



860 DSP VoIP Tests

This application note discusses two of the VoIP tests that can be performed with the 860 DSPi. The first test utilizes a server-side application that the 860 DSPi can communicate with, producing both upstream and downstream test results. The second test utilizes any network device on the system and produces a single result, which assumes that the upstream and downstream are equivalent. Both tests are controlled by the same setup menu, described below.

VOICE OVER IP (VOIP) SETTINGS		
CODEC Type	G.711	
CODEC Delay	21 msec	
CODEC Bit Rate	64000 bps	
Packet Rate	20 msec	
Packet Size	160 Bytes	
Jitter Buffer Delay	40 msec	
Call Length	20 sec by 1000 pkts	
Max Latency	Max Jitter	Min MOS
140 msec	40 msec	4.00
Press Enter, Up, or Down to Adjust		
<input type="text"/>	<input type="text"/>	<input type="text"/>

The setup menu allows you to choose the G.711 CODEC. The associated bit rate and delay are displayed. The meter also displays standard packet rate and packet size. You can then adjust the packet rate, packet size, jitter buffer length, and overall call length. The final items to set are the associated VoIP limits.

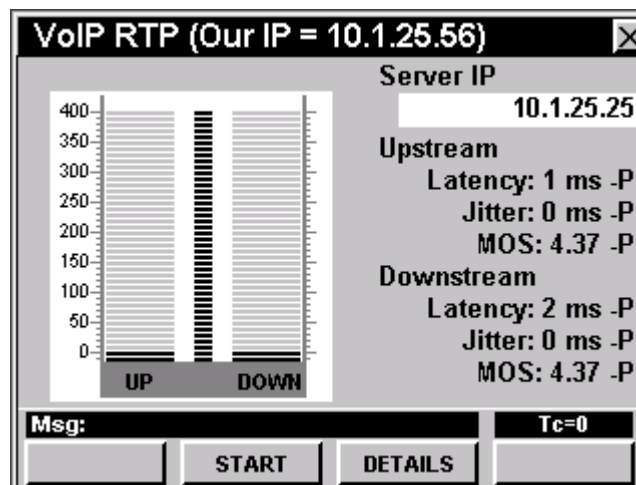
For Additional Help Contact
Trilithic Applications Engineering
1-800-344-2412 or 317-895-3600
support@trilithic.com or
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860 DSP VoIP Tests
P/N 0010275060 – Rev 01/10
1 of 3

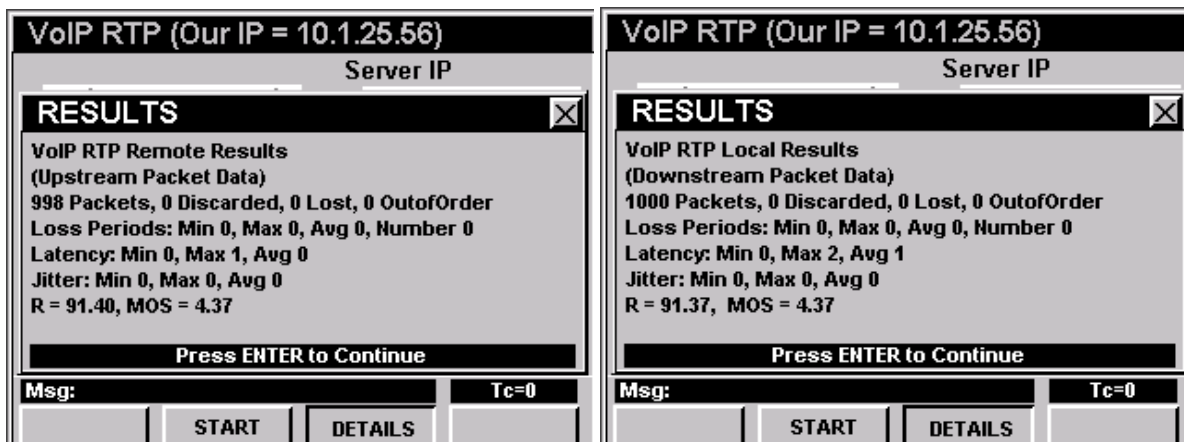
VoIP RTP Test



The first test utilizes a network protocol called RTP (real time protocol). The user presses the START button to begin the test. The 860 DSPi first establishes a connection to the server (IP address on the menu) on port 24007. The meter and the server negotiate the test parameters from the setup menu as well as synchronize their clocks. The server and the meter then communicate using RTP over the UDP protocol on a port that is negotiated in the first step. Once all of the packets are transmitted and received by both the server and the meter, the results are calculated and compared to limits.



Once the test is finished, the user can press the DETAILS button to view both upstream and downstream detailed information. This includes min, max, and average latency and jitter as well as packet arrival information.



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860 DSP VoIP Tests
 P/N 0010275060 – Rev 01/10
 2 of 3

VoIP ICMP Test


The second test utilizes a network protocol called ICMP (internet control message protocol). The user presses the START button to begin the test. Packets are sent to the designated IP address and then echoed back at the packet rate and packet size specified on the menu. The test ends when the user presses stop.

Ping (Our IP = 10.1.25.56)			
Host Name			
or Host IP	10.1.25.25		
Pkt Delay	20 msec	Pkt Size	256 bytes
Sent	1031	Min Time	1 msec
Received	1031	Avg Time	1 msec
Lost	0	Max Time	2 msec
LPR	0.00e+00	Latency	61 msec
LPR (%)	0.0 %	Jitter	0 msec
Msg:			
GOTO	START	STOP	MODEM

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860 DSP VoIP Tests
 P/N 0010275060 – Rev 01/10
 3 of 3