Feature Group IP

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Marlene H. Dortch, Secretary
Federal Communications Commission
445 12th Street SW
Room CY -B402
Washington, D.C. 20554

RE: CC Docket No. 01-92, *In the Matter of the Missoula Intercarrier Compensation Reform Plan; Missoula Plan Phantom Interim Process and Call Detail Records Proposal*; Written *Ex Parte* presenting method to uniquely identify, represent and allow callback to an IP endpoint from the Legacy Public Switched Telephone Network

Dear Ms. Dortch:

We request that this filing be brought to the attention of the Commission and appropriate Staff. UTEX Communications Corp d/b/a Feature Group IP ("Feature Group IP") has previously submitted comments in this proceeding and in particular in the comment cycles related to the so-called Missoula Plan and Interim Process/Call Detail Records Proposal. The purpose of this Written *Ex Parte* is to present for the record a recently devised method to uniquely identify, represent and allow callback to an IP endpoint from the Legacy Public Switched Telephone Network ("PSTN"). This method is responsive to but more reasonable than the interim and permanent Missoula Plan proposals, which would require representation and presentation of SS7-based Calling Party Number ("CPN") in the form of an E.164 address in the SS7 ISDN User Part ("ISUP") Initial Address Message ("IAM") CPN parameter, purportedly to allow identification of the originating endpoint on IP based networks if a session is addressed to a Legacy PSTN endpoint.

While the Missoula Plan portrays itself as a technical solution, the CPN requirement in fact represents a pure political ploy that will single out IP-based services and selectively impose intrastate or interstate switched access charges on them, in advance of more comprehensive changes to the entire, broken intercarrier compensation regime. More important, it will not even solve the Incumbents' perceived "Phantom" problem related to call control information where the originating endpoint is IP-based and does not use an E.164 address. Call control information is useful in that it can support interoperation of features, functions and services that use the information in the CPN parameter (*e.g.*, Caller ID, Call Return), but the CPN requirement is proposed purely for billing, and will not in any way serve or support interoperation of these functionalities between Legacy networks and IP networks, nor will it even provide the identity information the ILECs say they want.

The proposed Missoula CPN requirement purports to resolve the perceived issue by attempting to force modern IP networks to emulate Legacy networks through use of a Legacy address. But this does not in fact present a workable or reasonable technical solution. The Universal Tele-traffic Exchange ("UTEX") specification presents a much better way to answer the demand for information about the identity of the party initiating a call session involving the PSTN at one or more endpoints. It does so by representing IP-based addresses within the Legacy SS7 protocol, rather than forcing IP networks to emulate Legacy addressing. Legacy networks can recognize and act on this information if they take a few simple steps.

There are only two things the Legacy carriers must do. First, they must be interconnected with the UTEX, either directly or indirectly. Second, they just need to look for in the SS7 ISUP Internet Address Parameter described in the specification. If they do these things they will receive the information they insist they must have, and will be able to use that information for whatever purpose they may deem appropriate – subject, of course, to applicable regulatory rules and/or their interconnection agreements with other carriers. More important, from our perspective, is that this method will allow for call-back (Call Return) from the PSTN to IP endpoints that do not have E.164 addresses, which is not presently possible. Further, this method will expand interoperation of CPN-based features, functions and services (including the Privacy Indicator) now available on the PSTN so they can transparently work when one or more endpoints are IP-based.

All the Legacy carriers need to do is interconnect and look at the information in the Internet Address Parameter to establish identity of any party using IP-based services who seeks to initiate a session that involves a PSTN endpoint. That is what they proposed to do with the CPN requirement; the only change is that they will need to look at parameter different than the one they propose to use. The benefits of using this approach rather than mandating CPN – which leads to issues concerning waste of numbering resources, interconnection rights and other rights, duties and obligations of non-carriers and almost by necessity forces the Commission to engage in piecemeal changes to the intercarrier compensation regime – should be patently obvious.

Feature Group IP is pleased to present this information, and we hope it will resolve any perceived need for interim steps and will allow the Commission to turn its consideration back to a global set of solutions that incorporate the initial goals and criteria that were stated in ¶¶ 29-36 of the March 3, 2005 FNPRM in this case.[1]

Sincerely,

W. Scott McCollough
General Counsel
Feature Group IP

[1] Further Notice of Proposed Rulemaking, *In the Matter of Developing a Unified Intercarrier Compensation Regime*, CC Docket 01-92, FCC 05-33, 20 FCC Rcd 4685, ¶¶ 29-36 (rel. Mar. 2005).

UTEX Technical Specification 01.1 (TS-01.1-2007)

Initial Specification for the Theory and Operation of the Universal Tele-traffic EXchange (UTEX)

Abstract

This document describes a proposed standard and enumerates a set of proposed standard documents to interoperate Internet telephony networks at a peer to peer level with each other and with Legacy telephony networks. This specification seeks to address a number of issues that have been raised against the legitimacy and ability of Internet telephony endpoints to interoperate with the Public Switched Telephony Network or Integrated Subscriber Digital Network. All existing specifications for addressing interoperability focus on solving the problem of representing Legacy numbering in Internet telephony protocols. The manifestation of these inter-op problems has been labeled as a “Phantom Traffic Problem” by the incumbent Legacy carriers of the PSTN in North America.

The apparent importance of this issue in the industry belies the fact that Internet telephony protocols in general logically encapsulate Legacy protocols, and as such, attention should instead be paid to the issue of inserting Internet telephony addressing into Legacy telephony networks. The UTEX, as described in this specification, solves the interoperability issue of representing Internet telephony addressing in Legacy protocols through the creation of a Universal Global Title. In essence the UTEX technology has reverse engineered tie-ins to the Legacy Signaling network and the Legacy methods of originating and terminating communications to eliminate the “Phantom Problem.” The only condition is that the Legacy Network (which claims to have the Phantom Problem) must interconnect and exchange traffic with the UTEX. The solution is completely voluntary. If a Legacy carrier has no “Phantom” issues it need not interconnect; if it does have ‘Phantom” issues it can solve its problem by directly or indirectly exchanging traffic with the UTEX and use the information it will receive as a result.

Worldcall Inc. has a patent pending for inventing the Universal Global Title (“UGT”). The UGT is a Uniform Resource Identifier (URI) [RFC 2396] which represents and uniquely identifies a telephony endpoint. UGTs are required to be globally unique within the UTEX primarily for the purposes of routing and reversibility into and from the UTEX and secondarily to reverse engineer and allow the current capabilities of Legacy communications to interoperate with new technology. UGTs are not designed for “Retail” applications; rather, they are used between different service providers and between and among different technologies so as to preserve retail capabilities among and between differing networks.

Worldcall Inc. also has a patent pending in inventing the “IAP” (Internet Address Parameter) which populates the UGT in a Signaling System 7 (“SS7”) field and into the SS7 signaling stream.

1 Introduction

1.1 Addressing and Signaling

The Public Switched Telephony Network (PSTN) in the United States provides exceptionally reliable service to approximately 173 million wireline and 217.4 million mobile telephony subscribers despite the fact that it uses technology that is over thirty years old. As technology has evolved over time, the telecommunications industry has as a matter of practice initiated standardization efforts to integrate new technologies into the PSTN and ensure interoperability of all telecommunications networks. Generally speaking, these standardizing efforts have preceded the widespread deployment of these technologies. As a result, the industry as a whole has operated under consensus on the interpretation and interoperability standards for issues such as signaling, transport, addressing, and jurisdiction.

The rapid proliferation of Internet Protocol (IP) [RFC791] based communications systems in the last twenty years has engendered a number of standardizing initiatives to enable IP-based telephony, among them: SIP [RFC3261], MGCP [RFC3435], and H.323 [H.323]. The degree of involvement and commitment from the traditional telecommunications industry has varied by standard. While there has been effort to standardize (at least informally) the interpretation and interoperability of these new standards with the PSTN, efforts have lagged in the areas of signaling and addressing. This has left the industry in an increasingly and critically uncertain operating environment, particularly when it comes to inter-provider settlements.

One of the key issues involves the nature and interpretation of the identifier used by a service provider to represent the telephony endpoint initiating the call. In North America, network operators typically represent PSTN telephony endpoints via a telephone number, expressed as an E.164 address. These addresses are assigned in the US by a national number administration body, the North America Numbering Plan Administrator (NANPA). In the context of identifying the initiator of a call, the address is often referred to as the Calling Party Number (CPN), although the separate parameter for Automatic Number Identification (ANI) also serves the same purpose. As technology has developed and new types of telephony endpoints have emerged, efforts have been made to ensure that the addressing of these endpoints is compatible, with the predominant NANPA-based numbering scheme (c.f. E.212 for mobile GSM networks). Without exception, the nature of the interoperability in such schemes exclusively addresses the issue of conveying PSTN numbering to Internet telephony protocols.

In contrast, in the evolution of IP communications systems, particularly in the technologies of email (SMTP) and world-wide-web (HTTP), protocol endpoints are formally addressed in a very different way. IP-based communications systems almost uniformly use the Uniform Resource Identifier (URI) syntax [RFC 2396]. This standard specifies the identification of a resource as:

user@domain

where user and domain may or may not have further internal structure. Following this practice, the Internet telephony protocols all have provisions for, if not exclusive use of, URI-based identifiers for representing telephony endpoints.

The use of a URI-based scheme to identify telephony endpoints creates obvious PSTN interoperability issues. A number of standards have been proposed to unambiguously encapsulate NANP-based addresses in IP-telephony derived URIs. Despite these many efforts, a non-standardized practice has developed where the E.164 address is represented as the user part of the URI, while the network element receiving the signaling is represented in the domain part. Obviously, this mapping makes sense only where both URI and NANP addresses are available and assigned for a particular user or domain. However, nothing has yet been successfully proposed to interoperate networks where the telephony endpoint lacks a NANP-based address that can be inserted in the URI.

A number of issues have been raised by some network operators regarding the validity of addresses derived from IP-based telephony. First, the addressing flexibility inherent in the IP-based telephony protocols has created a perception among some network operators that any CPN that is derived from an IP-based telephony protocol is inherently susceptible to misrepresentation.

Moreover, the ubiquity of the NANP-based addressing scheme has created an apparent conceptual misunderstanding in the nature of telephony endpoints, such that the assignment of a NANP-based address is sometimes wrongly considered to be a logically necessary condition for a telephony endpoint to exist.

Finally, the inability of currently deployed SS7-based signaling elements to convey any kind of address other than a telephone number, has led some to a charge that URI-based schemes are illegitimate, or insufficient for telephony. In fact as can be readily seen, URI-based schemes are more general than the NANP scheme: any number represented in NANP can be conveyed into an infinite number of URI-based schemes. Conversely, only a trivial set of URI-based address can be expressed in the NANP-scheme. The inability of currently deployed SS7-based network elements to handle the URI syntax is a significant technical problem. This standard proposes a solution for this exact issue.

1.2 **Settlements**

Telecommunications providers and IP-based service providers peer in different ways, and the two industries have radically different predominant inter-provider charging schemes for the traffic exchanged between networks.

1.2.1 Telecommunications

Telecommunications interconnection and settlements (now more commonly known as "inter-carrier compensation") is regulated by the FCC and states, pursuant to federal and state legislation and administrative rules. Generally speaking carriers are required to directly or indirectly interconnect (establish links that will support calls) with other carriers on request, and the terms must be reasonable. Dominant firms such as the larger incumbent telephone companies must allow other providers to interconnect at any technically feasible point within the ILEC's network, for the most part within each LATA where calls will be originated and/or terminated. Facility charges must be cost-based, and are apportioned between the two carriers based on percent originating use. When traffic is passed from one carrier to the other, the originating carrier is responsible to pay the terminating carrier for the cost of transporting and terminating the call. If the call is deemed to be part of a "telephone toll service" the toll provider is required to compensate the LEC on whose network the call originates and the LEC on whose network the call terminates.

Some calls are deemed to be subject to "access charges" which are generally higher, while others are part of a "reciprocal compensation" regime, with lower cost-based charges. The difference between these two charging levels has lead to significant litigation, regulatory attention and, according to some participants, gaming in the form of masking the true nature of a call in order to avoid assessment of high cost access, in favor of low cost reciprocal compensation. This is the result of the fact that the rules are full of exceptions and conditions, and it is not always clear whether any particular call session that originates or terminates on the PSTN is subject to the higher cost "access" regime or the lower cost "reciprocal compensation" regime.

The rules do allow carriers to mutually waive cost recovery through bill and keep arrangements, but such arrangements are disfavored by incumbent LECs since they believe they would be net losers. Under the current default rules, the rate for PSTN inter-carrier compensation depends on three factors: (1) the type of traffic at issue; (2) the types of carriers involved; and (3) the end points of the communication. For instance, a long-distance call carried by an IXC is subject to a different regime than a local call carried by two LECs. Moreover, CMRS providers and LECs are subject to different inter-carrier compensation rules, and Enhanced Service Provider handled calls are subject to yet another regime. These distinctions create both opportunities for regulatory arbitrage and incentives for inefficient investment and deployment decisions. The FCC has long recognized that its current rules make distinctions based on artificial regulatory classifications that cannot be sustained in today's telecommunications marketplace. However, the efforts of the FCC to move to a unified and rational compensation regime have yielded little progress, more litigation, and more intra-industry debate and contention. The local exchange carriers are currently pressing the FCC to impose specific rules related to the signaling information a carrier must provide to downstream carriers. These companies believe that this signaling information will allow them to better identify the call type and classification and then the upstream carrier who may be responsible for the terminating charges. This standard does not comment on the propriety of the LECs' desire for information or the utility of the information they desire. It does, however, provide a means for IP telephony providers using the UTEX to provide the information the LECs seek.

1.2.2 IP providers and telephony

IP providers use a different regime. The larger providers have private agreements with each other for what is known as “peering” and “transit.” The larger providers typically use a settlement-free approach with their peers, based on an assumption that the traffic burden each peering partner imposes on the other and the value each receives are roughly equal. Smaller providers purchase connectivity and access to the larger provider networks, and thereby obtain the ability to communicate with any address on the larger provider’s network, or on another network that is in turn connected with the larger provider’s network. There are certain locations where smaller players have established exchange points in an attempt to avoid charges from the upstream provider. Where charges are assessed as between peers, for transit or to or between the smaller providers, they are usually based on bandwidth rather than individual “sessions” or packets or communications, or the type of application being applied in any session. Nor are charges assessed based on geographic endpoints. There is no analogue to the “access charge” or “inter-carrier compensation” regimes used for traditional PSTN communications. The general consensus is that the cost of metering, rating, and billing individual sessions far outweigh any benefit or incremental revenues that could be received in a competitive market. The ability to demand and receive above-cost transfer charges – such as are contained in the “access charges” usually demanded by ILECs – arises only where one player has market power. Signaling and addressing are still necessary, of course, but only for routing and identity purposes.

1.2.3 Basis and purpose of standard

This document proposes a solution to these issues and a standard that can fill the current need for ensuring interoperability and interpretation of addressing information. It will support and facilitate settlement-free transactions between participants but also ensure preservation and delivery of the type of signaling information currently demanded by LECs for call sessions that have an end-point on the PSTN. This solution provides (1) a technical proposal for the exchange of addressing in a general context, and (2) for the creation of a set of geographically distributed and settlement-free exchange points. Participation in these exchange points will require adherence to the proposed standard for the transmission of addressing and other signaling information. Adherence to this standard will then support the ability of each provider to fulfill the information demands associated with PSTN carriers’ perceived need to classify and therefore rate (if they so desire) a particular call session based on the location of the Telephony Endpoint (TE), the type of call, the type of user and the type or technology of the user’s provider.

2 Definitions

2.1 *Telephony Endpoint (TE)*

A network appearance that initiates and receives messages and signals related to telephony service. Endpoints may be real or virtual in nature and many may be present in a given platform. A TE is a participant in a Call Session; the user supported by the Responsible Service Provider (RSP). There will be at least two kinds of End Points. (1) The Originating End Point (OEP), which sends the call control set-up message for a call session, or on whose behalf the RSP sends the call control set-up message for a call session; and, (2) the Terminating End Point (TEP), to whom the call session is addressed, perhaps through one or more Service Providers.

2.2 *Internet Endpoint*

A Telephony Endpoint that uses Internet Protocol (IP) based protocols to establish telephony service.

2.3 *Legacy Telephony Endpoint*

A Telephony Endpoint that exclusively uses FXS signaling to connect to the Integrated Subscriber Digital Network (ISDN) or Public Switch Telephony Network (PSTN).

2.4 *Public Switched Telephony Network (PSTN)*

The interconnected set of common carrier wire and radio switched networks, including local exchange carriers, inter-exchange carriers (IXC) and mobile service providers, that use the North American Numbering Plan in connection with the provision of switched services.

2.5 *Universal Global Title (UGT)*

A unique identifier assigned by an RSP to identify an Internet Endpoint. A given UGT has a mandatory representation as a variable-length UTF-8 encoded string as specified by URI [RFC2396]. The structure of the UGT is described in detail in 3.2.

2.6 *Service Provider (SP)*

All call sessions take place between the UTEX member Service Providers. Traffic flows from ingress service provider to egress at either the RSP or a Transit Service Provider (TSP).

2.7 *Responsible Service Provider (RSP)*

The SP which obtains the direct customer relationship with the End-User represented by the UGT is deemed the Responsible Service Provider.

2.8 *Transit Service Provider (TSP)*

In the event that the RSP for a particular UGT does not or cannot participate in the UTEX, other service providers may choose to represent those UGTs to the UTEX. In this capacity such service provider is said to be operating as a Transit Service Provider, and they operate under the Transfer of UGT requirement as described in 3.2.

2.9 *Universal Routing Information Base (URIB)*

The database of all representations made to the UTEX by member SPs. The route selection policies detailed in this and other TSs are used to select specific routes from the URIB. The URIB contains all information that the UTEX uses to make routing decisions. The data contained in the URIB is exported to the Universal Routing Guide (URG) as described in TS 02.01. [TS02.01]

2.10 *Universal Routing Guide (URG)*

All parameters that are required for the operation of the UTEX to route from ingress SP to egress SP. Representations of UGTs made by SPs to the Exchange are expressed in the URG. The URG is made available to participant SPs on both a monthly and real-time basis.

2.11 *UTEX Technical Specification (TS)*

Documents which describe the theory and operation of the UTEX. Current and planned elements in the series are listed in TS00.1. [TS00.01]

2.12 *Session Initiation Protocol (SIP)*

As specified by IETF RFC 3261. [RFC3261]

2.13 *Uniform Resource Identifier (URI)*

A variable length string which identifies a resource according to IETF RFC 2396. [RFC2396]

3 Theory of Operation

3.1 General Description

The UTEX is a geographically distributed and settlement-free exchange point where service providers pass telephony traffic and associated signaling information to other peer service providers. In general terms, call sessions are established when an originating telephony endpoint initiates signaling with an upstream service provider. This service provider will usually but not necessarily be the Responsible Service Provider (RSP) for that endpoint. The RSP is the service provider who possesses the direct customer relationship with the entity operating the respective telephony endpoint. The RSP has special responsibilities at the UTEX which will be described in detail below.

On receipt of origination signaling from a telephony endpoint, the RSP will initiate signaling with the UTEX in order to further the initiation of the session and deliver the call. In signaling the UTEX, the RSP is required to pass the originating party Universal Global Title (UGT) to the UTEX. The UGT is the fundamental concept in the addressing scheme proposed here. The UGT is a URI which represents and uniquely identifies a telephony endpoint. UGTs are required to be globally unique within the UTEX primarily for the purposes of routing and reversibility.

The originating RSP may or may not pass the address of the terminating endpoint as a UGT. In the event that this address is not passed as a UGT, the UTEX will construct a UGT from the information elements passed by the RSP, and will make a best-effort attempt to establish the session. The default behavior of the UTEX will be described below.

Once the UTEX has a terminating UGT, either directly from the originating RSP, or from default operation, the UTEX will perform a lookup in the Universal Routing Information Base (URIB) database to determine the egress point. If the RSP for the terminating endpoint is a participant in the UTEX, then that RSP will have populated the URIB with its routing information, and the UTEX will pass signaling to the terminating RSP. Furthermore, the UTEX will require that all traffic passing directly to the RSP for the terminating UGT be passed in a settlement-free manner.

In the event that the RSP for the terminating endpoint does not participate in the UTEX, other service providers can opt to accept termination signaling for the session. In this capacity the service provider is acting as a Transit Service Provider (TSP). The TSP is required unconditionally to pass the originating UGT to the eventual RSP. This requirement is fundamental, and ensures the satisfactory flow of addressing and other signaling information. If a participating TSP is found not to be passing the originating UGT to the RSP, the UTEX will summarily suspend operations of the offending TSP until proper compliance can be achieved.

A TSP may register UGTs for transit termination under one of two operating regimes. First, a TSP may register UGTs which are identical to those registered by the RSP. In this case the TSP is said to have registered degenerate UGTs, and the TSP is required to terminate from the UTEX under the settlement-free requirement. Second, the TSP may register UTGs which are related to the UGTs represented by the RSP through the user-part only. In this mode of operation, the TSP is not required to provide settlement free termination. These issues are covered in detail in 4.

3.2 Construction of the Universal Global Title

3.2.1 General Syntax

Universal Global Titles are UTF-8 encoded strings and have the following format:

user-part@domain-part

The UTEX will not recognize any internal structure or semantics in either part. However, the URI representation of the UGT must conform to [RFC2396]. Neither the user-part nor the domain-part have any internal structure or semantics: UGTs are compared in a bit-wise manner. The only requirement is that the identifier as a whole represents a unique string in the URIB.

The domain-part specifically does not have an interpretation via the Domain Name System (DNS) [RFC1034] system. However, upon initial registration for operation in the UTEX, a service provider will be permitted to select a limited number of domain-part identifiers which will be reserved for the exclusive use of that service provider.

3.3 Generation of Universal Global Titles for Non-participating Networks

It is anticipated that certain networks will either choose not to participate or be unable to participate in the UTEX. The UTEX will attempt to ensure universal interoperability and backwards-compatibility by automatically generating UGTs with a domain-part appropriate to the RSP for the UGT. These UGTs will then be inserted into the URIB without egress route and will be marked as inactive, until such time as the RSP obtains membership and initiates representation of the UGT to the UTEX.

The UTEX will solicit from the non-participating networks their preference for the domain part they wish to represent. If the non-participating network does not specify a preference, the UTEX will construct a domain-part in the following way:

3.3.1 UGTs Without Represented User-Part

The UTEX will automatically construct UGTs for a large number of service providers based on that service provider's generally accepted DNS domain name.

3.3.2 UGTs Without Represented Domain-Part

If the UGT is derived from the Local Exchange Routing Guide (LERG) [LERG], the domain will be represented as:

UTEX-OCN-XXXX

where XXXX is the Operating Company Number (OCN) of the service provider in question.

If the SP does not have an OCN, then their associated Domain-Part will be represented in the URIB as:

UTEX-NOOCN-XXXX

where XXXX will be unique and numbered sequentially.

3.4 Universal Global Title Responsibility

A fundamental intent of this specification is to provide a mechanism for the disambiguation of the nature of the source of every possible call session passed through the UTEX. To this end, every member of the UTEX must without exception pass the UGT to the RSP, even if that SP is not a member of the UTEX. As mentioned above, TSPs are required unconditionally to pass the originating UGT to the RSP for the terminating UGT.

3.5 Transfer of Universal Global Title on Egress

As described above, the UTEX requires that the originating UGT be unconditionally passed to the appropriate RSP. If the signaling paths involve IP-based telephony protocols, it is anticipated that the transfer will be trivial. However, if Legacy networks are encountered in the signaling path, significant care will be required.

3.5.1 Transfer in SIP Networks

If SIP signaling is used to establish the call session, the network elements passing the originating UGT will place that address in the Request URI [RFC3261] of the SIP Transaction that establishes the call session.

3.5.2 Transfer in SS7 ISUP Networks

The original designers of the ISDN User Part (ISUP) [T1.113.1-1995] envisioned the protocol to have sufficient flexibility to accommodate a wide variety of future technologies. However, ANSI ISUP (ISUP) made a number of stipulations for the extension of the protocol [T1.113.1-1995--3.5]:

> (a) Existing protocol elements, i.e., procedures, messages, parameters and codes, should not be changed unless a protocol error needs to be corrected or it becomes necessary to change the operation of the service or network capability that is being supported by the protocol.
>
> (b) The semantics of a message, a parameter, or of a field within a parameter should not be changed.
>
> (c) Established rules for the formatting and encoding of messages should not be modified.
>
> (d) The addition of parameters to the mandatory part of an existing message should not be allowed. If needed, a new message should be defined containing the desired set of existing and new mandatory parameters.
>
> (e) A parameter may be added to an existing message as long as it is allocated to the optional part of the message.
>
> (f) The addition of new octets to an existing mandatory fixed length parameter should be avoided. If needed, a new optional parameter should be defined containing the desired set of existing and new information fields.
>
> (g) The sequence of fields in an existing variable length parameter should remain unchanged. New fields may be added at the end of the existing sequence of parameter fields. If a change in the sequence of parameter fields is required a new parameter should be defined.
>
> (h) The all zeros code point should be used exclusively to indicate an unallocated (spare) or insignificant value of a parameter field. This avoids an all zeros code, sent by one protocol version as a spare value, to be interpreted as a significant value in another version.

The specification provides a parameter that could in theory be used to convey a URI-based address, the Generic Address Parameter (GAP). In North America, the GAP is the parameter used to implement LNP [GR-2936-CORE]. Unfortunately, the designers of the implementations of LNP on network equipment in widespread use generally had a more limited view of the extensibility of the ISUP protocol than its originators. In general, most LNP-compliant switching equipment will treat the presence of multiple GAP parameters as a protocol violation, even though such messages are not explicitly forbidden by the protocol.

As a consequence, this standard proposes the creation of a new ISUP protocol element the Internet Address Parameter (IAP). The UTEX or the TSP servicing a session that involves SS7 endpoints will pass the UGT to the RSP via an Internet Address Parameter (IAP). Without exception, the UTEX or the TSP servicing a session will pass the IAP as an additional parameter, and will not substitute it for any parameter currently required for interoperability or call establishment under industry standards or FCC regulations. This requirement is critical for reverse interoperability, as switching elements that cannot properly interpret the IAP can choose to ignore it without effecting call processing. However, by ignoring the parameter, they will be discarding all of the informational content that the UTEX provides to satisfy the LECs' stated desire and need for this information.

3.5.2.1 The Internet Address Parameter (IAP)

The Internet Address Parameter is a variable-length optional parameter extension to the ISUP protocol

3.5.2.1.1 Parameter Name Code

Parameters in ISUP messages are identified by Parameter Name Codes defined by 8-bit words. The UTEX will petition relevant standard bodies for formal assignment of a new parameter, but will provisionally use the currently unassigned code point

1 1 0 0 1 0 0 0

This definition is consistent with current usage as it formally follows code points for the Generic Address Parameter, and the Generic Name Parameter.

3.5.2.1.2 Parameter Definition

The parameter will be encoded according to the following scheme:

<table>
<tr><th></th><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td>1</td><td>Odd/ Even</td><td colspan="2">Screening Indicator</td><td colspan="5">Nature of address indicator</td></tr>
<tr><td>2</td><td colspan="8">1st UTF-8 Octet</td></tr>
<tr><td></td><td colspan="8"></td></tr>
<tr><td>n</td><td colspan="8">nth UTF-8 Octet</td></tr>
</table>

The parameters have the following definition:

3.5.2.1.2.1 Odd/Event

The number of octets conveying the UGT has odd or even multiplicity.

3.5.2.1.2.2 Screening Indicator

This standard defines the following code point to describe the screening indicator:

2	1	
0	0	Reserved
0	1	Presentation Allowed
1	0	Presentation Disallowed
1	1	Reserved

3.5.2.1.2.3 Nature of Address

This standard defines the following code points to describe the nature of the address:

5	4	3	2	1	
0	0	0	0	0	Reserved
0	0	0	0	1	Universal Global Title per UTEX TS 01
0	0	0	1	0	
		through			No Interpretation
1	1	1	1	0	
1	1	1	1	1	Reserved

4 Practical Operational Issues

4.1 *Settlement Free Requirement*

If the terminating Service Provider is acting as a TSP, then that SP will likely face significant technical and/or operational challenges in passing the UGT to the RSP. Because of this, when terminating non-degenerate UGTs, the TSP is not required to terminate for the originating service provider under the settlement-free regime. However, when the TSP provides termination for degenerate UGTs, i.e. the TSP has registered a UGT which is identical to the one registered by the RSP, the TSP is operating under the settlement-free regime. Nonetheless, the UTEX does not prohibit the TSP from establishing a for compensation arrangement with the RSP, so degenerate UGTs provide greater availability for the RSP network.

4.2 Deliverables to Exchange members

4.2.1 Universal Routing Guide (URG)

4.2.1.1 Monthly and Quarterly Reports

The UTEX will distribute to UTEX members the URG in electronic format. The syntax and semantics of the documents included in the distribution will be described in TS 02.1 [TS02.01].

4.2.1.2 Real-time Query of URIB

The UTEX will support real-time querying of the URIB via SIP NOTIFY/SUBSCRIBE methods. The semantics of the queries will be described in TS 03.2 [TS03.02].

4.3 Routing

4.3.1 Routing Flow

On ingress, the UTEX will initiate target analysis by extracting the called-party UGT from the ingress protocol. If the called-party UGT cannot be extracted, the UTEX will immediately release the call and signal back to the originating SP a '01 -- Unallocated' for ISDN networks or '404' for SIP networks, or equivalent cause code. Once the UGT is extracted analysis will continue in order to determine the routing target.

The UTEX will reserve a set of UGT domain-parts to represent wildcard routing from the ingress SP. These wildcard domain-parts will be divided into two classes. (1) RSP-only, and, (2) global, and all will be specified in the URG. Wildcard domain-parts will be described in more detail in 4.2.2. If the domain-part of the UGT matches one of these wildcard domains, then the UTEX will strip the domain part. Likewise, if the ingress SP uses a Legacy protocol to signal with the UTEX, then the user-part will be the only extractable information. Unrelated to the routing logic, but central to the operation of the UTEX, it is important to note that if the ingress SP uses a Legacy signaling protocol, then they will be required to include an IAP parameter to represent the calling-party UGT.

In either case, the resultant user-part will be used to query the Legacy lookup engine (LLE). The LLE will be populated automatically by the UTEX with 16-digit prefixes which map to numbers assigned in the LERG and other Legacy routing guides. Representations by TSPs will also be present in the LLE. In general, the lookup will return the UGT user-part for the original called-party UGT mapped to the RSP domain-part, as well as UGTs provided by any TSPs present at the UTEX. If the wildcard part of the original called-party UGT was of the RSP-only class or the user-part was derived from a Legacy protocol, then the RSP UGT will be returned. If the wildcard was from the global class, then the TSP UGT will be returned if no RSP UGT is active in the URIB. If multiple TSPs are present in the LLE, then the UGT will be selected by rules identical to those presented in 4.2.3 for route multiplicity. In any event, the returned UGT will be fully qualified, and the UTEX will use it to translate the UGT from the original called-party UGT in the ingress message. The UTEX will then restart analysis based on the translated called-party UGT.

In general, called-party UGTs represent complete routing keys to the UTEX. As such, once the UTEX analysis engine has a fully-qualified called-party UGT, all that is left to complete routing is to perform a mapping of UGT to route list, which is performed in the URIB. A route list is an ordered collection of egress points. Cases where the multiplicity of the route list is greater than one are treated in 4.3.3.

In the case of termination from an originating SP to a TSP, the domain-part of the called-party UGT will be that of the TSP, and not that of the RSP. If the user-part is represented in the LERG or other Legacy routing guides, there will likely be an inactive route, pre-populated by the UTEX, for that user-part mapped to the domain-part of the RSP. However, in the case of call setup from the originating SP to a TSP, this inactive entry will never be referenced. If both TSP and RSP entries are active in the URIB, it will be to the discretion of the originating SP to decide where to send the call. Because RSPs are required to provide settlement-free interconnect, and TSPs are not, it is most likely that the originating SP will elect to send directly to the RSP. However, the UTEX does not mandate this behavior.

The UTEX will maintain real-time availability of each egress point in each route list. In the event that an egress point becomes unavailable, the next available egress point will be selected. In the event that no destination points are available in the route list, the UTEX will send back '34 – No circuit available' to ISDN-originated calls, '503' to SIP originated calls, or equivalent cause code. Since RSP routes have primary selection as described below, it is likely that the originating SP will then implement route advance logic to select a UGT of a TSP, and resend the call.

4.3.2 Wildcard Domain-Parts

Wildcard domain-parts will belong to one of two classes (1) RSP-only and (2) global. RSP-only wildcard parts will be of the form:

UTEX-RSP-XXX

Where XXX is of the form 000-999. Global wildcard parts will be of the form:

UTEX-GLOBAL-XXX

Where XXX is of the form 000-999.

4.3.3 Route Multiplicity

The URIB will contain two forms of route multiplicity: degenerate and non-degenerate. For degenerate multiplicity, this means that a TSP has registered a UGT identical to that registered by the RSP. For non-degenerate multiplicity, the UGTs registered by the TSP and RSP will have the same user-part, but divergent domain-parts. For non-degenerate UGTs, the LLE provides the logical link between the TSP's and RSP's UGTs.

In the presence of either degenerate or non-degenerate multiplicity in a URIB route list, or in multiple entries in the LLE, selection will take place with the following preferences: RSP over TSP, and geographically local over geographically remote. Furthermore, if route preference cannot be established based on the previous two criteria, the UTEX will pass traffic equally among possible SPs in coldest-first, first in first out (FIFO) routing allocation.

RSP represented routes will always be selected prior to TSP represented routes in the LLE. However, originating SPs may choose to send directly to a representing TSP for non-degenerate UGTs. While the presence of TSP routes will not necessarily be apparent to an originating SP by examining the decisions of the LLE, this information will be available both offline in the URG and in real-time via the URIB query [TS03.02].

4.4 Other Inter-working Considerations

4.4.1 Anonymity

FCC rules require carriers to preserve CPN and present the CPN to downstream carriers. SS7 has a parameter that is used to signify that the calling party seeks to keep their calling number and name (as stored in a relevant Line Information Data Base) private from the called party. If the privacy indicator is flagged, FCC rules require the terminating carrier to suppress delivery of the CPN to the called party, unless the called party falls into specific customer classifications entitled to obtain that information but required to limit use of it. Since the UTEX and TSPs will pass Legacy CPN unchanged to the RSP, the operation of the UTEX will voluntarily conform to existing FCC rules. Additionally, the IAP contains a screening parameter which will have the intent of allowing or disallowing the presentation of the UGT to the terminating endpoint. The proper enforcement of the interpretation of the IAP screening indication will be under the jurisdiction of the UTEX until such time as relevant regulating bodies establish appropriate rules and procedures.

4.4.2 Unsolicited calling (Voice Spam)

In the initial launch of the UTEX, Unsolicited Calling Providers (Voice Spammers) are prohibited from directly exchanging traffic with the UTEX. While there is no legal requirement to take this action (or any prohibition against doing so), the founders of the UTEX believe that public policy regarding appropriate use of new technology significantly lags the ability of the technology; and as such the UTEX will prohibit voice spam until a public policy is set either by its majority members or by the FCC related to allowed participation by Voice Spammers.

5 References

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[RFC3261] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M. and E. Schooler, "SIP:Session Initiation Protocol", RFC 3261, June 2002.

[RFC3435] Andreasen, F. and B. Foster, "Media Gateway Control Protocol (MGCP) Version 1.0", RFC 3435, January 2003.

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[TS00.01] UTEX Technical Specification TS00.01, Description of the UTEX TS Series of Specifications.

[TS02.01] UTEX Technical Specification TS02.01, Description of the Universal Routing Guide.

[TS03.02] UTEX Technical Specification TS03.02, Semantics For Use of SIP SUBSCRIBE/NOTIFY Methods For Real-Time Queries of the Universal Routing Information Base.

[LERG] Telcordia Technologies LERG Routing Guide.

[GR-2936-CORE] Local Number Portability (LNP) Capability Specification: Service Provider Portability, Telcordia GR-2936-CORE, Telcordia.

Federal Communications Commission

The FCC Acknowledges Receipt of Comments From …

UTEX Communications Corp. d/b/a Feature Group IP

…and Thank You for Your Comments

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