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RECENT ADVANCES IN SPEECH INTELLIGIBILITY AND SOUND SYSTEM DESIGN
Part I: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

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ABSTRACT

After a brief review of the latest advances in computer-assisted acoustical design packages developed by others, this paper describes A-CADX, a programme specifically tailored for the performance predictive analysis of acoustically difficult venues, where user-defined parameter flexibility is enhanced, with the application of a multitude of methods of analysis. These features include aspects relating to system design, coverage, reverberation time, time/frequency analysis (naturalness/annoyance), intelligibility predictions, gain-before feedback, in addition to overall system equalisation and control. Only a brief description of the data processing capability of objective measurements is given, these being discussed in more detail in the second part of this paper.

1. INTRODUCTION

Various PC-based computer programmes have been developed in recent years to assist in the analysis and design of sound reinforcement systems. This introduction describes some of the recent releases.

Altec Lansing's *AcoustaCADD*, is described by Lanphere [1]. Special features of the programme include general acoustics and sound system design, with specific facilities for cluster systems, further to powerful graphics and statistical ray-racing. The system can handle up to 20 clusters, with up to 40 loudspeakers in each. This is an extremely powerful programme, which has only been recently released.

JBL's Central Array Design Program, *CADP*, is described by Eargle & Kalmanson [2-3]. It is particularly useful in central-cluster system design, and is capable of direct-field calculations, direct-to-reverberant ratio calculations, and intelligibility predictions, in addition to mechanical aspects of array design. Up to 20 loudspeakers may be handled by this programme.

Bose Programmes, *Modeler* and *Speaker-CAD* are described by Birkle and Jacob [5-6], including acoustical space modelling, sound field calculations, intelligibility predictions and reverberation time analysis.

Prohs [4] describes the latest generation of *the PHD* programme. Other programmes and computer analysis techniques are described by McCarthy [7] and Anhert [8], amongst others [9-11].

2. OBJECTIVES OF A-CADX

In the course of carrying out the design of a complex sound reinforcement system with severe constraints, comprising several sub-systems of different characteristics with a total of over 3000 loudspeakers of various types, the Authors were confronted with the following problems:

(a) lack of a clearly defined set of *"quantitative" criteria*: in one sub-system for example, this related to analysis of intelligibility, involving numerous discrete echo contributions under near-free-field conditions. Although there appears to be general agreement on the effects of noise and reverberation on intelligibility, the effects of echo are not at all clear [12-13], and it was necessary therefore to retain some flexibility in the design criteria, with continuous refinement.

(b) several complex sub-systems involving *large numbers of loudspeakers*: in one sub-system for example, well over 100 loudspeakers needed to be considered at a time, so that the iterative calculations were far too laborious, due to the excessive degrees-of-freedom involved.

It became apparent very early on in the project that a specially-tailored programme needed to be developed, as the accuracy of manual methods was simply impossible to sustain, in addition to the fact that other available packages at the time were unable to provide the flexible features required (work actually carried out in 1987).

In this context, *Dar Al-Handasah Consultants* developed a programme entitled *A-CADX (Acoustic - Computer Assisted Design - XBASIC)* to provide the necessary features. Figure 1 gives a basic schematic diagram of the system functions, comprising the following components:

(a) *Main-menu*: this offers the user a choice of features, essentially comprised of "predictive analysis" and "objective measurement analysis" functions.

(b) *Template*: this feature enables the user to obtain appropriate "over-lays", based upon the scale of architectural plans being used, whereupon a proper choice of loudspeakers may be made, in addition to their tentative positioning. The latter parameters can then be refined by use of the main programme.

(c) *Data-entry*: user enters the above data, as appropriate. A memory-resident set of data is provided, for the loudspeaker models generally used, and the user can store additional data as necessary.

(d) *Plan generation*: a plan of the study area is generated here, giving location of each loudspeaker, with the aiming point being indicated.

(e) *Predictive analysis*: several features are included, such as:
- uniformity of coverage (direct-field)
- reverberation time
- time/frequency domain analysis (echo annoyance, naturalness criteria)
- intelligibility analysis
- gain-before-feedback analysis
- other features (optional loudspeaker parameters): operating level assessment, electronic time-delay, and equalisation).

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

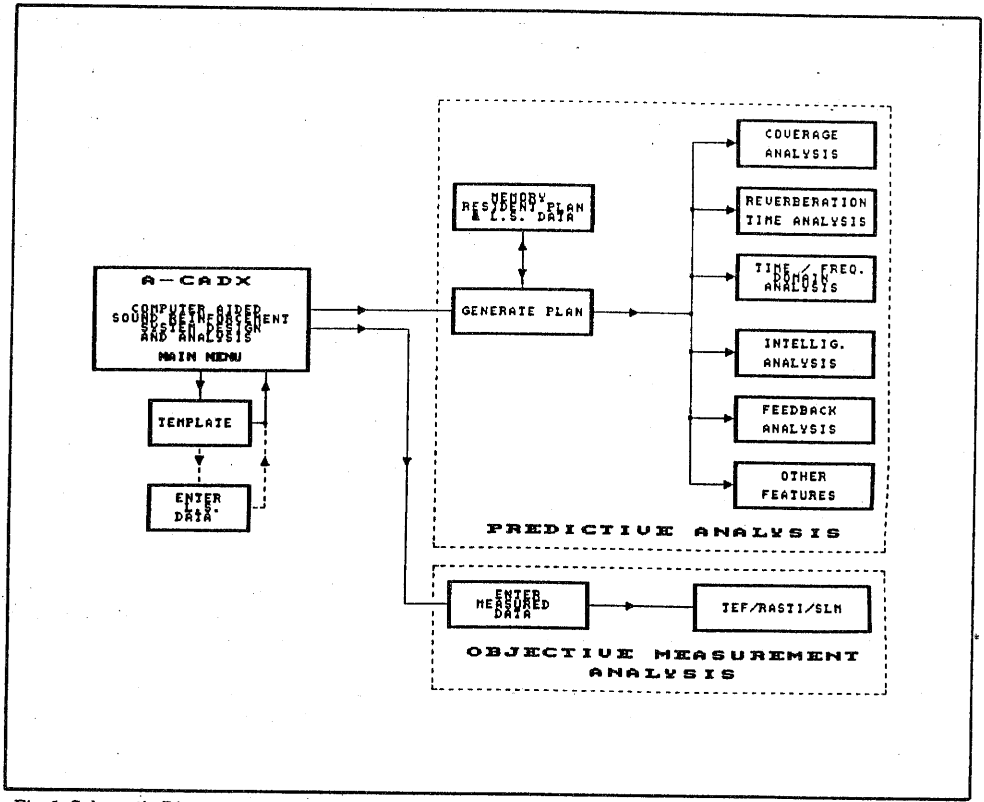


Fig. 1 Schematic Diagram of the Main ACADX Functions

(f) ***Objective measurement analysis***: this includes analysis of measurement results noted by TEF/RASTI/SLM, mainly relating to parameters of intelligibility analysis.

A-CADX has been written in XBASIC, and runs on IBM PC, PC XT, PC AT, PS/2 or 100% compatible machines running MS-DOS. The original version of A-CADX was written for Hercules graphics, but recent versions are also available for EGA and VGA color graphics.

3. TEMPLATE DESIGN AID

"Template" is a simple graphical method developed by Tappan [14], which is useful in arriving at suitable choices of loudspeakers based upon available dispersion characteristics, with tentative definition of positioning and orientation, to achieve some coverage criteria. This is based upon the use of a transparency "overlay" which shows the loudspeaker aiming angles, the attenuation of sound pressure level (SPL) due to inverse square law, and angular dispersion. The difficulty in the application of this method is that a different graph is needed for each scale and loudspeaker mounting height encountered. With the help of A-CADX a template printout can be obtained virtually immediately and a transparency/photocopy made to suit any scale and mounting height being used. Figure 2 gives a sample printout.

T. Uzzle [15] describes another method, which may be easily incorporated. This is based upon spherical mapping (light projection method). The authors have opted for the template method.

4. DATA ENTRY PARAMETERS

The basic entry parameters are:
- *choice of loudspeaker* (polar characteristics in various bands, normally 500Hz, 1kHz and 2kHz),
- *loudspeaker sensitivity*, and frequency response: SPL (1W,1m), in each of the above octave bands.
- *loudspeaker operating power* (with individual attenuation as a user-option).
- *choice of mounting position* (co-ordinates)
- *choice of aiming point* (orientation)
- *options* (artifial time delay and equalisation)

Based upon the layout of the coverage area being studied, a plan is generated on the screen, with further allowance for the inclusion of *environmental data*. Figure 3 gives a sample plan for a simple area (ten loudspeakers), with menu options of that specific mode being indicated.

The user also enters *ear-height*, enabling assessment of various conditions. Auditory directional characteristics may be considered through the microphone sub-programme. Other user data entry relates to reverberation analysis and microphone characteristics.

5. SYSTEM PERFORMANCE EVALUATION FEATURES

Having entered the parameters, the programme is now ready to carry out any of the user-defined tasks.

5.1 Calculation Subroutine (Direct Field Calculations)

The main subroutine calculates the direct SPL (dB) and associated acoustical time delay for each loudspeaker contribution at the defined observation points. Each calculation consists of obtaining the inverse square law attenuation, adding the off-aim dispersion loss, further to the air-absorption loss in the various octave bands. This procedure is repeated for every loudspeaker present in the system, resulting in a multi-dimensional array of size equal to the number of loudspeakers for every observation point considered.

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

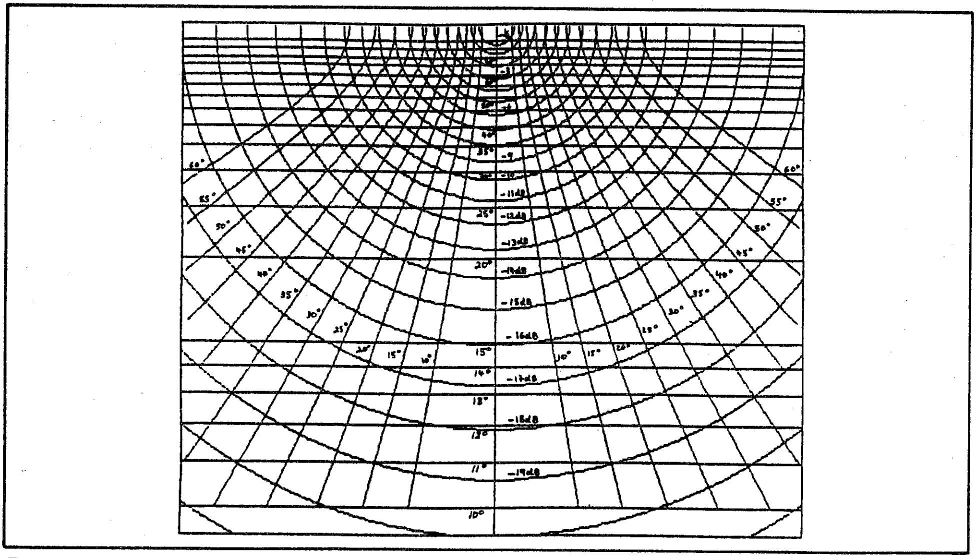


Fig. 2 Sample Template Print-out

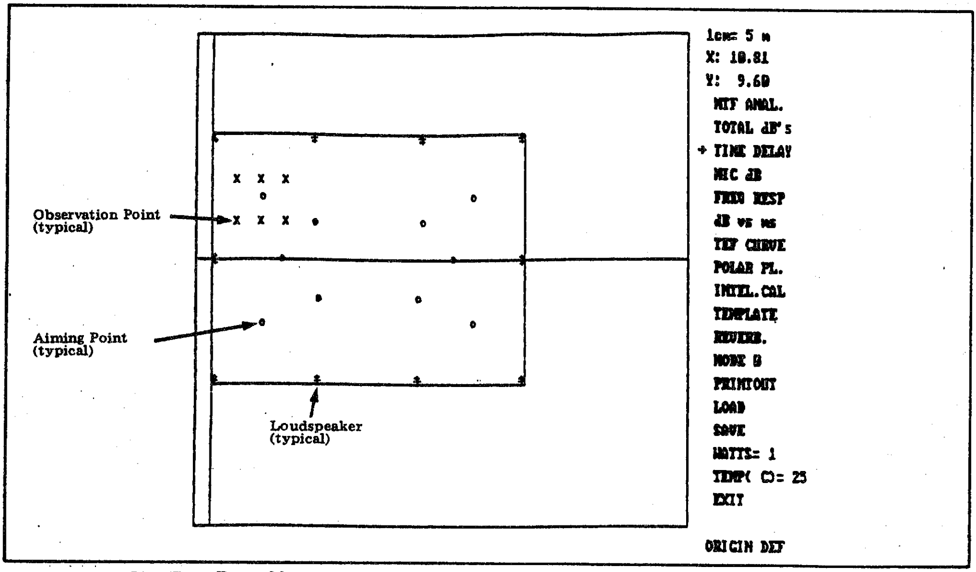


Fig. 3 Sample Plan/Data-Entry Menu

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

5.2 Uniformity-of-Coverage

The total direct contribution at any specific point consists of integrating the direct SPL at each observation point, as defined by the user. From these results, it is possible to check overall uniformity, in addition to checking the frequency response flatness criterion (calculations performed at 500Hz, 1kHz and 2kHz, with further extension to the bands from 250Hz to 8kHz, as required). This also allows the user to check whether or not equalisation will yield the desired performance criteria, under near-free-field conditions.

5.3 Reverberation Time Calculations

The basic models used in these calculations relate to the classical formulae of Sabine and Fitzroy [16], with allowance for other imperical results, such as Friberg [17]. In some of the sub-systems of the study, it has been necessary to develop other models of analysis. This has been necessary in cases where the sound reinforcement system is expected to have a marked effect on reverberation, further to difficulty in coping with a virtually "boundless" volume. For the latter, a three dimensional coupling method [18] was applied, which is the subject of a paper to be published shortly. The programme gives these options to the user, as suited to the problem being studied.

5.4 Time-Frequency Analysis

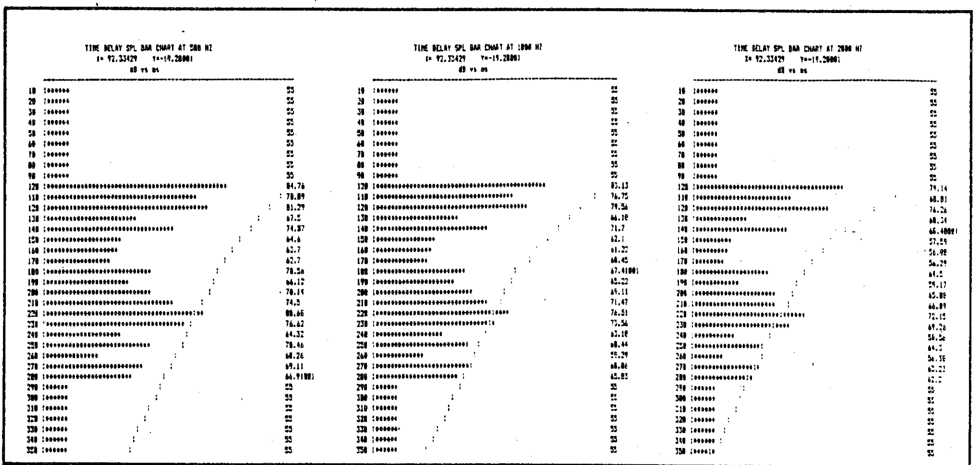
This feature of the programme provides the user with analysis options in both the time domain and the frequency domain. These are essentially suitable for near-free-field conditions only, with special provisions being required for diffuse-field conditions by consideration of reverberation time and direct-to-reverberant ratio, D/R.

(a) Time Delay/Early Integration: this feature of the programme is a simple extension of the fundamental direct field calculations. It helps locate the time delay annoyance areas and identify which loudspeakers are responsible for this annoyance. The calculations are performed for a single point at a time, and the output is given in the form of a table. Figure 4 gives an example for a complex study area comprising 98 loudspeakers. To assist the user in the analysis, integrated values are given for three durations: 0-50ms, 50-100ms, > 100ms, further to the full tabulated results. These parameters can be easily changed by the user, for analysis as required.

(b) Energy Time Chart (ETC): the above tabulated results are also presented in the form of a bar chart as shown in Figure 4. This consists of plotting the SPL contributions integrated over defined intervals (1-10ms) as a function of time. Superimposed on this plot is a 20% annoyance curve which indicates the maximum allowable delayed level to attain a 20% listener disturbance performance, (based upon a user-selected pseudo-reverberation time, applied to the results of Nickson at Al [19] and Haas [20]). The background noise entered by the user in this example is 55dB. Shorter integration periods than the indicated 10ms can be used for improved accuracy, at the expense of processing time.

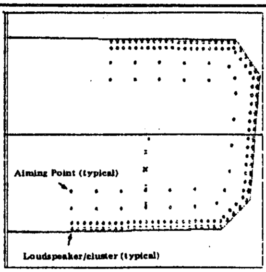
(c) Energy Frequency Chart/Curve (EFC): this calculation is done in 1Hz steps. The polar responses of the loudspeakers in the regions between the 500Hz, 1kHz and 2kHz octaves are approximated by interpolation. The output can be presented in bar-chart form (Figure 5a), where the resultant SPL value is given for each frequency increment, or as a condensed graph (Figure 5b).

(d) Energy Frequency Time (EFT) Curve: the energy frequency time graph is a 3-dimensional plot of SPL against time/frequency. Figure 6 gives a sample plot.



POINT NO.: 3 X= 92.33 Y=-19.2
MIN DELAY TIME IS FROM SPEAKER NO. 13 = 99.95814 ms

SPKR NO.	L500	L1000	L2000	DELAY(ms)	
1	66.22	60.19	56.07	38.85	***
2	66.72	66.10	64.06	38.85	***
3	62	59.44	52.76	38.85	***
4	63.24	61.12	54.45	31.64	***
5	64.25	62.92	55.09	24.12	***
6	65.93	64.07	58.07	17.37	***
7	74.63	71.71	68.43	11.51	***
8	74.74	73.69	71.10	11.51	***
9	68.01	66.94	59.91	11.51	***
10	69.22	68.64	60.53	6.72	***
11	71.31	68.97	68.04	3.12	***
12	72.24	69.18	61.04	.05	***
13	83.04	81.96	77.99	0	***
14	79.05	76	72.45	0	***
15	72.33	69.26	61.13	0	***
16	72.28	69.21	61.08	.61	***
17	71.22	69.05	60.91	2.64	***
18	69.24	68.55	60.63	6.03	***
19	74.32	71.48001	67.79	11.06	***
20	74.51	73.46	71.04	11.06	***
21	67.78	66.71	59.75	11.06	***
22	65.73001	64.65	57.77	17.49	***
23	64.23001	62.72	55.77	24.04	***
24	63.15	60.92	54.25	31.30	***
25	66.16001	60.08	55.99	39.30	***
26	68.63	66.03	63.91	39.30	***
27	61.97	59.29	52.57	39.30	***
28	61.07	58.09	51.02	47.94	***
29	60	56.98	49.16	56.97	
30	58.04	55.94	47.17	66.38	
31	63.46	57.14	51.02	76.12	
32	64.45	61.71	56.75	76.12	
33	57.76	54.98	45.37	76.12	
34	61.1	56.21	51.2	47.6	***
35	60.11	57.00	49.51	56.0	
36	58.94	56.03	47.54	66.37	
37	66.86	64.09	62.63	76.25	
38	57.05	55.86	45.75	76.25	
39	63.37	62.17	54.21	81.02	
40	62.05	61.42	53.91	87.00	
41	69.42	60.32	65.24	95.6	
42	61.06	60.11	53.32	95.6	
43	59.39	58.42	52	104.81	
44	63.72	60.47	52.06	105.17	
45	63.67	60.39	52.8	106.23	
46	71.9	60.8	64.22	106.28	
47	63.67	60.39	52.8	106.28	
48	63.63	60.36	52.75	107.28	
49	63.34	60.22	52.62	110.21	
50	62.40	60.1	52.5	113.05	
51	70.71	60.51	63.92	113.05	
52	61.29	59.96	52.35	116.75	
53	60.35	59.41	52.11	122.25	
54	60.05	66.95	63.31	127.54	
55	59.07	58.24	51.9	127.54	
56	58.66	57.38	51.05	133.56	
57	57.51	56.21	49.99	141.14	
58	64.64	63.52	60	146.42	
59	56.41	55.11	48.59	148.42	
60	55.63	54.05	47.53	156.26	
61	55.00	53.01	46.73	165.44	
62	66.57	65.77	61.08	167.76	
63	58.47	57.10	50.16	167.76	
64	58.19	56.63	50.2	167.57	
65	58.3	56.07	50.15	168.50	
66	57.68	56.11	49.05	169.14	
67	66.16001	65.04	61.81	171.15	
68	56.62	55.09	49.13	171.15	
69	55.3	53.57	47.14	163.59	
70	55.0	54.46	47.07	156.38	
71	56.72	55.39	49.02	149.54	
72	64.63	56.05	52.92	143.12	
73	64.67	63.55	59.15	143.12	
74	57.67	56.34	50.22	143.12	
75	58.64	57.31	50.90	137.14	
76	59.63	58.31	51.71	131.63	
77	60.1	59.06	51.91	126.64	
78	70.34	62.18	61.11	122.19	
79	67.7	66.08	61.03	122.19	
80	60.7	59.60	52.09	122.19	
81	61.75	59.04	52.26	118.32	
82	62.59	59.9	52.33	116.41	
83	63.32	60.81	52.44	113.8	
84	74.56	68.19	66.25	111.06	
85	70.4	67.3	61.46	111.06	
86	63.4	60.09	52.52	111.06	
87	63.46	60.15	52.58	110.59	
88	63.48	60.17	52.61	110.01	
89	63.48	60.17	52.6	110.13	
90	63.45	60.14	52.57	110.94	
91	74.64	60.33	66.32	110.94	
92	70.45	67.34	61.51	110.94	
93	63.39	60.08	52.51	112.43	
94	62.7	60	52.42	114.6	
95	61.04	59.00	52.3	117.42	
96	70.45	62.29	61.25	120.06	
97	67.79	66.95	61.1	120.06	
98	60.79	59.75	52.16	120.06	



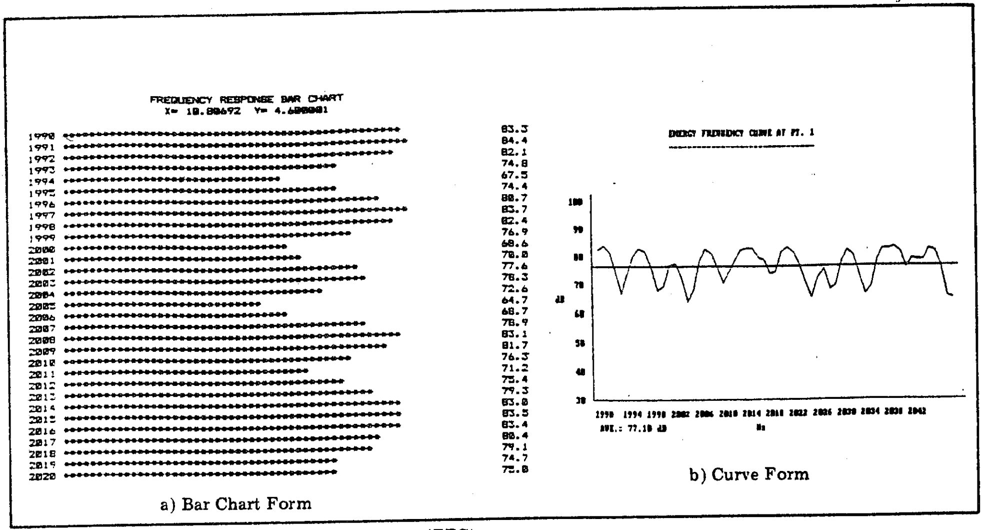
SUMMARY

	500	1000	2000
TOTAL dB LEVEL CONTRIBUTION BELOW 50 ms :	87.39	85.58999	85.58999
TOTAL dB LEVEL CONTRIBUTION ABOVE 50 ms & BELOW 100 ms	74.54	72.51	72.51
TOTAL dB LEVEL CONTRIBUTION ABOVE 100 ms :	83.36	79.96	79.96
PROGRAM DEPENDENT SNR VALUES:	3.5	4.91	4.91

x Observation Point

Fig. 4 Sample Result Illustrating ETC and Expected Annoyance Region

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS



a) Bar Chart Form

b) Curve Form

Fig. 5 Sample of Energy Frequency Chart/Curve (EFC)

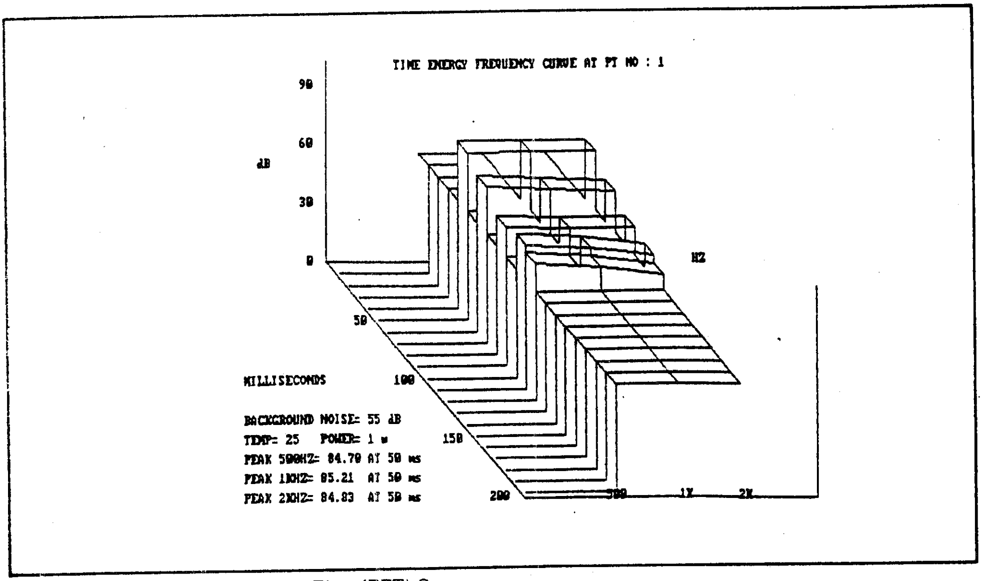


Fig. 6 Sample of Energy Frequency Time (EFT) Curve

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

5.5 Speech Intelligibility

The various methods of speech intelligibility analysis are outlined by Smith [13], amongst others. These methods are further expounded by Doany [21] and Mapp/Doany [22], in the context of this paper, including ALcons, STI and the modified-S/N method, as briefly described below:

(a) ALcons Method: ALcons, the Articulation Loss of Consonants method is based upon the work of Peutz [12], where ALcons is given as a function of the D/R, S/N and reverberation time. The diffuse field is computed as a function of room constant with various modifiers as appropriate [11]. Alternatively, the programme may also perform the calculation as a function of acoustical power and loudspeaker efficiency. The calculation is performed in the 2kHz band.

(b) STI Method: STI, the Speech Transmission Index, is based upon the work of Steeneken & Houtgast [23]. A-CADX offers two methods here. The first is based upon the work of Steeneken and Houtgast, but considers D/R, as derived by Doany [21], noted to yield results fairly close to those by Peutz [12]. The second method is based upon a direct computation of MTF, the modulation transfer function, based upon the specular components, somewhat similar to the work of Van Rietschote and Houtgast [24]. For both choices, the user can also select between 1/1 octave MTF (0.5-16Hz), 1/3 octave MTF (0.63-12.5Hz), and RASTI. A sample of a typical output is given in Figure 7, for a 2kHz band (1/3 octave). Similar plots may then be derived in other bands of interest, further to addition of noise "floor" values [23], as required by the user. It is also possible to carry out weighting in the frequency domain, if factors similar to those used in the Articulation Index (AI) are considered appropriate.

(c) Modified-S/N Method: This method is based on the Articulation Index (AI). In this method the speech peak-to-rms noise ratios are weighted and summed to obtain an index [25-26], from which intelligibility may be easily determined. This however, is strictly applicable to "additive" noise. For reverberant conditions, a modified version of this method can be applied, where the early part of the system square-impulse response (reflections/contributions) is treated as "signal", and the remaining part as "noise", thereby defining a programme-dependent S/N ratio [27]. This is normally implemented for an assumed exponential decay.

5.6 Gain-Before-Feedback Analysis

This feature of the programme assists in the analysis of gain before feedback. It takes into account various microphone types, and is mainly intended for near-free-field conditions. Under reverberant conditions, a simple calculation is done as a function of direct-to-reverberant ratio and reverberation time, with a frontal-to-random parameter, as appropriate.

A choice of various microphone types is given to the user, including omnidirectional, cardioid and figure-of-eight. Pick-up levels are computed by noting the direct-field contributions from each loudspeaker, in the various octave bands of interest, after applying the necessary microphone attenuation. Where long delays are present, the user may also obtain an ETC of the pick-up level, for analysis. A sample is given in Figure 8, for 44 loudspeakers, with one open cardioid microphone. Integrated levels are also given in the 50-100ms range.

Based upon the speaker level and choice of an appropriate margin, the operating level can then be set as required, thereby defining the expected SPL values in the coverage area. This routine proved particularly useful in cases where the microphone was required to be located in the midst of the listening audience, with the further constraint of an excessive talker-to-microphone distance.

SPEECH INTELLIGIBILITY: COMPUTER-ASSISTED PREDICTIVE ANALYSIS

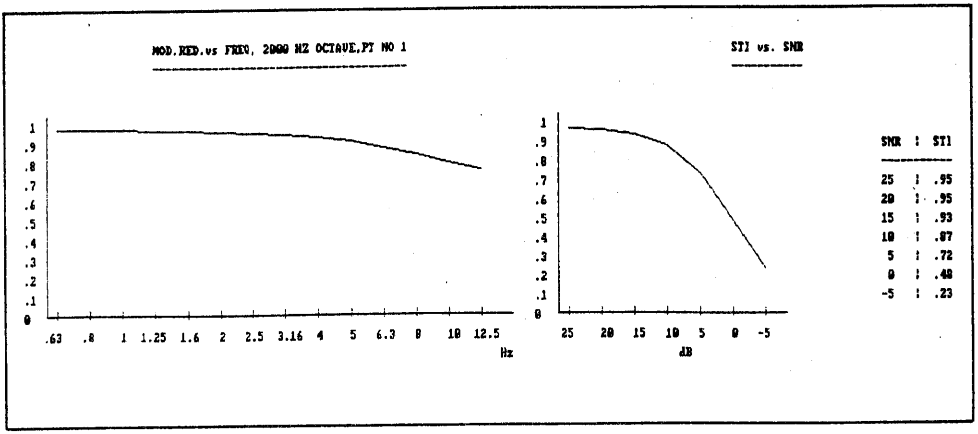
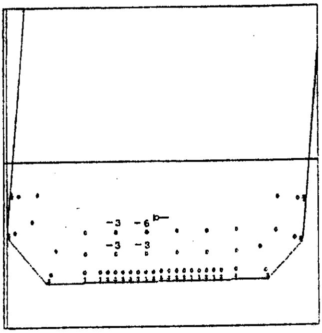


Fig. 7 Sample of MTF & STI Print-out



MICROPHONE Cardioid

ATTENUATION Low Attenuation Setting

REMARKS Attenuation Settings of Individual Units Shown at Aiming Point Indications

‡ Loudspeaker (cluster)

o Aiming Point

MIC POSITION: X= 82.6513 Y=-24.40
MIN DELAY IS FROM SPEAKER NO.: 34 = 85.30491 ms

SPKR NO.	L500	L1000	L2000	DELAY(ms)
1	60.65	57.46	57.25	4.45
2	53.34	52.87	50.73	4.45
3	47.05	46.27	38.42	4.45
4	46.82	41.64	39.07	29.24
5	43.95	41.57	41	29.24
6	37.60	34.98	28.63	29.24
7	44.97	39.77	34.91	65.13
8	40.35	37.84	34.05	65.13
9	34.15	31.28	21.51	65.13
10	69.79	66.35	64.49	3.55
11	64.29	61.98	59.62	3.55
12	61.82	58.35	58.47	3.55
13	62.27	58.41	55.51	25.06
14	60.23	56.54	57.52	25.06
15	56.84	54.91	48.43	25.06
16	63.73	57.72	53.67	60.08
17	59.76	57.01	54.27	60.08
18	52.34	50.38	42.24	60.08
19	40.41	37.54	35.93	108.98
20	32.23	29.64	22.7	108.98
21	49.27	46.78	42.57	147.05
22	38.1	35.27	28.42	147.05
23	58.91	56.09	54.63	108.94
24	50.67	40.18	41.51	108.94
25	65.29	64.69	60.5	155
26	54.13	53.19	46.35	155
27	45.43	44.17	37.53	8.95
28	42.7	41.44	34.60	14.71
29	39.38	37.22	30.81	21.54
30	37.02	33.9	27.2	37.64
31	37.02	33.9	27.2	37.64
32	36.1	32.94	25.35	46.62
33	49.4	46.6	38.71	1.42
34	49.90	46.73	38.84	0
35	60.74	57.52	49.63	1.61
36	59.51	58.77	50.96	7.04
37	58.95	57.75	51.1	11.92
38	57.49	56.29	49.59	17.99
39	55.91	53.4	47.05	32.94
40	55.12	52.22	45.65	41.40
41	49.56	46.7	42.45	150.75
42	38.41	35.17	28.36	150.75
43	67.60	64.79	60.54	150.48
44	56.54	53.26	46.46	150.48

TOTAL dB LEVEL CONTRIBUTION BELOW 50 ms :
73.64 70.79 68.07
TOTAL dB LEVEL CONTRIBUTION ABOVE 50 ms & BELOW 100 ms
65.52 60.06 57.18
TOTAL dB LEVEL CONTRIBUTION ABOVE 100 ms:
70.43 68.43 64.29

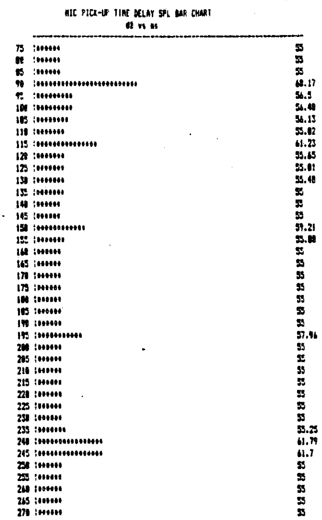


Fig. 8 Sample of Microphone Pick-up Level Analysis

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5.7 Loudspeaker Parameters

(a) Loudspeaker Level Attenuation: it is possible here to attenuate the operating level of some loudspeakers, as suited to specific requirements. In feedback analysis, for example, this may be advantageous due to relative microphone proximity. Figure 8 gives an example, where the selected settings are actually shown, for a specified open microphone location.

(b) Electronic Delay: it is possible here to incorporate artificial delay for a group of loudspeakers, the parameter being changed to obtain some desired objective. This feature proved useful in some study areas, where delay-annoyance was inevitable, and thereby enabled its reduction to a reasonable minimum. This could also be useful in the speech intelligibility analysis of some systems, as suited to the method of analysis selected by the user.

(c) Equalisation: it is possible here to include octave equalisation settings to a set of loudspeakers, which is sometimes useful in the analysis of distributed systems under near-free-field conditions, involving highly frequency-dependent overlapping patterns. This enables an assessment of the expected quality of frequency response in the coverage area, and as to whether or not equalisation can actually achieve the flatness response required.

6. CONCLUSION

A number of computer-assisted packages are now available on the market, and these should be seriously considered by professionals involved in the analysis and design of sound systems, prior to considering in-house development of their own programmes.

In-house development of such facilities must be justified. Involvement in projects with severe constraints may provide an incentive towards such a consideration, however caution should be exercised against "over-designing" such systems.

A-CADX has been developed with the express aim of enabling a multitude of methods of analyses to be considered by the user where it is appropriate to do so. The plans for the future include designing an interface to a powerful graphical package, in addition to statistically ray-tracing. Other features being considered for future incorporation include: subjective articulation test analysis; multi-way loudspeaker analysis and optimisation; noise and vibration analysis.

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