Exhibit No. ____ (TAM-8)

BEFORE THE WASHINGTON UTILITIES
AND TRANSPORTATION COMMISSION

WASHINGTON EXCHANGE CARRIER ASSOCIATION, et al. Complainants, v. LOCALDIAL CORPORATION, an Oregon corporation, Respondent.	DOCKET NO. UT-031472

REPLY TESTIMONY OF

TERRENCE A. MARTIN

ON BEHALF OF

THE

WASHINGTON EXCHANGE CARRIER ASSOCIATION

March 29, 2004

Q. WHAT IS THE PURPOSE OF YOUR TESTIMONY?

A. I am replying to certain portions of Mr. Montgomery's testimony.

Q. AT PAGE 20 OF HIS TESTIMONY, MR. MONTGOMERY STATES THAT LOCALDIAL'S GATEWAYS ARE "GENERATING, STORING, RETRIEVING AND CONVERTING INFORMATION." DOES THIS MAKE LOCALDIAL'S SERVICE AN INFORMATION SERVICE?

A. From an engineering perspective, not a legal viewpoint, I would have to say no. While there is no argument that when a LocalDial caller reaches the voice prompt to enter the destination phone number the call is converted from Time Division Multiplexing or TDM, see explanation below, to an Internet Protocol (IP) which is then routed over an IP local area network (LAN) to the destination gateway where it is converted back to TDM. As I explain below, the call starts and ends as a voice call.

Q. ARE THERE DIFFERENCES IN HOW A VOICE CALL IS HANDLED BY TRADITIONAL METHODS AND THE METHOD USED BY LOCALDIAL?

A. There are differences in form, but not end result. The following explains how a call is processed over the Public Switched Telephone Network (PSTN) using traditional means:

1. In a PSTN network, the phone is picked up and a number is dialed. The local exchange company (LEC) switch finds the best route to forward that call, though either:
    a. Another local connection on the switch on the PSTN, or
    b. A dedicated connection to an interexchange carriers (IXC) point of presence (POP) on the PSTN, or
    c. A tandem switch in the PSTN.
2. Once the call route is determined, the voice signal, which is still characterized as an analog signal up to this point, is then converted into a digital format using a technique called Pulse Code Modulation (PCM).[1] How the conversion works is the voice message is sampled 8000 times per second. The sampled information is then converted into an 8 byte digital word.[2]
3. This converted information is put into a TDM technology frame, such as T1 frames or ISDN Primary Rate Interface Frames, to be sent to another location for further routing which can be either a business location or it is sent to a tandem switch on the PSTN network.
4. When the call is received at the final switch location, it is converted back to an analog signal and sent to the customer's phone.

The following explains how the same call is processed by LocalDial:

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[1] Harry Nyquist of Bell Labs developed the Pulse Code Modulation (PCM) algorithm in 1928 to translate analog voice signals into digital equivalents. The following web site explains PCM http://www.cisco.com/warp/public/788/signalling/waveform_coding.html.
[2] PCM standard is the ITU-T G.711.

1. The call starts the same way as in the PSTN scenario, by someone who subscribes to the LocalDial service, picking up the phone and calling the LocalDial access number.
2. The LocalDial Server answers the call and authenticates the user from the ANI supplied from the LEC switch. Once authenticated, then the user will dial the destination number.
3. The call is now converted from an analog signal into a digital format using the same technique that was used to convert the call in the PSTN scenario. Using the standard PCM encoding method, the voice message is sampled 8000 times per second and each sample is converted into an 8 byte words and then put into packets and sent across LocalDial's LAN or, in the case of traffic from Clark and Cowlitz Counties, the LocalDial Wide Area Network. With LocalDial's equipment, the designer has the option to compress several of these 8 byte words into one packet, which allows more information to be sent using less bandwidth.[3] The compression process will be discussed in more detail later.
4. At the same time the packets are made, the telephone number of the caller is converted into a source Internet Protocol address and the number that the caller has dialed is converted into a destination Internet Protocol address.
5. Now the packets are identified, this allows the LAN to route the packet to the destination gateway. Once it is received at the destination gateway,

[3] ITU-T G.726, G.729, G.728, G.723 and G.7.11.

the information is converted back to analog and the call is completed as a voice call over the PSTN.

This clearly demonstrates that both the TDM and IP conversion of an analog voice message to digital message is done the same way. Both methods even use the same protocol to convert the information.

Q. IS THERE A TECHNICAL DIFFERENCE BETWEEN HOW VoIP GATEWAYS AND TDM SYSTEMS HANDLE *STORING AND RETRIEVING* OF INFORMATION?

A. Technically both systems handle these mechanisms the same way. To demonstrate how this works using T1 or PRI TDM circuits, a discussion on how TDM systems handle traffic must occur. T1 and PRI TDM system systems support 24 channels, each channel with a rate of 64Kbps of information that is transferred to the other end. Externally, someone assigns one of the 24 multiplexer logical channels to this service. When the first PCM encoded sample byte is completed, it is then put into the channel and the stream of 24 filled channels is sent across the network. The next sample goes into the next stream of filled channels and it is sent to the other side. After doing so, it takes the next byte and so on.

This implies that the total output stream rate of the multiplexer should be at least 24*64Kbs or 1544 Kbps. This means the TDM multiplexer took the 1/8000 second which is needed for transferring a single byte of a single channel, and divided it between the 24 channels by increasing the rate so that each byte of a channel will take 1/(8000 * 32) per sec or 15 Milliseconds to send.[4] The TDM equipment must be able to accept, packetize and send the continuous stream of call information every 15 milliseconds. Depending on how busy the multiplexer is this may require the device to store/forward and retrieve information from the equipment buffers.

LocalDial's gateways use the same process of storing/forwarding and retrieving of data before it is put into a packet and sent to its destination. This is how it works - the heart of the LocalDial system is the Ethernet frame, which is used to deliver data between computers or gateways. The frame consists of a set of bits organized into several fields. These fields include address fields, a variable size data field that carries from 46 to 1,500 bytes of data, and an error checking field that checks the integrity of the bits in the frame to make sure that the frame has arrived intact.

Each voice call sets up a session through the VoIP gateway, the same way the TDM multiplexer sets up a channel in a signal stream. The information from each session is put into the next available packet and is sent over the LAN which routes the packet to the destination device. The information is removed from the packet and then the information is converted to TDM and is delivered to the end user.

[4] http://www.hardware-guru.com/SystemDesign/E1_T1_Tutorial.htm

The total frame for one VoIP PCM encoded packet is 40 bytes but the minimum Ethernet packet size is 46 bytes. The voice encoded samples are sent every 1(8000*8 [PCM words] + 38 [Ethernet frame overhead]) seconds or 27 milliseconds. As these sessions are established through the gateway, this equipment must be able to accept and packetize multiple streams of voice information every 27 milliseconds. Depending on the configuration of the gateway, some voice session may be stored briefly in a buffer before it is packetized ad sent out on the Internet where the Internet is used for transport, which does not appear to happen very often with LocalDial. Care must be taken in the design of theses gateways to ensure that the time the voice data spends in the buffer is set to a small amount of time or it will affect the signal quality. To conclude, both the PSTN and VoIP technologies go through the same process and even the same encoding methods of "*converting, storing, forwarding and retrieving*" the call information from its inception as an analog message signal to its conversion to a digital message, then being transferred through the equipment over the network, back through the equipment and then converted back to analog signal to be delivered. It must also be added that neither system adds any information to the voice information or change the characteristics of voice conversation. Then, by definition, these systems should not be considered Information Services since there is no net change in protocol to the end user. If LocalDial's service is an information service, then services using TDM would also be information services.

Q. Mr. Montgomery states that the LocalDial network is an overlay of the existing telecommunication infrastructure. Do you agree with that statement?[5]

A. No, I do not. The design of LocalDial's network, as it relates to this case, defines two collocation facilities, one in Seattle and one Portland. The Seattle facility has VoIP gateways and Atlas PRI concentration devices to support the termination of all the calls to and from areas within the state of Washington, except the Vancouver area which is routed to the Portland equipment. The Portland collocation facility has VoIP gateways and Atlas PRI concentration devices to support the termination of all the calls to and from areas within the state of Oregon, and the routing of calls from Clark and Cowlitz Counties in Washington to LocalDial's Seattle equipment.

It is my opinion the IP traffic is converted to VoIP ONLY within the collocation space (i.e., the LAN) and between collocation facilities (i.e., the WAN) and it is not an overlay of telecommunication services. The access and transport portion, which is the majority of LocalDial telecommunication services is not a part of the VoIP LAN or WAN. Instead, it consists of traditional elements of the DSTN.

[5] ISPA - Expert testimony from William Page Montgomery_files on page 22 lines 15-19

Q. Does most of LocalDial's traffic using the public Internet?

A. No. It is my opinion, based on the design of LocalDial's network, that most of the traffic stays within the LAN within the collocation facility. The only traffic that will use the public internet is the traffic that will be sent through the other collocation facility over the WAN, which recently converted to using Internet transport between facilities.

Q. When VoIP packets are mixed in with other types of data, can it be identified?

A. Yes it can. Packets are identified by a uniquely port number which is defined by the application [the Session Initiated Protocol VoIP protocol uses port 5061]. All applications are identified with a unique port ID that is associated with the destination IP address. These unique identifiers are not made up. Equipment manufacturers and application designers register the port identification at the Internet Assigned Number Authority (IANA)[6]. For example the VoIP Session Initiated Protocol (SIP) uses a registered port ID as 5061. This allows network devices on the network to identify packets and be able to classify them as well as prioritize them for delivery.

Q. Why is it important to identify specific application packets, like VoIP?

A. It is important to have this capability because it allows the network designer to program routers to allow certain packets to be routed first before any others. This

[6] http://www.iana.org.

ensures the delivery of critical information when it is intermixed with several other types of packets or during times of network congestion.

Q. What is the difference between LocalDial's service and the Free World Dialup's end-to-end IP communications offered by pulver.com which was deemed by the FCC to be an information service?[7]

A. There is a major difference between these two products. The pulver.com solution does not require the customer to dial a local access number to access the service. The pulver.com solution requires the customer to have a broadband Internet network connection at their house. When the service is purchased, an ATA device will be placed at the home. One side of the ATA is connected to the Internet side of the broadband connection; the other side of the ATA is connected to an analog phone. The purpose of the ATA device is to convert the analog call message to digital IP message and route the call over the Internet to another pulver.com customer on the Internet.

When the pulver.com service offering is compared to the LocalDial service, the pulver.com service communicates peer to peer over the Internet with minimal intervention of the VoIP gateway. The PSTN is not used. The LocalDial service requires the caller to access a VoIP gateway over the existing LEC infrastructure (PSTN). The Internet is only used to communicate between gateways for a

[7] A copy of the order can be found at http://pulver.com/reports/pulver-decision.pdf.

limited portion of the traffic. All traffic originates and terminates on the PSTN.

Q. DOES THAT COMPLETE YOUR TESTIMONY AT THIS TIME?

A. Yes.