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VIA ECFS

Ms. Marlene H. Dortch
Secretary
Federal Communications Commission
445 12th St SW
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Re: In the Matter of Petitions for Waiver of Commission's Rules Regarding Access to Numbering Resources, CC Docket 99-200; Further Notice of Proposed Rulemaking on IP-to-IP Interconnection Issues, WC Docket No. 10- 90; GN Docket No. 09-51; WC Docket No. 07-135; WC Docket No. 05-337; CC Docket No. 01-92; CC Docket No. 96-45; WC Docket No. 03-109; WT Docket No. 10-208; Connect America Fund, WC Docket No. 10-90; A National Broadband Plan for Our Future, GN Docket No. 09-51;

Dear Ms. Dortch:

The Petitions for Waivers of the Commission's Rules regarding access to Numbering Resources raises novel questions that have significance far beyond the immediate needs of the petitioners. Numbering issues are becoming central to the Transition of the PSTN to all IP networks.

The Commission has made it clear in the National Broadband Plan and its Report Order and FNPRM [FCC 11-161] on Intercarrier Compensation and Universal Service Fund Reform, that it desires progress as well as "good faith negotiation" among service providers on SIP/VoIP Interconnection.

I use the term SIP/VoIP interconnection instead of IP to IP Interconnection here, since there has been considerable confusion within the Commission and the industry on what this means. SIP/VoIP Interconnection should not be confused with the normal voluntary private agreements for general "best efforts" IP Interconnection that have successfully grown the Internet to where it is today. The Commission should take steps to end this confusion as soon as practically possible. SIP/VoIP interconnection presumes the use of North American Numbering Plan [NANP] naming in SIP session establishment. Other forms of real time communications that do not use NANP resources should be completely out of scope for regulatory oversight, as they are covered by the Commission's Pulver Order.

I recommend and support the idea that the Commission should undertake a Further Notice of Proposed Rulemaking on the full range and scope Numbering issues with particular emphasis on the relationship of Numbering to SIP/VoIP Interconnection and the PSTN Transition, irrespective of the merits of the Petitions for Waivers for Numbering Resources.

I would not recommend a specific NPRM on the issues raised by the Petitioners as advocated by NARUC (June 1, 2012) and others. To focus specifically on the request of the Petitioners would "kick the can" down the road on the Numbering issues that inevitably need to be addressed. A clear national policy on the Future of Numbering is, in my judgment, an essential precondition for further progress on the National Broadband Plan, SIP/VoIP Interconnection and the inevitable transition to all IP networks

The Petitioners and their Opponents have clearly demonstrated the extrodinary monetary value attached to US numbering resources. The Petitioners and their Opponents both make valid points. Innovative service providers, such as the Petitioners, have and continue to use telephone numbers in novel and innovative ways. This process of innovation needs to be accelerated. This is not just about voice. There are now several service providers that are using NANP resources for alternative SMS services, in partnership with licensed carriers. The Opponents correctly point out that they have made substantial capital investments in order to have direct access to NANP resources and fully accept the obligations direct NANP access entails.

The Commission can easily assume that if the current Petitions were granted, then the "floodgates" would be opened to dozens of companies with similar issues. What is not completely clear from the record on the Petitions is what would be the effect on numbering resources in the intermediate term given the absence of a comprehensive national policy on how such non-licensed companies should be treated.

Much like the Universal Service Fund and Intercarrier Compensation, the system of NANP access and allocation is clearly not working. This argues that a full and public inquiry is called for.

The Commission should seek comment on a wide range of issues, among them:

- What is the role of Telephone Numbers in an all IP world?
- Who has access to Numbering Resources and why?
- What public obligations should holders of the US portions of the North American Numbering Plan have beyond requirements to support Number Portability, Number Pooling and Number Resource Utilization Reporting?
- How do Number holders plan on using these resources in SIP/VoIP network interconnection strategies?

- What technical numbering resource techniques are being used in SIP/VoIP Interconnection today?
- What new and innovative services that use North American Numbering Plan [NANP] resources could be enabled by encouraging the industry to move to all SIP/VoIP interconnection?
- How will the 800 SMS be transitioned to SIP/VoIP?
- What are the new requirements for numbering for Machine to Machine and Telematic applications?
- Are new numbering databases for SIP/VoIP Interconnection necessary? What data should those databases contain? Are application specific URI's required at the Telephone Number level?
- Should the United States move immediately to national 10 digit dialing, if for no other reason than to increase the size of the NANP?
- Will National 10 dialing and the proposed end of LATA's in the ICC/USF reform create the technical preconditions for National Geographic Number Portability?
- What are the benefits to consumers and businesses of National Geographic Number Portability?
- Does anyone really care about the Geographic nature of existing telephone numbers? Are telephone numbers just a name?
- Is the existing allocation model for Phone Numbers relevant anymore?
- Should phone numbers be treated as "domain names"? What are the risks?
- How would the Federal/State partnership over management of Numbering Resources be affected?
- What are the new security and authentication models associated with NANP resources in an all IP world?
- Do future numbering databases need bindings to digital certificates as we now have with DNSSEC?
- Are new provisioning technologies needed for Numbering Databases in an all SIP/VoIP world?

It is not noted that the North American Numbering Council [NANC] has had a Working Group on the Future of Numbering for many years, but its discussions have not received significant attention nor has its output come to definitive conclusions or recommendations for the Commission to act on. In addition, the FCC's Technical Advisory committee is also looking into these issues.

Further, the Commission should seek comment on its Authority to Act under Section 251(e)[1] which governs its Authority over telecommunications numbering in general. That Authority has been successfully tested in the Federal Appeals Courts [267 F.3rd 91 2001]

- Can the Commissions Plenary authority over Numbering in 251(e)[1] be used to encourage the Transition from the PSTN to all SIP/VoIP interconnection without affecting the existing voluntary, industry led agreements on best efforts IP interconnection?
- Can 251(e)[1] be used to facilitate SIP/VoIP interconnection agreements as discussed in FCC Order FNPRM 11-161 paragraph 1351-1358 as opposed to the use of 251(a)[1], 251(c)[2] or Section 706?
- Could SIP/VoIP Interconnection agreements be a precondition to access to numbering resources?

Telephone Numbers are and will continue to be a vital and essential part of the overall real time communications landscape of the United States and global communications for the foreseeable future. They are simple, trusted and linguistically neutral. Telephone numbers, along with Internet Domain names, represent the two communications naming and addressing systems on the planet that are completely universal. The Commission has a wonderful opportunity here to review the existing system and create the preconditions for allowing phone numbers to be an engine for innovation and not an impediment.

I attach for the Record a Paper I presented at a Conference on the End of the Phone System at The Wharton School of Business in May of this year where these issues and other SIP/VoIP related technical issues were discussed.

I'm available to answer questions from staff as needed.

Respectfully submitted,

Richard Shockey
Principal
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Technical Challenges in the PSTN Transition from Plain Old Telephone Service (POTS)[1]

Presented at the End of the Phone System Workshop Held at the Wharton May 16, 2012

http://mackcenter.wharton.upenn.edu/eventpage/2012/05/16/134-the-end-of-the-phone-system

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ABSTRACT

The Federal Communications Commission and the Canadian Radio-Television and Telecommunications Commission have issued Orders and directed Policy that will begin the long process of transitioning from classic Time Division Multiplexing [TDM] and Signaling System 7 [SS7] to all IP technologies over a ubiquitous National Broadband System and all Session Initiation Protocol [SIP]/Network to Network Interconnection [NNI] for key legacy services such as Voice, SMS and 91, new innovative service such as point to point video calling and enhanced services for the disabled. This will be the most radical technical transition the traditional Phone System has ever seen. The transition will be a highly complex process that will retire billions of dollars of existing infrastructure and replace it with better, faster and less expensive technology that will create a more reliable, as well as functional, Real-Time Communications Network for services using E.164 [phone number] naming.

The rationale for this transition is not only the promise of new service delivery and reduced operational costs for service providers, but it is increasingly evident that the existing PSTN equipment in the network is operating well beyond its projected 25 to 30 year End of Life with parts and software patches increasingly hard to procure. The risk of network failure and the implications for Public Safety are of increasing concern.

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[1] Richard Shockey is the Principal of Shockey Consulting, a private firm advising telecommunications companies and the investment community on any number of issues related to Next Generation Networks, Voice over IP, Communications Provisioning, Peering, Numbering and Signaling. He is also is Chairman of the Board of Directors of the SIP Forum, an IP communications industry association that engages in numerous activities that promote and advance SIP-based technology [RFC 3261]. Mr. Shockey was a co-founder and long time co-chair of the Internet Engineering Task Force (IETF) ENUM Work Group [RFC 6116] and an author of several IETF RFCs. He also co-founded the IETF working group DRINKS on the provisioning of data for VoIP Peering Federations. Mr. Shockey is also a member of the Federal Communications Commission Communications Security, Reliability and Interoperability Council

Opinions in this paper are purely those of Mr. Shockey and do not necessarily represent those of the SIP Forum or any of its member companies.

This paper makes a clear distinction between the social contract inherent in the PSTN and the underlying POTS technology currently deployed. Among the technologies that have not been addressed is the role of centralized numbering databases in Session Initiation Protocol [SIP] session establishment and SIP/VoIP interconnection discovery.

The Federal Communications Commission may have Authority to Act for SIP/VoIP Interconnection under Section 251 (e) [1] which governs numbering administration.

1. Introduction

Many of the core technologies involved in the PSTN transition are starting to be deployed, especially in Cable and 4th generation Long Term Evolution [LTE] mobile networks, but there are key technical elements of the existing PSTN that have not been addressed and that are not fully understood by policy makers. It is essential that the existing social contract, features, functionality and quality of service of the current PSTN be preserved in any technical transition.

The migration of the PSTN to a SIP-based architecture has the promise of enabling a variety of business and public goods, including video calling, location-based information, advanced emergency services, and enhanced services for the disabled, but the predicate to all of this promise is ubiquitous SIP/VoIP interconnection. As long as we view the use of TDM/POTS as the default least common denominator for service interconnection, we will preclude the introduction of new and innovative real-time communications services in the future.

This paper will review how we got to this point, the key technologies that must be replaced, strategies for managing this technical transition, and policy implications for such areas as numbering and network management.

2. Recent Regulatory Events Driving the PSTN Transition

With the publication of The National Broadband Plan[2], the FCC has stated its goal for a transition to all IP networks at some point in time, especially for the essential core PSTN/POTS services, such as voice and SMS. The FCC's USF/ICC order[3] and the FCC's Technical Advisory Committee[4] reconfirmed that policy goal. The FCC is also aware, based on its own Form 477 data, that both managed, as well as OTT [Over the Top], VoIP services are proliferating at an accelerated rate of adoption.

The American people rely on real-time communications services based on E.164 numbers capable of delivering reliability, affordability, accessibility and ubiquity consistent with principals of fairness in pricing, privacy and suitable institutions for the redress of grievances. It is essential that policy makers make a clear and demonstrable

[2] http://www.broadband.gov/plan/

[3] http://transition.fcc.gov/Daily_Releases/Daily_Business/2012/db0206/FCC-11-161A1.pdf

[4] http://transition.fcc.gov/oet/tac/TACJune2011mtgfullpresentation.pdf

distinction between national communications goals and objectives and the issue of a technical transition of POTS from TDM to SIP.

In addition, the Canadian Radio-Television and Telecommunications Commission (CRTC) issued a Policy directive in January 2012 that came to many of the same conclusions the FCC did.[5] Central to the CRTC Policy framework was a mandate for SIP/VoIP interconnection for voice. Canadian carriers must provide SIP/VoIP Interconnection for core voice services within 6 months of a request. In addition, the policy order mandated development of a master SIP/VoIP voice interconnection agreement, SIP/VoIP voice interconnection test plans, and the establishment of a Carrier ENUM database.

3. Current State of the PSTN Infrastructure

As a practical matter, the existing TDM/SS7 switching/signaling infrastructure among incumbent land line and mobile carriers has reached its practical End of Life (EOL). Most of this equipment cannot be modified or upgraded, nor does it match to future SIP interconnection strategies. Carriers and the telecommunications equipment supplier community are painfully aware of this.

The current state of the POTS infrastructure is now complicating aspects of the USF/ICC order. Virtually all of the legacy land line service providers have petitioned for Limited Waivers from the Order on the phantom traffic call signaling rules.[6] Phantom traffic refers to traffic that terminating networks receive that lacks certain identifying information necessary for billing. The cost of this problem to carriers has been estimated at hundreds of millions of dollars. The FCC has ordered carriers to include Calling Party Number or Charge Number in the SS7 signaling stream. The petitions for Waiver indicated that compliance with this rule is due to the lack of technical support or End of Life service for critical components in the Class 4 TDM (tandem access) infrastructure. The first 4ESS Class 4 switch was introduced in 1976.

The aging condition of Class 4 components potentially exposes possible deterioration in the critical Class 5 (5ESS, DMS) access infrastructure that provides the essential link to 911 emergency calling for both legacy wireline services, as well as all current 3G HSPA/CDMA mobile voice communications. The first Class 5 5ESS was introduced into service in 1982.[7]

There is ample anecdotal evidence that critical TDM-based equipment parts are in short supply, as a large secondary market has emerged for used parts, and the trained personnel that understand the POTS infrastructure are retiring at an accelerated pace. At some point neither of these resources will be available. The question is when?

[5] http://www.crtc.gc.ca/eng/archive/2012/2012-24.htm

[6] http://hraunfoss.fcc.gov/edocs_public/attachmatch/DA-12-34A1.pdf

[7] http://en.wikipedia.org/wiki/5ESS_switch

In my judgment, the need for an orderly transition from POTS to all SIP/VoIP networks is not just a laudable goal, but a critical national goal since the End Of Life state of the PSTN infrastructure indicates an emerging Public Safety problem if these components fail.

4. The State of IP-based Real-Time Technology in the Evolving Network

The Session Initiation Protocol[8] [IETF RFC 3261] and its protocol super set, the IP Multimedia Subsystem [IMS], are the next generation of IP-based Real-Time Communications Systems. There is no engineering dispute about this.

Real-time communications, especially voice, are not going away any time soon. In fact, voice communications minutes are increasing.[9] Enterprises have spent billions of dollars deploying SIP-based IP PBX systems which now totally dominate the market.[10] In addition, SIP Trunking, which is the replacement of classic, channelized T1 or Primary Rate Interfaces for PSTN connection in enterprises, grew 88% during 2011 from the previous year.[11] Cloud Telephony, which is the IP version of traditional POTS hosted CENTREX services, is also seeing rapid market expansion.

Cable Operators are now nearly 100% SIP/IMS based. Advanced offerings from ATT and Verizon such as FIOS and uVerse are all based in part on SIP/VoIP/IMS for real-time communications services. OTT providers such as Vonage have been SIP-based from their inception and Competitive Exchange Carriers, such as Level 3, XO and CenturyLink, have significant SIP/VoIP/IMS infrastructure in place.

The mobile industry is very close to deploying Voice over LTE [VoLTE] over 4th Generation networks, with the first interoperability tests occurring in late 2011.[12] VoLTE will use SIP/IMS, with High Definition Voice being one of the many new services envisioned by mobile operators.[13]

5. Technical Issues in the Transition

The ICC/USF order and the CRTC Policy statement, in my judgment, have correctly identified all SIP/VoIP Interconnection as the first area to begin the PSTN Transition. Though the Order encourages carriers to "Negotiate in Good Faith" SIP/VoIP Interconnection, a key missing ingredient is a fundamental technical understanding of how ubiquitous IP-to-IP interconnection for real-time communications using E.164 phone numbers would work *on the wire*.

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[8] http://www.ietf.org/rfc/rfc3261.txt

[9] http://www.businessinsider.com/long-live-phone-calls-2011-2

[10] http://www.infonetics.com/pr/2012/4Q11-Enterprise-UC-VoIP-TDM-Equipment-Market-Highlights.asp

[11] http://finance.yahoo.com/news/infonetics-research-enterprise-session-border-003800194.html

[12] http://www.msforum.org/interoperability/VoLTE.shtml

[13] http://en.wikipedia.org/wiki/Rich_Communication_Suite

Without a common stakeholder consensus on what the technical profile for SIP/VoIP Interconnection is, there will be no progress on achieving the FCC or the CRTC's stated goals.

The transition to all SIP/VoIP Interconnection has enormous advantages for the telecommunications industry as a whole. The costs of Media Gateways and Signaling Gateways between POTS and managed SIP/IMS networks are a huge financial burden on all operators. The PSTN/POTS network operates as a parallel network universe that increases OPEX for legacy carriers that could better be deployed bringing ubiquitous broadband to consumers.

SIP/VoIP Interconnection is NOT IP to IP Interconnection. It is vitally important to distinguish between the existing voluntary IP to IP Interconnection agreements that govern "best efforts" IP traffic from the requirements for SIP/VoIP Interconnection. They are very different since real time communications traffic is latency sensitive and will require a very different form of network management and a different set of interconnection agreements in order to maintain the Quality of Service that existing business and residential customers are familiar with and have come to expect.

a. *The transition from POTS to SIP/VoIP must eventually eliminate Signaling System 7.*

The transition from the classic PSTN to an all SIP/VoIP infrastructure will mandate the end of Signaling System 7[14] and the entire infrastructure that supports it. This is a substantial undertaking, the consequences of which are not fully understood. There is a variety of data transmitted by the SS7 network that currently has no equivalent in an SIP/IMS network configuration.

An example of this is the Line Information Database [LIDB][15]. LIDBs store an array of subscriber and service information that is critical to call completion. Calling Party Name [CNAM] depends on LIDB records, as does single-number service, such as 311 or local mappings for regional or national 800 numbers, and to block certain calls, allow collect calls, allow international calls, validate account information, etc.

b. *All SIP/VoIP interconnection must include SMS.*

The ubiquitous SMS [Short Message Service] is an extremely important and profitable service for mobile operators and, increasingly, land line and cable operators who can deliver SMS messages on multiple devices such as televisions. Since the SMS service is integral to SS7, this service must be transitioned to an all IP Interconnection system as well. How this will be done is not well understood.

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[14] http://en.wikipedia.org/wiki/Signaling_System_7

[15] http://telecom-info.telcordia.com/site-cgi/ido/docs.cgi?ID=SEARCH&DOCUMENT=GR-446&

c. Fax.

Fax is a global communication service using E.164 numbering over existing POTS circuits. Many service providers have reported substantial problems transmitting fax over SIP/IMS networks. The reason for this is that the ITU T.30 protocol for fax is unusually sensitive to latency in the network. The technical issues in fax failure analysis are not well understood. The SIP Forum, among other associations, has technical task groups looking at the problem.[16] As of this date, there is no resolution of these issues.

d. International Call Completion.

There are no current equivalents in the SIP/VoIP Interconnection standards for International Call Completion or Global Title Translation, in PSTN terms.[17] Several industry groups have looked at various means to accomplish this goal, including the creation of a Global Service Provider Identification Code[18], but as of this date there is no international consensus on how to proceed.

e. Are there sufficient "technically feasible" points of Interconnection for all SIP/VOIP real-time communications?

Logic would argue yes, unless you are a rural carrier in Alaska or Montana, for instance. The reality is no. Certainly, some incumbent carriers are using existing carrier hotels in well-understood *urban* locations to segment out real-time communications traffic and interconnect today. However, some segments of the industry are complaining that even for the existing "best efforts" IP traffic interconnection, rural carriers are required to buy expensive special access circuits in order to interconnect. This is an area that requires additional study.

f. In an all SIP/IMS interconnection agreement, who is responsible for what?

Any Master Technical Agreement on SIP/VoIP Interconnection needs to reach consensus on multiple issues, such as what is the appropriate Layer 1 or Layer 2 means of interconnection, such as Ethernet or Multi-Protocol Label Switching. There are complicated issues that would have to be resolved, such as how is CNAM to be transmitted and how are existing legal privacy requirements to be respected or enforced. A particularly contentious issue will be which party in the transaction is responsible for transcoding media. The promise of all IP communications means that there will be multiple devices capable of transmitting audio and video codecs of various types. There is no current industry consensus on how this problem should be addressed. Current agreements are strictly on a bi-lateral basis.

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[16] http://www.sipforum.org/content/view/310/252/
[17] http://en.wikipedia.org/wiki/Global_Title
[18] http://datatracker.ietf.org/doc/draft-pfautz-service-provider-identifier-urn/

g. The state of technical standards in SIP/IMS Interconnection

The good news is that, unlike the Digital Television Transition, we may have up to 80% of the relevant technical standards in place to deploy a POTS transition. The bad news is that the last 20% either don't exist or would be subject to considerable differences of opinion within the technical community. Technical standards are not immune from economic or political considerations in their development and those factors must be taken into consideration.

Some technical work has been undertaken on what is commonly referred to as the Network to Network Interface [NNI]. The NNI is a "profile" of existing IP communications standards that sets out the minimum set of implementation requirements and implementation guidance on the wire. At last count, there are multiple different interfaces that have been documented by standards bodies associated with different parts of the community, including those from ATIS-PTSC, CableLabs, 3GPP, GSMA, i3Forum, and the ITU-T.[19] [20] [21] [22] [23] [24]

I believe many of these technical profiles may be mutually incompatible and must be reconciled or progress cannot be made on a framework for implementing the Transition. In addition, I believe all discussions about the technical aspects of the Transition should be made in an open multi-stakeholder process that has been successfully used in bodies, like the Internet Engineering Task Force, to achieve the Internet we now know.

Of larger concern is what we do not know. Namely, what elements of the traditional PSTN/POTS infrastructure have no equivalent in the SIP based world? Public policy experts should be deeply concerned that no fundamental "gap analysis" has occurred, identifying elements of the TDM/SS7 world that may require direct action by the relevant Standards Development Organizations. It is my judgment that a comprehensive review of existing NNI profiles and a technical gap analysis of omissions in the technical standards would take a concentrated effort of not less than 18 to 24 months.

h. Current SIP/VoIP Interconnection models

Considerable SIP/VoIP Interconnection is occurring now for basic POTS traffic. Cable Operators are already using managed packet labeled technologies for SIP/VoIP

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[19] i3Forum - IP international interconnections for voice & other related services Technical Interconnection Model for International Voice Services (http://i3forum.org/library).

[20] GSMA -Inter-Service Provider IP Backbone Guidelines (http://www.gsma.com/go/download/?file=).

[21] ITU-T -Q.3401 NGN NNI signaling profile (http://www.itu.int/itu-t/recommendations/index.aspx?ser=Q).

[22] 3GPP -TS 29.165 – Inter-IMS Network to Network Interface (http://www.3gpp.org/ftp/Specs/html-info/29165.htm).

[23] PacketCable Interconnect Guidelines Specification (http://www.cablelabs.com/specifications/PKT-SP-IGS-I01-110228.pdf).

[24] http://www.atis.org/0191/ngcitf.asp

Interconnection to their competitive advantage. Cable Operators are delivering the exact level of Quality of Service that consumers and businesses have been used to from traditional land line operators. Consumers simply do not know, nor do they care, that their voice communications are using state of the art technology based on SIP/IMS.

Cable Operators are now capable of avoiding Inter-Carrier Compensation among themselves and eliminating the use of the PSTN as the default network, as well as SS7 signaling when E.164 traffic is exchanged among them.

i. So how do they do it?

One technical technique for SIP/VoIP interconnection uses some clever engineering and the Number Portability Administration Database [NPAC].

1. As a call enters an Operator's network, the operator must first discover the terminating provider of record Service Provider Identification Code [SPID] from a locally cached Number Portability Administration Center and the Local Exchange Routing guide databases, and then establish an alternative VoIP trunking mechanism for call termination.

2. As the call comes into the originating network's SIP/IMS proxy, the Operator reads the TO: field in the SIP signaling headers to find the destination phone number. At that point, the originating network performs a localized Local Number Portability look up to find the true Local Routing Number [LRN] of the terminating party and additionally performs a look up to find the SPID associated with that LRN. These look ups are accomplished using locally cached and highly redundant databases within the Operator's network. These queries do not use SS7, but IP based ENUM (RFC 6116) or SIP Redirect queries.

3. The SIP/IMS originating network proxy performs a Policy function to determine if the SPID corresponds to a terminating network where the originating network has a bilateral agreement to reciprocate each other's VoIP traffic.

4. Instead of using the SS7 network or localized SS7 data to determine the Destination Point Code for the terminating Class 5 TDM switch, an alternative trunk determination mechanism is used by the originating networks local policy server to identify one or more IP entry points to the terminating network as defined by a Uniform Resource Indicator [URI].

5. The originating carrier sends the SIP session INVITE signaling message out over that "VoIP trunk" and the terminating carrier's edge Session Border Controller authenticates the session establishment data and ultimately completes the call.

Some mobile operators are now using this technique as well.

This technique is useful for a preliminary phase of the POTS transition, but will not, in my judgment, be sufficient to provide the level, quality and granularity of services an all IP network can provide. The reason for this is that the aforementioned technique assumes that the destination is some form of traditional POTS. Only the interconnection method has been altered.

To achieve a true transition to an all IP network will require a new look at numbering issues in the network.

6. Issues in Numbering

Of unique concern in the Transition are issues involving E.164 number translation and Service Discovery. E.164 telephone numbers are now and will continue to be an essential part of the National Communications System. North America relies on two essential numbering databases. The first is the LERG. This is the central routing database for the PSTN. In addition, there is the NPAC, which is technically referred to as the "exception database", since its data overrides information in the LERG to discover the ported Local Routing Number and corresponding Destination Point Code necessary for SS7 to accurately signal the network to complete the call.

With the introduction of VoLTE and other advanced services, it becomes even more imperative to discover at the point of call/session origination if the endpoint is SIP/IMS reachable, accessible, or capable in order to ultimately deliver the service via IP end-to-end. This discussion is even more critical, since it will also include the entire 800/Service Management System and could include locally specific N11 services, such as 311, 411, etc.

In addition, with the rapid adoption of SIP-based IP-PBX systems and SIP Trunking services, enterprises are not gaining the full benefit of the billions of dollars of investment in these systems, since there is no mutually agreed to database(s) among the service providers that can specifically identify SIP/VoIP endpoints based on their E.164 number across Autonomous System [AS]/carrier boundaries.

The industry has known for some time that there was a necessity to design one or more new databases that could translate an E.164 phone number into IP URIs or IP query models for metadata associated with that phone number.

Though the existing system of SIP/VoIP Interconnection works, and works well, it is based on a rather crude direct bilateral exchange of data among operators, often by simple spreadsheet. This can continue to work among a limited number of operators but, in my judgment, will not scale if it is to deploy among the 1200 licensed operators in the United States. SIP/VoIP Interconnection will ultimately require industry agreement on one or more new centralized numbering databases containing IP specific data.

One technique that has been contemplated for some time has been IETF ENUM (RFC 6116).[25] In fact, ENUM has been in wide deployment for many years now, as service providers first used it to eliminate costly SS7 data queries from their networks, then

[25] The author co-founded and co-chaired the IETF ENUM working group until 2010 when it was dissolved.

adapted ENUM to MMS routing for pictures between mobile phones. Recently, the FCC itself has deployed ENUM as part of its I-TRS service for the disabled.[26]

This is not to suggest that ENUM is the ultimate solution to IP numbering databases – only that the technical community understands the problem and that next generation numbering databases are a fundamental requirement for a PSTN transition.

7. Authority to Act

Many discussions of the Transition and the USF/ICC Order have centered on the Commission's Authority to Act. In my judgment, the Commission has not looked carefully at the possibility of using 251(e)[1] as a basis for action. I believe that using interconnection authority under other sections of 251 or section 706, creates an unusual and unnecessary complication for the Commission. Those rules were created during a different era and may not be appropriate for the current state of the network. The Commission, correctly in my judgment, has chosen not to open up a Pandora's Box by declaring Interconnected VoIP a Title II service. The FCC's authority in Numbering is plenary, though it has traditionally shared responsibility for numbering with the states. Canada's authority over its portions of the North American plan is even more extensive. (*See*, Appendix A attached hereto).

In addition, the FCC's plenary authority for the Numbering plan has been successfully upheld by the US Second District Court of Appeals in a case involving Number Pooling in 2001.[27]

Focusing regulatory attention specifically on the E.164 named traffic *only* has the advantage of "boxing" the problem into a specific subset of SIP/VoIP traffic without disturbing the existing, voluntary and well-functioning IP Interconnection agreements that govern "best efforts" IP service. Over the Top (OTT) Voice providers would not be subject to mandatory interconnection agreements unless they chose to originate or terminate E.164 named traffic.

The Commission ruled in the Pulver FWD order[28] that real-time IP communications that did not use NANP addressing or traverse the PSTN were an Information Service and not subject to regulation as a Telecommunications service under Title II of the Act. Therefore, it is logical to conclude that the current ICC/USF Order is *only* addressing issues where NANP numbering is being used to establish a communications session across any network platform.

If the Commission chose to do so it could simply rewrite the existing North American Numbering Plan or Number Portability Administration Center [NANPA] user

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[26] http://neustarcare.org/infrastructure/itrs_enum.php

[27] *People of the State of New York & Public Service Commission of the State of New York v Federal Communications Commission, et al.*, 267 F.3d 91. Decided September 28, 2001.

[28] http://hraunfoss.fcc.gov/edocs_public/attachmatch/FCC-04-27A1.pdf

agreements, and require those carriers that have access to the North American Plan to agree to new requirements as directed.

The North American Numbering Council has had a "Future of Numbering" committee in place for some time, but it is my opinion that it has been ineffectual and is not constituted in such a manner to understand the value future numbering will have in an all SIP/VoIP world.

The Commission should consider a FNPRM's on any and all Numbering issues that could speed the transition to all SIP/VoIP networks and simplify network operations for all service providers.

The Commission should consider working with the State Public Utility Commissions to mandate national 10 digit dialing across the United States. In my estimation, nearly 70% to 80% of all E.164 transactions now use 10-digit dialing. All mobile terminals use 10-digit dialing, and those NPAs with Overlay codes use 10-digit dialing for land line use as well. By mandating 10-digit dialing, it simplifies future E.164 network routing queries.

Mandating 10-digit dialing also opens up the NANP by allowing the use of the "D" digit in NPA-NXX plans to use "1" or "0" in the NXX. This would instantly increase the size of the NANP by 20%, providing significant relief to those NPA areas that are or could be threatened with number exhaustion.

Second, since the USF/ICC reform order contemplates a "bill and keep" methodology for future E.164 traffic, this effectively eliminates the need for LATA boundaries in billing.

The combination of national 10-digit dialing and LATA boundary elimination creates the technical preconditions for National Geographic Number Portability. This would allow for the provision of "one number for life" within the United States. Though Number Portability has been enormously successful in creating competitive telecommunication markets, the fact that one has to change numbers if you move out of LATA boundaries or change mobile carriers out of LATA boundaries is a significant and continuing annoyance to consumers and businesses.

Combined with new phone number to URI numbering databases, these initiatives could create vast opportunities for service providers and the general telecom supplier community to create new innovative and profitable services.

CONCLUSION

We Need a Technical Plan for the Transition without Delay

There have been some private discussions among the carriers on business-level agreements for a Transition, but what has not happened since the National Broadband Plan or USF/ICC Orders were issued, are substantive technical discussions between the stakeholders on what are the practical technical barriers to the implementation of SIP/VoIP interconnection that would permit the eventual migration of the PSTN to an all SIP/IMS-based architecture.

How long will it take? What will it cost? Is there sufficient investment capital available to affect the transition?

State regulators must have an ongoing opportunity to inspect and, if necessary, participate in technically-oriented discussions involving SIP/VoIP interconnection and the evolution of the real-time network since critical aspects of the transition will logically come under their authority. It is not clear the States or the relevant Canadian Provincial authorities are currently involved at the technical level.

This is not, in any way, suggesting that traditional regulation is the goal – far from it – but the reality is that state and provincial regulators are important partners with the FCC and the CRTC in making sure the migration to SIP/VoIP is a success for all involved.

It is time to send real and demonstrable signals to the telecommunications industry that the regulators – state and federal – are serious about the transition to all SIP/VoIP interconnection.

The industry needs to begin an **open multi-stakeholder process** that defines the underlying technical requirements for nationally critical E.164 named real-time communications services and the requirements for new numbering databases that contain E.164 to IP data.

The Technical Advisory Committee of the FCC has done an excellent job of laying out broad general principals involved in the transition to SIP/NNI interconnection, but now we need to go a step further.

The FCC could sponsor a new round of Technical Workshops involving the telecom network operations experts within the carrier and the SIP/IMS vendor community that would specifically focus on issues in SIP/IMS real-time service interconnection.

The goal of such workshops would be to bring the issues to the forefront, while ultimately encouraging the participants to move forward with **all deliberate speed** in whatever multi-stakeholder forum the participants feel is most appropriate. The Commission need not actively oversee such activities, but only encourage or “nudge” the participants to “make it happen”.

Focusing on the transition of the core network requirements for SIP/IMS Interconnection, the Future on Numbering policy and the transition of legacy support systems is essential for the next step. Other discussions about rich communication, location-based and emergency services should be left to later phases when harmonization will be possible to the all IP core capabilities.

Quality of Service, Network Management and packet prioritization issues should be in scope for those discussions, since it is essential that the existing consumer experience be preserved in any transition.

I believe that E.164 named real-time communications can continue to be a profitable product for service providers as part of a ubiquitous broadband communication package if the underlying technical infrastructure can be aligned with the cost realities in the industry.

Appendix A

47 USC § 251(e)[1]

(e) **Numbering administration**

(1) **Commission authority and jurisdiction**

The Commission shall create or designate one or more impartial entities to administer telecommunications numbering and to make such numbers available on an equitable basis. The Commission shall have exclusive jurisdiction over those portions of the North American Numbering Plan that pertain to the United States. Nothing in this paragraph shall preclude the Commission from delegating to State commissions or other entities all or any portion of such jurisdiction.

(2) **Costs**

The cost of establishing telecommunications numbering administration arrangements and number portability shall be borne by all telecommunications carriers on a competitively neutral basis as determined by the Commission.

NUMBERING AUTHORITY IN CANADA

Section 46.1 of the *Telecommunications Act* grants the CRTC the authority to administer numbering resources in Canada.

46.1 The Commission may, if it determines that to do so would facilitate the interoperation of Canadian telecommunications networks,

(a) administer

(i) databases or information, administrative or operational systems related to the functioning of telecommunications networks, or

(ii) numbering resources used in the functioning of telecommunications networks, including the portion of the North American Numbering Plan resources that relates to Canadian telecommunications networks; and

(b) determine any matter and make any order with respect to the databases, information, administrative or operational systems or numbering resources.