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May 20, 2008

By electronic filing:

Marlene Dortch
Secretary
Federal Communications Commission
445 12th Street, SW
Washington, DC 20554

Re: Ex Parte Presentation
CG Docket No. 03-123

Dear Ms. Dortch:

On May 16, 2008, Brian Rosen and I had a telephone discussion with Nicholas Alexander of the Wireline Competition Bureau and Richard Hovey of the Public Safety and Homeland Security Bureau to discuss technical aspects of NeuStar's Telephone numbers for Relay Users ("TRU") plan. Our discussion was consistent with the attached document that discusses the signaling flow used in TRU.

Sincerely,

Richard L. Fruchterman, III
Public Policy and Regulatory Counsel

cc: Nicholas Alexander
Richard Hovey

TRU Signaling Flow

Brian Rosen

May 19, 2008

NeuStar's Telephone Numbers for Relay Users (TRU) proposal is designed to make maximum use of off-the-shelf components created for the Voice over Internet Protocol (VoIP) industry. By utilizing standards based, common industry practice, TRU lowers costs, simplifies implementation, significantly reduces implementation time by relay providers, and offers the potential to have complete functional equivalence for Relay users.

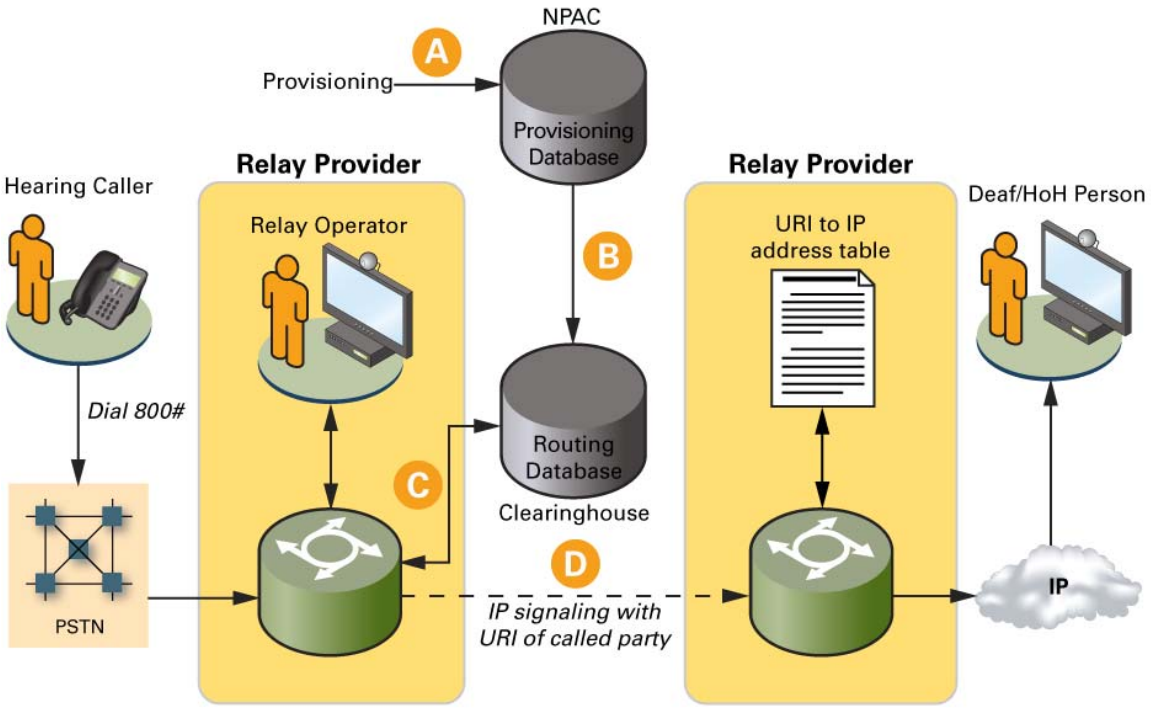

Figure 1: Hearing to deaf call using an alternate relay provider

Figure 1 illustrates a complex signaling flow using two relay providers that occurs when there is a call between a hearing person that chooses to use a relay provider other than the default provider of the deaf person being called. TRU uses the Number Portability Administration Center (NPAC) as the common central database. NPAC is already used by all U.S. telephony carriers of all types to route calls. NPAC is an off line provisioning database: carriers provision NPAC using a variety of methods. The provisioning changes are broadcast to carriers and to specialized clearinghouses by the NPAC, typically using a Local Service Management System (LSMS) to receive NPAC data and in turn, to update their real time routing databases. For traditional carriers, the real time database is a Service Control Point (SCP). The specialized clearinghouses provide real time database query service to smaller carriers, and particularly to VoIP providers. Although the NPAC is certainly capable of updating routing databases of each relay provider, NeuStar suggests that, consistent with the INC report, the relay providers obtain routing data

through a real time query service provided by one of the specialized clearinghouses. NeuStar operates such a clearinghouse and there are competing services available from several other providers.

Provisioning flow **"A"** in Figure 1 uses one of three mechanisms to provision the NPAC.

- The Service Order Administration (SOA) system is an automated, high volume operations support system to supply updates to the NPAC. As TRU requires a new field be added to the NPAC, and field updates require corresponding updates in the SOA system, it may be some time before the SOA system is used to update TRS fields. Ultimately, however, NeuStar believes that SOA systems will be updated, and the normal process to port a number from one service provider to another will be used to provide the TRS URI data along with the other porting information.
- The Low Tech Interface is a web server used by carriers for low volume, manual NPAC updates. If carriers who provide PSTN services to relay providers elect to provision the TRS fields themselves, they would most likely use the LTI until their SOA systems were updated. The LTI is often used by carriers as a permanent provisioning mechanism for low volume processes such as IP TRS support, and their SOA systems may never be updated.
- The Help Desk accepts telephone and email requests for updates, and help desk support agents typically access the LTI to make updates. The underlying telco can use the help desk itself, or it could issue a Letter of Agency (LOA) to the relay provider so it can access the Help Desk directly.

Provisioning flow **"B"** in Figure 1 is the standard NPAC data distribution protocol used today by carriers and clearinghouses. This protocol documentation can be found at: http://www.npac.com/documents/documents.shtml.

A call flow, as illustrated in Figure 1 starts when an interpreter at a relay provider initiates an IP call to a deaf or Hard of Hearing (HoH) person who is served by another relay provider. If SIP signaling were used, as the TRU proposal suggests, this would be a normal INVITE transaction. The Request URI and To: field would typically be a sip URI of the form

sips:2025551212@originating-provider.net;user=phone.

The From: field (and, after passing through the originating relay providers softswitch, the P-Asserted-Identity) would be a similar sip URI with the telephone number of the calling party.

The softswitch, or, more likely, the "gateway Session Border Controller" (gateway SBC) which faces other relay providers, would query the real time database at the clearinghouse using ENUM on flow **"C"**. The typical query string would be 2.1.2.1.5.5.5.2.0.2.1.e164.arpa (although the right most two labels are likely something specific to the clearinghouse). The return would be a NAPTR which yields a sip URI of the form

sips:2025551212@terminating-provider.net;user=phone.

The softswitch/gateway SBC would replace the Request-URI with the URI obtained from the clearinghouse, resolve the "terminating-provider.net" portion using the global DNS and forward the call to that IP address on flow **"D"**. The terminating provider would typically receive the calls on its gateway SBC (and filter IP addresses to limit access to only bona fide relay providers. The gateway SBC would pass the call to its softswitch, which would query its registrar database to obtain the IP address of the destination relay user.

A sample complete message at point **"D"** would be:

```
INVITE sips:2025551212@gateway.terminating-provider.net;user=phone
  SIP/2.0
 Via: SIP/2.0/TLS client.access-provider.com:5061
  ;branch=z9hG4bK74bf9a
 Via: SIP/2.0/TLS incoming.originating-provider.net:5061
  ;branch=d5y7Ho3sC99s
 Max-Forwards: 68
 From: 6175551234;tag=1234567
 To: sip:2025551212@originating-provider.net;user=phone
 Call-ID: 12345601@originating-provider.net
 CSeq: 1 INVITE
 Contact: sips:a998677@originating-provider.net
 P-Asserted-Identity sips:6175551234@originating-
   provider.net;user=phone
 Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
 Supported: replaces
 Content-Type: application/sdp
 Content-Length: ...
    <SDP not shown>
```

Signaling of other cross-provider calls such as VP to VP calls, when observed at point "D" would be the same as the above example.

TRU suggests that SIP based signaling as in the above example be required at point **"D"** between providers. This allows transition to SIP based signaling from the current requirement that end devices implement H.323. This inter-provider SIP based signaling requirement could be considered sufficient, allowing devices to implement any protocol, or it could be augmented by requiring devices to support SIP signaling at a later date. Establishing SIP between providers at the onset of IP-based TRS numbering implementation will facilitate the adoption of SIP without requiring immediate upgrade of devices.