Verizon
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Short Term Public Notice of Network Change Under Rule 51.333(a)

Verizon VoIP Interconnection Service

August 31, 2004

Type of Change:

The initial release of Verizon's VoIP Interconnection Service provides an IP interface for service providers to terminate calls in an IP format without protocol conversion to Time Division Multiplexing (TDM) for Public Switched Telephone Network (PSTN) termination in Verizon service areas. The product will also enable conversion of TDM end user traffic to an IP format for delivery directly to service providers. VoIP Interconnection Service consists of the following components.

Media Gateway Controller / Signaling Gateway (MGC/SG)
The MGC/SG converts an IP signaling stream to SS7 format and the reverses the process by converting SS7 into an IP signaling stream. The protocols used for this interface are Session Initiation Protocol (SIP) and Session Initiation Protocol for Telephones (SIP-T).

Trunk Media Gateway (TMG)
The TMG will convert an IP voice stream into a TDM analog voice stream and also reverse the process by converting an analog TDM voice stream into an IP voice stream.

Transport
IntraLATA IP traffic will traverse Verizon's IP Multi Protocol Label Switching (IP/MPLS) network. VoIP providers will acquire a tariff offered IP/VPN IP Port in each LATA intended for IP traffic transport. InterLATA IP transport will be completed over the customer's network or by a Verizon affiliate as determined by the customer.

Technical References:
The following technical references and subsequent versions shall apply:

MGC/SG - IETF RFCs:

. RFC 2976 - INFO Method

. RFC 3087 - Control of Service Context using SIP Request-URI

. RFC 3261 - SIP Specification

. RFC 3262 - Reliability of Provisional Responses

. RFC 3265 - Event Notification (Subscribe/Notify)

. RFC 3311 - UPDATE Method

. RFC 3326 - Reason Header (used by the History-Info)

. RFC 3420 - Message/sipfrag (carries the status code for the Refer)

. RFC 3515 - Refer (basis for Call Transfer implementation)

. RFC 3204 - MIME media types for ISUP and QSIG Objects

. RFC 3264 - An Offer/Answer Model with the Session Description Protocol (SDP)

. RFC 3372 - Session Initiation Protocol for Telephones (SIP-T) Context and Architectures

. RFC 3263 - Session Initiation Protocol (SIP): Locating SIP Servers

. RFC 3324 - Short Term Requirements for Network Asserted Identity

. RFC 3312 - Integration of Resource Management and Session Initiation Protocol

. RFC 3313 - Private Session Initiation Protocol (SIP) Extensions for Media Authorization

. RFC 3329 - Security Mechanism Agreement for the Session Initiation Protocol

. RFC 3398 - ISDN User Part (ISUP) to Session Initiation Protocol (SIP) Mapping

IP/MPLS - RTP

RFC 3550 - RTP: A Transport Protocol for Real-Time Applications.

To obtain IETF documents, visit the following website:
http://ietf.org/home.html

Date Changes are to Occur:
February 14, 2005

Location Changes are to Occur:
The interface will be phased-in beginning in February and throughout 2005 in locations across the Verizon footprint where IP/VPN service is offered. Consult the Verizon contact listed below for specific LATA availability.

Reasonable Foreseeable Impact of Change:
VoIP Interconnection Service offers wholesale customers a seamless IP interface to Verizon's PSTN network.

Verizon Contact:
For additional information regarding this network change, please contact:

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