

Table I: SNR Contrast for Random noise removal.

Sl. No	Sample No	Before Filtering	LMS		Kowngs VSSLMS		RVSSLMS		MRVSSLMS	
			After	Imp	After	Imp	After	Imp	After	Imp
1	I	0.7523	5.9077	5.1553	6.5145	5.7621	9.0738	8.3214	10.1064	9.3542
2	II	-2.1468	4.1468	6.6975	5.7103	8.2610	6.6617	9.2154	7.9232	10.473
3	III	-4.1554	1.4826	5.6380	1.539	5.6944	3.1546	7.3100	4.7609	8.9163
4	IV	-3.6941	1.9213	5.6154	2.0417	5.7358	3.5682	7.2623	5.1431	8.8372
5	V	-5.6992	0.5443	6.2435	2.3337	8.0329	2.6920	8.3912	3.8539	9.5531
Average Improvement				5.8699		6.6972		8.1000		9.4269

Table II: SNR Contrast for High voltage murmuring removal.

S.No	Sample No	Before Filtering	LMS		Kowngs VSSLMS		RVSSLMS		MRVSSLMS	
			After	Imp	After	Imp	After	Imp	After	Imp
1	I	-1.5937	2.0034	3.5971	3.0735	4.6672	4.2078	5.8015	4.6311	6.2248
2	II	0.0705	1.7646	1.6940	1.9657	1.8951	5.9283	5.8577	6.5044	6.4338
3	III	2.6032	4.3508	1.7476	5.5225	2.9193	7.4302	4.8270	7.9161	5.3129
4	IV	3.0644	4.9673	1.9029	6.6277	3.5633	7.4096	4.3452	8.5129	5.4485
5	V	0.9671	2.8560	1.8888	3.0758	2.1086	7.1156	6.1484	7.9817	7.0145
Average Improvement				2.1660		3.0307		5.3959		6.0869

Table III: SNR Contrast for Crane noise removal.

S.No	Sample No	Before Filtering	LMS		Kowngs VSSLMS		RVSSLMS		MRVSSLMS	
			After	Imp	After	Imp	After	Imp	After	Imp
1	I	0.5244	3.2108	2.6863	4.1024	3.5770	4.2822	3.7577	4.6914	4.1669
2	II	-1.8459	3.2714	5.1173	5.7327	7.5786	6.0373	7.8832	6.7004	8.5463
3	III	-2.1790	3.3691	5.5481	4.2556	6.4346	4.3284	6.5074	4.9409	7.1199
4	IV	-1.6394	2.3560	3.9954	4.2422	5.8816	4.4689	6.1083	5.1134	6.7528
5	V	-3.6823	0.7695	4.4518	4.9700	8.6523	5.8311	9.5134	6.7282	10.410
Average Improvement				4.3597		6.4250		6.7540		7.3993

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