**Contribution**

**TITLE:** IP Interconnection Routing Outline

**SOURCE\*:** AT&T

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**ABSTRACT**

This document proposes a new baseline for the IP Interconnection Routing Document to be distributed in June 2014 by the ATIS/SIP Forum IP-NNI Task Force. The new baseline organization is intended to reflect task force agreement to focus on documenting current and per-TN solutions under study and their interworking. The proposed baseline moves other proposals to an appendix.

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\* CONTACT:

Penn Pfautz; email: pp3129@att.com; Tel: +1732-420-4962

**ATIS-0x0000x.YYYY**

American National Standard for Telecommunications

**IP Interconnection Routing**

**Alliance for Telecommunications Industry Solutions**

Approved Month DD, YYYY

**American National Standards Institute, Inc.**

**Abstract**

Abstract text here.

**Foreword**

The information contained in this Foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI’s requirements for an ANS. As such, this Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the Standard.

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ANSI guidelines specify two categories of requirements: mandatory and recommendation. The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, [**COMMITTEE NAME**], 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, [**COMMITTEE NAME**], which was responsible for its development, had the following leadership:

[**LEADERSHIP LIST**]

The **[SUBCOMMITTEE NAME]** Subcommittee was responsible for the development of this document.

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# Scope, Purpose, & Application

## Scope

This document was developed under a joint ATIS/SIP Forum collaboration. The document discusses the existing in-use and proposed routing solutions to facilitate the exchange of traffic associated with IP-based services between North American service providers.

Many options and issues were previously investigated by an ATIS Inter-Carrier VoIP Call Routing Focus Group (IVCR-FG), which issued its final report in February 2008. At that time, the IVCR-FG report noted that a number of vendor proposals have been made, but no initiative exists to develop the necessary standards needed to enable VoIP call interconnectivity [1].

Subsequent to the formation of the ATIS/SIP Forum collaboration, the Federal Communications Commission authorized the creation of a Numbering Testbed to “spur the research and development of the next generation standards and protocols for number allocation, verification, and call routing.”[[1]](#endnote-1) The Commission also held a workshop to initiate a Numbering Testbed on March 25, 2014. Discussion at the Workshop focused on ideas for a “future integrated registry” that would support number allocation, verification, and call routing across all types of NANP numbers in a post TDM environment.

It should be noted that this first report does not address the development of such an integrated registry, but instead focuses on the identification of existing in-use and proposed “interim” solutions to facilitate call routing across IP interconnections between now and the deployment of the future integrated registry envisioned at the Workshop.

## Purpose

As Service Providers introduce and expand IP-based service offerings, there is increasing interest in identifying the opportunities for the industry to facilitate IP routing of VoIP traffic using E.164 addresses. The ATIS/SIP Forum Task Force has taken on the initiative to develop the necessary standards and is publishing this first report to describe the candidate proposals for circulation and comment.

The purpose of this first report is to:

1. Provide an overview of the in-use and proposed architectures with the provisioning processes and calls flows to facilitate the exchange of VoIP traffic associated with IP-based services using E.164 addresses.
2. Present criteria that provide an overview of the routing information elements required to recognize the comparative characteristics of each of the approaches.

Based upon the output of this first report, further analysis will be presented in a final report that includes:

1. Refinement of solution(s) and criteria that includes consideration of feedback obtained from the first report.
2. How existing in use and proposed interim solution(s) may be adopted and/or coexist, and evolve for transition to a future integrated registry envisioned at the Workshop.
3. Finalization of criteria requirements
4. Development of analysis leading toa recommendation of an interim solution or set of solutions.

## Application

This standard is defined for North America deployments, but may be applicable for deployments outside North America.

# Informative References

[1] ATIS-I-0000017, ATIS Inter-Carrier VoIP Call Routing (IVCR) Assessment and Work Plan, February 2008

[2] ATIS-0x0000x, *Technical Report*.

[3] ATIS-0x0000x.201x, *American National Standard*.

# Definitions, Acronyms, & Abbreviations

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < <http://www.atis.org/glossary> >.

## Definitions

**AAA**: xxxx.

**Bbbb**: xxxx.

## Acronyms & Abbreviations

[this list was copied from Protocol document]

3GPP 3rd Generation Partnership Project

ALG Application Level Gateway

ATCF Access Transfer Control Function

B2BUA Back to Back user agent

BGCF Border Gateway Control Function

CSCF Call Session Control Function

IBCF Interconnection Border Control Function

I-BGF Interconnection Border Gateway Function

I-CSCF Interrogating-Call Session Control Function

ICSS IMS Centralized Services

II-NNI Inter-IMS Network to Network Interface

IM-CN IP Multimedia Core Networks

IMS IP Multimedia Subsystem

IMS-ALG Multimedia Subsystem Application Level Gateway

IP Internet Protocol

IPSec IP Security

IPv4 Internet Protocol Version 4

IPv6 Internet Protocol Version 6

MGCF Media Gateway Control Function

MGF Media Gateway Function

MIME Multipurpose Internet Mail Extensions

MSC Mobile Switching Center

NAT Network Address Translation

NAT-PT Network Address Translation—Protocol Translation

NNI Network to Network Interface

P-CSCF Proxy Call Session Control Function

RTP Real-Time Protocol

SBC Session Border Controller

S-CSCF Serving-Call Session Control Function

SCTP Stream Control Transmission Protocol

SDP Session Description Protocol

SGF Signalling Gateway Function

SIP Session Initiation Protocol

SIP URI SIP protocol Uniform Resource Identifier

SIP-I SIP with encapsulated ISUP

SIP-T SIP for Telephones

SLA Service Level Agreement

SRVCC Single Radio Voice Call Continuity

TCP Transmission Control Protocol

tel-URI Telephone Uniform Resource Identifier

TRF Transit and Roaming Function

TrGw Transition Gateway

TLS Transport Layer Security

UA User Agent

UDP User Datagram Protocol

URI Uniform Resource Identifier

VoIP Voice over IP

# Current Routing Approach Using Existing LERG & NPAC Data

Some service providers are already exchanging voice traffic over IP facilities. This section details how routing for such exchanges has been implemented based on existing industry data in the LERG and NPAC supplemented with the bilateral exchange of information to map LERG and/or NPAC identifiers to IP addresses.

Existing approaches to IP interconnection routing rely on NANP constructs for aggregating telephone numbers into groups and then associating a route (SBC URI or IP address) with the TN group. Most commonly TNs are aggregated via association with a common Location Routing Number (LRN) in the NPAC, but elements such as OCNs, CLLIs, or central office codes (NPA-NXXs) may also be used,

Placeholder:

This section illustrates some of the mechanisms currently in use and/or being deployed to facilitate the exchange of traffic associated with IP-based multimedia services (e.g., VoIP) between North American service providers.

See IPNNI-2014-045R1.

# Telephone Number Registry (per-TN) Approach

## Per-TN Use Case

A number of service providers have identified a need to move beyond routing based on NANP aggregation elements as discussed in the previous section.

In general these needs arise where TNs may share common point of interconnection (PoI) for TDM interconnection (and are thus associated with the same LRN or CLLLI) but need to be treated differently for IP interconnection.

For example, wireless SPs are migrating their existing 2G/3G subscribers to VoLTE – from TDM to IP based user equipment (UE). For VoLTE to VoLTE calls IP interconnection makes sense for a number of reasons – support for high definition (HD) voice and other Rich Communication Services (RCS) features and elimination of needless IP-TDM and TDM-IP conversions as would be required for TDM interconnection. SPs must still offer TDM interconnection for VoLTE TNs since not all SPs are capable or willing to provide IP interconnection. And because the migration will be gated by customer adoption of VoLTE capable UE, SPs may want to maintain existing TDM PoIs for both 2G/3G and VoLTE TNs and maintain existing TDM routing to those PoIs. Moreover, it may be desirable not to use the IP interconnection serving VoLTE TNs for 2G/3G TNs. First, additional network equipment must be deployed sooner than if IP interconnection scales with VoLTE adoption and, second, 2G/3G calls will be forced to go through unnecessary TDM/IP and IP/TDM conversions. These issues can be avoided if an SP can specify IP interconnection routing for VoLTE TNs separately from the associated LRNs.

A related case cited during Task Force discussions occurs in the deployment of RCSe capabilities outside North America in situations where voice calls and sessions using other RCS features need to be routed differently. This may be particularly the case where number portability methods may not support aggregation via methods like porting to different LRNs.

There may be other use cases for TN routing as well. It has been suggested that per-TN routing could be used to either avoid routing calls to fax numbers over IP interconnections using incompatible compression or taking other measures to insure adequate transmission quality.

## Per-TN Routing Implementation

Service providers wishing to provide per-TN routing perform the following provisioning activities:

1. As part of bilateral interconnect negotiations provide mappings for SIP URI hostnames to SBC IP addresses.
2. Populate registry records for TNs available for IP interconnection with the appropriate SIP URI. The URI will be a full SIP URI (e.g., <sip:+13036614567@example.mso-a.com;user=phone> ) but without number portability information.

The registry must insure that only the provider of record for the number as defined by LERG/NPAC can populate a corresponding record.

Service providers electing to use the per-TN routing information will:

1. Provision the hostname – IP address mappings into their internal DNS (A or AA records).
2. Provision TN-URI mappings from the Registry into their internal routing servers. If the routing server is accessed via a SIP query, the SIP URI may be directly populated. If the routing server is accessed via an ENUM query, the SIP URI is encapsulated into a NAPTR record.

This provisioning process is illustrated in Figure 1 below. The Figure shows the registry instantiated in the NPAC but alternate registry implementations (using different provisioning mechanisms than the SOA/LSMS) are possible.

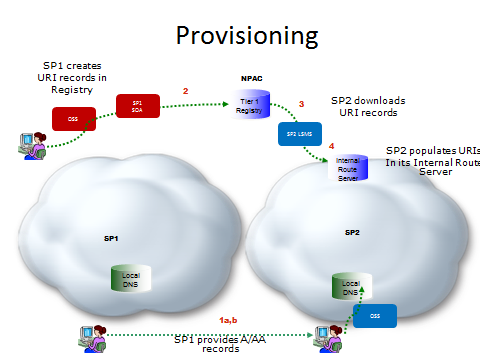
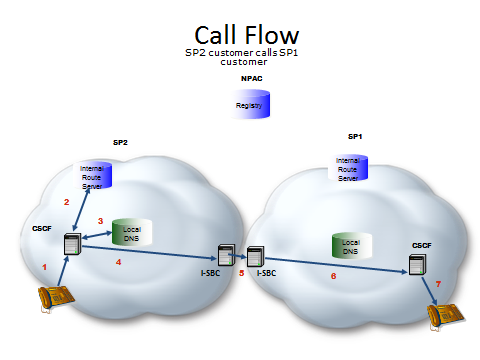


Figure 1

On call origination, the originating service provider will query their routing server and obtain the corresponding SIP URI for numbers available for IP interconnect. They will resolve the hostname from the URI in their internal DNS to obtain the IP address of the terminating provider’s ingress SBC.[[2]](#footnote-1) The call flow is shown in Figure 2 below:



1. SP2 Caller dials destination number
2. SP2 S-CSCF queries internal route server and SP2 route server responds with a URI passed back to S-CSCF
3. SP2 S-CSCF resolves the hostname in the SIP URI to obtain the IP address of an agreed upon SP1 ingress SBC
4. A SIP INVITE is sent to egress SBC of SP2 that has layer 3 connectivity to the ingress SBC of SP1
5. The SIP INVITE is forwarded to the SP1 ingress SBC.
6. and 7. SP1 terminates the call to its end user.

# Interworking between Current and Registry based approaches

This section discusses how interworking may take place between service providers using different routing approaches.

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# Next Steps

# Appendix A – Other Solution Proposals

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# Independent ENUM Registry

Placeholder:

This section describes an independent ENUM Registry, for the exchange of data for IP routing and interconnection for routing of E.164 Addressed Communications over IP Network-to-Network Interconnection (NNI). See IPNNI-2014-043R1.

# Appendix B - Routing Criteria Tables

# Appendix C – Data Exchange Worksheet Example

1. FCC 14-5, released January 31, 2014. [↑](#endnote-ref-1)
2. There may be alternate approaches to combining the bilaterally exchanged URI-IP address mappings and the TN-URI mappings obtained from the Registry and combining them in a routing server for session establishment. [↑](#footnote-ref-1)