Dyaptive SYSTEMS

Using the DMTS-8000 for Voice over Data Planning and Measurement

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1 Introduction

This document presents a number of measurements and test factors that are critical in determining how a network operator can plan, measure, and optimize the impact of carrying voice services over CDMA data networks. The two main sources of such traffic considered are Push-To-Talk (PTT) and Voice-over-IP (VoIP). Two network topologies are considered, a CDMA2000® 1X network in which the packet data traffic is sharing the same carrier resources with circuit-switched voice traffic, and a 1xEV-DO network in which all resources on a carrier are dedicated to carrying packet data traffic.

1xEV-DO[1] networks are still in their early stages, often still very much in the planning and concept phase. Also, these networks are usually being considered as extensions to existing CDMA2000 1X networks. As such, carrying voice over the combined circuit and packet resources of two networks should be considered. In the initial phase of planning, measurement of the current capabilities of the 1X network should be carefully taken. As the DO network becomes available, measurement of the DO network capabilities for identical data traffic patterns should then be taken. This should be undertaken as early as possible so that the benefit of the combined network can be validated and the optimization strategy can be planned.

As one or both networks evolve, the measurements of each networks capacity by traffic type and traffic mix are repeated. The measurements give relative strengths of each network's ability to carry traffic by class. These strengths then used to derive a set of per-traffic class weights which are used to develop an optimization that may include load balancing strategies between the 1X and DO carriers.

2 Background

Dyaptive Systems Inc. offers CDMA2000 network performance measurement tools and services. The core of these tools and services is the DMTS-8000, a comprehensive load generation and measurement tool. The tests described in this document reference the DMTS-8000 as the load and measurement source for evaluation of the network capabilities.

3 End-To-End Latency and Capacity

These tests are used to establish the baseline performance of the networks for carrying voice traffic as circuit-switched or packet-switched traffic. These tests include background load, but only of the same type.

3.1 Test Requirements

The test capability required for proper traffic modeling and network performance measurement includes at least:

3.1.1 Statistical Specification of the CDMA2000 Traffic Descriptors

- Call arrival process (e.g. Poisson)
- Call duration process, or call hold time
- Voice Activity Factor process (duration of talk versus listen spurts)

[1] In this document, where a reference is made to 1xEV-DO, the reference generally applies to 1xEV-DO revision A unless specifically stated otherwise.

- Packet Generation Process (e.g. a voice application typically generates a packet every 10 or 20 ms during a talk spurt)
- Packet Size Distribution (this is usually dependent on the selection of the voice codec selection and head compression scheme)

3.1.2 Monitoring and Reporting of the Grade of Service Parameters

- Call blocking rate (determined by Call Admission Control configuration)
- Call drop rate
- Handoff call blocking rate
- Call setup delay
- Call throughput capacity given that the packet loss and delay is within the specified bounds for the traffic type (e.g. less than 2% packet loss and 160 to 260 ms delay)

3.1.3 Monitoring and Reporting of the Quality of Service Parameters

- Frame error rate
- Bit error rate
- Signal quality
- Packet loss
- End-to-end packet delay (e.g. Maximum Perceptually Tolerable one-way mouth-to-ear delay is 160 ms according to ITU-G.114)
- Packet jitter

3.1.4 Quality of Experience Parameters

- Mean Opinion Score (MOS)

3.2 Test Cases

3.2.1 Comparative Evaluation of Resource Utilization Efficiency of VoIP over DO

1. Establish a set of live rate-set I or rate-set II conversation sessions connected to a 1xRTT carrier.
2. Generate background voice traffic and proportional signaling channel load using the DMTS-8000. Increment the background load until the MOS of live voice sessions, or the GoS / QoS of the aggregate traffic drops below acceptable levels. Note the carried traffic as reported by the DMTS-8000.
3. Repeat the previous steps and measurements with VoIP over the 1xRTT carrier.
4. Establish a set of live mobile-to-mobile VoIP sessions on a DO carrier.
5. Generate background VoIP BHCA traffic and proportional load on signaling channels using the DMTS-8000. Increment the background traffic until MOS of live VoIP sessions or GoS / QoS of aggregate traffic drops below acceptable levels. Note the carried traffic as reported by the DMTS-8000.

6. Compare the observed carried traffic (Erlangs) during above test iterations to quantify the spectrum and resource utilization efficiency of VoIP over DO with respect to both the Circuit Switched voice over 1xRTT and VoIP over 1xRTT measurements.
7. Note other GoS / QoS parameters such as call set up delay.
   a. In 1xRTT arrival of a data burst subsequent to the establishment of a (PPP) session would involve verifying if an SCH is available and could be assigned; reserving the Walsh Code; and finally assigning SCH to support the burst. If during the burst the signal fades, the BTS increases power causing congestion (lower SINR for other connections).
   b. In 1xEvDO, as the data burst arrives, the BSC forwards the burst to the BTS. The BTS checks the requested bit-rate over Data Rate Control (DRC) and transmits the appropriate data. Thus much less setup time is expected as compared to 1xRTT. If during the burst, a fast fade occurs, the DRC recommends that no data be transmitted and thus the BTS would buffer and send the data when the channel conditions are appropriate. Thus no power control is needed, eliminating the congestion in the forward link.

3.2.2 Measurement of the VoIP Pole Capacity

This test uses an identical setup to the previous test but drives the traffic levels to the point of failure to setup a voice connection.

3.2.3 End-to-end packet delay and jitter characterization of VoIP over DO

1. Monitor end-to-end packet delay and jitter of a single, mobile-to-mobile VoIP call. Typical measurements include:
   a. Encoder delay (around 35ms)
   b. Modem delay (reverse link plus forward link, around 10 ms)
   c. RAN and PDSN delay (reverse link plus forward link, around 40 ms)
   d. Decoder delay (around 20ms)
   e. De-jitter buffering delay (around 15ms)

   Thus end-to-end delay is approximately 110 ms without including the network delay which is proportional to the hop-count on the IP path. The key advantage of VoIP is that it does not suffer from any transcoding latency as there is no need for any transcoding at the PCF or media gateways.
2. Configure the DMTS-8000 to increment the VoIP traffic in small steps and record the resulting end-to-end delay and jitter statistics of the aggregate traffic.
3. Analyze the results to determine the delay and jitter bottlenecks. For example:
   a. scalability of RAN and PDSN (with respect to load)
   b. scalability, optimality and fairness of the schedulers

4 Impact of VoIP on CDMA System Capacity for Mixed Voice and Data Traffic

These tests are used to measure the tradeoffs between system capacity and speech quality for a variety of network settings given a traffic mix that includes a combination of circuit and/or packet switched traffic types in addition to voice.

4.1 Test Requirements

4.1.1 Statistical Specification of Independent Traffic Parameters by Traffic Type

- Call arrival process
- Call duration process
- Forward Link Activity Factor process
- Reverse Link Activity Factor process
- Packet Generation Process
- Packet Size Distribution

Sets of these parameters can be established to model a number of different traffic types such as voice, web browsing, e-mail, gaming, etc.

4.2 Test Cases

4.2.1 Comparative Analysis of DO and 1xRTT Performance under Mixed Traffic Types

1. Configure the DMTS-8000 to generate the desired composition of mixed mode traffic including:
   a. Non real-time data traffic e.g. FTP and HTTP sessions, and SMS load
   b. Non real-time audio / video streaming sessions
   c. Real-time VoIP calls
2. Load the 1xRTT carrier with the above traffic and observe the steady state GoS / QoS statistics through DMTS-8000 reports
3. Load the DO carrier with the above traffic and observe the steady state GoS / QoS statistics through DMTS-8000 reports
4. In the 1xRTT test bed, replace VoIP calls with conventional rate-set I or rate-set II CS Voice calls and observe the steady state GoS / QoS statistics through DMTS-8000 reports.
5. Compare the effectiveness of the underlying technologies in accommodating (multiplexing) traffic from a wide spectrum of service types each with a unique set of QoS requirements and traffic descriptors. Compare the throughput and GoS / QoS measurements.
6. Configure the DMTS-8000 to iteratively adjust the traffic composition to determine the optimum portions of traffic types that maximize operator revenue while achieving acceptable levels of GoS / QoS. Use these results to verify the optimality of the forward link schedulers and reverse link rate and power allocation schemes in place (refer to the test cases 5.2.3 and 5.2.4 below).

Alternatively, use these results to derive resource allocation (reservation / prioritization) policies and offer differentiated services.

4.2.2 Develop Methods to Improve Speech Quality under Realistic Traffic Conditions

These measurements may require collaboration with phone vendors to establish the handset-specific factors affecting speech quality. Such factors might include antenna design, microphone and speaker design, handset ergonomics, as well as others.

1. Establish a set of live VoIP sessions on a DO carrier.
2. Configure the DMTS-8000 to generate the expected VoIP background traffic.
3. Explore improvements in Speech Quality and Delay by applying techniques such as Adaptive Jitter Buffering and Speech Coder Resynchronization.

4.2.3 Measure the Impact of Handoff Latency on VoIP Packet Delay and Jitter

1. Configure the DMTS-8000 to generate VoIP calls that undergo at least one handoff per call. The DMTS-8000 achieves this by directly controlling each mobile's perceived pilot strengths of the connected sectors without any additional external faders or attenuators.
   a. DO supports hard handoffs in the forward link.
2. Measure handoff latency and its impact on the end-to-end delay and jitter of VoIP calls.

4.2.4 Evaluation of the Tradeoffs Between Speech Quality and RLP Parameters in the Presence of Channel Noise

1. Configure the DMTS-8000 to inject noise into the forward and / or reverse links of a VoIP call.
2. Measure the speech quality under different settings of non-transparent RLP parameters.

5 Evaluation of the Impact of Fast Handoff and Multi-Flow Applications

5.1 Test Requirements

The basic requirement for the following tests is supported for the relevant MAC features.

5.2 Test Cases

5.2.1 Performance Evaluation of Applications Capable of Multi-Flow

1. Configure the DMTS-8000 to provision VoIP as a multi-flow application in which the signaling (IP/UDP/SIP) and the voice (IP/UDP/RTP) packets are transmitted as separate flows and perhaps with different priority levels.
2. Generate multi-flow VoIP traffic and compare the performance statistics with single-flow VoIP traffic.
   a. In particular, study the impact of multi-flow configuration of VoIP calls on call setup delay.
3. Repeat the tests for other applications such as Push-to-Talk and FTP, and with mixed traffic types.

4. Configure the DMTS-8000 to provision TCP applications such that SYN, ACK and FIN packets are also sent as a separate (high priority) flow.
    a. Study the impact on TCP connection set-up time.

5.2.2 Performance Tuning and Optimization of Multi-Flow Capability

1. Optimize the maximum number of RLP flows per application i.e. increase (N_{RLPMax}) until there is no further improvement in throughput.
2. Optimize the maximum number of Reverse Traffic Channel MAC flows per mobile i.e. increase $RTCMAC_{FlowMax}$ until no further improvement in throughput.
3. Similarly optimize other negotiable attributes identified in DO specifications (e.g. sections 4.8.1 & 4.8.2).
4. Dimension reverse link Access Channel
    a. Generate peak BHCA and increase the numbers of reverse link Access Channels until the call setup delay is within the acceptable limits.

5.2.3 Verification of Optimality and Fairness of Transmission Rates and Power Allocations in the Reverse Link

1. In DO, the Access Terminals (AT) monitor Reverse Activity Bit (RAB) in the forward link to detect the congestion at the BTS and adjust their reverse link transmission rate in terms of Transmit to Pilot Power (T2P) ratio.
2. Configure the DMTS-8000 to establish multiple connections with diverse QoS requirements
3. Monitor the throughput and/or T2P estimates at each AT.
4. Verify that the ATs transmit in the reverse link in a ‘fair-share’ manner i.e. in proportion to the QoS requirements (or SLAs) of the Class of Service (CoS), and equally among the connections belonging to the same CoS.
    a. For example, for real-time VoIP services, the T2P estimations and hence transmission rates shall stay consistent. Non real-time data connections such as Web or FTP have elastic QoS requirements. For them, the T2P estimations and hence the transmission rates should be varied to minimize the impact on the real-time connections that are in progress in parallel.
5. Confirm the optimality of the reverse link MAC by configuring the DMTS-8000 to periodically change the traffic mix to include a number of Classes of Service and the traffic load of each CoS. The DMTS-8000 can then monitor and verify that the reverse link transmission rates, and hence T2P estimates, converge towards minimizing the Mean Square Error between the *accurate* resource allocation and the *actual* resource allocation. This verification can be performed to include the path gains of each connection, the constraints such as the overall rise-to-thermal threshold at the BTS, the SINR threshold for each CoS at the BTS, and the maximum battery power limitation of the AT.

Note: The terms ‘optimality’ and ‘fairness’ are used here in their most conventional forms, their definitions are indeed subject to revisions by Wireless Service Providers.

The DMTS-8000 can be used to evaluate optimality in terms of various utility functions such as overall Revenue or throughput.

5.2.4 Verification of the Optimality and Fairness of the Forward Link Schedulers

1. DO schedulers exploit multi-user diversity in which users are assigned forward link transmission rates according to the forward link channel conditions.
2. Configure the DMTS-8000 to establish multiple connections.
3. Introduce fast fades in randomly selected connections at random times.
4. Monitor the throughput.
5. Verify that the scheduler compensated the connections that experienced fast fades (and hence resource starvation) fairly after the channel conditions improved.
6. Configure the DMTS-8000 to establish multiple connections with diverse QoS requirements.
7. Monitor the throughput.
8. Verify that the scheduler allocated the resources in a ‘fair-share’ manner i.e. in proportion to the QoS requirements (or SLAs) of the Classes of Service and, equally among the connections belonging to the same CoS.
9. Confirm the optimality of the scheduler by configuring the DMTS-8000 to periodically change the traffic mix by varying the number of CoSs and the traffic load of each CoS.
10. Verify that the scheduler converges towards minimizing the Mean Square Error between the *accurate* resource allocation and the *actual* resource allocation.

Note: The terms ‘optimality’ and ‘fairness’ are used here in their most conventional forms, their definitions are indeed subject to revisions by Wireless Service Providers.

The DMTS-8000 can be used to evaluate optimality in terms of various utility functions such as overall Revenue or throughput.

5.2.5 Performance Evaluation of Dynamic Rate Control

1. Configure the DMTS-8000 to establish multiple connections.
2. Introduce fast fades.
3. Monitor the performance of DRC in terms of prediction error
   a. Mean Square Error between the predicted rate (based on the forward channel’s SINR Prediction) and the actual, possible rate.
4. Verify that DRC tracks the variations in channel conditions correctly and that the response time is respectable.
   a. Verify that in response to any stimulus such as fast fades or a random spike in SNR etc. the DRC converges in finite amount of time and the MSE during the convergence is acceptable.
5. Use the results to fine tune the prediction of forward channel at the AT and the frequency of DRC

a. If required, update SLA agreements to accommodate such deviations.

5.2.6 Performance Tuning of Dynamic Rate Control

1. Use the DMTS-8000 to optimize the tradeoffs between the DRC Length, DRC Channel's relative (to pilot) gain, or erasure rate and the noise level on the reverse link.
2. Achieve the desired forward link capacity to reverse link capacity ratio.

5.2.7 Performance Evaluation of Fast Handoff schemes and features

1. Use the DMTS-8000 to generate desired number of calls that undergo handoff at least once or more during their life span.
2. Use the DMTS-8000 to generate expected channel conditions and fine tune the fast channel estimation (sector selection) schemes as well as handoff thresholds and the associated timers.
3. Use the DMTS-8000 to generate inter-PDSN and/or intra-PDSN handoffs and measure the performance of fast handoff schemes in terms of handoff latency and packet loss etc.

6 Evaluation of Compression Schemes on Capacity

These test cases are targeted at measuring the system capacity benefit gained from potential header compression schemes.

6.1 Test Requirements

Support within the load generator for the header compression scheme of interest. Candidates include:

- ROHC
- LLA ROHC
- SIGCOMP

6.2 Test Cases

6.2.1 Evaluation of Capacity Gain under Noisy Channel Conditions

1. Use the DMTS-8000 to generate VoIP calls that utilize Header Compression
2. Use the DMTS-8000 to generate expected channel conditions
3. Monitor header update rate (i.e. refresh frequency)
4. Verify the correctness of compression scheme
    - For example, for LLA ROHC, verify that the MAC layer is reliably detecting packet losses and assisting ROHC
5. Evaluate the capacity gain in terms of pole VoIP capacity when such compression schemes are being utilized.

6.2.2 Evaluation of the Scalability of the PDSN and Other Impacted Network Nodes

1. Use the DMTS-8000 to load the PDSN with VoIP calls
2. Increase the load in small steps.
3. Monitor the packet delay and jitter.
4. Use the DMTS-8000 to load the PDSN with VoIP calls employing ROHC, LLA ROHC and/or SIGCOMP.
5. Increase the load in small steps.
6. Monitor the packet delay and jitter.
7. Evaluate how well PDSN scales to the increased processing load due to header decompression.

7 Quality of Service Verification

7.1 Test Requirements

7.1.1 Push-to-Talk Traffic Configuration

- Number of PTT Groups in the network
- PTT Group Sizes (i.e. membership per group)
- Location and Mobility of group members
    - with respect to each other
        - to study the impact (latency) of unequal distance of listeners from the speaker and distinct mobility patterns of groups members
    - with respect to the BTSs of their serving cells
        - i.e. different channel conditions
- Percentage of On-Demand vs. Prearranged talk Sessions
- On-Demand Talk Session Origination Process (e.g. first member initiating a call with other members)
- Distributions of participants joining or leaving a PTT group
- Percentage of participants in multiple PTT sessions
- Talk Burst Arrival Process
- Talk Burst Duration Process
- Priority of Talk Bursts (e.g. preemptive or queuing etc.)
- Packet Generation Process (Packet generation during the Talk Spurt is usually constant - every 10ms or 20 ms)
- Packet Size Distribution (usually dependent on the type of voice codec as well as header compression scheme assumed)

7.1.2 Push-to-Talk QoS Measurements

- Right-to-Speak (RTS) time

   - Start of session - from first sending invite and receiving RTS indication should be less than 2 sec
- End-to-end channel delay
   - Delay from speaking to being heard by other participants should be less than 1.6 sec
- Signal quality
- Packet loss
- Packet jitter

7.1.3 Push-to-Talk QoE Measurements

- Mean Opinion Score (e.g. typically > 3 at BER = 2%)

7.2 Test Cases

7.2.1 Service Level Agreement Verification

1. Using the DMTS-8000, emulate mobiles with diverse SLAs such as throughput and QoS requirements.
2. Configure the DMTS-8000 to monitor for SLA violations
3. Produce reports when SLAs are violated indicating the percentage of users of each service type whose SLAs were violated and the fraction of time that the SLAs were violated.
4. Use the above results to derive CAC (Call Admission Control) policies to limit the number of connections accepted into the system so that SLAs for certain classes of users could be assured, particularly if the scheduler at the BTS or BSC only supports differentiated service and not QoS Guarantees.