CADP2 Design Applications
Calculating Gain-Before-Feedback

Gain-Before-Feedback

Maximum system gain before the occurrence of feedback is a function of:

- The characteristics of the electronics and transducers.
- The acoustical characteristics of the room.
- The source distances to a microphone.
- The source distances to a listener.

If the characteristics of the electronics are assumed to be flat response and minimum phase, their effects can be disregarded. For the same purposes, the microphone is designated omnidirectional, flat response and minimum phase. Any other microphones are defined as non-contributing and the ambient noise floor is assumed to be at least 25 dB lower than the talker.

With these conditions met, the onset of steady state oscillation will generally occur when the sound pressure level from a talker at the microphone is equaled or exceeded by the sound pressure level from speaker sources and room reflections [1][2]. In order to provide an appropriate margin below this oscillation point the speakers should then be reduced an additional 6 dB [3]. Once the speaker levels have been adjusted to this stable level, comparisons between the talker and speaker sources can be made in the listening area. The difference between the original source SPL (the talker) and the combined speaker SPLs will be the Gain-Before-Feedback value. (See Figure 1.)

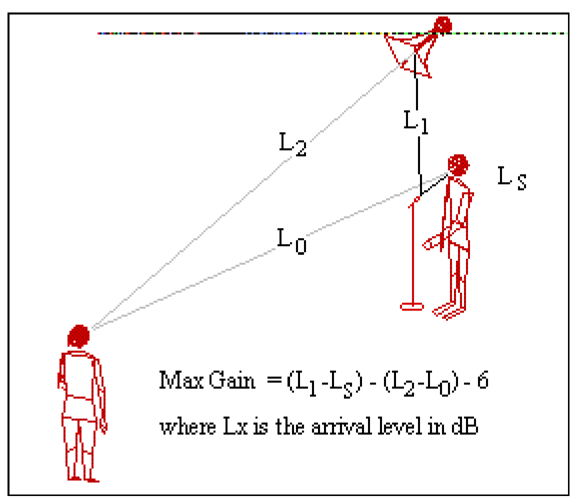

Figure 1

Calculating Gain-Before-Feedback in CADP2

In the example shown in Figure 2 the sound system has been located in a room and the proposed system has satisfied all of the other aspects of a good design: even coverage, flat response, and minimal phase anomalies, and appropriate frequency response. An imaginary microphone position was designated downstage, center and a talker source located 2 feet behind this position, aimed at the microphone. The 2 foot setting is an arbitrary but often used value for this type of calculation. Most people will stand within this distance of a microphone and so this distance represents a worst case scenerio. A “talker” is included with the CADP2 device files and has a sensitivity value of 74 dB (1W-1M), the level of an average orator. The talker source is positioned 5 feet above the stage floor, at mouth height.

In this example the speaker sources are three JBL2365 horns and three JBL4648 bass cabinets, placed within a ‘cluster’. The ‘cluster’ type is used so that the average complex sum can be used to examine phase effects and so the results can be merged to only one arrival.

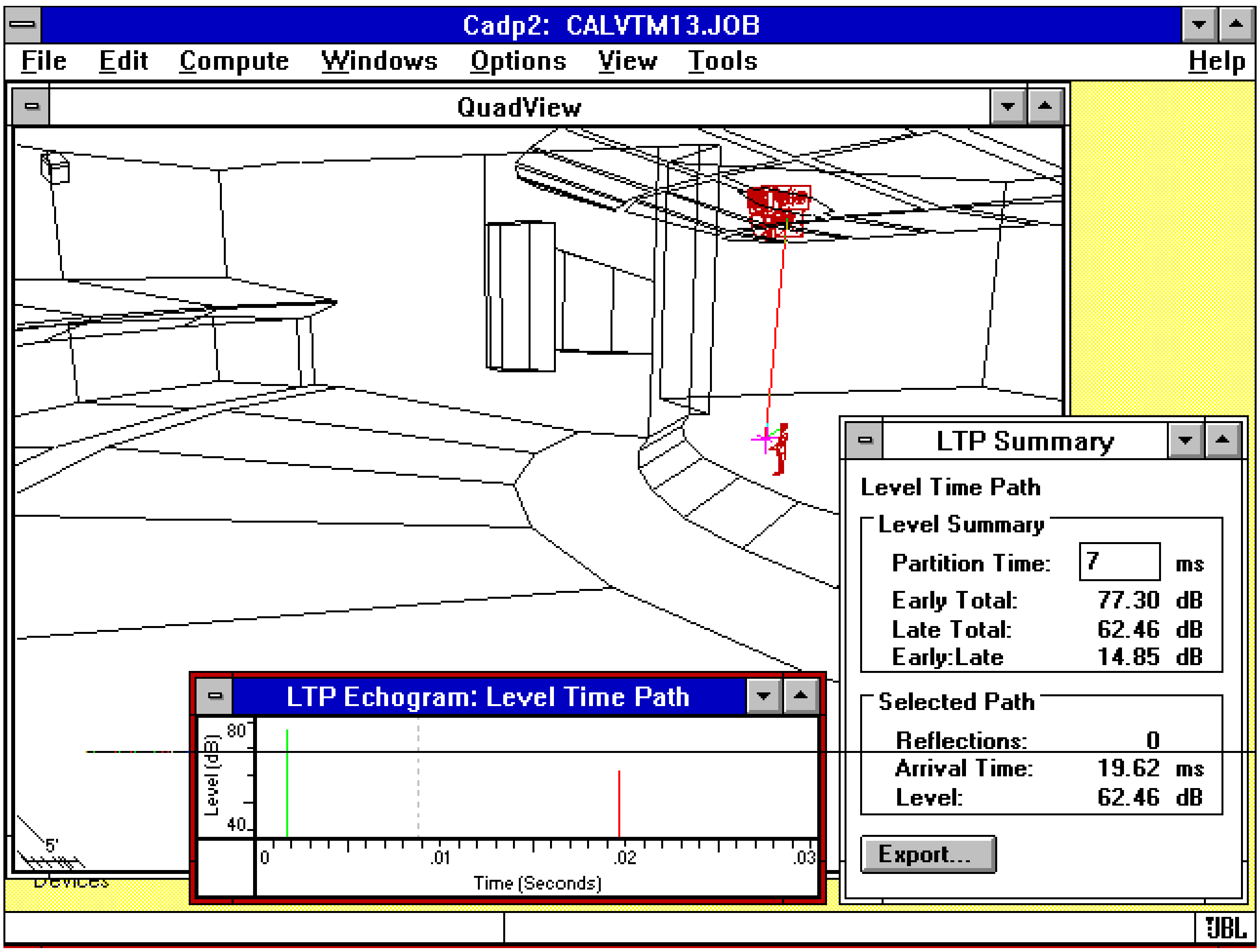

Figure 1: Arrivals at the microphone location.

The first LTP computation was set up at the imaginary microphone position by following these steps:

- Select the LTP computation
- The 'Listener Position' window was brought forward by selecting it and the listener cursor was positioned at the microphone by using the shift-right mousekey in an appropriate view. (The values can also be entered or fine tuned by keyboard.)
- The 'LTP Calculation' setup window was brought forward and was set for:

  Frequency Band 2 kHz (The Talker only includes the 2 kHz band)

Direct Merge Average Complex Sum
Reflections 0
Other values remain as defaulted.

The computation was then begun. To simplify the example, only direct arrivals were computed; The room reflections were ignored but there is no reason why they could not be included in the LTP calculation. After the new layer was added to the Layer List window the echogram and summary windows were opened by double-clicking the layer's line number, and positioned.

The arrivals from the talker and cluster were compared in Figure 1 by clicking on each arrival and noting each arrival's sound pressure level value in the summary window.

Cluster-to-Microphone 62.5 dB @ 19.6 ms
Talker-to-Microphone 77.3 dB @ 1.8 ms

As shown, in order for the sound from the cluster to arrive at the microphone at a level equal to the talker, the cluster level should be increased by 14.9 dB. To provide the appropriate margin below feedback this level should then be reduced 6 dB yielding a total increase of 8.9 dB.

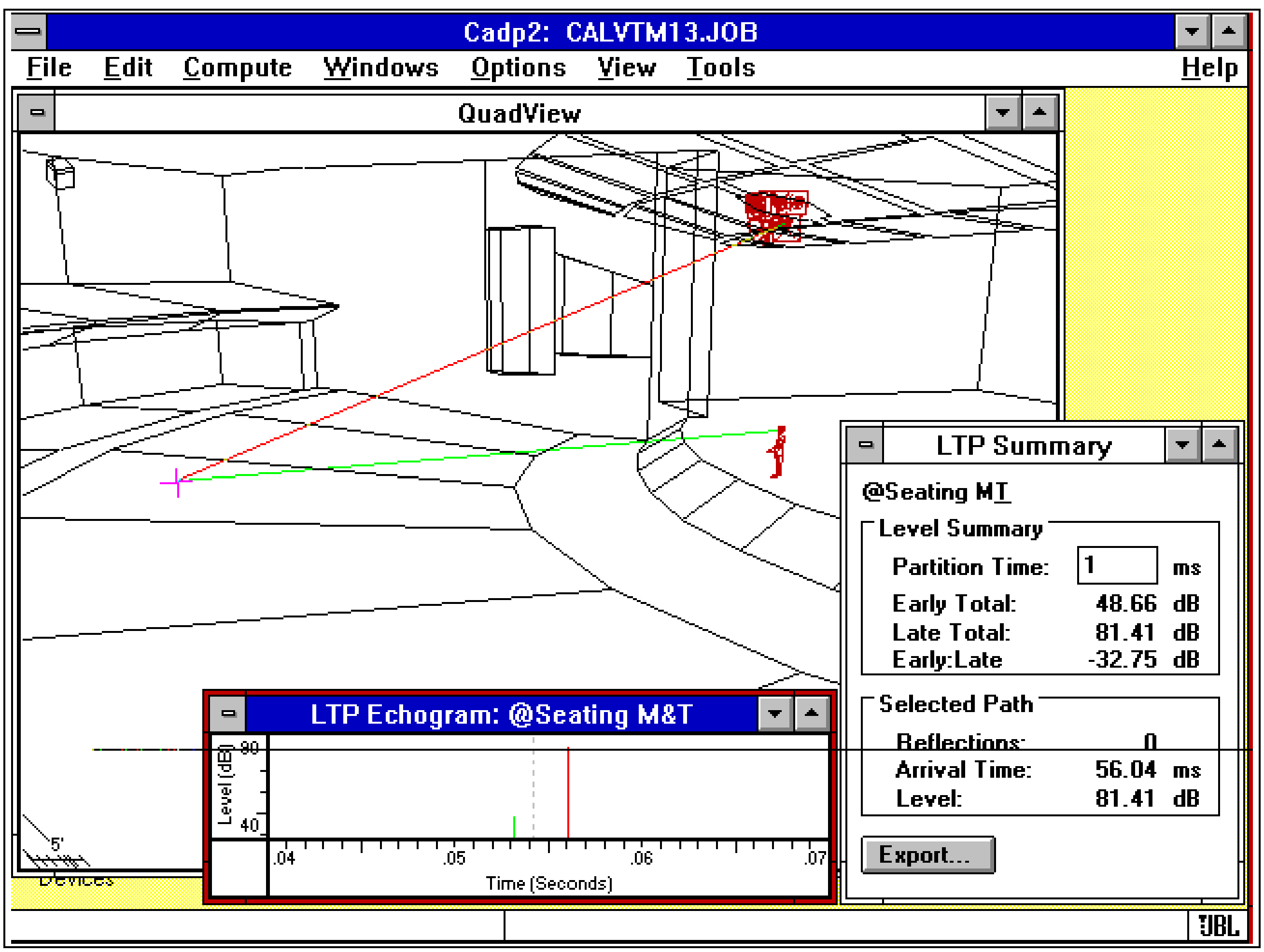

Figure 2: Arrivals at an audience location.

With the cluster adjusted another LTP computation was performed at an audience location as shown in Figure 2. Again the arrivals from the talker and cluster were compared by clicking on each arrival and then viewing each arrival's sound pressure level value in the summary window.

Cluster-to-Audience 81.4 dB @ 56.0 ms
Talker-to-Audience 44.7 dB @ 53.2 ms

The talker level can now be then subtracted from the cluster level. The resulting value of 32.8 dB is the system gain over the unaided talker at this listening location. (Note: The Partition Time can be chosen to give this value automatically)

It should be pointed out — since by definition of gain-before-feedback — the drive level to the cluster cannot be raised further without incurring ringing or feedback. The audience level of 81.4 dB is the maximum level of system reinforcement from the talker at this room location for the conditions given.

Expanding the Concept...

The performance of a microphone other than omnidirectional can be estimated. Assume that the chosen directional microphone is on-axis to the talker and from the microphone's polar pattern, obtain the dB attenuation value at the cluster's arrival angle. This value can be subtracted from the cluster arrival level at the microphone. For example, a cardioid microphone is -6 dB at 90 degrees off axis. The resulting gain-before-feedback value will most likely increase by 6 dB.

A lectern, as a free-floating volume, can be added to the room to determine it's effect on gain. The LTP is then calculated for the microphone position using first order reflections. The early-to-late partition can be set to merge all signals after the arrival of the talker to obtain a value for the combined level of the late arrivals.

References

[1] JBL Sound System Design Reference Manual, March 1986

[2] The Gain of a Sound System, C. P. Boner and R. E. Boner, *J. Acoust. Soc. Am.*, April 1969

[3] Frequency Characteristics of a Sound-Reinforcing System, William B. Snow *J. Acoust. Soc. Am.*, April 1955