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Data Throughput Analysis on IS-2000 Wireless Networks

October 8th, 2003

presented by:

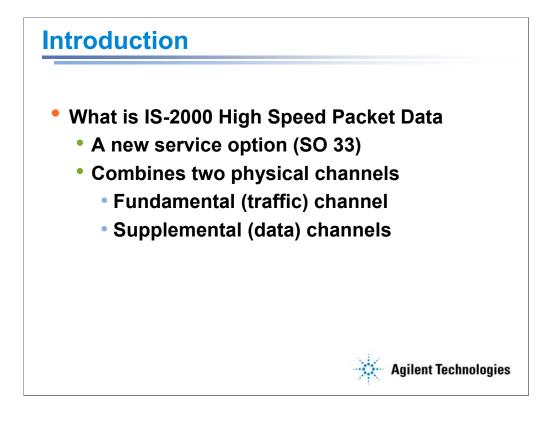
Paul Dohrman

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Agenda

- Introduction
- What is packet switching?
- Network models
- Upper layer (end-to-end) protocols
- Lower layer protocols
- Analyzing data throughput



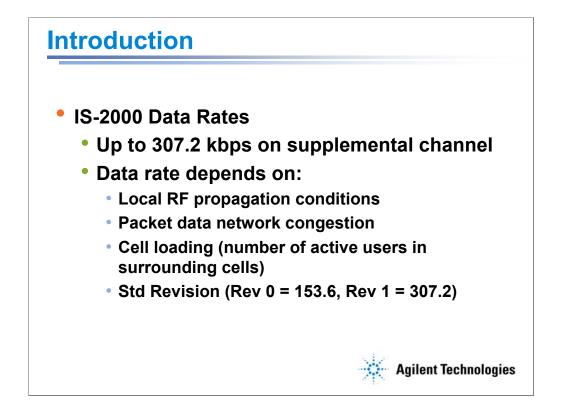


Service options specify how traffic bits are processed by the wireless device and the base station. SO 33 configures the baseband processing software to handle high speed packet data.

SO 33 is available with radio configurations 3 and above. With SO 33, forward and reverse supplemental channels can be added to the fundamental (traffic) channel data rates to increase bandwidth.

The supplemental channels are requested by the mobile station when high speed operation is needed. The base station assigns the supplemental channel and controls the duration of time that the supplemental channel is granted.

The test set is limited to one supplemental channel.



As higher data rates are granted, less Walsh code space is available for other traffic causing cell loading and reducing network capacity.

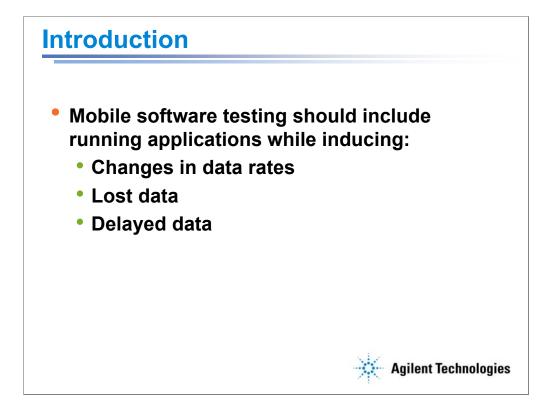
RF propagation conditions can disrupt the flow of data traffic by reducing the signal to noise (Eb/Nt) to a level that makes it impossible to decode data.

Network congestion can also reduce data flow, causing data to be transmitted multiple times before the receiving end acknowledges it.



Interactive and multimedia applications are appearing more often on wireless devices as data rates go up.

Java (J2ME) and Brew are a couple of popular programming languages approved by cell phone manufacturers. Wireless Application Protocol (WAP) provides standard specifications for displaying graphics on a small screen.

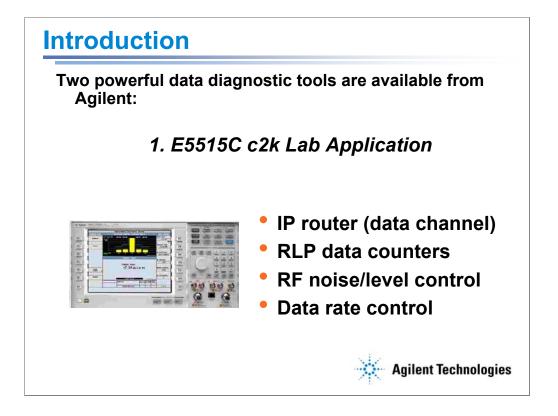


Software is often written by small companies with inadequate equipment to test the effects of variations in RF propagation conditions. In some cases, qualification labs are available to test software prior to releasing it to service providers for integration.

Effects of data rate variations include unacceptably long file downloads, graphics images that take too long to display, and streaming video images freezing.

Delayed data causes the mobile station to buffer data while waiting for the late packets to arrive.

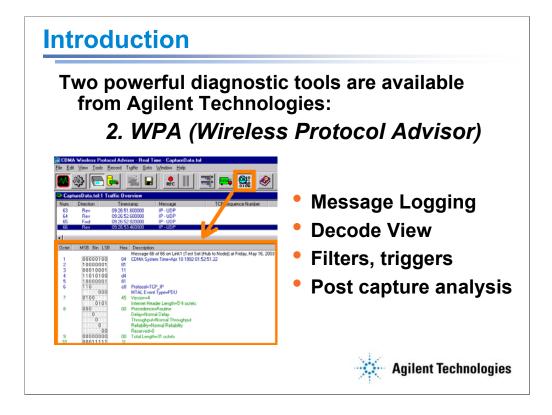
Delayed data (due to retransmissions) can cause the mobile station's application to perform at a level that is unacceptable to the end user.



The C2K Lab Application is the "network on a bench". It is useful when the performance of application software needs to be tested in a controlled environment.

IP data can be routed through an Ethernet port to/from a wireless device through the RF port. Data transfer is monitored by RLP (Radio Link Protocol) counters, and the data rate can be changed while on a call.

RF power can be adjusted to simulate poor channel conditions and an AWGN (Additive White Gaussian Noise) noise source can be added to the signal to simulate signal interference.



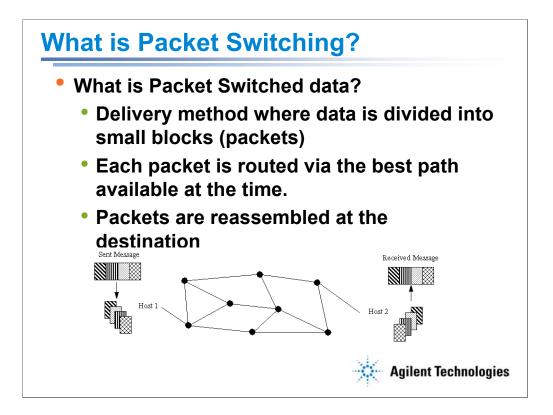
The WPA can monitor messages at various tap points in the protocol stacks. Unwanted messages can be filtered out and triggers can limit logging to time periods beginning just prior to or at a particular protocol event.

Data can be "unpacked" by opening the Decode View. Every byte is listed and labeled.

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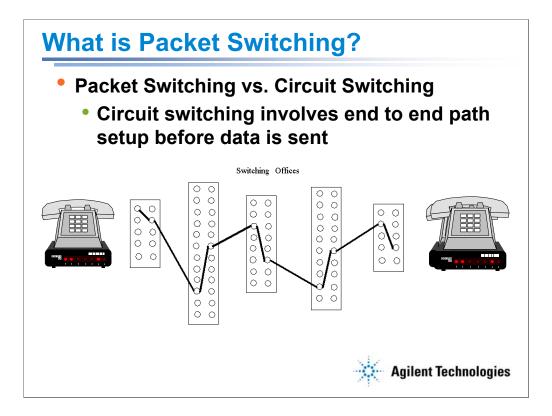




A quick overview of packet switched data will be helpful in understanding how lost/delayed packets affect throughput.

Packet data usually refers to the network layer peer to peer exchange of packets of data. Packet size is limited (typically not much larger than 1 kB) and each packet is sent with enough control information for the network to route each packet individually to the receiver on the other end.

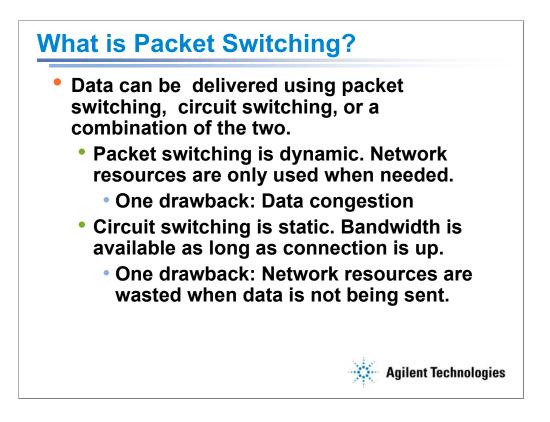
Protocols are designed to receive packets in a different order than they were sent and to re-assemble them in the correct order.



Circuit switching takes time to set up, sometimes on the order of 10 seconds.

ISDN is a good example of circuit switching, although ISDN data can be routed to packet switched connections.

Once a call is made voice and data can be transmitted on the same circuit. The medium over which the data flows is conceptually referred to as a "digital bit pipe"

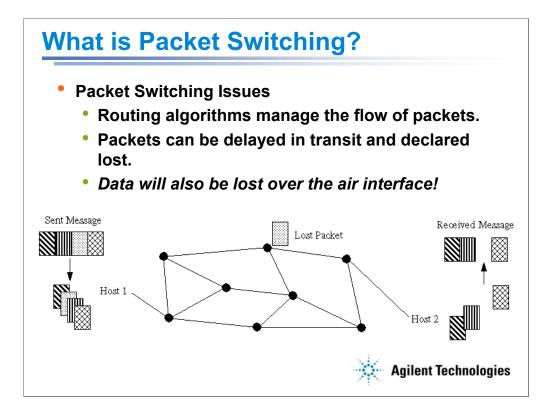


For bursts of data, packet switching makes more efficient use of network bandwidth than the alternative, circuit switching.

Circuit switching requires a constant connection (data path) which takes time to set up and wastes bandwidth during quiet periods. Once it is established, however, there is no problem with congestion or data getting out of sequence.

If data is being sent by a mobile station over the wireless medium, a dedicated physical connection is established, but it is usually relaying packet data bound for the packet internet.

The wireless medium is an inherently unreliable path and data loss is likely to occur.



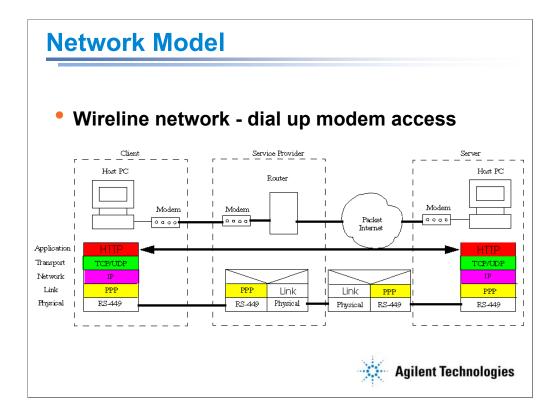
Complex routing algorithms are used to make decisions about the best path to send packets on. If the algorithms work, packet data travels on the most efficient, least congested route. If not, data is slowed dramatically. This can occur when too many packets are routed to one particular node, causing network congestion. Packets can be delayed so long that they are declared lost.

In a wireless application, data loss is mostly due to RF propagation conditions. Network problems such as congestion only add to the problem of data delivery.

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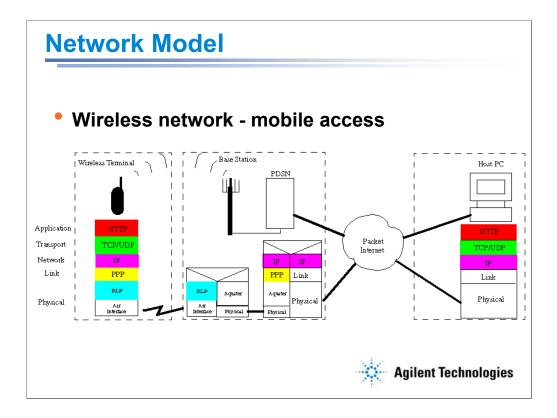




This graphic illustrates an "end to end" packet data communication network.

At one end is a typical PC (not on a LAN) accessing the Internet via a dial up modem connection to a local ISP (Internet Service Provider). All data transfer is over "copper" (no wireless links).

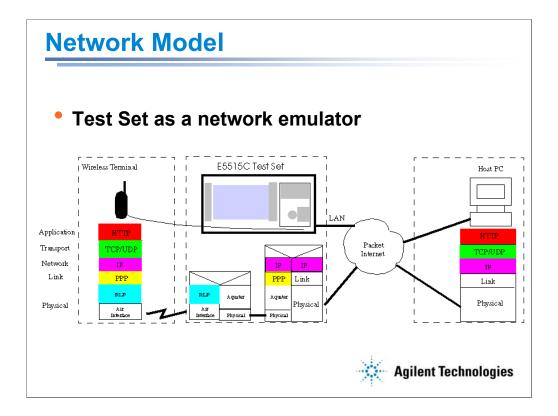
The color coded blocks highlight the protocol layers with peer to peer communication that is particularly relevant to each network entity.



In contrast with the previous slide, this model introduces the air interface, in place of the modem-modem link in the wireline version.

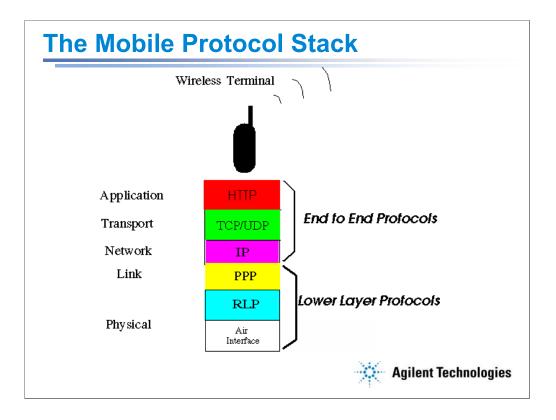
The physical layer in this network model is referred to as the relay layer to the PDSN. Central to the relay layer is the RLP (Radio Link Protocol), the protocol that is used to frame the data for streaming onto the CDMA physical channels. The combination of RLP and the air interface can be thought of as analogous to the modems in the prior slide.

The air interface represents the weak link in the chain of interconnections necessary to transport data from sender to receiver. This is where the test set can simulate real world network problems in a bench top environment.



The test set performs the relevant functions of the PDSN, base station, mobile station controller, and PCF (Packet Control Function). IP datagrams are routed through the ethernet connection and through the RF interface to the mobile station.

The test set also has an internal http server. There is a home page and the capability to serve screen images.



The network model for the SO33 protocol stack in the mobile station can be viewed in two groups.

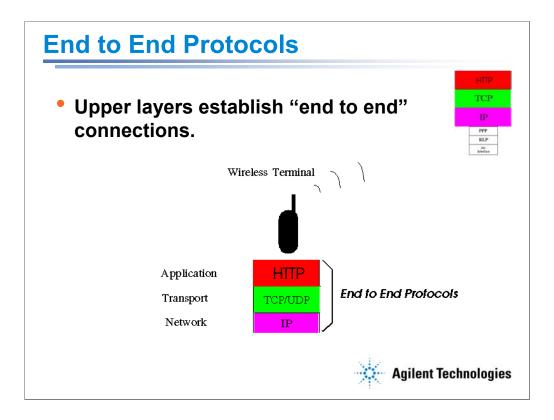
The upper layers must include control information in messages that support the routing of data to and from its final destination.

The lower layers are only concerned with establishing a "single hop" communication channel.

Agenda

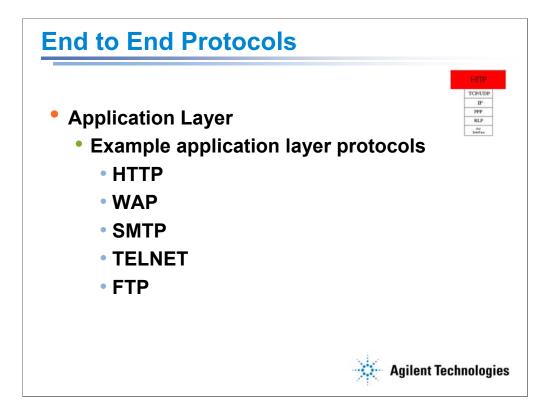
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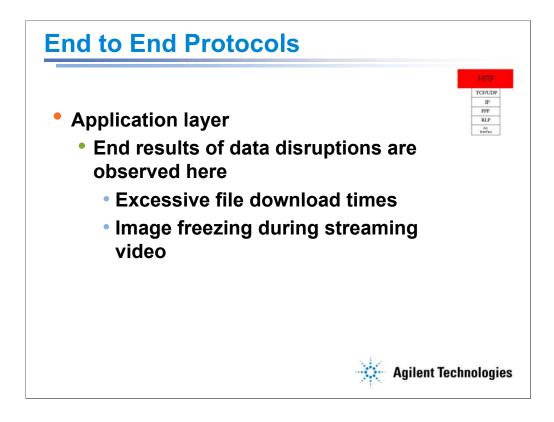


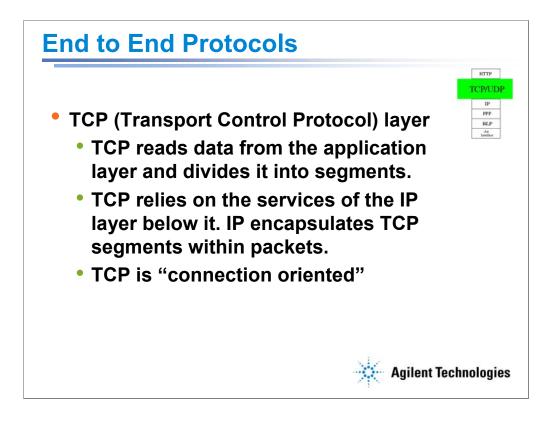


For one application to communicate with another, the services of all layers below it are required.

The application layer interfaces directly with TCP/UDP through protocol ports. TCP/UDP passes data to the IP network layer for packetization and routing to its destination.

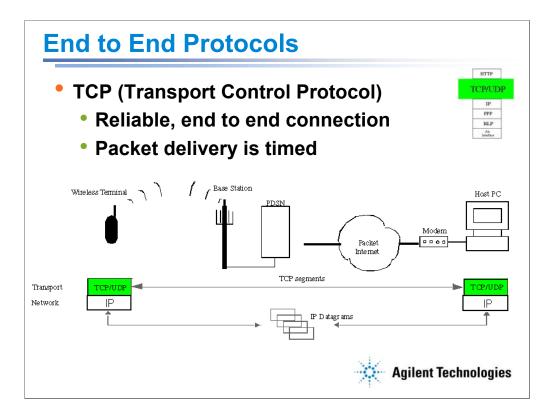






TCP is a connection oriented protocol. By connection oriented, it means that data that is declared lost gets retransmitted. There is data acknowledgement and flow control. This attribute relieves the upper (application) layer of the complexities associated with fixing problems that come up as a result of unreliable channel characteristics.

In the TCP layer, each segment is sent with header information. In the header is a hexadecimal number that identifies the first byte of information. In the next segment, this number is incremented by the number of bytes sent in the last segment so that every byte is essentially numbered. Also sent in each segment is a "piggybacked" acknowledge number. The acknowledge number lets the other end know the sequence number of the next byte it is expecting.

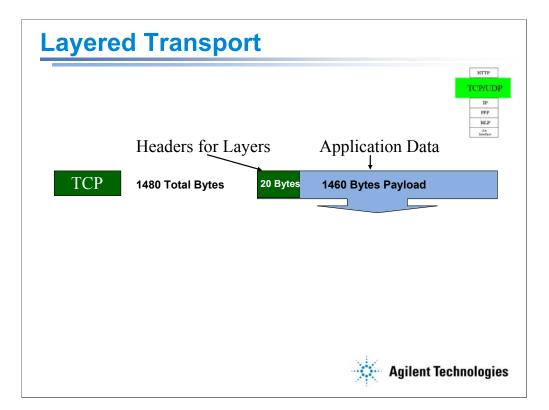


TCP is a full-duplex, connection oriented protocol.

Data from an application is divided into segments, assigned a sequence number, and sent to a receiver host, encapsulated in IP datagrams.

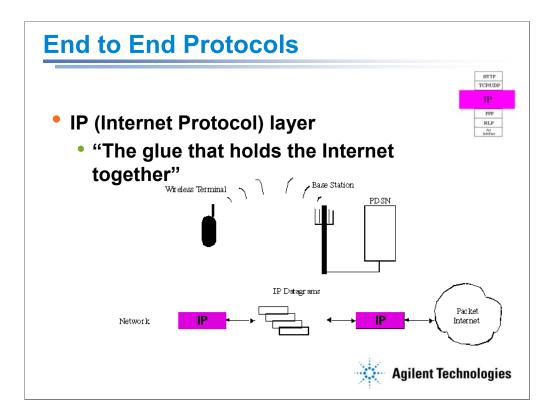
If a segment is not acknowledged from the receiver by a predetermined length of time a timeout occurs. The TCP sender assumes there is congestion and waits a required amount of time (determined by a congestion window) before attempting to send the next segment.

With each segment sent, the TCP sender indicates how much data it can currently accept (flow control) when it receives its next segment. If this receive window goes to 0, data transfer is suspended until the TCP sender determines the receiver once again is ready to receive.



The TCP segment consists of a 20 byte header and 1460 bytes of application layer "payload".

TCP does not care about frame boundaries established by the layer above. It simply divides the application layer data up into the size of blocks that are most efficient for transport and provides reliable, connection oriented service.



The IP layer packetizes datagrams sent from the upper layers.

It provides a "best effort" way to transport datagrams from the TCP/UDP layer across networks. Part of this task is figuring out the path with least congestion.

Packet ordering and flow control are not performed by IP. It is connectionless.

The headers added by IP provide instructions about how to handle the packets:

*What is its source address

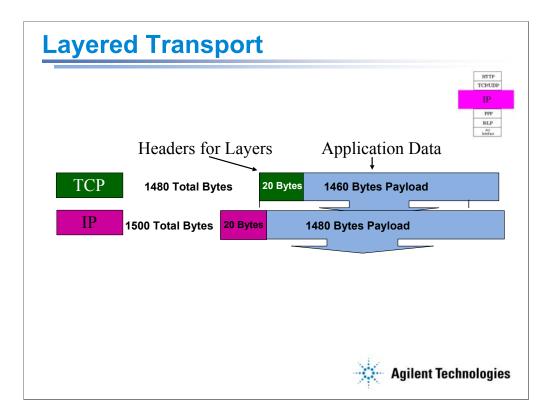
*What is its destination address

*Which piece of a datagram is it?

*Is speed more important than reliablility?

*Is it TCP or UDP?

Mobile IP adds the ability to manage hosts that move from one location to another. Home agents keep track of hosts whose home is in their area, but who are currently visiting another area. Foreign agents keep track of hosts who are from another area. When a message arrives for the travelling host, the home agent forwards it to the foreign agent and the foreign agent delivers it to the host.

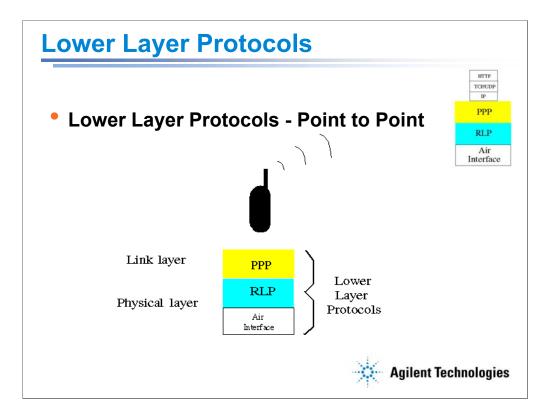


IP adds 20 bytes of header information to form an IP datagram.

Agenda

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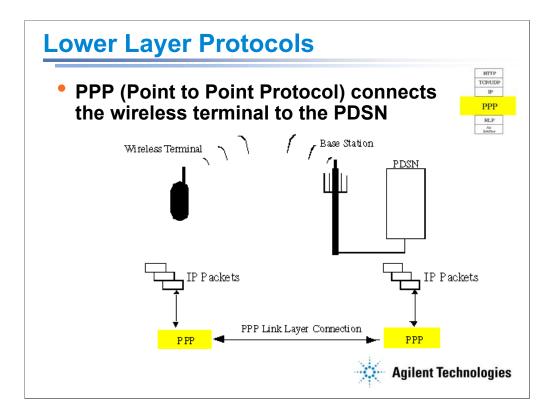




The lower layer protocols support the exchange of data over a single hop. In the case of the wireless terminal the single hop is to/from the PDSN.

The RLP layer is similar to TCP in many ways. It divides PPP frames into smaller RLP frames and streams the data bytes from sender to receiver while providing error control within the limits of its ability to buffer data.

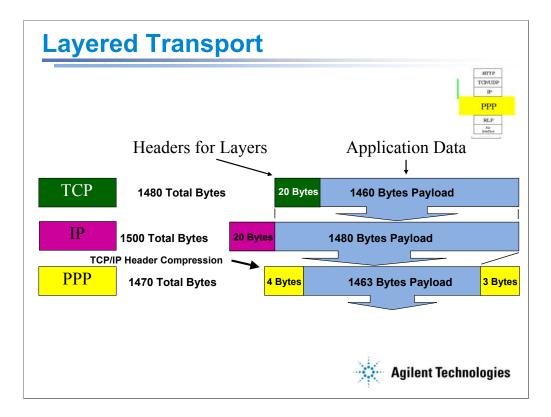
The air interface (IS-2000) is the lowest level protocol and specifies the method that binary data gets from one radio to another.



PPP establishes a connection between the wireless device and the PDSN and sends IP packets over the link layer connection. PPP is the same protocol that dial-up modems use to connect to the Internet service provider's router, and it is PPP that handles IP address negotiation for Internet service providers.

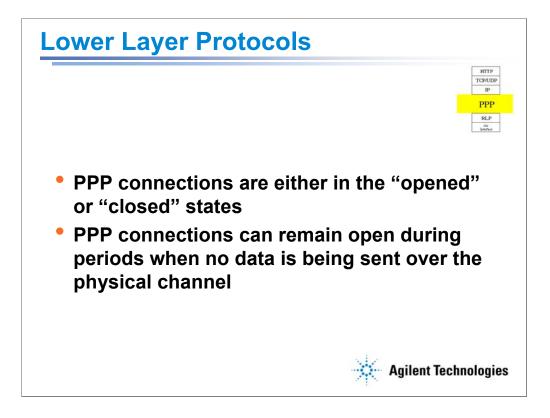
Point to point means that link layer peers communicate over a channel that directly connects the sender and receiver. There is no endpoint addressing or routing as with the IP network layer.

Also, the data link assumes that data is delivered in the same order it is sent unlike end-to-end packet data protocol layers like TCP, which expects segments to get out of order in transit.



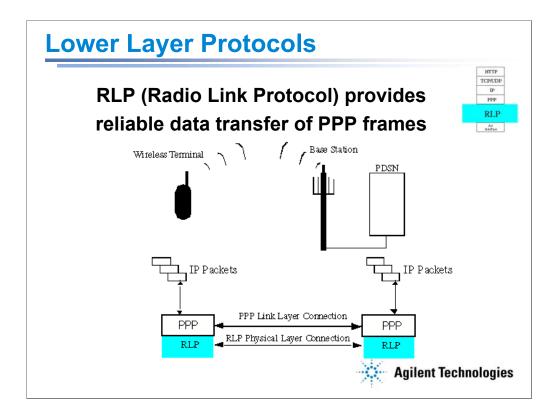
PPP accepts IP packets (IP datagrams are encapsulated within the packets), and maps them to PPP frames.

PPP can apply header compression. In this example, Van Jacobsen compression reduced 40 Bytes of TCP/IP header into 4 Bytes.



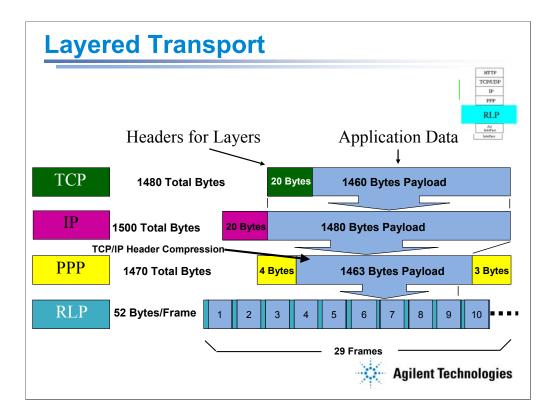
When you are using a phone as a modem, the phone display may show "PPP" after the PC has requested a network connection.

Timers may (optionally) determine when data transfer has stopped and put the connection into a dormant state. The PPP connection remains opened during this time but the physical channel resources are released to handle other forms of traffic.



RLP divides PPP frames into RLP frames.

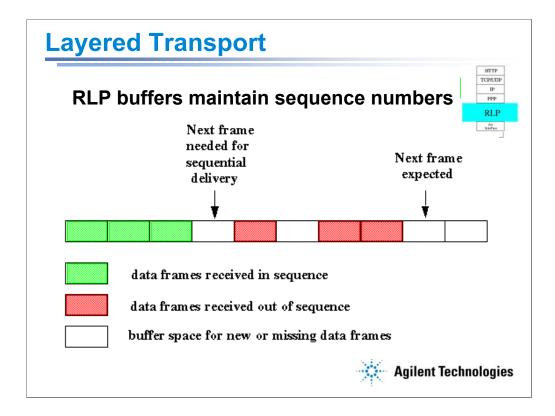
RLP peers synchronize, then exchange control and data frames.



RLP further divides the data sent down from PPP into a frame size suitable for transport over the air interface.

It is expected that RLP frames will be lost over the air interface. In fact, the wireless device's signal to noise (Eb/Nt) is controlled to a level where the average FER (frame error rate) is often at a level that leaves room for improvement in error rate during periods of exceptionally good RF reception. This makes more efficient use of cell capacity by lowering the average noise interference.

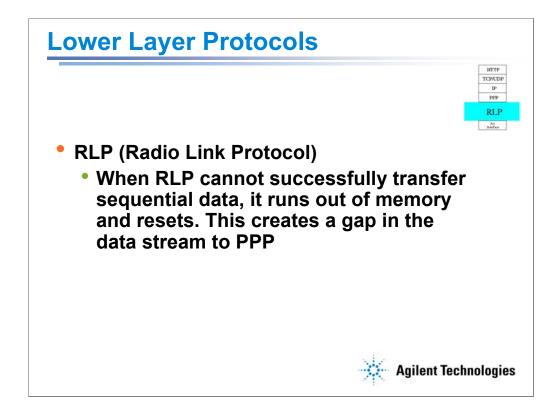
RLP does not care about frame boundaries established by the layer above. RLP simply divides the data up into the size of blocks that is most efficient to stream data from sender to receiver.



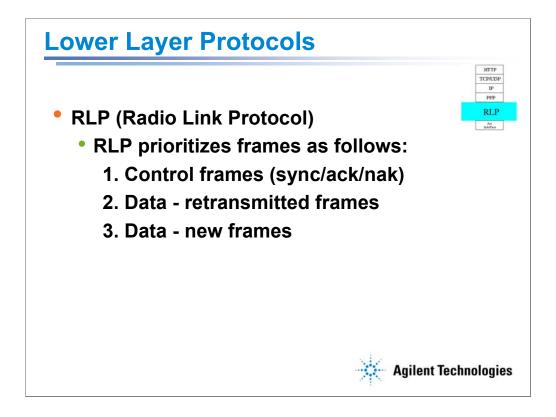
RLP senders and receivers buffer frames of RLP data.

If an RLP frame is received out of sequence, new frames are still received but the receiver lets the sender know that a frame did not arrive in sequence (through a NAK, Negative AcKnowledge).

If the missing frame shows up before storage space runs out, the receiver can put it in the correct sequence and deliver sequential data to PPP. Delivering data to PPP restores buffer space.

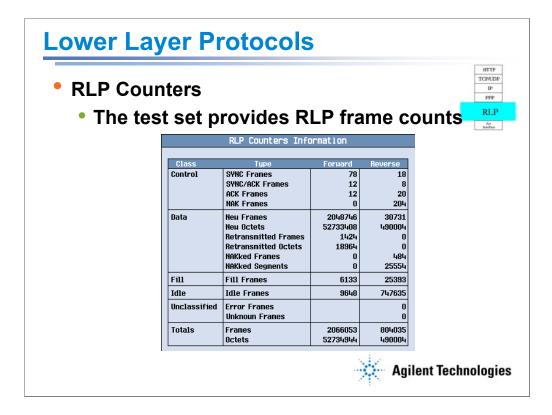


If RLP runs out of buffer storage, it performs a reset and flushes memory. All frames not delivered to PPP at this point are lost and it is beyond RLP's ability to recover them.



The highest priority frames are the control frames. They are used to synchronize communication and let the sender know when RLP frames need to be retransmitted.

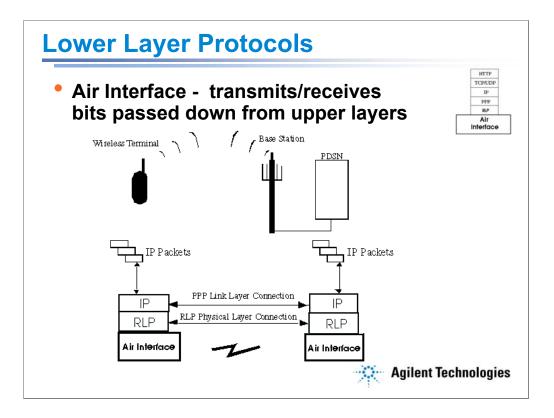
Retransmitted frames have a higher priority than new frames



During periods where there is no data passed down from PPP, but the physical air interface is still connected, RLP transmits idle frames.

When data inactivity causes the data connection to go dormant, the physical air interface is disconnected (idle) and the RLP entities stop sending idle frames. The test set will display Idle on the Call Setup Screen.

Lowe	r Layer Prot		s										
 RLP Summary The test set summarizes IP and RLP frame counts 													
		Counte	rs										
		IP	IP	RLP	RLP								
	-	ackets	Bytes	Frames	Octets								
	Foruard Data (NS RX):	35940 5	2976224	2066053 5	52734944								
	Reverse Data (NS TX):	28488 (1147936	804035	490004								
				27									
	Agilent Technologies												



RLP sends packets down to the air interface where interleaving, coding and modulation is applied to the bits. The physical channel is established as needed, and may include one or more supplemental channels to handle the higher data rates.

The test set sends forward channel data on the fundamental or the fundamental + supplemental channels.

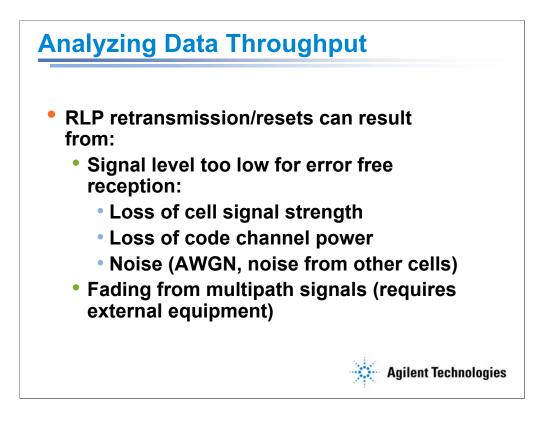
The test set provides controls for varying data rate, RF level, and interference level to simulate the mobile's environment while on a packet data call.

Error detection and correction at this layer is performed by turbo/convolutional coders.

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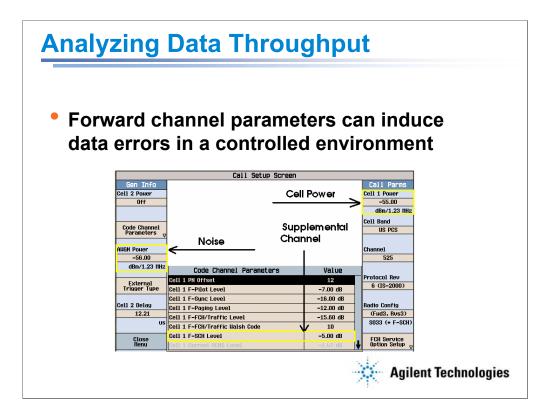


The coding gain that results from convolutional or turbo encoding at the physical layer provides the first line of defense against data loss due to RF degradation. If the coding gain fails to correct errors, the task of data recovery moves up the protocol stack to RLP.

RLP is designed to provide reliable data delivery of PPP frames. The test set can simulate RF degradation and monitor RLP activity.

The test set provides level control of the cell power and level control of a noise generator.

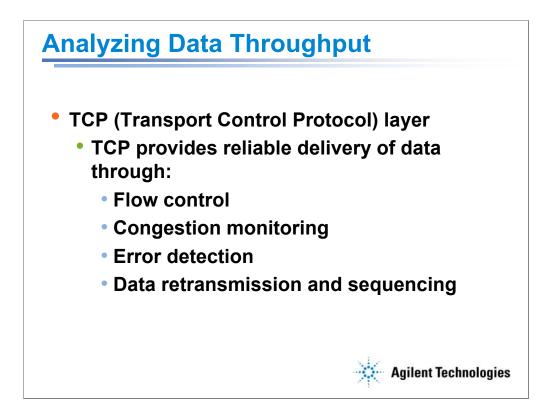
The supplemental channel is also directly settable to a level relative to cell power. Lowering the power level on the forward supplemental channel will eventually interrupt data flow, especially at the higher data rates.



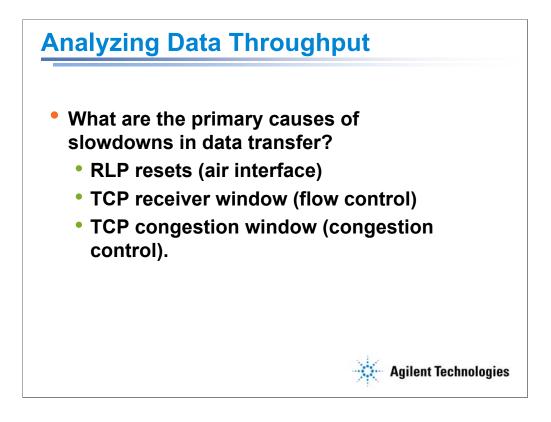
The test set provides adjustments to vary relative signal/noise power levels.

This capability can be used to introduce a controlled amount of data loss beginning at the physical channel. The amount of physical channel data loss on the forward supplemental channel can be measured by bringing up a call in service option 32, which is a data loopback service option, and measuring TDSO FER. A description of this measurement is beyond the scope of this paper but can be found by going to the Agilent.comwebsite and searching on TDSO FER.

Eventually, RLP retransmissions will occur, then RLP resets. At this point, TCP retransmission will occur because all layers below it have failed to deliver data sequentially.



Let's turn our attention back to the TCP layer. Since this is the layer that is ultimately responsible for delivery of data to/from the application layer, monitoring TCP gives valuable information about network performance.

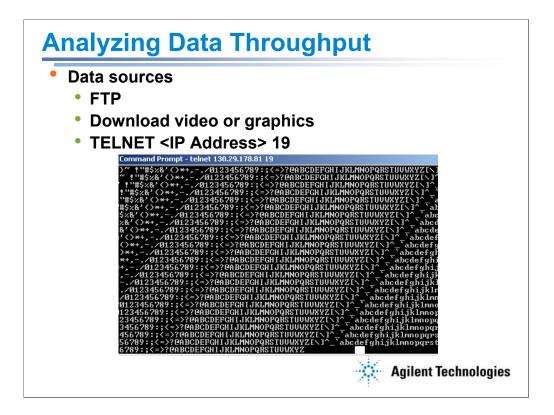


The areas of interest when analyzing the effects of lost data are:

RLP - if the circular buffer is overrun it performs a reset.

TCP receiver window - receiver runs out of buffer space

TCP congestion window - TCP senses what it thinks is network congestion and slows down.

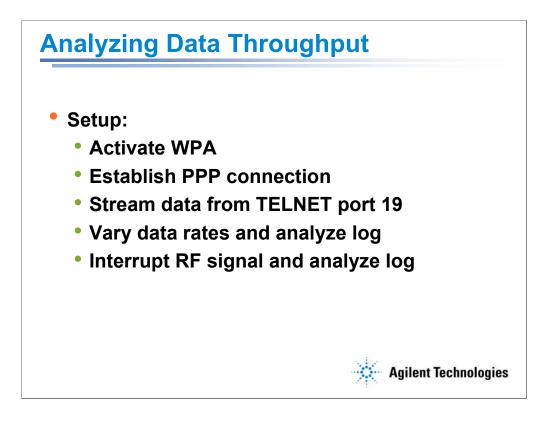


To observe date throughput a significant amount of data must be sent to the wireless device.

CDG Stage two testing specifies a file that can be sent to the wireless device using the FTP protocol. Some video streaming files or graphics files also provide adequate data.

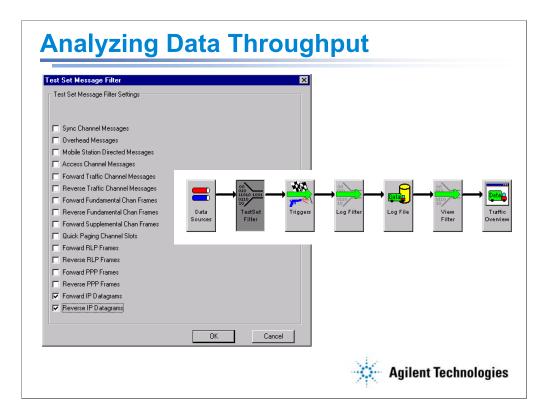
For this demonstration, a laptop PC was used to request a TELNET connection with a UNIX computer. The wireless device was a phone functioning as a modem.

One of the protocol ports available through TELNET is port 19, which sends a stream of alphanumeric characters until the session is ended by the user. The characters are displayed in a window and make it easy to see when data flow increases or decreases.



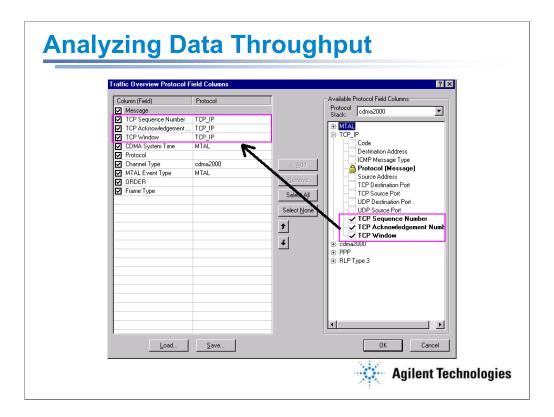
A TCP connection to TELNET port 19, which is available on all unix machines will provide a visual means of monitoring the flow of data to the device that requests this port.

Transmitted characters will automatically scroll across the command prompt window.



The Test Set Message Filter allows filtering messages that are sent to the WPA software. In this setup, we are only interested in the Forward and Reverse IP Datagrams, which carry encapsulated TCP segments, so these boxes are checked.

Filtering can also be performed at the Log Filter and View Filters stages.



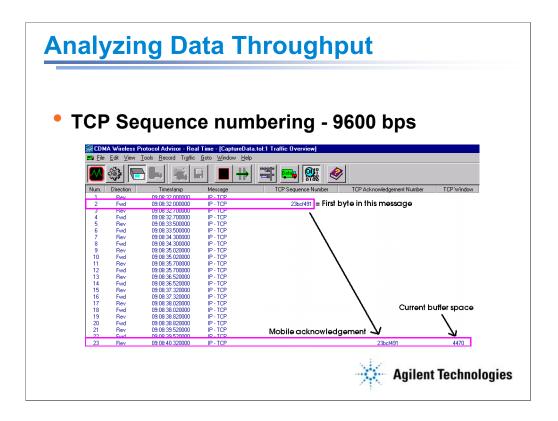
The WPA allows customization of columns displayed in the Traffic Overview window. Since we are interested in analyzing TCP_IP protocol, we will select three fields that the sender and receiver use to control the rate of data flow at the TCP layer.

The window on the right selects the message fields of interest and the window on the left specifies which column they will be displayed in.

		Call Setup Sc	reen			
RLP Counters		RLP Counters Info	rmation		Call Parms	
					Cell 1 Pouer	
	Class	Туре	Foruard	Reverse	-55.00	
	Control	SYNC Frames	17	5	dBm/1.23 MHz	
		SYNC/ACK Frames ACK Frames	3	2 5	Cell Band	
		NAK Frames	3 0	а 0	US Cellular	
	Data	Neu Frames	93	39		
	Data	Neu Octets	1798	720	Channel	
Clear Counters		Retransmitted Frames	0	0	384	
counters		Retransmitted Octets	0	0		
		NAKked Frames	0	0	Protocol Rev	
		NAKked Segments	0	0	6 (IS-2000)	
	Fill	Fill Frames	30	1	0 10 2000/	
	Idle	Idle Frames	3950	3983	Radio Config	
	Unclassified	Error Frames		0	(Fud3, Rys3)	
		Unknoun Frames		0	S033 (+ F-SCH)	
	Totals	Frames	4096	4040	0000 (+1=0011)	
Return		Octets	1798	720	FCH Service	
nevan					Option Setup _V	
		Active Cell	1			
Ì		Connected + C				
		IntRef Offset			1 of 3	

When a service is requested, whether it be TELNET port 19, requesting a web page using HTTP protocol, or initiating the transfer of a file using FTP, a PPP connection must be established and the physical channel resources allocated to permit data transfer over the air interface.

The test set displays Connected +Data when a service option 33 call is connected. If data transfer is stopped for a long enough period of time, the test set will display Idle, indicating the physical channel has been released. The PPP connection may remain, however, so that the PPP connection setup does not have to be repeated the next time data is requested.



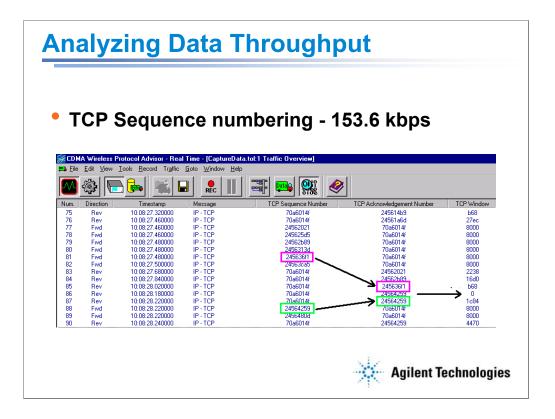
This display was generated by sending data to the mobile at 9.6 kbps using TELNET port 19. WPA Test Set Message Filter Settings was set to include Forward and Reverse IP Datagrams only. This representation of the WPA display has been altered to highlight areas of particular interest.

Note that the pattern here is alternating between forward and reverse channel messages. Forward channel messages are occurring a little more frequently than one per second.

No data is being sent on the reverse channel.

The first message shows a Fwd (forward) TCP Sequence Number of 1bb48f23. The TCP Sequence Number identifies the first byte of data in this segment being sent .

Eight seconds later, the reverse channel (mobile) sent a message acknowledging that it had received all bytes up to 1bb48f23. The mobile also let the sender know that it has 4470 bytes of buffer space in the TCP Window field, which was max for this phone.

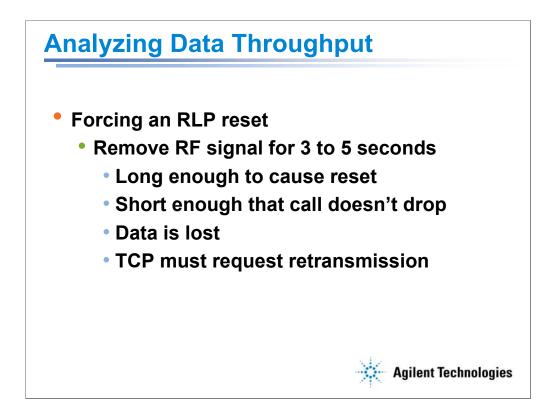


In this display, the data rate has been increased to 153.6 kbps.

Notice that the number of messages per second has greatly increased. Now forward channel messages are occurring at a rate of approximately eight per second. The mobile is reacting much faster in sending acknowledgements.

Also, notice that in message 86 the TCP Window shrunk to 0, prompting the TCP sender to slow down. The message from the mobile station highlighted in line 87 shows that the TCP Acknowledgement Number did not increment due to the TCP Window going to zero in message 86. The mobile is still asking for data beginning at 24564259 but indicates it now has buffer space.

The sending TCP has held off sending this segment of data and ends up sending it in line 88.



When data is lost over the air interface, RLP will send NAKs (negative acknowledgements) to the sender and will have the frames retransmitted. This will work up to a point. Due to limits on buffer space RLP will perform a reset if its ability to recover lost data cannot keep up with the rate of data loss. At this point, the job of recovering lost data gets pushed to the upper layer protocol TCP.

To force an RLP reset, open the RF path to the wireless device for a period of 3 to 5 seconds. This will be long enough to force a RLP reset, but not long enough to drop the call. For reference information, this is the technique used in:

CDG Stage 2 Interoperability Tests

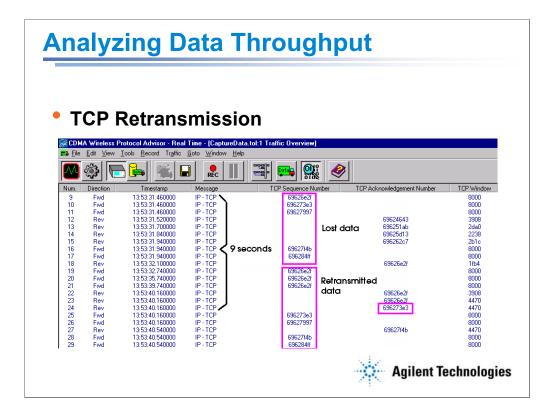
(TIA/EIA/IS-2000)

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CDG 57

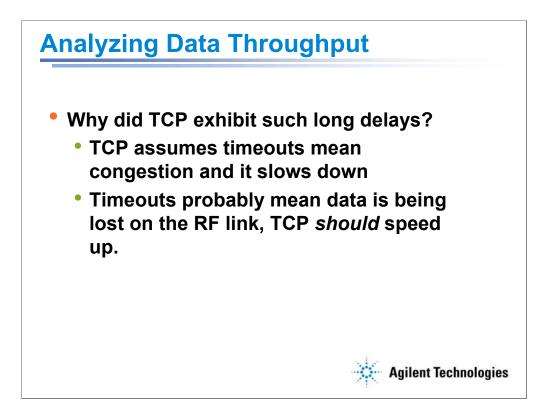
10.1.12 RLP Abort and TCP Retransmit Test



In this display the RF path to the mobile was removed for 3 seconds. RLP does not buffer enough data to handle the interruption so the TCP layer has to retransmit lost segments.

The data in the first message in this display, message 9, begins at 69626e2f. In message 23 the mobile station acknowledges receiving all data before 69626e2f, but it is not until message 24 that the mobile station indicates it received data beyond 69626e2f. The process of retransmitting lost TCP segments took over 9 seconds.

After message 24, the forward channel retransmitted the next four 1500 byte segments.



When the RF path was disconnected, RLP did not have enough buffer space to fill the gap, so it reset. When the TCP sender's retransmission timer timed out, it assumed that there was congestion in the network and adjusted its congestion window, which determines how large a burst of data it can send, to let network traffic clear up.

Messages 20 and 21 were only one-segment bursts, delayed 3 and 4 seconds respectively. No messages from the mobile station were received during this time.

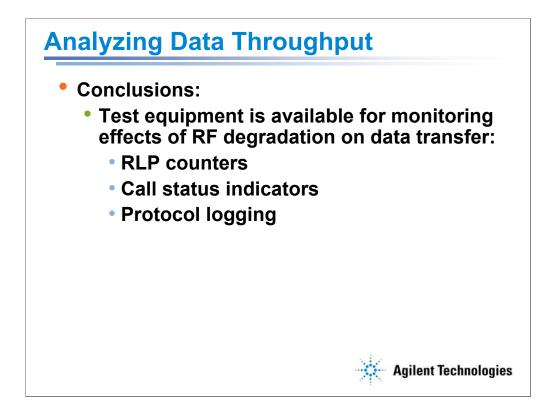
After acknowledgments resume, the TCP receiver will gradually increase the congestion window up again until another timeouts or the receiver TCP window says its buffers are full.

Since the data loss was caused by disruption to the RF signal, not network congestion or receiver buffer overflow, it would be better for throughput if TCP would speed up. This is an inefficiency in networks that include an RF subnet that is caused by the TCP congestion algorithm.

Analyzing Data Throughput

- Conclusions:
 - The wireless portion of the network is inherently unreliable.
 - TCP/IP is not designed to handle data loss due to RF degradation.
 - It is important to test end user applications during data slowdowns.





RLP counters to to see if retransmissions are happening

Call status indicator to see if a call has been dropped.

Protocol logging to monitor specific layers of protocol during RF degradation.