DCB, Inc.
2949 CR 1000 E
Dewey, Illinois
61840

217.897.6600 Tel
800.432.2638 Toll Free
217.897.1331 Fax
www.dcbnet.com

Using Corporate Networks for Telex Dispatch
Verify Suitability Using the ETTA

September, 2018

Bosch-Telex radio dispatch console systems are installed in commercial dispatch centers and Public Service Access Point (PSAP), or 911, dispatch centers. These often communicate with radio transceivers located at remote "high-point" or rural tower sites. Often one dispatch center communicates with a half dozen or more remote transmitter sites. In the past, this communication between the dispatch center and remote transmitter was typically through a four-wire analog line leased from the local telephone company. New installations, however, are communicating through agency-owned private IP networks or the Internet because telephone companies are no longer offering four-wire analog lines; or in those rare cases they will install one, the lines are expensive.

DCB authored a white paper ***Using Enterprise Networks for ROIP and VOIP, Verify Suitability Using the ETTA*** [1], in which the network characteristics were discussed and a general case was made for qualifying an IP network prior to installing ROIP or VOIP equipment. Proper vetting of the circuit saves installation time, prevents repeat service calls for marginal service, and provides a baseline for future problem analysis.

This paper discusses the specific case of Bosch-Telex dispatch system products and how the ETTA can be configured to mimic Telex dispatch console traffic when testing an IP connection. Note that all information we present about the Telex protocols was determined empirically or through Telex published documentation. However, this information has been used many times to assist Telex resellers install IP-223, IP-224, C-Soft, Hardware Consoles, and other radio communications equipment.

First let's look at some details of the ROIP protocol in use. Telex equipment supports several codecs from 16K to 64K, requiring UDP payload packets of 40 to 160 bytes. Audio packets are sent every 20 msec. And there are other parameters that may be configured to insure a successful installation… including "delay before playback", which is essentially related to network jitter. Knowing the actual network jitter and transfer characteristics helps the installer determine optimum settings for the Telex equipment.

Typically, any modern IP network can provide adequate bandwidth for the basic dispatch console user's needs. In some cases that might be low DSL speeds for a single channel remote radio, or as low as 1 to 10 Mbps for a few audio channels. In others with multiple audio channels, bandwidth requirements rapidly grow. If a network is engineered for 10 simultaneous audio channels in full duplex (that's 20 one-way audio paths) and one of those remote locations also has redundant network recorders, it's conceivable that the network path requires bandwidth for 40 or more simultaneous audio streams, even if normal usage is only a few audio streams. Since bandwidth is always specified when ordering circuits, adequate bandwidth should be assumed. Reliability is also assumed to be adequate for the job as poor reliability will be obvious.

However, there are additional important network path characteristics. After reliability and bandwidth, the two most important network characteristics in a ROIP transport are latency and jitter. Another issue we've seen is the packets per second limit on some circuits.

Latency, or delay, is the time it takes for a packet to transit the network and arrive at its destination. That's the delay we measure when running a typical PING test. It should be consistent and low enough to not affect the quality of traffic being transmitted. Although most ROIP connections work poorly with long delays, Telex protocols tolerate longer than average delays. Extremely long delays cause pauses in "turning around" the conversation, which is not well received and inconvenient for the users. However, we are aware of Telex installations with latency of 350 msec or longer that are working well.

Jitter is simply the difference or variation in packet delay(latency). In other words, jitter is measuring time differences in packet inter-arrival time. For an example, consider that a sending side transmits packets evenly spaced apart in time. In a perfect network, they would arrive with exactly the same time spacing. But in the real world, that time often varies due to network congestion, configuration errors, or queuing problems. So, the steady drum beat of packets becomes more erratic… some are closer together, some are farther apart. A "jitter buffer" is used to minimize the effect of transmission jitter, but it can only do so much towards curing the problem… and large buffers cause other problems such as more delays.

ROIP connections require adequately low latency and jitter along with suitable bandwidth and reliability.

How good is good enough? While specifications published by different VOIP manufacturers are often unique to their product characteristics there are some commonly used values. Commonly seen specifications are:

Packet loss: < 0.1% (some less-critical systems will tolerate <1%)
Packet delay: < 40-50 msec (some long-haul systems can tolerate much greater delays, with degradation)
Packet jitter: < 20 msec (some long-haul systems can tolerate much greater jitter with degradation)
Bandwidth: Approximately 200Kbps per active radio or about 80 Kbps per audio channel in each direction. Telex recommends 50 Kbps per 32Kb ADPCM channel, 34Kbps per 16Kb ADPCM audio channel and double that to 100Kbps of bandwidth for full duplex. Most new installations are full duplex. Some manufacturers recommend designing a network for three times the anticipated traffic.
MTU: 1500 bytes, with larger packets required for 802.1Q or encryption. Some as large as 1522 bytes
Ring Convergence: If a physical ring topology is used, 50 msec convergence is required.
DSCP: Required or preferred by almost all manufacturers
Multicast Transport: Required by most installations. May be provided over a WAN by DCB's UT encrypted bridges.

Should we test and "prove" that the network is valid BEFORE installing equipment? Of course! Assuming that the network you ordered is the one they delivered can cause a longer and difficult installation and future problems.

The ETTA is ideal to "proof" the network before installation and recording the results, you also have a baseline measurement to test against if you see audio or network degradation in the future.

The tests are simple to run so any technician can perform them. Some tests may be performed with a single ETTA, but thorough analysis requires a pair, one ETTA at each end of the circuit.

We should configure the ETTA to mimic Telex ROIP traffic when performing some of the tests.

First perform the basic test steps as you would for any ROIP/VOIP network path. Reading the ETTA manual[2] is always a good first step!

Basic test steps:

1. Verify connectivity. Use the ETTA's ping test with variable size packets to insure that there is basic connectivity. Be sure to include a test with the maximum size packets that you'll be using.

2. Verify bandwidth using configurable packet size, preload, rate limiting and intervals to learn the rate, PPS, minimum and maximum PPS capability of the network.

3. Measure network jitter. Run a longer term jitter test to insure that jitter meets maximum specs for the equipment you're installing.

4. Save the test results on a USB drive and transfer to your PC. Some organizations require an initial test and the results saved before putting a circuit into service. These results can used to meet their internal record keeping requirements. Even if the customer doesn't require them, retain these in spread sheet form for later analysis if there are communications problems.

Then, configure the ETTA to use packet traffic that is similar to actual Telex traffic and see how it works.

5. Determine the size of the Telex jitter buffer ("Delay before playback" in their terminology). Use the jitter test with appropriate packet size and the transmit delay. For packet delay, use 0.02 seconds (that's the 20 msec audio slice). If your Telex is using 32K codecs, then the UDP payload is about 80 bytes; for 64K codecs, it's a 160 byte UDP payload.

Because the Telex setting is in units of "packets", after you determine the average jitter in msec using packets that mimic theirs, divide it by the size of the packets to determine how many packets to configure in the Telex "Delay before playback" field. Divide then jitter delay in msec by the packet timing to scale it appropriately. A minimum value to use would be the "average jitter", a more conservative value would be "maximum jitter" from the tests. We recommend the latter, but each installer should determine the appropriate value for the specific location. Simply divide the jitter in msec by the packet frequency of 20 msec per packet.

Entry value [in packets] = (jitter test results [in msec]) / (20 msec [the frequency of packets transmissions])

For example, if the jitter test result is 40 msec, the entry value in packets would be (40/20) or two packets.

For a greater number of audio channels, the Bandwidth Transmit or Bandwidth Received tests could be used to determine an upper limit the network can support. This limit is determined by the maximum bandwidth of the link in raw bandwidth of Mbps as well as maximum packet rate in pps. To determine these values, set the appropriate packet size to match the codec packet size in use and test.

The resulting PPS divided by 50 would give the approximate upper limit based upon maximum packets per second.. 50 is the number of packets per second required by 1 audio channel. Calculating the maximum bandwidth due to raw bandwidth availability is just as simple. Since we're already using the typical packet size for the bandwidth test, that PPS result also indicates maximum raw bandwidth availability for packets based upon the expected traffic.

Again, document the results for acceptance and later comparisons.

---

[1] Read the ETTA paper, "Using Enterprise Networks for ROIP and VOIP, Verify Suitability Using the ETTA" at https://www.dcbnet.com/notes/ETTA-ROIP.pdf

[2] Read the ETTA manual at https://www.dcbnet.com/manuals/etta.pdf

Read the ETTA data sheet at https://www.dcbnet.com/datasheet/ettads.html

The ETTA (Ethernet Traffic Test Appliance) is available from Data Comm for Business, Inc (DCB) either directly or through resellers such as Graybar or Anixter. At only $995, and small enough to carry along in a laptop case, each technician should have one, and a second one for the team to use at remote sites.