of new work into SIPPING WG
  - Difficulty in completing existing chartered work
  - Inability to prioritize disparate work items
- Two new WGs established to handle changes to core SIP Protocol (SIPCORE) and to screen new work (DISPATCH).

SIPCORE WG

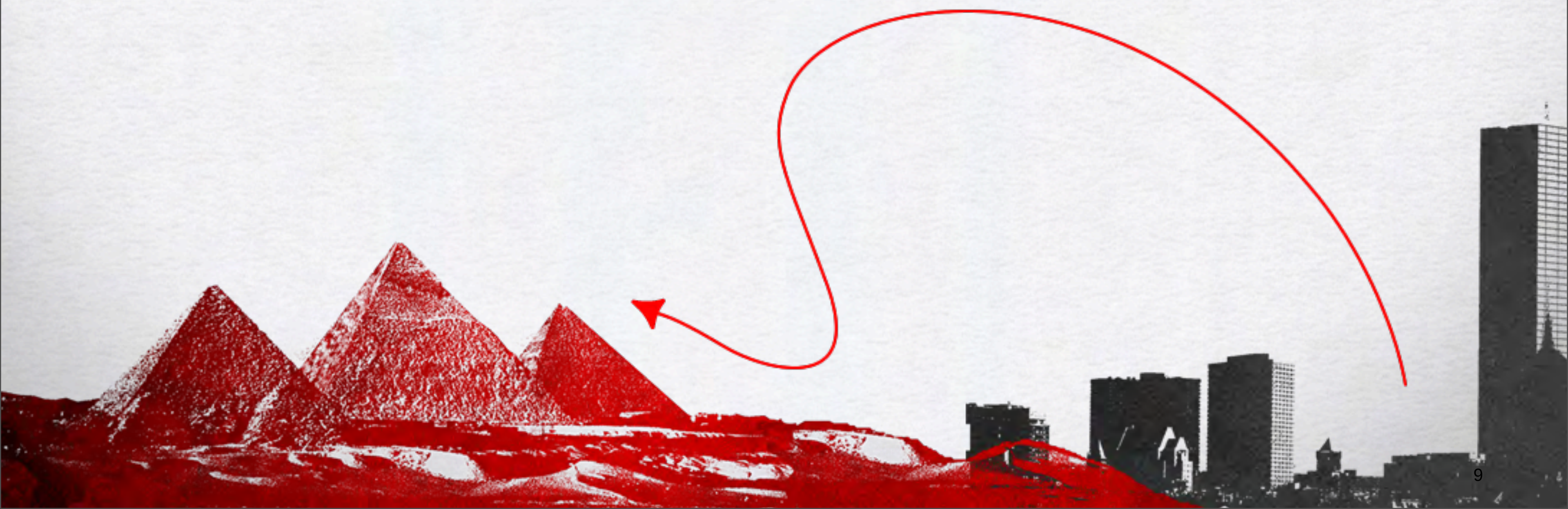
- Protocol enhancements, updates and extensions to core SIP protocols - i.e., RFC 3261- RFC 3265, such as:
  - 199 response code indicating terminated session
  - Throttling of SIP Event notifications.
  - Location header - e.g., for emergency services
  - Proxy capabilities - indicating features supported by a proxy *(individual item)*
  - RFC4244bis (updates to History-Info header field) - driven by service provider feature/application requirements

DISPATCH WG

- Screens new work items for Real-time Applications Infrastructure area – either charter new WG, assign item to existing WG, recommend AD sponsor or kill the work item.
- Objective: Scope of new work items is narrow, with well defined deliverables and clear milestones.
- Work items instigated for a variety of reasons, such as:
  - Robustness/enhanced infrastructure (e.g., overload control),
  - Standardization of operational interfaces (e.g., logs)
  - Interoperability
  - Feature/application requirements
- Work items/activity is tracked on WG wiki:
  - http://trac.tools.ietf.org/wg/dispatch/trac/wiki
- No protocol development - only milestones are for the dispatching of incoming work.

COMPLETED DEVELOPMENT

- MARTINI (Multiple AoR reachabiliTy InformatioN Indication)



Multiple AoR reachabiliTy InformatioN Indication (MARTINI WG)

- Motivation: In current deployments, dynamic reachability mechanisms based on the SIP REGISTER method are commonly used. Effectively, a single REGISTER request registers the AoR of the SIP-PBX, so that any request targeted at a SIP URI for which the SIP-PBX is authoritative can be delivered from the SSP to the SIP-PBX. Issue is that implementations vary in details, leading to interoperability issues.
- Objectives: standardize a multiple-AoR registration mechanism - initial focus is E.164 AoRs
- Deliverables:
  - Requirements: RFC 5947 (based on SIP Connect 1.1 needs)
  - Registration for multiple E.164 AoRs (FQDNs): RFC 6140
- Timeframe: Completed (in just over one year)

DEVELOPMENT UNDERWAY

- SOC (SIP Overload Control) WG
- SIPCLF (SIP Common Log Format) WG
- SIPREC (SIP Recording) WG
- CUSS (Call Control UUI Service for SIP) WG
- CODEC WG
- CLUE WG (ControlLing mUltiple streams for tElepresence)
- VIPR WG (Verification Involving PSTN Routability)
- RTCWEB (Real-Time Communications over the WEB)*Proposed* WG

SIP Overload Control (SOC WG)

- Motivation: SIP protocol provides a minimal mechanism for overload control (503, Server unavailable response code), whereas, overload is a serious problem in SIP servers.
- Objectives: Develop mechanisms for overload control for both SIP users and SIP servers:
  1. Mechanism to prevent overload in SIP servers by adjusting the incoming load using implicit and/or explicit feedback to identify overload condition
  2. Mechanism to prevent overload in SIP servers by distributing load control filters to SIP servers that throttle calls based on their source or destination domain, telephone number prefix for a specific user.
- Deliverables:
  - Design Considerations: draft-ietf-soc-overload-design
  - SIP Overload Control mechanism: draft-ietf-soc-overload-control
  - SIP Overload control Event package: draft-ietf-soc-load-control-event-package
- Timeframe: expect to complete deliverables August 2011

SIP Common Log Format (SIPCLF WG)

- Motivation: No common log format defined for SIP - diverse elements produce distinct log formats which makes it difficult to develop tools to analyze them.
- Objectives: produce a format suitable for logging from any SIP element
- Deliverables:
  - Problem statement, motivation and use cases: draft-ietf-sipclf-problem-statement
  - Common log format (indexed text encoded): draft-ietf-sipclf-format
- Timeframe: currently past initial proposed completion date - estimate that work will complete mid 2011

SIP Recording

- Objective: Develop a SIP Based protocol for controlling a session media recorder.
- Scope:
  - Recorder Control
  - Session metadata content and format
  - Security mechanisms, including transport and media encryption
  - Privacy concerns, including end-user notification
  - Negotiation of recording media streams
- Requirements & architecture are near completion:
  - draft-ietf-siprec-req
  - draft-ietf-siprec-architecture
- SIP extensions for recording and meta data under discussion:
  - Session Meta-data model: draft-ietf-siprec-metadata
  - Session Meta-data format/content (proposal): draft-ram-siprec-metadata-format
  - Protocol proposal: draft-portman-siprec-protocol
- Timeframe: Deliverables proposed to be completed in Sept. 2011

Call Control UUI Service for SIP (cuss)

- Motivation: ISDN User to User Information Service, defined by ITU-T Q.931, is extensively deployed in the PSTN today supporting such applications as contact centers, call centers, and automatic call distributors (ACDs). A major barrier to the movement of these applications to SIP is the lack of a standard mechanism to transport this information in SIP.
- Scope of the mechanism to be defined:
  1. The information is generated and consumed by an application during session setup using SIP, but the application is not necessarily even SIP aware.
  2. The behavior of SIP entities that support it is not significantly changed (as discussed in Section 4 of RFC 5727).
  3. User Agent Clients (UAC) and User Agent Servers (UAS) are the generator and consumer of the UUI data. Proxies may route based on the application tag.
  4. Intermediary elements or proxies can remove or insert UUI information
- Deliverables:
  - Problem statement: draft-ietf-cuss-sip-uui-reqs
  - SIP Call control UUI specification: draft-ietf-cuss-sip-uui
  - ISDN UUI Application usage
- Timeframe: Deliverables proposed to be completed in WG June 2011

CODEC WG

- Objectives: Define an Internet codec meeting the following conditions:
    1. Optimized for use in interactive Internet applications.
    2. Published by a recognized standards development organization (SDO) and therefore subject to clear change control.
    3. Can be widely implemented and easily distributed among application developers, service operators, and end users.
- Deliverables:
    1. Technical Requirements: draft-ietf-codec-requirements
    2. Codec specifications:
        - Definition of the Opus audio code: draft-ietf-codec-opus
    3. Codec Standardization Guidelines defining the work processes: draft-ietf-codec-guidelines
- Timeframe: WG documents completed (Oct. 2011), Liase with other SDOs (Nov. 2011)

CLUE WG

- Motivation: Current telepresence systems are based on open standards such as RTP, SIP, H.264, the H.323 suite. However, they cannot easily interoperate without operator assistance and expensive additional (vendor specific) equipment.
- Objective: Create specifications for SIP-based conferencing systems to enable communication of information about media streams so that a sending system, receiving system, or intermediate system can make reasonable decisions about transmitting, selecting, and rendering media streams.
- Requirements and use cases available:
  - draft-romanow-dispatch-telepresence-requirements
  - draft-ietf-dispatch-telepresence-use-cases
- Other deliverables:
  - Framework/model (may be combined with requirements doc)
  - Description of protocols to achieve functionality (syntax and transport mechanism)
- Work targeted to be completed by November 2011.

Verification Involving PSTN Routability (VIPR)

- Motivation: Phone numbers are currently used to connect to SIP islands via PSTN Interworking. However, (Public) ENUM (mapping E.164 numbers to SIP URIs) is not yet widely deployed. Thus, services between SIP entities are limited by the PSTN.
- High Level Solution Proposal:
  1. Initial call via PSTN.
  2. Use P2P SIP Distributed Hash Tables (DHTs) to determine if the called party in a PSTN call supports SIP.
  3. Uses P2P validation to create a “ticket”, which is stored in a cache.
  4. Subsequent calls securely bypass PSTN using the “ticket” cached by the call agent for validation.
- Deliverables (current timeframe is completion in April 2012):
  - Requirements, Problem statement, and architecture
  - VIPR P2P protocol specification (using RELOAD)
  - PSTN based number validation techniques
  - Specification of authorization tokens to mitigate SPAM
  - Protocol for call agents to exchange call and routing information

RTCWEB *Proposed* WG

- Motivation: There are many applications that use a Web browser to support direct, interactive communications, including voice, video, collaboration, and gaming. However, these applications typically require the installation of plugins or non-standard browser extensions.
- Proposal: Standardize the functionality, so these applications can be run in any compatible browser and allow for high-quality real-time communications experiences within the browser.
- Proposal: Work will be done in coordination with W3C (who will develop the API)
- Likely to be chartered prior to IETF-81.
- Tentative timeline is completion of the work mid-2012.

Summary

- Lots of new work related in the Real-time Applications infrastructure area related to SIP and other protocols.
- Keys to success:
  - Participation of network operators/service providers increases relevance.
  - Participation of product development primes encourages evolution of standards to meet market demands.
  - Input from participants and liaisons from other SDOs/forums broadens market applicability and improves interoperability.
  - Prototyping (i.e., “running code”) as the standards evolve enhances interoperability and improves quality of protocol specifications.

Questions ?

- What do you want the IETF to be working on?

THANK YOU

BACKUP

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- Problem:
  - Phone numbers are currently used to connect to SIP islands via PSTN Interworking.
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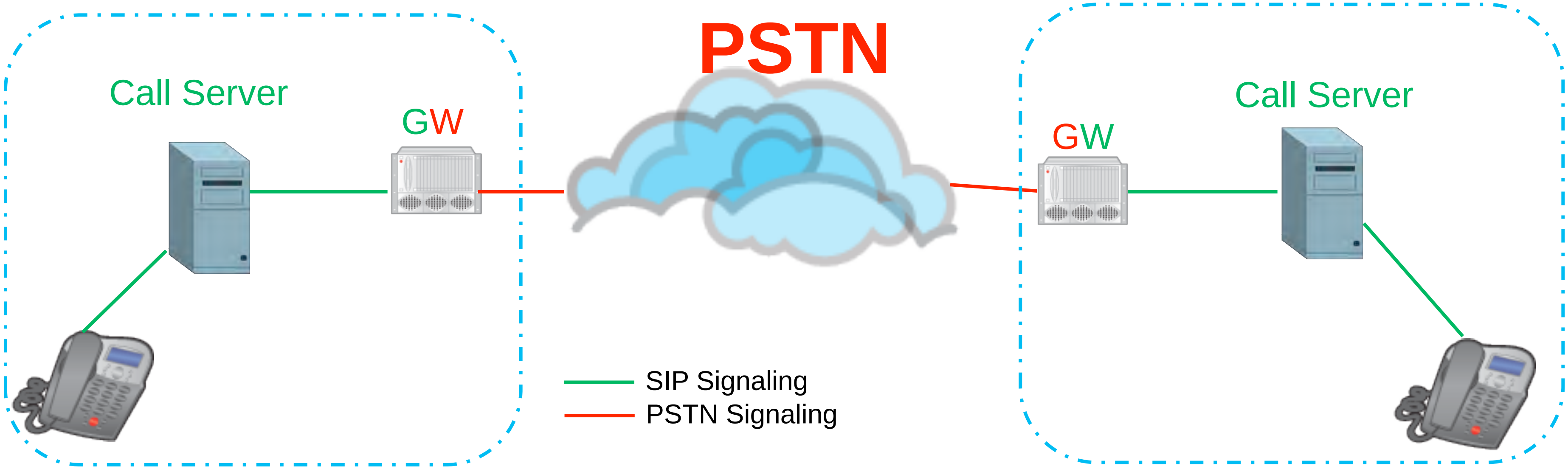
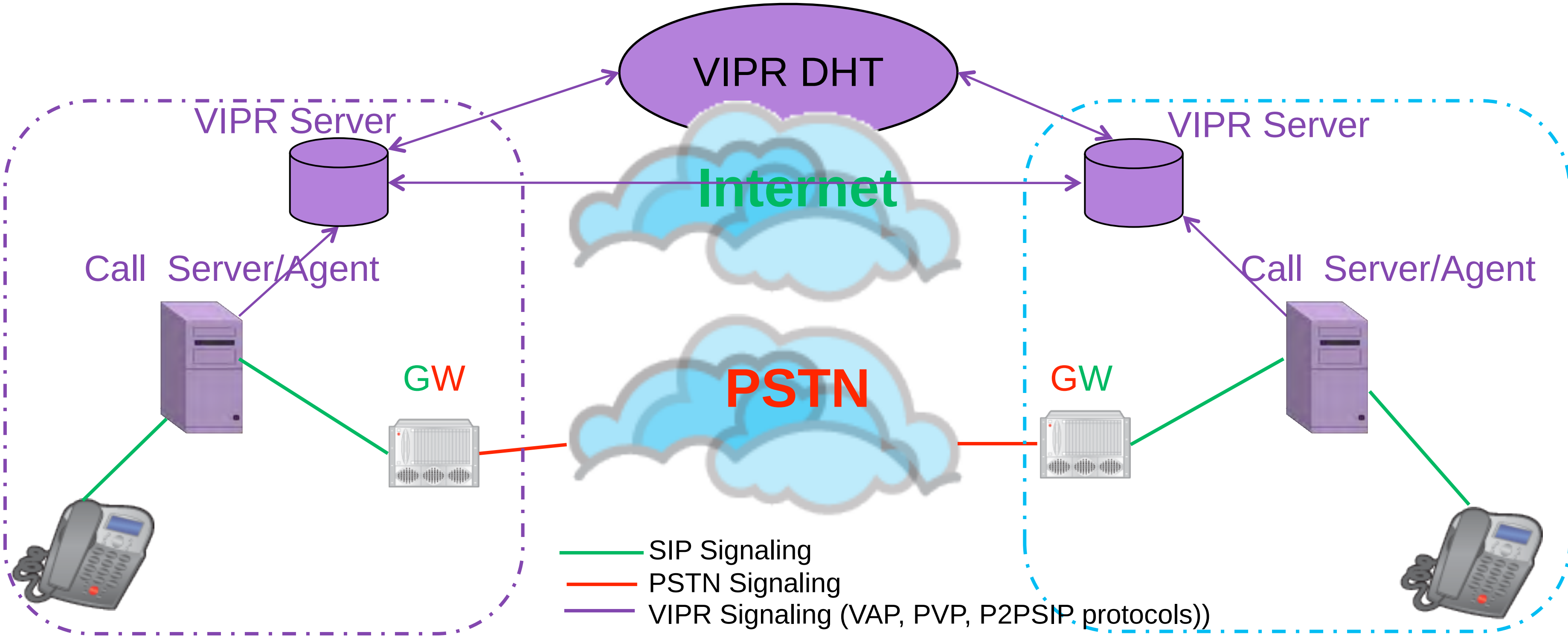


Verification Involving PSTN Routability (VIPR)

- Uses P2P SIP Distributed Hash Tables (DHTs) to determine if the called party in a PSTN call supports SIP.
- Uses P2P validation - returns a “ticket” (phone #, domain & signature).
- Stores SIP URI and a “ticket” in a cache.



Verification Involving PSTN Routability (VIPR)

- Subsequent calls securely bypass PSTN - the “ticket” cached by the call agent is used for validation.
- Enables video (and other media) calls in cases where only voice was previously available.