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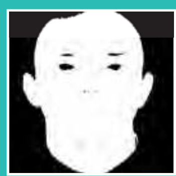

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No Room for Complacency

ENVIRONWISE ON-LINE DESIGN FORUM – page 38

It's been a tough year or so, for companies and employees alike. As the credit crunch took further bites from the economy, those firms who are still surviving are leaning even more heavily on its workforce. And employees have been responding with a genuine will to help, and not only so to save their jobs, but to save their companies too. And it's everywhere: employees are taking on more responsibilities, doubling up their workloads, and in many cases doing the jobs of two. This has caused many to extend their working hours, and at a time when companies have curtailed even the most minimal of annual pay-rises.

We see that firms can rely on their employees, but when times change employees should see better returns as companies reciprocate.

So it's kind of worrying when reports such as '*Agility in Services – Capabilities for Difficult Times*' by the Institute of Customer Service, in association with the Advanced Institute of Management Research (AIM Research), claim that in the UK, organisations show a big shortfall in agility. The study says that fewer than one in four UK organisations are fully equipped to react to changes in today's fast changing, competitive marketplace; only 22% of organisations have the necessary 'agility' to respond quickly to customer demands, adapt to solve problems or offer flexibility in their approach. Overall the study shows that agile organisations have higher productivity, sales growth and satisfaction than non-agile organisations.

The study reinforces the customer service model that requires all aspects of an organisation – strategy, culture, people and processes – to be focused on, to enable rapid response to the marketplace and customer demands.

As Jo Causon, Chief Executive of The Institute of Customer Service, says: "Customer service is the only sustainable competitive differentiator in the marketplace today. It's not just about surviving the recession, but making UK plc more competitive in the global marketplace going forward. Being customer-focused is not enough, organisations need to be agile across multiple fronts."

So if a company cannot find a way to respond to a changing marketplace; find a way to cater for customers in the most possible manner, effectively securing its own future; then how is it going to look after its employees properly?

It's time to adjust business practices – and there is no room for complacency.

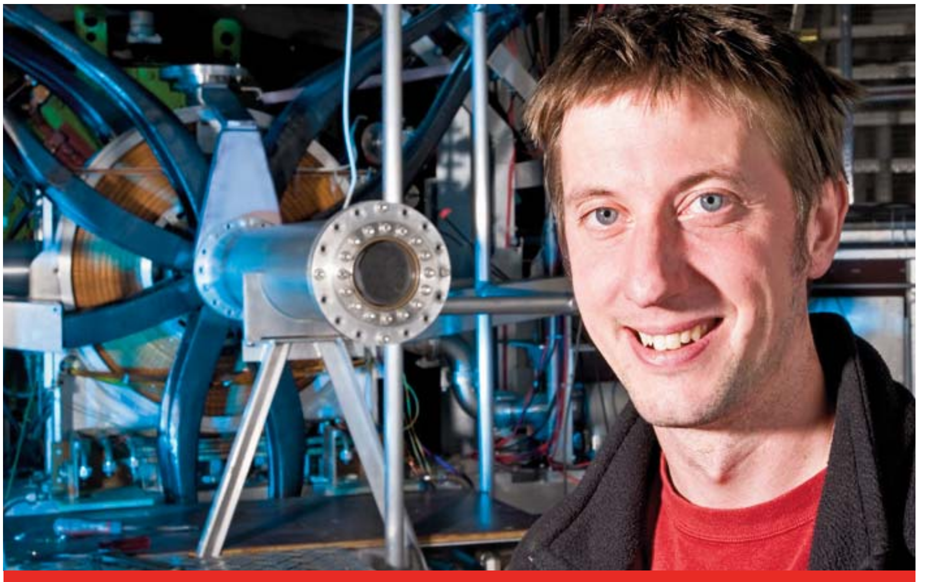
Editor
Svetlana Josifovska

'Magnetricity' Observed and Measured for the First Time at ISIS

Magnetic charge can behave and interact just like an electric charge in some materials, according to a new research conducted at STFC's ISIS facility and led by the London Centre for Nanotechnology (LCN).

The research is claimed to prove the existence of atom-sized 'magnetic charges' that behave and interact just like more familiar electric charges. It also demonstrates a perfect symmetry between electricity and magnetism – a phenomenon dubbed 'magnetricity' by the authors from the LCN and the Science and Technology Facility Council's ISIS Neutron and Muon Source. The findings could lead to a reassessment of current electromagnetism theories, as well as significant technological advances.

"This is a very important step to establish that magnetic charge can flow like electric charge. It is in the early stages, but who knows what the applications of magnetricity could be in 100 years' time," said Professor Steve Bramwell, part of the LCN team.



Dr Sean Giblin in front of the EMU instrument at ISIS where the magnetricity experiments were conducted

In order to prove experimentally the existence of magnetic current for the first time, the team mapped Onsager's 1934 theory of the movement of ions in water onto magnetic currents in a material called spin ice. They then tested the theory by applying a magnetic field to a spin ice sample at a very low temperature and observing the process using muons at ISIS.

This allowed the team to detect magnetic charges in the spin ice ($Dy_2Ti_2O_7$), to measure their currents and to determine the elementary unit of the magnetic charge in the material. The monopoles they observed arise as disturbances of the magnetic state of the spin ice and can exist only inside the material.

"The results were astounding, using muons at ISIS we are finally able to confirm that magnetic charge really is conducted through certain materials at certain temperatures – just like the way ions conduct electricity in water," said Dr Sean Giblin, instrument scientist at ISIS.

"The unequivocal proof that magnetic charge is conducted in spin ice adds significantly to our understanding of electromagnetism. Whilst we will have to wait to see what applications magnetricity will find in technology, this research shows that curiosity driven research will always have the potential to make an impact on the way we live and work," added Professor Keith Mason, Chief Executive of STFC.

ARGONNE SCIENTISTS FIND NEW SET OF MULTIFERROIC MATERIALS

Scientists from Argonne's materials science division and Center for Nanoscale Materials, along with scientists from Pennsylvania State University, University of Chicago and Cornell University, used piezoresponse force microscopy, optical second harmonic generation and magnetometry to show ferroelectricity at and below room temperature and weak ferromagnetism below 120 Kelvin for polycrystalline $FeTiO_3$ synthesized at high pressure.

The effort took place after Argonne scientist Craig Fennie's principles of microscopic materials design predicted that the high pressure form of $FeTiO_3$ would have both weak ferromagnetism and ferroelectric polarization, an unusual combination in a single material.

Multiferroic materials (such as $FeTiO_3$) show both magnetism and polar order, which are seemingly contradictory properties. Magnetic ferroelectrics may have applications in memory, sensors, actuators and other multifunctional devices by acting as magnetic switches when their electric fields are reversed.

Argonne scientist John Mitchell assembles the high pressure furnace used in creating the multiferroic properties of $FeTiO_3$

Funding for the project came from the US Department of Energy, Office of Science.

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Another Radio Specification To Be Worried About – INTERMOD

Myk Dormer

WHEN SPECIFYING a radio device, there are already a fair few parameters to worry about: sensitivity and maximum transmitter power, which define range; baseband path bandwidth, which limits data rate; switching times; and the rejection parameters (adjacent, discrete spurious responses and blocking), which characterize how robust the system will be in the face an interfering signal.

Unfortunately, there is another effect to worry about.

When two signals of sufficient amplitude are applied to a non-linear device, a multiplication of sine-functions, or 'mixing', occurs. If we simplify into frequency domain terms (naming the higher frequency f1 and the lower f2) we will get the following: First order products at the sum and the difference of the two carrier frequencies:

$$fs = f1 + f2$$
$$fd = f1 - f2$$

This is the effect employed in the frequency mixers, or up/down converters, necessary to the operation of superheterodyne receivers.

Second order products – of considerably lower power – appear (as a result of the fs term mixing with the fundamentals) at:

$$fh = f1 + (f1-f2)$$
$$fl = f2 - (f1-f2)$$

Third order, and higher, terms are also generated, as a comb of progressively calculated sum and difference products of ever decreasing amplitudes.

But these second order products are the ones that pose a particular problem. Imagine that the receiver is subjected to two interfering signals. Assuming that the amplitudes are below the blocking threshold, and neither interfering frequency falls on a spurious response, then no trouble is to be expected?

Not necessarily. If the frequencies of the two interferers are such that the second order product (fh or fl) falls on the wanted channel frequency, and the amplitude of the interferers is high enough to cause the receiver circuits to exhibit unwanted non-linear behaviour (usually in the LNA or first mixer), then another source of interference has appeared. This is Intermodulation.

To give a real world example (from the UK industrial narrow-band allocation):

Signal frequency: 458.600MHz

Interferer 1 at 458.650MHz, interferer 2 at 458.700MHz

Interferer 1 is on the 'alternate' channel. A competently designed receiver on this band can easily achieve a rejection of 70dB at this offset. Interferer 2 is further off-carrier, and over 80dB rejection is expected here.

But 458.700MHz – 458.650MHz = 50kHz and 458.650MHz – 50kHz = 458.600MHz

"IN RADIO ENGINEER FOLKLORE, THERE ARE REPORTS OF INCIDENTALLY FORMED NON-LINEAR JUNCTIONS IN METAL STRUCTURES ('RUSTY BOLT' INTERFERENCE) GIVING RISE TO INTERMODULATION EFFECTS"

On increasing the unwanted signal amplitudes, the intermodulation product of the two interferers will have swamped the wanted signal long before the receiver's rejection limit is reached.

Typical high specification designs (EN 300-113 PMR radios, or 300-220 Class 1 units) meet a 65dB intermodulation specification, while exceeding 70dB is the province of fairly specialized designs (high quality shortwave sets, military radios, RF test gear).

Regrettably, most low-power ISM band radio designs fall far short of these figures. 55dB is typical of FET mixer based units, while simple bipolar designs or single-chip integrated receivers are frequently worse than 40dB.

At sufficiently high signal levels, intermodulation can occur in

"GAIN DISTRIBUTION ALONG THE RECEIVER SIGNAL PATH MUST BE CAREFULLY PROFILED, TO LOCATE AS MUCH OF THE SYSTEM GAIN AS POSSIBLE AFTER THE FILTERING STAGES"

the output stages of two co-sited transmitters, at which point the interfering signal is irrelevant of the receiver specification altogether. In radio engineer folklore, there are also reports of incidentally formed non-linear junctions in metal structures ('rusty bolt' interference) giving rise to intermodulation effects.

Realizing good intermodulation performance takes considerable care. The linearity of the circuits ahead of the intermediate filters must be maximized, with particular attention given to the front end amplifiers and the mixer.

Gain distribution along the receiver signal path must be carefully profiled, to locate as much of the system gain as possible after the filtering stages. Compromises must be made between minimum noise (requiring high LNA gain) and maximum linearity, and the relevant circuits will need as much supply current as the designer can afford it.

Intermodulation has an influence on the overall system design, too. Since it is not practical to design receivers with intermodulation figures as good as their blocking specs, the placement of occupied channels in a multiple transmitter band-plan must avoid – as much as possible – the intermodulation product of any two transmitters falling on the channel occupied by another. Where only a few frequencies are considered this is an easy task, but as the number of transmitters rises, the task complexity escalates out of control.

There are software vendors who base their entire business on the supply of tools to plan such bands.

To conclude: this should not be taken as a council of despair. Despite intermodulation and all other performance limitations and effects, radio links function successfully in their millions. What is necessary is to remain aware of this potential effect, and design to minimise it.

When selecting radio hardware, be aware of this specification among all the others, and remember: If the manufacturer isn't mentioning it, then it's likely to be embarrassingly poor.

Myk Dormer is Senior RF Design Engineer at Radiometrix Ltd
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Steve Mowry, an independent researcher and consultant in transducer, loudspeaker, headphone design and new product development, explains the role of signal processing in active loudspeaker designs

Active Loudspeakers and DSP: A Simplified PATH TO PERFECTION

IN THE VAST majority of loudspeakers, the low level output signal from a CD or DVD player is the input to a receiver or preamplifier (control centre), with the output from the preamplifier being the input to the power amplifier(s) ~ 1.0 V. In these cases, each loudspeaker is connected to the power amplifier's output terminals using a single pair of "cables".

However, between the power amplifier and the audio transducers, there is a passive RLC network. The power amplifier sees the RLC network and the audio transducers (see **Figure 1**). To implement a reasonable crossover and any equalization and/or impedance compensation is known to be a difficult task that usually becomes an empirical process (build and try) at some point. Degradation and variability of the high level audio signal is simply unavoidable. One alternative is a full range audio transducer that eliminates the need for a crossover network but not for equalization and/or impedance compensation.

ACTIVE LOUDSPEAKERS

Another alternative is active loudspeakers. In these cases, multiple power amplifiers are used and the crossover(s) and equalization are implemented electronically at low level and typically between the preamplifier and the power amplifier with regards to the audio signal path. This is sometimes referred to as "Bi-amp" or "Tri-amp" for active two-way and three-way loudspeakers respectively.

The power amplifiers are connected directly to the loudspeakers, with cables but in multiple pairs. Signal processing, including but not limited to crossover, equalization and phase compensation, can be implemented in the analogue (continuous time) domain, typically using RC networks and operational amplifiers, or in the digital (discrete time) domain using software.

Some companies have implemented operational amplifiers using discrete components or even valves to implement low-level operational amplifiers. The fact is that, unlike in the analogue domain, in the digital domain the signal is a number and that number can be preserved and/or processed to a high precision.

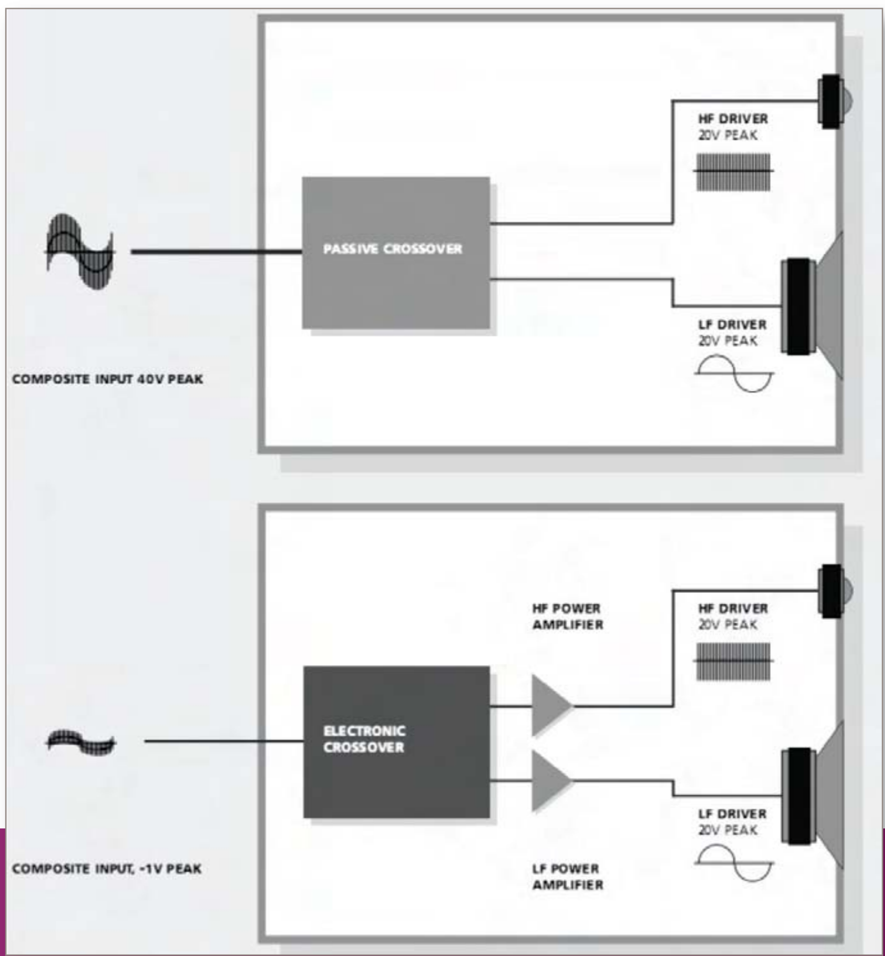


__Figure 1:__ Block diagrams of passive (upper) and active (lower) loudspeakers

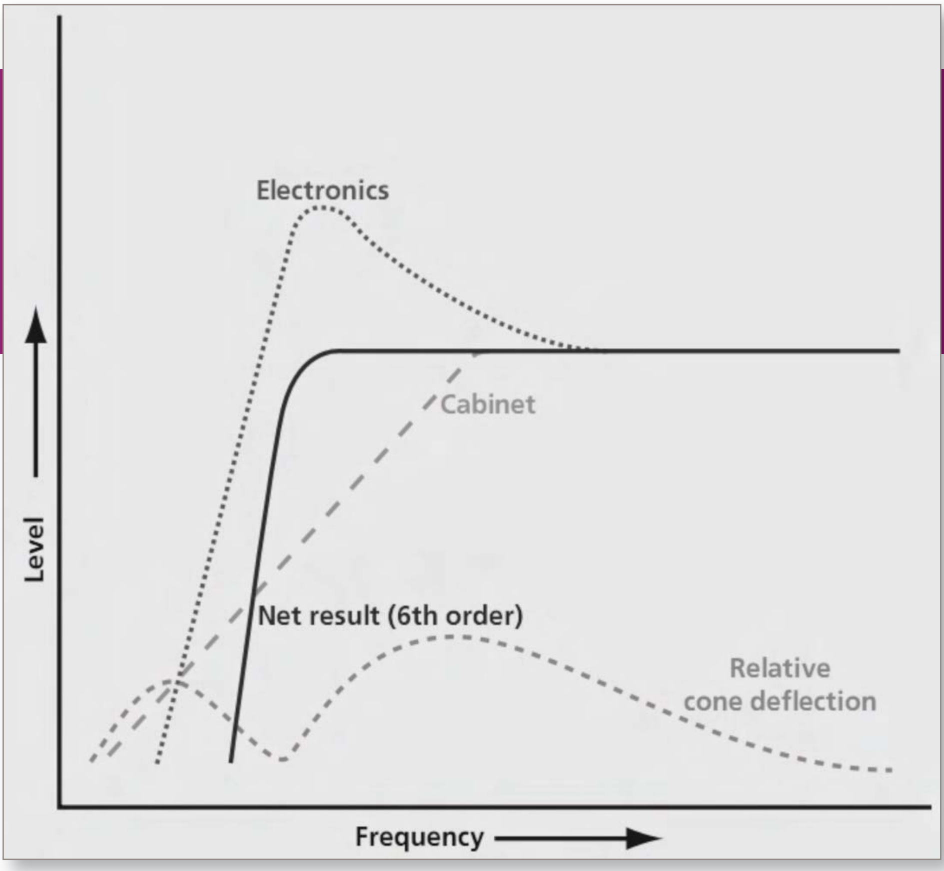
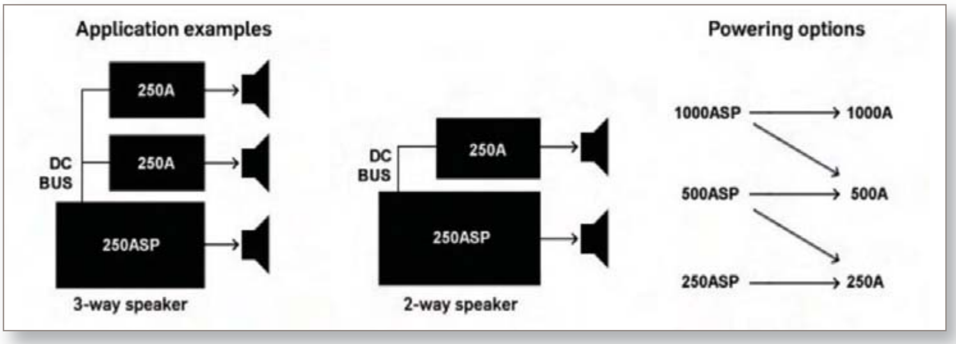


Figure 2 (left): *Illustration of an actively equalized ported box woofer response*
Figure 3 (below left): *Illustration of the configuration of ICEpower modules for 3-way and 2-way loudspeakers*
Figure 4 (below): *The Meridian DSP8000 active loudspeaker*



Although an analogue active crossover can result in significant loudspeaker performance gains, implementing digital signal processing (DSP) essentially eliminates any degradation of the low level signal and adds levels of flexibility and capability to signal processing that are just not possible or cumbersome for analogue implementations with the inherent degradation of the analogue signal as the complexity of signal processing is increased.

With that in mind, what if the DSP setup was via user-friendly Windows software allowing implementations of relative gain, time delay, filters, limiters, and/or expanders with the mouse on the user's PC? And what if the user could import loudspeaker frequency response LMS acquisitions or equivalent that can be used to tailor filtering and/or equalization to obtain some target response, typically flat? This can and, in the limit, should be the 'in room' response and/or the 'in room' power response (an averaging of acquisitions at predefined on and off axis microphone positions throughout the listening room).

Finally, the user has the option to implement signal processing that best suits drivers, amps and enclosures in the listening room. DSP could also be used to equalize and/or filter and/or phase compensate a full-range (single transducer) loudspeaker, which would then only require one power amplifier per channel. In the limit and with respect to "less is more", but requiring a very wide bandwidth transducer, the later could result in very high performance with regards to active loudspeakers, because the power amplifiers do not see RLC networks and each amp is band limited to the bandwidth of the respective transducer. The result is that the amplifiers are used more efficiently and frankly, more effectively with a reduction in intermodulation and harmonic distortion. The passive RLC crossover network effectively disconnects the transducer from the power amplifier for out-of-band frequencies, leaving the transducers' cones and domes free to vibrate passively, driven by the relative sound pressure.

Now, if the amplifiers are compact and reasonable mass, then the DSP electronics and the amps can be mounted within the loudspeaker enclosure. This simplifies and effectively reduces the number of components within the audio system et al. Only the input components such as FM tuner, CD/DVD player, TV/monitor and/or PC or MAC and/or a preamplifier with low level out and these may include Home Theater DSP of its own, along with the active loudspeakers are all that is needed to complete the A/V system. Active loudspeakers facilitate enclosure design and alignments that consider the electro, mechanical and acoustic parameters.

Figure 2 illustrates a 6th order high-pass woofer response. The transducer is 2nd order high-pass; the ported box is 2nd order high-pass and the active filter is also 2nd order high-pass. The net result of the cascading of

the electro, mechanical and acoustic filters is 6th order high-pass response. The same basic methodology can be applied to an equalized sealed box with the result being 4th order high-pass response, without the tuning null in the displacement magnitude.

It would seem to be convenient to use an efficient class of power amplifier such as Class-D. This would limit power supply size and heat. Although, two or three amplifiers are required for 2-way and 3-way loudspeakers respectively, a single high-performance switching power supply can be utilized for the power amplifiers et al. One company stands out in the development of high-performance Class-D power amplifier modules: B&O ICEpower. The ASP modules include 115/230V @ 50/60Hz power supply (**http://www.icepower.bang-olufsen.com/en/solutions/speaker/ASP_SERIES**), while the A series modules come without power supply but utilize the power supply from the ASP module (**http://www.icepower.bang-olufsen.com/en/solutions/speaker/A_SERIES**).

Other amplifiers can be used, but the ICEpower modules offer high power ratings with the highest performance and very low output impedances, and the ability to comfortably drive a 2-Ohm load. This offers further flexibility in transducer design and/or selection.

Using very high power amplifiers (mega-amps) with very low output impedance (high damping factor) and low noise and distortion, connected directly to the loudspeaker transducers, represent the best possible system topology. Then the utilization of DSP for crossover and EQ et al bring the performance potential to the very highest levels.

One company that has been committed to active DSP loudspeakers since 1991 and has emerged as a leader in the development of active DSP loudspeakers is Meridian. **Figure 4** contains a picture of the Meridian DSP8000 active DSP loudspeaker with MSRP of US$79,999/pair. When I interviewed Neville Thiele in 2005, I asked him what company he thought was performing the most important R&D and he said: "This must be a very selective estimate, but I would say Bob Stuart, Peter Craven and Rhonda Wilson at Meridian on loudspeakers and lossless digital coding."

Here's a video of Bob Stuart discussing the benefits of active DSP loudspeakers **http://uk.cinenow.com/videos/318-meridian-bob-stuart-on-digital-dsp-loudspeakers**.

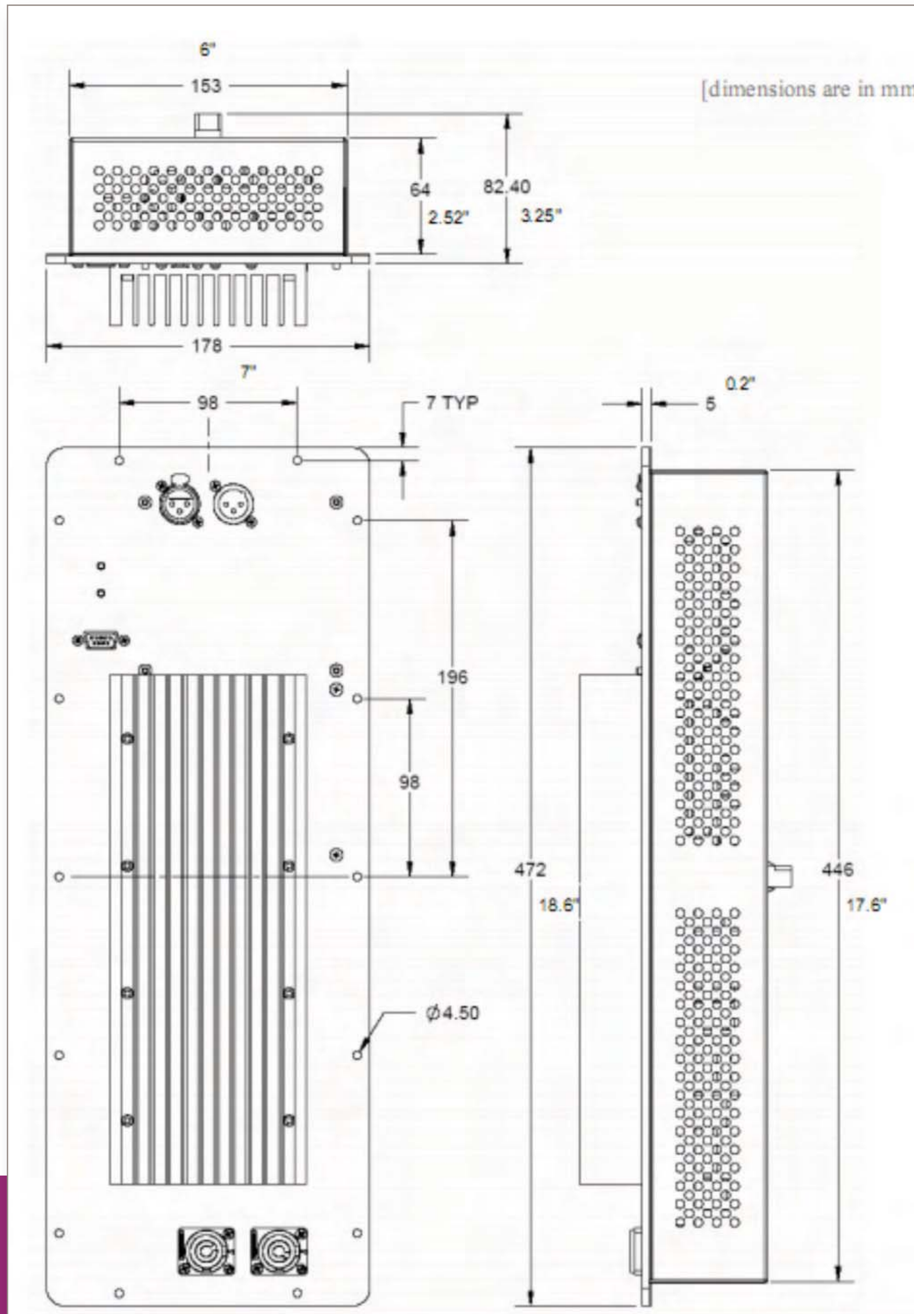
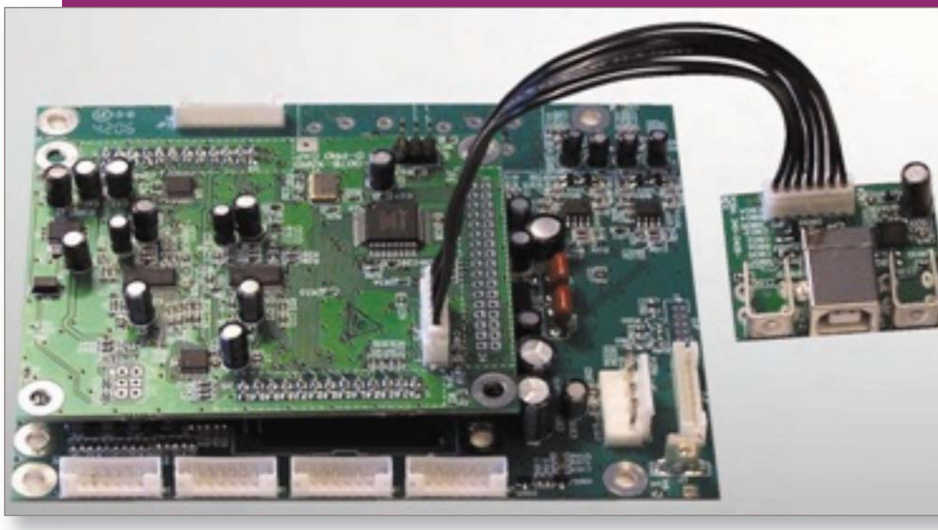
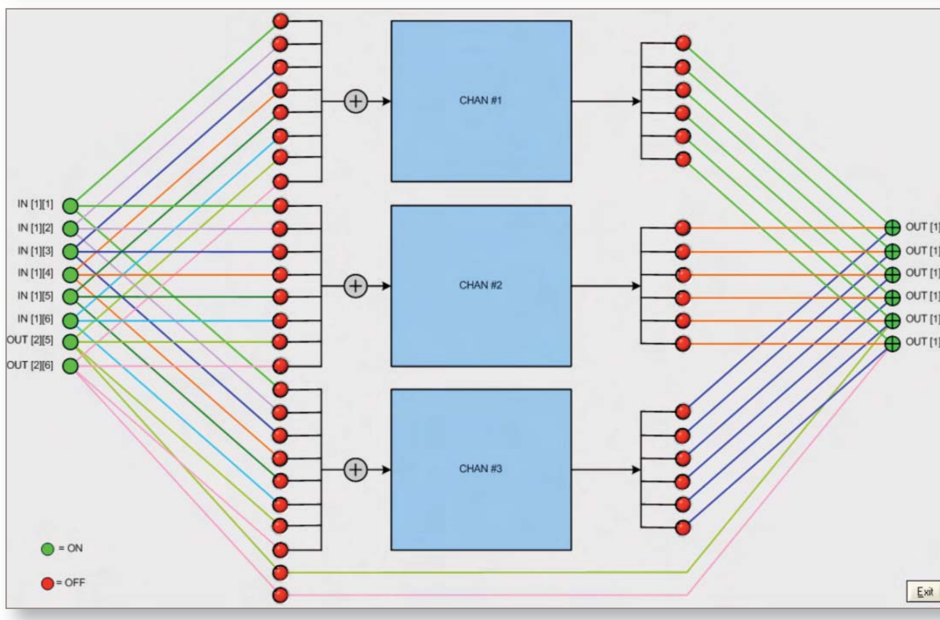


Figure 5 (above): *DIGMODA DDC-552*
Figure 6 (right): *Mechanical drawing of the DIGMODA DDC-552 with dimensions*

Figure 7 (top): *KSC DP240 DSP board (two channels in, four out) with A/D converter*
Figure 8 (bottom): *Input to outputs routing configuration screen*





Other mostly professional audio companies, such Genelec, have done some work in this area; however, most companies have yet to implement active DSP loudspeakers.

DIGMODA

KSC in San Diego, California has developed turnkey solutions to active DSP loudspeaker implementations. They have developed proprietary D-Pro Windows software and hardware DSP boards with A/D and D/A on board. The following DIGMODA models with the respective ICEpower modules are available (**http://www.digmoda.com**):

2-channel
DC-520, 500ASP + 250A
DDC-550, 500ASP + 500A
DDC-1050, 1000ASP + 500A
DDC-1100, 1000ASP + 1000A
3-channel
DDC-552, 500ASP + 500A + 250A
DDC-1150, 1000ASP + 1000A + 500A
Subwoofers
DDC-500SW, 500ASP
DDC-1000SW, 1000ASP
DSP boards with D-Pro software only
DP130, 1-channel in x 3 out
DP240, 2-channel in x 4 out

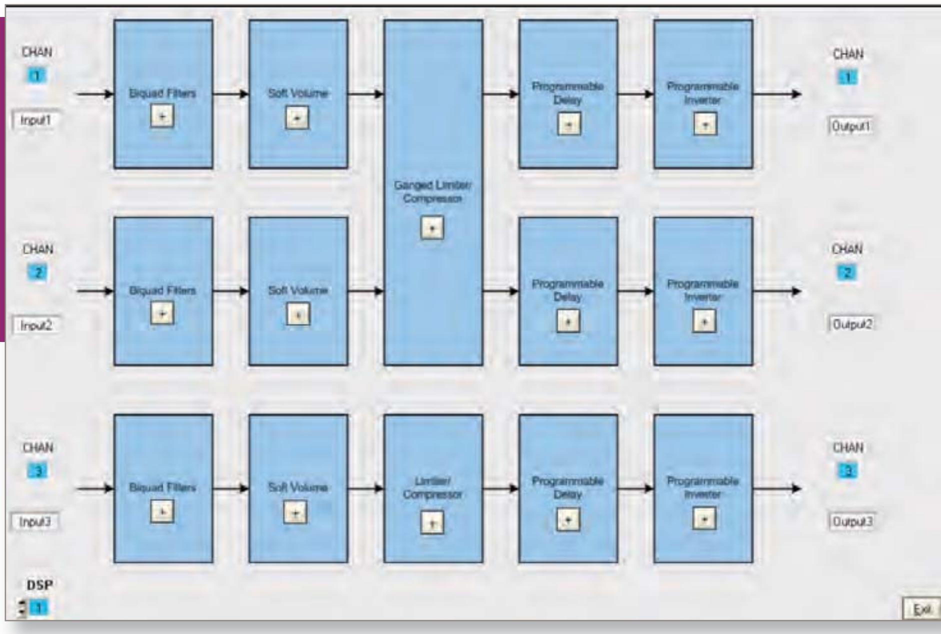
Figure 5 contains a picture of the DDC-552. The DIGMODA electronics are packaged with large integral heat-inks and a convenient mounting flange. Custom models are also available in reasonable minimum order quantities, and the DSP boards and D-Pro software are available separately, for those companies that wish to use their own amplifier designs.

A company can literally almost catch up with Meridian by implementing DIGMODA into new product development of active loudspeakers, including subwoofers and 2 and 3-way loudspeakers. The D-Pro software is friendly and good support is available from KSC and their representatives. Pricing is surprising low for such a high performance product and does depend on quantity purchased and must be quoted by KSC. In Asia and Australia/New Zealand, DIGMODA and D-Pro et al and support are available from S. M. Audio Engineering.

In addition to the performance aspects of an active DSP loudspeaker, the development of such products is simplified though the implementation of DIGMODA. So much can be done with the DSP with regards to equalization and/or compensation that the results are essentially assured. Upgrades of transducers and/or enclosures can be implemented with only changes to the software setup. Additionally, a single DIGMODA model can be used with several loudspeakers with again only a change to the software setup.

Finally, products can be brought to market more quickly and with lower development costs. Installers of custom A/V systems should find DIGMODA useful in implementing very high performance systems that can meet a wide variety of user needs. This includes, but is not limited to, in-wall and in-ceiling applications. These applications seem to also imply the DIGMODA would be an excellent solution for kit loudspeakers from small manufacturers.

Furthermore, the flexibility and capability of DIGMODA makes implementing other technologies, such as new transducers or an actively equalized trick enclosure, a reasonable task and with available support should not present any significant problems. The use of the limiter functions can save expensive transducers and eardrums and reduce warranty claims and, thus, improve product reliability.



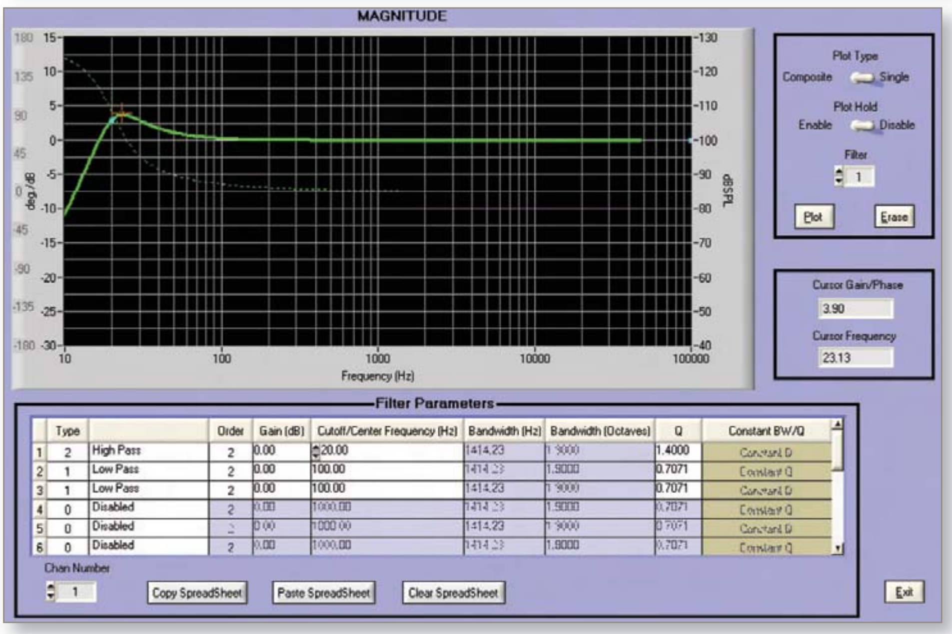


Figure 9 (top): *DSP block selection screen*
Figure 10 (middle): *Second order filter implementation screen*
Figure 11 (bottom): *Filters responses with LMS acquisitions overdrawn before EQ with and without invert tweeter*

The typical audio specifications/dimensions of the DDC-552, DDC-550 and DDC-520 are illustrated in **Figure 6**.

For those who want to use amps other than the ICEpower, the KSC's DSP Boards DP130 and DP240 are available. The DP130 is one channel in, three out, and the DP240 (**Figure 7**) is two channels in, four out.

D-PRO SOFTWARE

D-Pro software utilizes virtual instruments that are intuitive and easy to use. The user wires the DSP by connecting the input to the DSP channels and selecting the outputs with the PC mouse. The D-Pro software is typically downloaded from KSC's FTP server. Once D-Pro is installed (Windows XP recommended), the user connects his PC to DIGMODA with a common USB printer like interconnect.

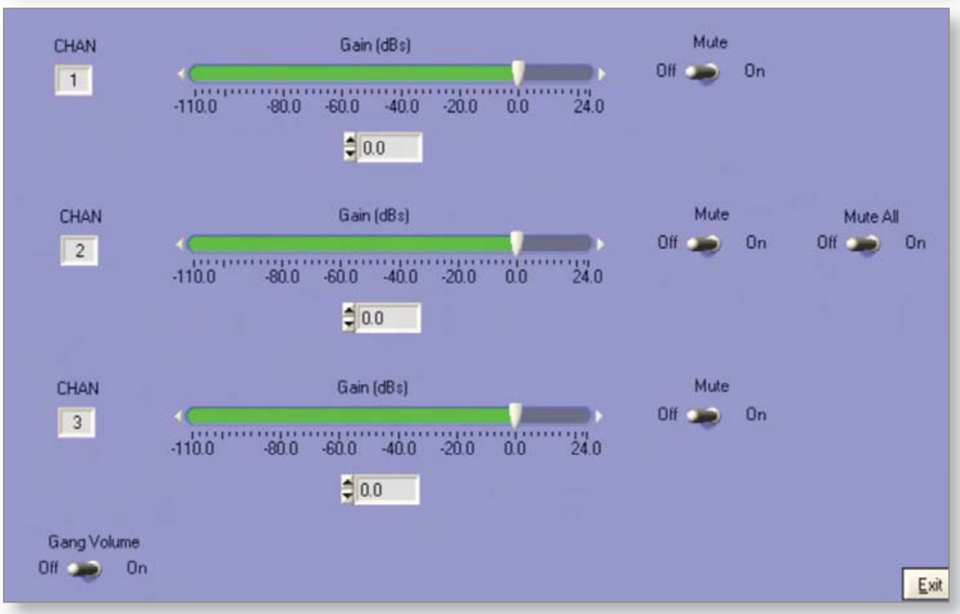
There are two main screens the require setup. The first screen is shown in **Figure 8**. Here the user wires the inputs with respect the DSP channels and outputs. The second screen contains the DSP blocks as shown in **Figure 9**. Clicking on the respective DSP block opens a control screen. The control screens for each block and sub-block are shown in **Figures 10, 11, 12, 13, 14, 15** and **16**.

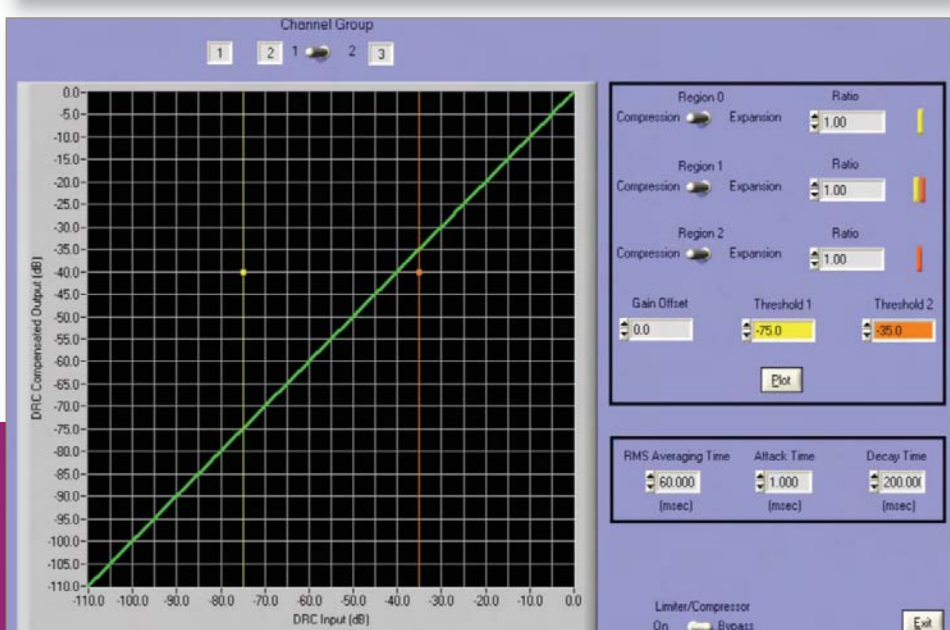
Figure 10 shows the user-selected FIR 2nd order filter setup screen. Up to 16 filters are available for each channel. The filter types are selected by number code:

1 = Low Pass
2 = High Pass
3 = Band Pass
4 = All Pass
5 = Parametric EQ (peak or notch)
6 = Bass Shelf
7 = Treble Shelf

Then either enter Gain, Frequency and Q or use the mouse to drag the curve to the desired filter response.

'In room' frequency response acquisitions can be utilized as an interim evaluation, as the appropriate filters are determined and applied with the PC mouse to equalize the





response with respect to some target response curve. Figures 11 and 12 show LMS response acquisitions before and after EQ and with tweeter phase invert.

The Gain Control, shown in Figure 13, allows tweaking the relative levels of transducer input signals.

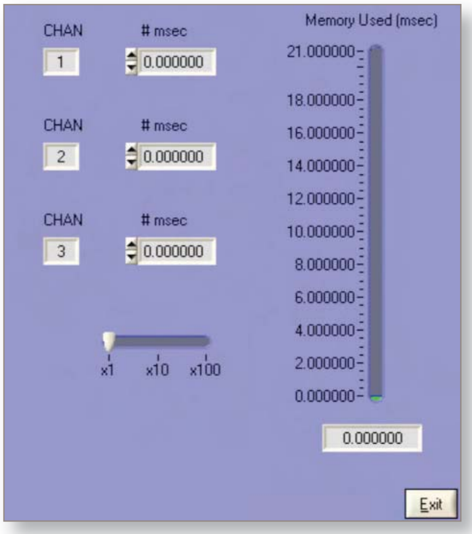
Figure 14 shows an interesting screen. This reminds me of the old DBX 3BX dynamic range expander that was sold from 1984-1986. These were similar to open-ended noise reduction but also allowed high power delivery to the loudspeaker on transient peaks. They were based on analogue compander chips.

Perhaps a more common use of these controls is to implement a gated limiter such that the transducer(s) will be protected dynamically, especially the tweeter. However, the possibilities from this control screen seem limitless.

Figure 15 shows the screen for time alignment that can be adjusted with the PC mouse using on-board memory. Each screen setup is independent of another. Once then setup is completed, a binary file containing the setup can be saved. D-Pro can read and write setup files. Suddenly, passive networks seem primitive and analogue electronics seem cumbersome. DIGMODA is an exciting and innovative product that is "fun" to use. Additionally, DIGMODA brings the A/V system closer to a convergence system that utilizes the family PC as a powerful but easy to use signal processing including effects and along with the Internet as sources of program material.

Figure 16 shows the controls for relative phasing of the transducers that can be implemented with the PC mouse.

Figure 12 (top): *Filters responses with LMS acquisitions overdrawn after EQ*
Figure 13 (middle): *Gain control screen*
Figure 14 (bottom): *Dynamic compression-expansion implementation screen*



OTHER PERFORMANCE CONSIDERATIONS

With state-of-the-art electronics now available in an easy-to-use turnkey solution, the requirements for transducers must also be at a high standard to realize the highest performance. Frankly, the transducers are still the weak link in the performance chain of an A/V system. S. M. Audio's solutions are STEALLUS X woofer, STEALLUS mid/bass and STEALLUS Air Core tweeter. However, there are several manufacturers and distributors of other high performance transducers.

High performance engineering materials can also play a role in the quest for A/V perfection. Diaphragm and cone materials such as Beryllium foil, Magonite (Keronite process applied to Magnesium) and sandwich composites can reduce mass and/or eliminate frequency-related bending. Carefully designed molded TEEE, treated Acrylic, and Nomex surrounds and spiders can improve suspension linearity.

Klippel GmbH is engaged in nonlinear system R&D regarding a proprietary chip that would compensate for nonlinearity in the large signal parameters Bl(x), Le(x,i), Kms(x), along with the thermal characteristics of the loudspeakers' transducers. First, these large signal parameters are acquired with a KLIPPEL DA and then the chip is "taught" these dynamic nonlinearities. A nonlinear correction and/or compensation signal is then convolved with the low-level input signal. The result is dynamic reduction in IM and THD at the root cause, the transducers' motors and suspensions. ■

Audio Specifications

Parameter	Conditions	Typical	Maximum	Unit
THD+N in 4Ω (AES17 filter)	f = 40 Hz, Po = 1W	0.009	0.011	%
	f = 500 Hz, Po = 1W	0.025	0.031	
	f = 1 kHz, Po = 1W	0.019	0.024	
THD+N in 8Ω (AES17 filter)	f = 40 Hz, Po = 1W	0.007	0.009	
	f = 500 Hz, Po = 1W	0.016	0.020	
	f = 1 kHz, Po = 1W	0.013	0.016	
Maximal THD+N in 4Ω (AES17 filter) -10dBFS	f = 40 Hz	0.006	0.008	%
	f = 500 Hz	0.008	0.010	
	f = 1 kHz	0.010	0.013	
Maximal THD+N in 8Ω (AES17 filter) -10dBFS	f = 40 Hz	0.006	0.008	
	f = 500 Hz	0.008	0.010	
	f = 1 kHz	0.010	0.013	
THD+N Low Level in 4Ω	f = 40 Hz, 100 mW	0.02	-	%
	f = 500 Hz, 100 mW	0.04	-	
	f = 1 kHz, 100 mW	0.07	-	
THD+N Low Level in 8Ω	f = 40 Hz, 100 mW	0.02	-	
	f = 500 Hz, 100 mW	0.04	-	
	f = 1 kHz, 100 mW	0.05	-	
Nominal Voltage Gain (A_v)	f = 40 Hz	29.92	-	dBV
	f = 500 Hz	29.97	-	
	f = 1 kHz	29.98	-	
Frequency Response (Po = 1W)	f = 20 Hz - 100 kHz	±0.5	±1	dB
	f = 100 Hz - 2 kHz	±0.5	±1	
	f = 2 Hz - 20 kHz	±0.5	±1	
Common Mode Rejection (CMR)	20 Hz - 200 kHz	-55	-	dB
Signal to Noise Ratio in 4Ω (SNR)	0db=1% THD, 40 Hz	-135	-	dB
	0db=1% THD, 550 Hz	-125	-	
	0db=1% THD, 1 kHz	-110	-	
Signal to Noise Ratio in 8Ω (SNR)	0db=1% THD, 40 Hz	-136	-	
	0db=1% THD, 500 Hz	-125	-	
	0db=1% THD, 1 kHz	-105	-	
Input Impedance (Z_{in})	Unbalanced	18	-	kΩ
	Balanced	36	-	
Input Sensitivity in 4Ω (Balanced) (V_{in})	f = 40 Hz, 500W	1.43	-	Vms
	f = 1 kHz, 500W	1.42	-	
	f = 40 Hz, 225W	0.99	-	
Input Sensitivity in 8Ω (Balanced)	f = 500 Hz, 225W	1.42	-	
	f = 500 Hz, 225W	1.40	-	
	f = 1 kHz, 112W	0.95	-	
DC Output Offset in 4Ω (DC_{off})	Low channel	3.0	-	mVms
	Mid channel	3.3	-	
	High channel	4.3	-	
DC Output Offset in 8Ω (DC_{off})	Low channel	9.5	-	
	Mid channel	4.0	-	
	High channel	4.3	-	
Input Clipping ($V_{in\text{-}clip}$)	f = 1 kHz	12	-	dBU
Damping Factor (DC_{dis})	f = 100 kHz, R_L = 8Ω	2000	-	-

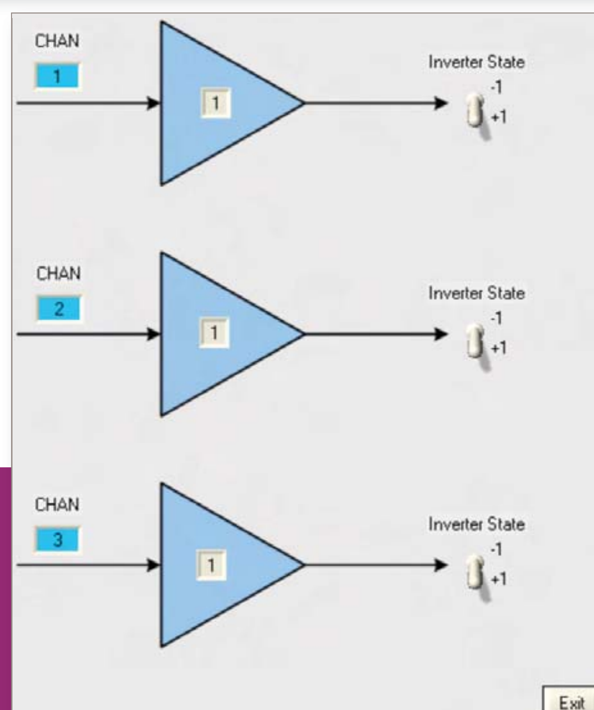


Figure 15 (above): *Time delay control screen*
Table 1 (above right): *Typical Audio Specifications for DIGMODA DDC-552*
Figure 16 (right): *180-degree phase invert screen*

Speech Processing in EMBEDDED APPLICATIONS

Professor Dr Dogan Ibrahim from the Near East University in Nicosia, Cyprus, gives examples of various speech processing chips available in the market, and describes a speech processing example using a microcontroller-based system with a text-to-speech processing chip

SPEECH PROCESSING, especially audio manipulation and sound processing are commonly used in many everyday electronic devices. Some examples are MP3 players, digital voice recorders, speaking GPS receivers, electronic toys, intelligent alarm systems, speaking clocks, radio and televisions, mobile phones, devices used by older and blind people, and many more similar devices.

In general, the speech processing capabilities that can be added to an electronic device are voice recording, voice playback, text-to-speech (TTS) synthesis and speech recognition (SR). Voice recording and voice playback are used in digital voice recorders to store speech in non-volatile memory and then replay it at a later time. Some intelligent recording systems have additional features such as searching for a particular speech, skipping speeches, organising recorded speeches in folders, and so on.

TTS involves reading a written text and converting it into spoken words that can be played through speakers. One typical application of TTS is that the computer can read a piece of text (e.g. via a keyboard or scanner), thus saving the user to look at the screen all the time. People with reading or visual difficulties (e.g. dyslexic or blind people) may find such systems extremely useful.

One of the recent applications of speech processing is in the field of speech recognition (sometimes called voice recognition, or VR), which basically gives a product the ability to "listen and understand". A speech recognition system takes a user's spoken words and interprets what has been said. Speech recognition systems are nowadays commonly used in mobile phones where the numbers to be dialled are read out by the user and the phone rings the required number without the touch of a button. Integration of speech recognition to a product makes that product more intelligent and also more marketable.

Speech Recognition

Speech recognition is mainly divided into two parts: speaker dependent (SD) and speaker independent (SI). Speaker-dependent speech recognition systems are trained by the person who will be using the system and these systems respond accurately only to the person who trained the system. Such systems can achieve 99% accuracy for word recognition and also have large vocabularies.

Speaker-dependent systems also have the advantages that they are language independent and can be trained in any language, or using any type of sound in place of words (for example for people with disabilities who can not speak words). Speaker independent systems are trained to respond to words independent of the user. As a result of this the accuracy of such systems are much lower and also their vocabularies are rather limited compared to speaker dependent systems. Most commercially available speech recognition devices are general purpose speaker independent systems.

The style of speech is also an important factor in most speech recognition systems. Some systems can only recognise words that are isolated from each other and have a short delay between the words. These are the most commonly found systems and low-cost devices, such as toys and simple speech outputting electronic devices, fall into such category. Some more sophisticated speech recognition systems can recognise multiple sets of words when spoken normally and with only natural delay between the words. Continuous speech recognition is the most desirable system as it can recognise natural speech. Such systems are, however, very complex as the words tend to merge with no delay between them and the system has to recognise the words even if they are spoken in a different order. Continuous speech recognition systems are complex but are continually being developed.

Speech Processing ICs

There are many types of commercially available speech processing chips and development kits. Some examples are given here. Magnevation Speakjet is a 20-pin IC designed to add speech and audio to embedded microcontroller applications. The chip is self-contained and requires just an external +5V supply and a speaker for its operation. A mathematical sound algorithm is used to control its five channel internal sound synthesizer to generate vocabulary speech synthesis and complex sound generation. The chip is low cost and is aimed for simple controllers and the hobby market such as robotics, toys and so on.

The Speakjet is programmed with 72 speech elements, 43 sound effects and 12 DTMF touch tones. In addition, sound effects such as the pitch, rate, bend and volume can be controlled. The chip can easily be controlled from a microcontroller.

The TTS256 is a text-to-speech processor chip in a single 28-pin package. This chip is a companion to Magnevation Speakjet and comes with a built in 600 rule database to convert English text to phoneme codes. Speech can easily be generated from ASCII text in microcontroller-based embedded applications, making the chip extremely easy to use in applications where it is required to generate speech. TTS256 is controlled from its serial port and, thus, it is compatible with any microcontroller with a serial port.

TTS-03 is a text-to-speech processor module with serial ASCII input and direct speaker interface. The module operates with a wide supply voltage (+3V to +9V) and has optional IrDA infra-red input for PDA interface. The

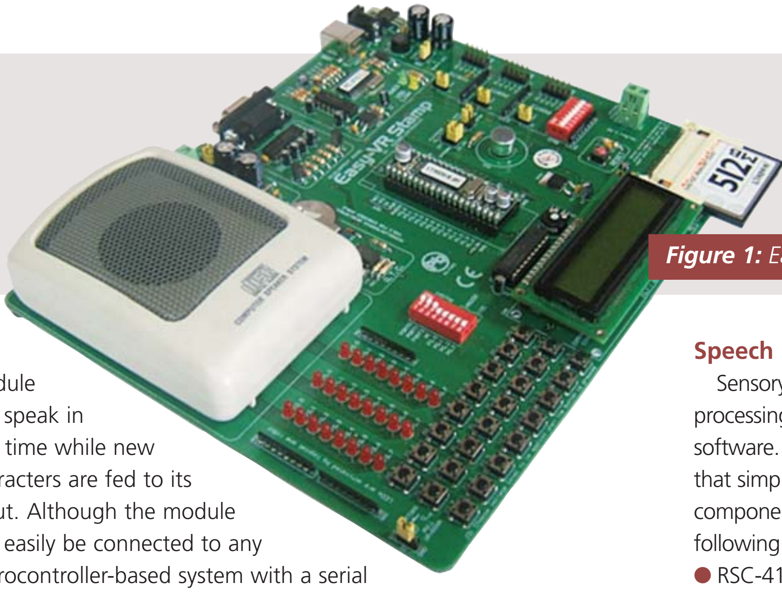


***Figure 1:** Easy-VR Stamp development kit*

module can speak in real time while new characters are fed to its input. Although the module can easily be connected to any microcontroller-based system with a serial output, a mini-motherboard is available with a PS/2 keyboard interface for testing and evaluating the module.

SP03 is a text-to-speech synthesizer chip which also includes an audio amplifier, voltage converter and a PIC microcontroller. Interface to SP03 is via its RS232 serial port, I2C bus interface or parallel interface can be used. The chip can be programmed to generate up to 30 predefined phrases, each phrase having a maximum length of 85 characters. A PC program (SP03.EXE) is available to load these predefined phrases into the chip. To speak any of the predefined phrases a command is sent to the chip in serial or I2C mode, or the required phrase is selected from the 5-bit parallel input port. The volume, pitch and speed of the sound can be selected by commands.

Emic text-to-speech module is based on the Winbond WTS701 speech processor chip and the module can handle values, sentences, numbers and common abbreviations with simple string sentences. The module is manufactured by Grand Idea Studio and distributed by Parallax. The module requires just a +5V supply and an external speaker for its operation. It is controlled from a serial TTL interface and in addition to speech output, the volume, speed and pitch of the sound can be changed by commands, and new abbreviations can be added or an existing abbreviation can be deleted. This module is used in many hobby electronic applications, including in robotics and electronic toys.

S1V30120 is another text-to-speech chip aimed for embedded applications, such as home appliances and office/industrial equipment. The chip is controlled from its serial port and requires an external crystal and an audio amplifier for speaker output. S1V30120 supports several languages and has nine predefined voices.

SSG01 is a sound coprocessor chip in an 18-pin DIP package. It is a 6-voice electronic music synthesizer, sound effects and voice synthesizer chip. This is a low-cost chip aimed for use in manufactured electronics, educational and home projects. SSG01 has a serial port connection and it can easily be connected to an embedded microcontroller with a serial port. The chip requires an external crystal and an audio amplifier to generate sound from a speaker.

HM2007 is a speech recognition chip where up to 40 words can be recognized by the chip and a multiple chip configuration is possible in complex applications. The chip is operated from a single +5V supply and a microphone can be connected directly to the chip.

ISD17120PY is a sound record and playback chip (or a chipcorder) that can be used to record and then play sound for 120 seconds. Different models of the chip can be used for shorter or longer times.

Speech Processing Modules and Boards

Sensory Inc is one of the leading companies manufacturing speech processing chips and hardware modules, and developing speech processing software. VR Stamp is a highly sophisticated module from the company that simplifies the design of speech processing by incorporating all key components into a convenient 40-pin DIP module. The module has the following features:

- RSC-4128 speech processor
- 1Mbit flash and 128kb serial EEPROM
- 14MHz and 32kHz clocks
- 24 I/O lines
- Microphone preamplifier
- Pulse width modulator for speaker output.

VR Stamp module supports the company's FluentChip technology and sound processing library with the following features:

- Speaker-independent and speaker-dependent speech recognition
- 2.4-7.8kbps speech synthesis and sound effects
- Many language modes for international use
- 8-voice MIDI compatible music synthesis
- Audio wakeup from sleep
- Touch tone (DTMF) output.

Some companies offer general purpose digital signal processing chips that can be programmed and used in embedded speech processing applications. For example dsPIC (www.microchip.com) digital signal controller (DSC) chips from Microchip are high performance 16-bit signal processing chips that can easily be used in sound processing as well as in image processing applications. Similarly, Texas Instruments (www.ti.com) TMS series chips, or Analogue Devices (www.analog.com) ADSP series chips can easily be used in speech and image processing applications.

Perhaps one of the easiest ways to learn how speech processing can be done in practice is to use a speech processor development board. The RSC-4x Demo/Evaluation Toolkit from Sensory is a speech development board that helps the user to be familiar with the current speech processing technology and also learn how to program Sensor's speech processing products.

Easy-VR Stamp (see **Figure 1**) is one of the popular speech processor boards manufactured by mikroElektronika. This board is based on Sensory's highly successful VR Stamp module. The board is very low cost and is a good starting point for a student or a professional who wishes to learn the capabilities of the VR Stamp module or to develop speech processing software using the C language and the FluentChip library of sophisticated speech processing functions. The Easy-VR Stamp module has the following specifications:

- Support for VR Stamp speech processing modules (DIP40 socket)
- Real-time clock chip with battery
- On-board speaker and microphone
- Audio amplifier
- 24 LEDs
- 24 buttons
- USB 2.0 programmer

- LCD display
- Compact Flash card slot
- External speaker connection
- RS232 port
- Power from USB port.

The board is normally connected to a PC via its USB bus. Power for the board is derived from this bus and, in addition, the VR Stamp chip is programmed from the PC via the USB bus.

A C-language based compiler, named as RSC-4x mikroC, and chip programmer software named as VR Stamp Loader are also given as part of the distribution. The RSC-4x compiler has no built-in assembler or a linker and, thus, in addition to the supplied software, the user should get a copy of the Sensory MCA-SE assembler and MCLINK linker from the Sensory Inc web site. In addition, for complex speech processing, music and voice recognition applications, it will also be necessary to obtain a copy of the FluentChip library from the Sensory Inc web site.

In this article we will use the Emic chip in a simple text-to-speech application to show how easy it is to include speech in embedded applications. The example given is a "talking thermometer", where the temperature will be read from an analogue sensor IC and will be output as speech every minute.

Digital Audio Effects

Before looking at the design of our "speaking thermometer", it is worthwhile to review the theory of some of the commonly used digital audio effects. Digital audio (sound or speech) processing is basically a branch of the well-known field of digital signal processing (DSP). In a typical digital audio processing application the audio signal is normally received by a microphone, converted into digital format and then fed into a digital processor. The digital processor runs a program where most of the processing takes place. The processed digital audio signal is either stored in the memory of the system as a digital signal, or it is converted into analogue format and is sent to a speaker as an output.

Some simple but commonly used digital audio effects are: audio delay, audio echo, audio equalizer and audio reverberation.

The highly popular MATLAB software (www.mathworks.com) package can be used for experimenting and learning various sound processing functions and algorithms as the package contains a very large number of sound processing functions. This approach is extremely useful, especially for students who may want to learn the basics of digital sound processing. For example, a ".wav" sound file "music.wav" can be read into MATLAB with the function:

```
[mytest, fs] = wavread('music.wav');
```

where fs is the sample rate (in Hertz) used to encode the data in the file. The sound can then be played using the "soundsc" command:

```
soundsc(mytest, fs);
```

where array "mytest" stores the stereo sound and "fs" is the sampling frequency. The two channels of the signal can be separated into two columns using "left" and "right" commands:

```
Left = mytest(:,1);
Right = mytest(:,2);
```

The left channel sound can be played as:

```
soundsc(Left, fs);
```

and the right channel as:

```
soundsc(Right, fs);
```

We can perform a simple processing on the sound files for example by reversing the elements of the sound vector:

```
NewLeft = flipud(Left);
```

and then play the sound in reverse order:

```
soundsc(NewLeft, fs);
```

We can change the sampling frequency and play the sound slower:

```
soundsc(Left, fs/2);
```

or play it faster:

```
soundsc(Left, fs*2);
```

In the following example, the sound file is opened and only the first two seconds sound is played:

```
[mytest, fs] = wavread('music.wav');
nsamples = 2 * fs;
[mytest, fs] = wavread('music.wav', nsamples);
sound(mytest, fs);
```

The following example shows how the sound wave in the file can be plotted (see **Figure 2**):

```
time = (1 / fs)*length(Left);
v = linspace(0, time, length(Left));
plot(v, Left);
xlabel('Time (sec)');
ylabel('Signal strength');
```

Figure 2: *Plot of the sound wave*

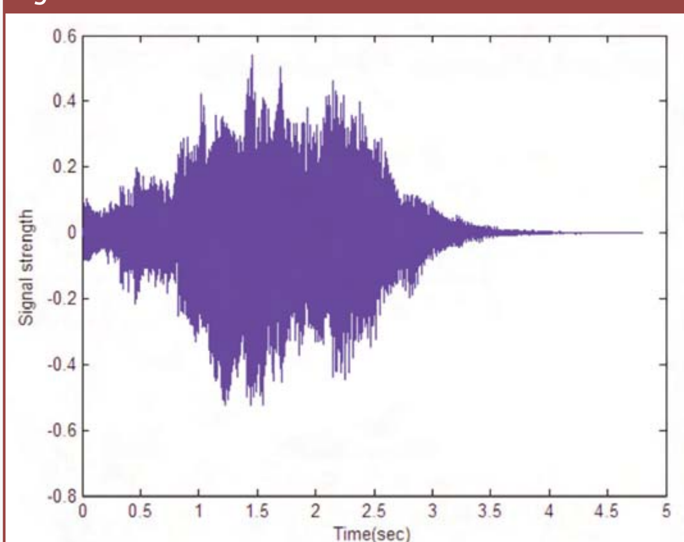
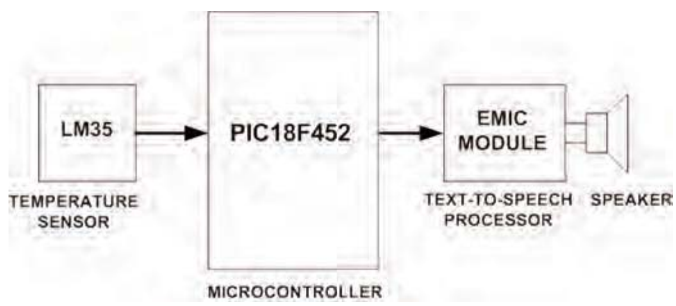


***Figure 3:** Block diagram of the "speaking thermometer"*



Finally, the following example shows how echo can be added to the sound file by adding echo to each sample. In this example V is the new vector that stores the new sound where samples from the original sound are added to the new sound after 1/2 second delay:

```
[mytest, fs] = wavread('music.wav', nsamples);
Left = mytest(:,1);
v = Left;
num = nsamples / 2;
for j = num + 1:length(Left)
        v(j) = Left(j) + Left(j – num);
end
soundsc(v, fs);
```

As can be seen from the above examples, students can easily experiment and learn the basic principles of sound processing by actually modifying the sound characteristics and hearing the resulting effects on the PC speaker.

Example Microcontroller Speech Application

An example is given to show how speech output can be added to a simple microcontroller-based application. In this application an electronic "speaking thermometer" is designed which tells the temperature every minute in the format:

The temperature is nn degrees centigrade

Figure 3 shows the block diagram of the thermometer. A PIC18 microcontroller reads the temperature from an analogue temperature sensor every minute and sends commands to an Emic text-to-speech processor module to tell the temperature via a speaker.

The circuit diagram of the thermometer is shown in **Figure 4**. An LM35DZ type analogue temperature sensor is used in the design and is connected to the analogue port RA0 of a PIC18F452 type microcontroller. LM35DZ can measure the temperature between 0-70°C and its output voltage is directly proportional to temperature, i.e. Vo = 10mV/°C. Thus, for example, at 10°C the output voltage is 100mV and at 30°C the output voltage is 300mV.

The Emic Module

Before describing the microcontroller program it is worthwhile to look at details of the Emic text-to-speech processor module.

Emic (see **Figure 5**) is a 16-pin module (not all pins are used) to provide speech from a written text string. The module is operated from a +5V supply and can drive an 8 ohm, 300mW speaker directly. Text to be spoken can be sent as commands, either in ASCII format or as hexadecimal command sequences. Interface to the module is via a pair of TTL compatible serial wires. The default communication speed is 2400 baud, 8 bits and no parity.

Emic module has the following pins:

VCC	-	+5V supply
GND	-	Ground
BUSY	-	Emic is busy, either receiving characters, or converting text to speech
SIN	-	Serial TTL compatible input
SOUT	-	Serial TTL compatible output
RESET	-	Emic reset pin (active low)
AOUT	-	analogue speech output for external amplification
SP-	-	Negative speaker connection
SP+	-	Positive speaker connection
AIN	-	Analogue input (for the AOUT or SP-/SP+ outputs)

***Figure 4:** Circuit diagram of the thermometer*

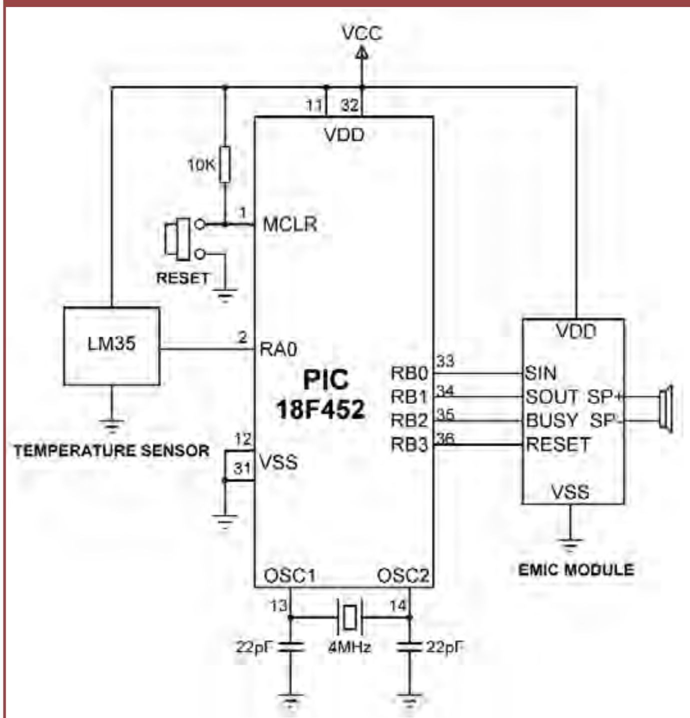
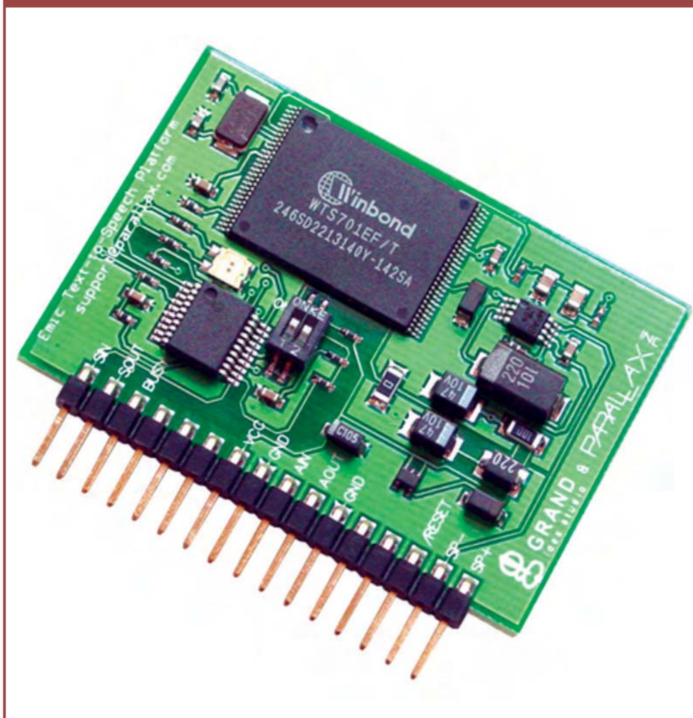


***Figure 5:** Emic text-to-speech processor module*



The microcontroller is connected to the Emic module via the following pins:

Microcontroller	Emic	Description
RB0	SIN	serial input
RB1	SOUT	serial output
RB2	BUSY	Emic busy signal
RB3	RESET	reset to Epic

Two DIP switches are located on the Emic module: SW1 determines the text mode and if set to ON then the module will accept commands in ASCII format. If SW1 is set to OFF, the module will accept commands in hexadecimal format. In this example SW1 is set to ON position.

Switch SW2 controls the host echo mode and if set to ON then each byte sent to the module will be echoed back to the host processor. If SW2 is set to OFF, the module module will not echo each byte to the host (it will only echo "OK" or "ERROR"). In this example SW2 is set to OFF position.

The Emic module has a 128 circular buffer and any command and text data sent to the module must be less than 128 bytes. A text must not be greater than a maximum length of 53 characters, otherwise it will be truncated. End of the user buffer is assumed when the termination character ";" (semicolon) is received in ASCII mode, or 0xAA in hexadecimal mode. In ASCII mode, if a valid command and data are received and the operation is successful, the Emic module will respond with "OK" (or 0x55 in hexadecimal mode). If an incorrect command or data is sent to the module or if the operation has failed, the Emic module will respond with "ERROR" (or 0xFF in hexadecimal mode).The example in this article uses the ASCII mode as it is easier to understand. **Table 1** gives a list of the Emic module command set when operating in ASCII mode. Some of the useful commands are summarised here:

say=text; - convert text to speech

volume=n; - set the volume. The volume can be 0 to 7 where n = 0 is softest and corresponds to -28dB, and n = 7 is loudest and corresponds to 0dB. Default setting is 4. The range can be incremented or decremented by 1 step using "+" or "-".

speed=n; set the speed. The speed can be 0 to 4 where n = 0 is the slowest and n = 4 is the fastest. Default setting is 2. The range can be incremented or decremented by 1 using "+" or "-".

pitch=n; set the pitch. The pitch can be 0 to 6 where n = 0 is the lowest and n = 6 is the highest. Default setting is 1. The range can be incremented or decremented by 1 using "+" or "-".

There are other commands to add abbreviations, delete abbreviations, list existing abbreviations, get the Emic module version number, soft reset the device, analogue audio input and a help command.

Program Description

Figure 6 shows the operation of the thermometer project in PDL. The actual program listing is given in **Figure 7**. In this example the program was developed using the mikroC compiler (www.mikroe.com). At the beginning of the program port I/O directions and the analogue port are configured and the Emic module is reset. Then the serial port is configured to operate at 2400 baud and the welcome message "welcome to speaking thermometer" is sent to the speaker via the Emic module. The main loop of the program in enclosed inside a 'for' loop which executes forever with one minute delay (by calling to procedure Wait_One_Minute) between each loop, and reads the temperature by calling to procedure Read_Temperature. The temperature is converted into ASCII using built-in function ByteToString and is stored starting at offset 19 of array TempMsg. Text string TempMsg is then sent to the speaker by calling to procedure Speak.

***Table 1:** Emic module command set (in ASCII mode)*

Command	ASCII sent
Convert text to speech	say=text;
Set speech volume	volume=n;
Set speech speed	speed=n;
Set speech pitch	pitch=n;
Add abbreviation	addabbr=abbr,text;
Delete abbreviation	delabbr=abbr;
List abbreviations	listabbr;
Version numbers	version;
Soft reset	reset;
Audio input	audio;
Help	help; (or ?;)

***Figure 6:** Operation of the speaking thermometer*

```
BEGIN
        Configure I/O ports
        DO FOREVER
                Read temperature from A/D converter
                Convert to Centigrade
                Store temperature in a text string
                Send text string to EMIC module
                Wait 1 minute
        ENDDO
END
```

```
/*-------------------------------------------------------------------------------------
------------
 SPEAKING THERMOMETER
        ====================
This is the software for the speaking thermometer project.
A PIC18F452 type microcontroller receives the temperature
from a LM35DZ type analogue sensor connected to port pin
RA0. The temperature is sent to a speaker via an EMIC text-
to-speech processing chip.

The temperature is output to the speaker in the following
format:

The temperature is nn degress centigrade

The microcontroller is operated with a 8MHz crystal The
connection between the microcontroller and external world
is as follows:
```

```
 RA0-> Vout (LM35DZ)
 RB0-> SIN (EMIC)
 RB1-> SOUT (EMIC)
 RB2-> BUSY (EMIC)
 RB3-> RESET (EMIC)

File:   THERMOMETER.C
Date: September, 2009
--------------------------------------------------------------------------------
-----------*/
#define TX_Pin 0
#define RX_Pin 1
#define RESET PORTB.F3
#define BUSY PORTB.F2

char WelcomeMsg[] = "welcome to speaking thermometer";
char TempMsg[] = "the temperature is nn degrees
centigrade";

//
// Send a text to EMIC text-to-speech processor module
//
void Speak(char *ptr)
{
 char text[53],i, j;
 text[0]='s'; text[1]='a'; text[2]='y'; text[3]='=';

 j = 4;
 while(*ptr)
 {
 text[j] = *ptr++;           // Insert message
 j++;
 }
 text[j]=';';                // Insert terminator, ";"

 for(i=0; i<=j; i++)         // Send to EMIC module
 {
 while(BUSY);  // Wait if Emic is busy
 soft_Uart_Write(text[i]);             // send to serial port
 }
}

void Wait_One_Minute()
{
 unsigned char i,j;
 for(j = 0; j < 60; j++)
 {
 for(i = 0; i < 4; i++)Delay_Ms(250);
 }
}

void Read_Temperature(void)
{
 unsigned long Vin;
 unsigned char Temperature,j;
 char txt[4];

 Vin = Adc_Read(0);                  // Read temperature
 Vin = Vin * 5000 / 1024;                      // Convert to mV
 Vin = Vin / 10;                     // Convert to Celsius
 Temperature = Vin;                  // As byte
 ByteToStr(Temperature, txt);        // Convert to string
 for(j=0; j<3; j++)TempMsg[19+j] = txt[j];     // Add to
TempMsg
}

void main(void)
{
 TRISA = 1;                  // RA0 is input
 ADCON1 = 0x80;                      // Configure A/D
 TRISB = 6;                  // RB1,RB3 are inputs
 RESET=0;                    // Reset EMIC module
 Delay_ms(200);
 RESET = 1;

//
// Configure serial port to 2400 Baud
//
 Soft_Uart_Init(PORTB, RX_Pin, TX_Pin, 2400,0);
//
// Send welcome message
//
 Speak(WelcomeMsg);
//
// Program loop. Read temperature and send to speaker
continuously
//
 for(;;)
 {
 Read_Temperature();                 // Read temperature
 Speak(TempMsg);                     // Send to speaker
 Wait_One_Minute();                  // Wait for 1 minute
 }
```

Figure 7: *Program listing of the speaking thermometer*

An Important Topic

Speech processing, such as speech recognition and text-to-speech, is becoming a very important topic in intelligent embedded applications. Nowadays most mobile phones include some form of speech recognition modules to help use the phone hands-free. Another very important application of speak recognition is to help people with disabilities. For example, a lift with a speech recognition processor can help a blind person to control the lift by speech. Similarly, an elderly person for example can turn a kettle ON or OFF by speech.

Text-to-speech is largely used in alarm and control systems with audio output and in hobby electronics, such as robotics and electronic toys. A text-to-speech processor based system can also help a person with reading and learning difficulties, for example by reading a book or a newspaper, or any other document.

This article has given examples of various commercially available speech processing chips and modules. An example text-to-speech application is given in the form of a “speaking thermometer” to show the simplicity and the beauty of including speech in microcontroller-based systems. ■

Alaa Eleyan, **Hasan Demirel** and **Hüseyin Özkaramanli** from the Electrical & Electrical Engineering Department in Northern Cyprus, Turkey, discuss a new system for recognizing faces from video sequences using the weighted majority voting (WMV) method

FACE RECOGNITION from Low Resolution Video Sequence Using Weighted Majority Voting

PERSON IDENTIFICATION and verification is widely used in security and authentication applications. This raises the importance of face recognition as one of the most significant biometric applications.

Although face recognition is not as accurate as the other recognition methods such as fingerprints recognition, it still grabs huge attention of many researchers in the field of computer vision. The main reason behind this is the fact that the face is the conventional way people use to identify each other. The earliest face recognition methods were introduced by Bledsoe the in mid-sixties where he proposed a semi-automated face recognition approach with a hybrid human-computer system that classified faces on the basis of marks on images entered by hand.

Fischler and Elschlager described an algorithm that used template matching to locate the local features and global pictorial structures to represent facial features. The robustness of face recognition systems mainly depends on the approach which is used for feature selection and extraction.

Appearance-based approaches seem more successful than the component-based ones. Principle component analysis (PCA) and linear discriminant analysis (LDA) are two powerful classical tools in the appearance-based category used for feature extraction and data reduction from high dimensional data sets. The main objective of PCA and LDA is to reduce the

Figure 1: *The first 16 eigenfaces with highest eigenvalues*

Figure 2: *Example from the video database training set*

dimensionality by transforming a number of correlated variables into orthogonal subspaces.

Making the Most of It

The approaches listed here are well-known, still-to-still, face recognition approaches. Almost all video-based recognition systems apply still-image based recognition to selected good frames. In real life, however, it is a rare chance to end up with a set of good quality frames, especially from low resolution videos.

So, the approach proposed here introduces a face recognition method that includes not only the selected good frames but also very low resolution, poor quality, frames. This article discusses a majority voting (MV) mechanism, where the decision from each frame of the video with different resolution and quality contributes to the final recognition. The method is tested in two ways: In the first approach the votes from each frame are equally weighted. This is referred to as the majority voting (MV). In the second approach, the votes from each frame are weighted with a value proportional to the information capacity of the respective frame. The size of the respective frame serves as a measure of the information capacity. This approach is referred to as weighted majority voting (WMV). The proposed WMV achieves excellent recognition rates reaching 100% with images ranging in resolution from 22×18 to 64×47 pixels. Due to the fact that the test frames are of low resolution and poor quality, the MV method produces a recognition rate of 80%, which is significantly lower than the performance of the WMV method. The performance of MV is even worse than a simple PCA method, which achieves 93.33% recognition rate applied to the highest resolution frame.

Recognition Model From Video

Here we introduce a new method of recognizing faces extracted from a video sequence of different resolution. The main objective of this method is to recognize faces from a sequence of video using a single eigenspace generated by using the PCA method.

The faces with differing resolution, extracted from a video sequence frames, are upsampled into the resolution of the eigenfaces and the recognition is then performed. With uniform majority voting, where the votes from each frame are equal, each face extracted from a frame is classified and the person with highest multiplicity throughout a video sequence is declared to be 'recognized' person.

Giving each frame a certain weight, depending on its information capacity, helps to suppress the errors that might appear through the recognition process. Using weighted majority voting (WMV), the recognized person is declared to be the subject with the highest accumulative weight throughout the recognition from video frames.

The proposed system is trained by using the PCA-based eigenfaces method with 300 face samples of same resolution extracted from video sequences of 30 people. Each subject has 10 frames in its respective training sequence with facial expression and limited pose variations. The system is tested by using automatically cropped face images with varying resolutions changing from 22×18 pixels to 64×47 pixels, from 1200 frames, where 30

Figure 3: *Frames of one video sequence with image size being 176×144 pixels, while the actual face size ranges approximately from 22×18 pixels in the first frame up to 64×47 pixels in the last frame*

***Figure 4:** Block diagram of the proposed system*

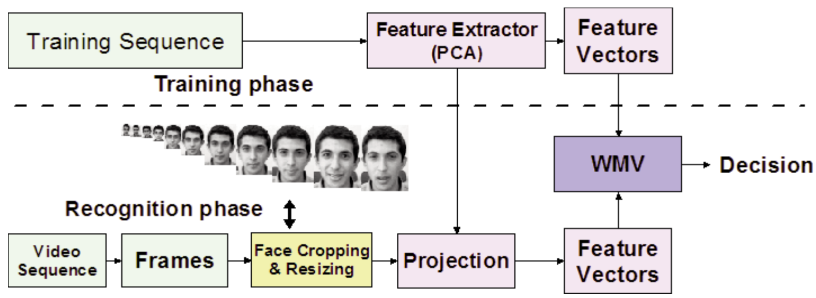
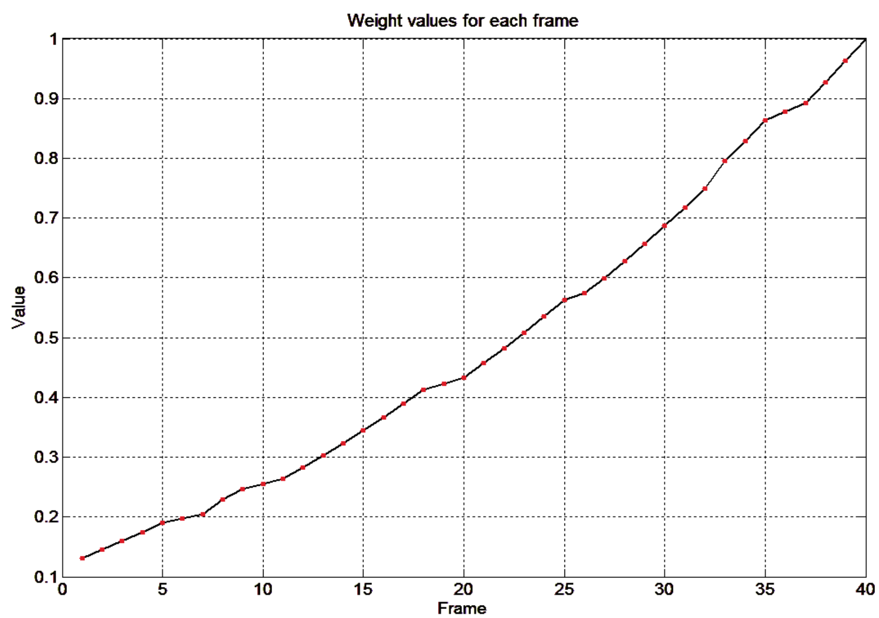


***Figure 5:** Weight values assigned to each frame*



subjects with 40 frames each, are used to obtain the system performance.

The proposed system successfully recognizes the subject in most of the frames, even though the resolutions vary. In some cases the number of misclassified faces in respective classes outnumbered the number of correctly recognized class. However, the proposed WMV method overcomes this problem by assigning higher cumulative weight to the correct subject, based on fitness values, which are proportional to the information capacity of each frame being recognized.

Principle Component Analysis

Principal component analysis is a standard technique used in statistical pattern recognition and signal processing for data reduction and feature extraction. As the pattern often contains redundant information, mapping it to a feature vector can get rid of this redundancy and preserve most of the intrinsic information content of the pattern. The extracted features have great role in distinguishing input patterns.

Every test image can be transformed to a lower dimensional feature vector projecting it onto the eigenface space which is obtained from the training set. This feature vector can then be compared with the set of feature vectors obtained from the training set.

Classification is performed by comparing the feature vectors of the training face images with the feature vector of the input/test face image by using a similarity measure such as Euclidean distance or cosine distance. **Figure 1** shows the first 16 eigenfaces with highest eigenvalues.

Segmentation and Histogram Equalization

The dataset consists of video sequences of 30 individuals. The training set consists of 10 face images 75×55 pixels in size for each individual facing the camera with considerable facial expression variations (**Figure 2**). The test set consists of 40 face images extracted from video sequences for each of the 30 individuals making a total of 1200 faces (**Figure 3**).

In this work, the local SMQT algorithm proposed by Nilsson has been adopted for face detection and cropping in the preprocessing stage. Cropped faces were converted to greyscale and a histogram equalized to minimize the illumination problems. Bicubic interpolation was used to upsample the resulting face images to the same size of the reference resolution (the size of training images).

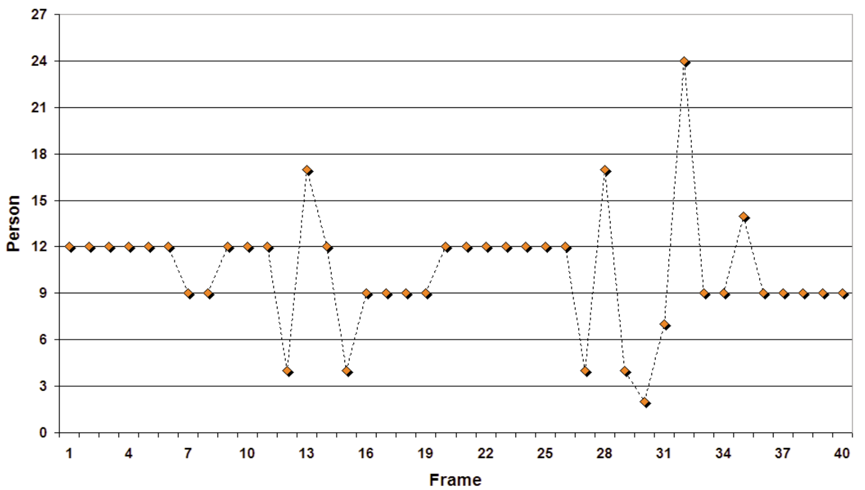
The resulting segmented face images have varying resolutions changing from 22×18 pixels to 64×47 pixels. Total of 1200 cropped face images were used for testing the system performance. The overall setup of the proposed recognition system is illustrated in **Figure 4**.

Weighted Majority Voting

Based on the discussion earlier in this article, it is clear that the recognition rate and resolution of test images are closely related. Specifically, the recognition rate improves when the resolution of the test images are higher. It is possible to explain this rather obvious relationship by using the information content of the test images (see *R. G. Gonzalez, R. E. Woods. Digital Image Processing, 2nd ed. Prentice Hall. NJ.2002*). It is, however, difficult to measure the information content. We can consider the information capacity as a measure of information content for images of similar structure, like the face images. The information capacity provides a measure of maximum information that can be conveyed by an image at a specific resolution. The information capacity is directly proportional to the resolution or number of pixels in a given image. This forms the basis for establishing the weights in the weighted majority voting method.

The proposed recognition system assigns weights to each extracted face from the respective video frame, depending on the information capacity of that face. The relative information capacity is calculated by using the ratio between the number of pixels for each frame and the number of pixels for the maximum sized frame. Then the frame with

Figure 6: *Recognitions throughout the frames*



lower resolution was assigned a lower weight and the frame with a higher resolution was assigned a higher weight. For example, frame 40 which has a resolution of 64×47 pixels was assigned the highest weight, which is one, and the first frame (22×18 pixels) was assigned the lowest weight.

The weight for each extracted test face can be calculated using the following:

$$\omega_k = \frac{M_k \times N_k}{M_L \times N_L} \qquad 1 \le k \le L \qquad (1)$$

where ω_k is the weight at k^{th} frame, M_k and N_k are the image dimensions of the k^{th} frame. L is total number of frames for each person. **Figure 5** shows the values of the weights assigned to each frame.

The weights are accumulated to the recognized subjects through the processing of the video frames. The final recognition is declared to be the subject that accumulated the highest weight through this process.

The weight vector is $\Omega_p = [\omega_1, \omega_2 \ldots \omega_k]$, which is the weights values of each frame. Having $\Delta_p = [\delta_1, \delta_2, \ldots, \delta_k]$ as the decision vector with the decisions made at each frame, we can define our weighted decisions ξ_p for each person as:

$$\xi_p = \sum_{k=1}^{L} \omega_k \quad if \quad \delta_k = p \qquad (2)$$

Defining the weighted decisions vector as $\Xi p = [\xi_1, \xi_2 \ldots \xi_p]$. Then the person with the highest cumulative weight ξ_{max} is declared the 'recognized' person.

Results and Discussions

A video sequence of 40 frames for each person, where a subject walks towards the camera to provide different resolution face data, is used for testing the video-based face recognition system. The faces in each frame with different resolutions are cropped as shown in Figure 3 for recognition. Then each cropped face is recognized in each frame. **Figure 6** shows the recognized person in each frame along the 40 consecutive frames throughout the video using PCA. In Figure 6 it can be seen that as the person walks towards the camera, the resolution of the cropped face increases and approaches to the reference resolution.

As the figure indicates, the correct recognition of this person is more probable in higher resolutions (i.e. after frame 35). However, the recognition of the person fails when we apply majority voting directly on these results as shown in **Figure 7**, where instead of the right person to be recognized, a wrong one was identified as 'recognized'.

Using the weighted approach based on the information capacity of the frames described in the previous section, the contribution of the higher resolution frames will be emphasized in proportion to the information capacity that they can potentially have. Hence, the weighting will help to suppress the adverse effects of low resolution frames and, thus, lead to an improved decision.

Carrying on with the example demonstrated in Figure 7, with the proposed WMV method, the right person was correctly recognized. This is demonstrated in **Figure 8**, where the cumulative weights are also indicated. Note that with the simple MV scheme most of the recognitions of the 'wrong' person occurred over the low resolution frames, which are not very reliable. It is also interesting to note that this person has four more votes than the 'right' one. However, the 'right' one has most of its votes in the high resolution frames of the test video. By emphasizing the higher resolution frames more, the correct decision can be easier reached, with the cumulative weights

Figure 7: *Wrong decision with Majority Voting*

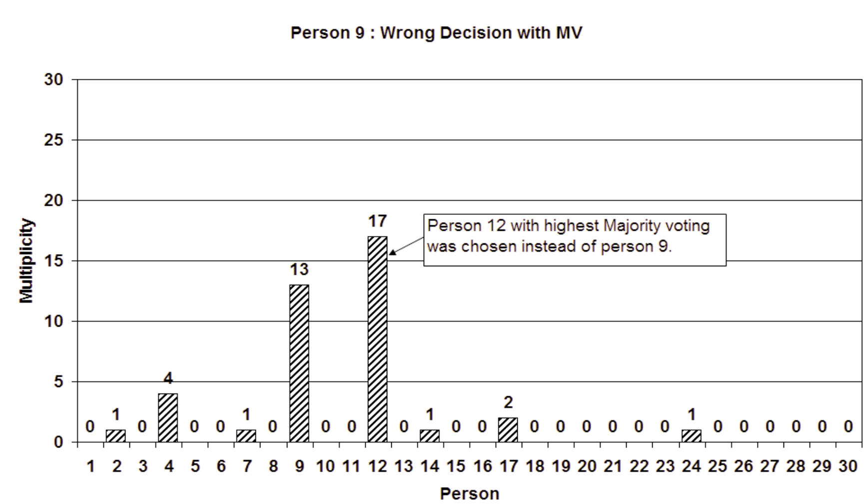


***Figure 8:** The correct decision with Weighted Majority Voting*

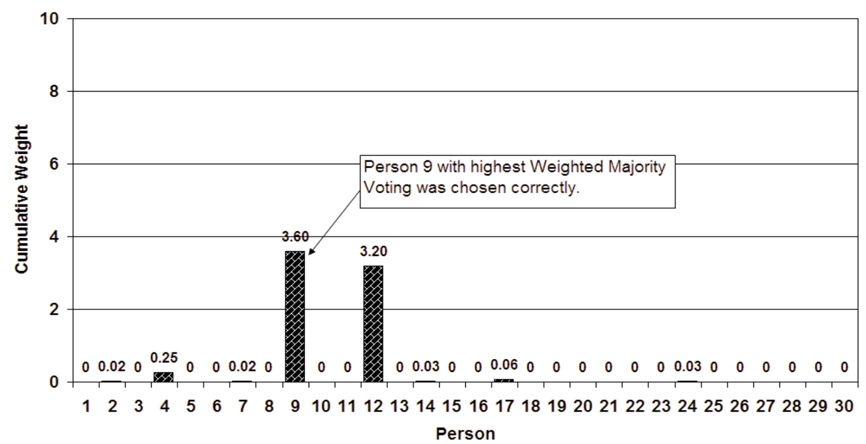
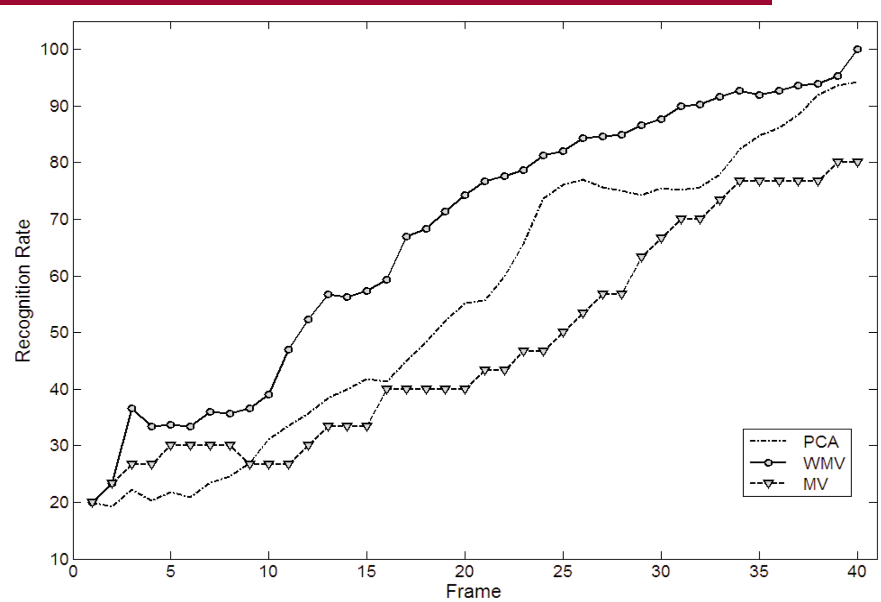


***Figure 9:** MV, WMV and PCA recognition rate using a different number of frames*



being 3.60, whereas for the 'other' person this is only 3.20 (see to Figure 8).

This method is also tested with a varying number of training frames for each subject. **Table 1** shows the recognition rates of the proposed system relative to the number of training frames. As the number of frames in the training sequence for each subject is increased, it is noticeable that the performance of both methods consistently improves with WMV and always outperforming the MV method. The recognition rate of the WMV reaches 100% with only 10 training images.

Figure 9 shows the performance of the proposed WMV and MV methods as a function of the number of frames used in making a decision. Also shown in Figure 9 is the recognition performance of PCA using a single test frame at the resolution of the best (highest resolution) frame employed by the MV method.

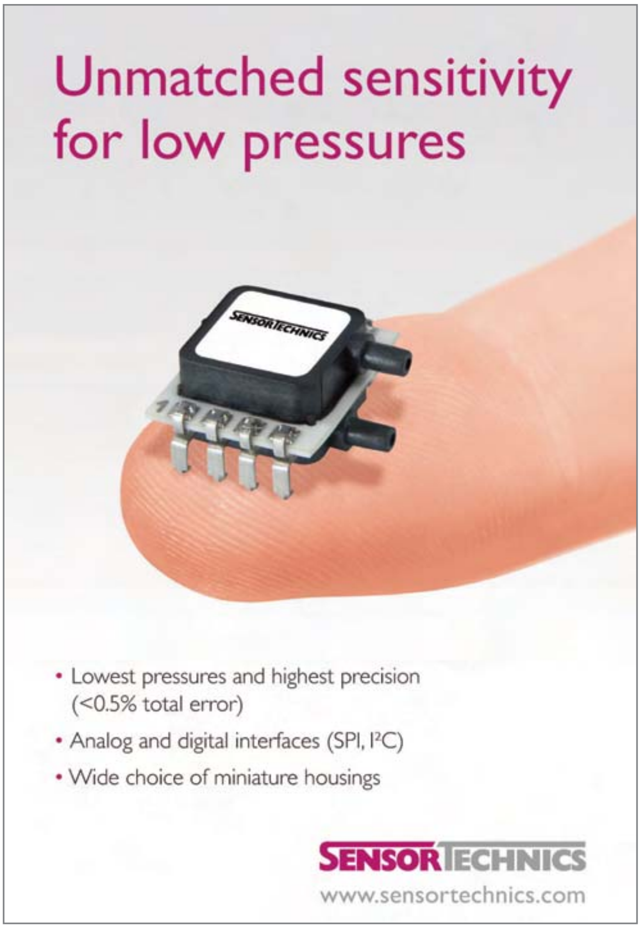
Figure 9 shows that the WMV outperforms MV and reaches to 100% after using the weighted decisions of the 40 frames with changing resolutions from 22×18 pixels in the first frame up to 64×47 pixels in the last frame. For example, if you consider frame 33 (42 x 25 pixels of resolution) of the multi-resolution videos of 30 subjects, the proposed WMV method, which combines the weighted decisions of the 33 frames through the video sequence, gives 91.67% where MV which equally uses the decisions of 33 frames gives 73.33%. The result of using the decision of only the frame 33 is 77.97%. The reason why the MV method produces poor results is due to the fact that the test videos are of poor quality with very low resolution face images. The WMV method performs better than both MV and the single frame PCA method by a significant margin due to the judicious use of weighting scheme that takes into account the suitability of each individual decision.

A New Face Recognition System

A new system for recognizing faces from video sequences using weighted majority voting (WMV) method is described here. This system is robust to scale changes. It uses PCA-based single eigenspace generated by sequences of 10 faces of each subject, of same resolution, in the training phase. The recognition of a face with changing resolutions in different frames of a given video sequence is performed by automatically extracting each face and upsampling it to the resolution of the eigenfaces, followed by assigning each recognized subject with a weight based on its suitability, proportional to the information capacity of the frame under test. The subject with the highest cumulative weight through the video sequence is declared to be the recognized person. The proposed WMV achieves excellent recognition rates reaching 100% with low quality video sequences containing face images of a resolution in the range of 22×18 to 64×47 pixels. ■

***Table 1:** WMV vs MV with different number of training images per person*

# Training images	MV	WMV
1	53.33	62.33
2	57.00	75.67
3	62.00	81.00
4	68.67	86.00
5	70.00	89.33
6	72.00	91.67
7	73.00	92.67
8	73.33	94.33
9	75.67	96.00
10	80.00	100.0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CONNECTING TODAY'S NEEDS

Histogram Based FACE

Hasan Demirel of the Department of Electrical and Electronic Engineering at the Eastern Mediterranean University in North Cyprus, Turkey, and **Gholamreza Anbarjafari** of the Department of Information System Engineering at the North Cyprus International University, Turkey, describe a practical face recognition system based on colour histogram matching

MOST RECENTLY, security systems have been taking a very important place. In this work a practical face recognition system, based on colour histogram matching, is proposed. Histogram matching is performed by using the correlation between the histograms of a given face and the histograms of the faces in the database. The high correlation between the histograms of the same faces at different resolutions makes the system robust to scale changes. A graphical user interface is developed in MATLAB environment for establishing links between project components.

A biometric system provides automatic recognition of an individual based on some unique characteristic owned by the individual. Biometric systems have been developed based on fingerprints, hand geometry and the face. The earliest work in computer recognition of faces is reported by Bledsoe. Kanade used a method of characterizing the face by geometrical parameterisation, whereby distances and angles between points are used. Statistical face recognition systems, such as the Principal Component Analysis (PCA) based eigenfaces method introduced by Turk and Pentland attracted a lot of attention by researchers. Relhumeur et al introduced the fisherfaces method, which is based on the Linear Discriminate Analysis (LDA) that minimizes the discrimination within a class and maximizes the discrimination between classes.

Another statistical descriptor can be considered to be the histogram of a grey level image that shows the distribution in terms of occurrence frequencies of grey level pixel intensities. Histogram of a face image can be considered as the signature of the face, which can be used to represent the face image in a low dimensional space. Images with small changes in translation, rotation and illumination still possess high correlation in their corresponding grey-scale histograms. This histogram characteristic prompts the idea of using not only the grey-scale histograms but also colour histograms for face recognition.

Yoo et al used chromatic histograms as a model of faces to perform face detection. Ojala et al divided a face into several blocks and then the Local Binary Pattern (LBP) feature histograms are extracted from each block and concatenated into a single global feature histogram which efficiently represents the face image. The recognition is performed by simple distance-based histogram matching. In our previous work, we have introduced a probability distribution function based face recognition system the uses the Kullback-Leibler distance.

Pre-Processing: Skin-Tone Detection Method

Face segmentation is one of the important pre-processing phases of face recognition. There are several methods for this task, such as skin tone based face detection for face segmentation.

Skin is a widely used feature in human image processing with a range of applications. Human skin can be detected by identifying the presence of skin colour pixels. Many methods have been proposed for achieving this. Chai and Ngan modelled the skin colour in the YCbCr colour space. Boussaid et al modified Chai and Ngan's methods in order to make it VLSI friendly. They normalized the RGB components, and skin colour was detected by using four threshold values, forming a rectangular region in new RGB space.

In the present work the detection is done by using the HSI colour space, since hue, saturation and intensity are three properties used to describe colour. The advantage of using the HSI colour space is its

Figure 1: Original image

Figure 1: Detected skin region

Figure 3: Skin region image after opening

Figure 4: Detected skin region

RECOGNITION SYSTEM

Figure 5: *Image where the skin region of the original image has been segmented*

Figure 6: *Typical face image from the HP database* ***(a)*** *before pre-processing and* ***(b)*** *after pre-processing*

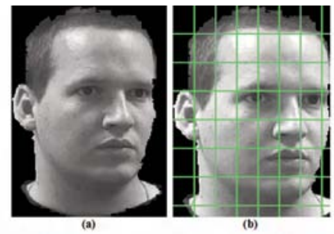

Figure 7: *A pre-processed face image from HP database (a) with its sub-images (b) where have been used for obtaining the LBP recognition rate*

independency of knowing the exact percentage of red, green or blue to produce a colour. Many applications, such as machine vision, use the HSI colour space in identifying the colour of different objects.

Kjeldson and Kender have stated a colour predicate in the HSI colour space to distinguish skin regions from other segments. Skin colour classification in the HSI colour space is based on hues and saturation values.

In this article the threshold for hue and saturation has been found by using 450 face samples from the FERET dataset and 1500 face samples from the Essex University database. According to these samples, the thresholds for hue and saturation to satisfy recognition of skin are:

$$(H < 0.17 \text{ or } H > 0.63) \text{ and } S > 0.1 \qquad (1)$$

According to thresholds mentioned before, the skin area of an image from the FERET database in **Figure 1** has been detected and shown in **Figure 5**.

Figure 2 shows the skin region of the given face. This image needs to be cleaned as there are some small pixels that have been detected in the skin region but are not necessary in being considered for the face recognition.

By using opening operation on the binary image shown in Figure 2, a cleaner image shown in **Figure 3** will be obtained.

To include the nose holes and eyes, the region has been filled (see **Figure 4**).

This binary image has been used as a mask to segment the original image to obtain the skin region, and ignoring the background and shadow effect for face recognition. In this article, each image, before being used for the recognition, has been pre-processed by this algorithm, and the face section has been segmented out for recognition.

PCA and LBP-based Face Recognition

The eigenfaces method is based on linear PCA where a face image is encoded to a low dimensional vector. All face images are decomposed into a small set of characteristic feature images called eigenfaces. Each face image is projected on the subspace of meaningful eigenfaces (ones with nonzero eigenvalues). Hence, the collection of weights describes each face. Recognition of a new face is performed by projecting it on the subspace of eigenfaces and then comparing its weights with the corresponding weights of each face from a known database.

Assuming that all face images in a database are of the same size *w*×*h*, eigenfaces are obtained as the eigenvectors of the covariance matrix of the data points.

Let Γi be an image from a collection of *M* images in a database. A face image is a 2-dimensional array of size *w*×*h*, where *w* and *h* are the width and height of the image, respectively. Each image can be represented as a vector of dimension *w*×*h* and the average image, Ψ, is defined as:

$$\Psi = \frac{1}{M}\sum_{i=1}^{M}\Gamma_i \qquad (2)$$

Each image Γi differs from the average image Ψ by the vector:

$$\Phi_i = \Gamma_i - \Psi \qquad (3)$$

The covariance matrix of the dataset is defined as:

$$C = \frac{1}{M}\sum_{n=1}^{M}\Phi_n\Phi_n^T = \Lambda\ \Lambda^T \qquad (4)$$
$$\Lambda = [\ \Phi 1\ \ \Phi 2\ \ ...\ \Phi M\ ]$$

Since there are *M* images in the database, the covariance matrix C has only *M-1* meaningful eigenvectors. Those eigenvectors u_l can be obtained by multiplying eigenvectors v_l of the matrix $L = \Lambda^T\Lambda$ (of size *M*×*M*) with difference vectors in matrix Λ.

$$u_l = \sum_{k=1}^{M} v_{lk}\Phi_k \qquad (5)$$

The eigenvectors, u_l, are called the eigenfaces. Eigenfaces with higher eigenvalues contribute more in representation of a face image. The face subspace projection vector for every image is defined by:

$$\Omega^T = [\omega_1 \quad \omega_2 \quad \dots \quad \omega_{M'}]$$
$$\omega_k = u_k^T(\Gamma - \Psi) \qquad k = 1,2,\dots,M' \qquad (6)$$

The projection vectors are indispensable in face recognition tasks, due to their uniqueness. The projection vector, which represents a given face image in the eigenspace, can be used for the recognition of faces. Euclidian distance, ε, between the projection vectors of two different images (Ω_1 and Ω_2) is used to determine whether a face is recognized correctly or not.

$$\varepsilon = \|\Omega_1 - \Omega_2\| = \sqrt{\sum_{i=1}^{M}(\omega_{1i} - \omega_{2i})^2} \qquad (7)$$

While for a perfect reconstruction of the face image all the coefficients may be needed, for recognition only the most significant coefficients play an important role.

A typical face image which has been used for PCA is shown in **Figure 6**. PCA also has been applied to the different colour channels (H, S, I, Y, Cb and Cr).

Table 1 and **2** show the performance of PCA-based face recognition system using data from the HP and the FERET face database in the HSI and *YCbCr* colour spaces respectively.

In this article we have suggested to use the majority voting (MV) technique to consider the votes of each colour channel in a PCA-based face recognition system.

For the LBP approach the pre-processed image has been divided into 64 sub-images and then the histograms of each sub-image in different colour spaces have been concatenated and used for recognition. **Figures 7a** and **7b** show one the pre-processed face images from the HP database and its sub-images in the I colour channel.

Tables 3 and **4** show the performance of an LBP-based face recognition system, using the HP and the FERET face databases in the HSI and YCbCr colour spaces. LBP is a time consuming process compared to the proposed algorithm which has been discussed in the next section.

Histogram-Based Pose Invariant Face Recognition

The histogram of an image is a statistical description of the distribution in terms of occurrence frequencies of pixel intensities. It can be considered as a feature vector representing the image in a lower dimensional space. The size of the image histogram depends on the number of quantization levels of the pixel intensities.

A typical monochrome image with 8-bit representation has 256 grey levels. In a general mathematical sense, an image histogram is simply a mapping η_i that counts the number of pixel intensity levels that fall into various disjoint intervals, known as bins. The bin size determines the size of the histogram vector. In this article the bin size is assumed to be 256 and the size of the histogram vector is 256. Given a monochrome image, a histogram η_i meets the following conditions, where N is the number of pixels in an image:

$$N = \sum_{i=0}^{255} \eta_i \qquad (8)$$

Then, histogram feature vector, H, is defined by:

$$H = [\eta_0, \eta_1, \cdots, \eta_{255}] \qquad (9)$$

The similarity between two images can be measured by using the cross correlation between the histograms of the respective images.

Cross correlation is a standard method of estimating the degree to which two vectors are correlated. Given to histogram vectors $x(i)$ and $y(i)$, where $i=0, 1, 2...\beta-1$ and β is the number of bins.

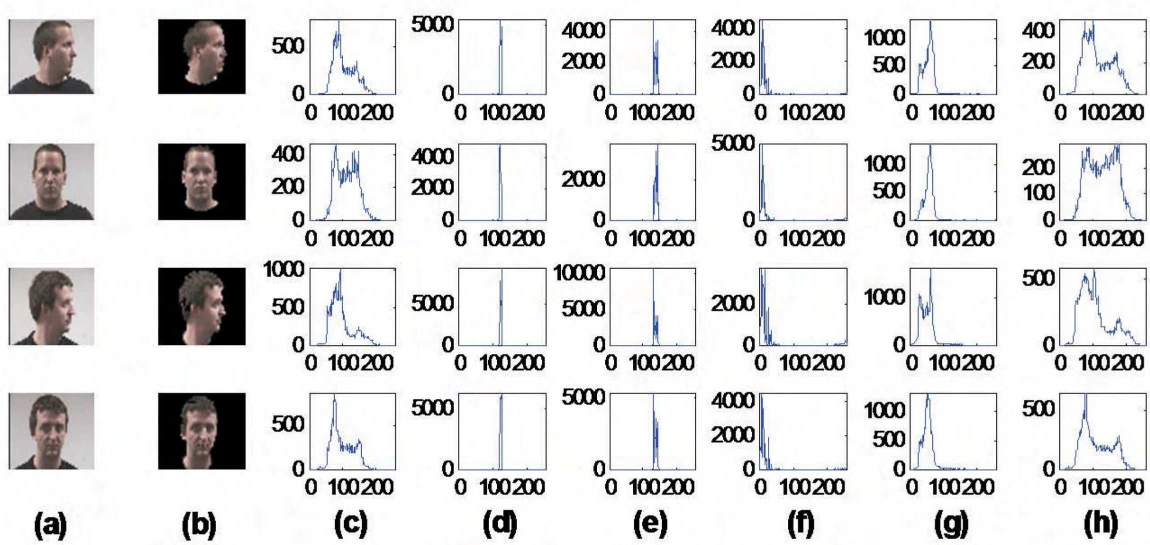

Figure 8: *Two subjects from HP database with two different poses: (a) their segmented faces; (b) and their histograms in Y (c), Cb (d), Cr (e), Hue (f), Saturation (g) and Intensity (h) colour channels respectively*

The cross correlation vector $r(t)$ is defined as:

$$r(t) = \left| \frac{\sum_i \left[\left(x(i) - \mu_x\right)\left(y(i-t) - \mu_y\right) \right]}{\sqrt{\sum_i \left(x(i) - \mu_x\right)^2} \sqrt{\sum_i \left(y(i-t) - \mu_y\right)^2}} \right| \qquad (10)$$

where μ_x and μ_y are the means of the corresponding vectors and t is the translation. The denominator in this expression serves to normalize the correlation coefficients such that $0 \leq r \leq 1$, where 1 is indicating maximum correlation and 0 indicating no correlation. The maximum correlation coefficient in the correlation vector is taken as the measure of similarity and used in the histogram matching process.

Let the histograms of a set of training face images be H_1, $H_2 \ldots H_M$, where M is the number of image samples. Then, a given query face image, the histogram of the query image Hq, can be used to calculate the correlation between H_q and histograms of the images in the training samples as follows:

$$\chi_i = \max(H_i \circ H_q), i = 1, L, M \qquad (11)$$

Here χ_i is the maximum cross correlation coefficient reflecting the similarity of the i^{th} image in the training set and the query face. The, image with the highest similarity measure is declared to be the identified image in the set.

Figure 8 shows two subjects with two different poses and their segmented faces from the HP face database. The images also have different backgrounds, with slight illumination variation. The intensity of each image has been equalized to minimise the illumination effect, where the shapes of the histograms are more or less preserved, while only the amplitudes are changing. The colour histograms used in the proposed system are generated only from the segmented face and, hence, the effect of the removed background regions is eliminated.

The performance of the proposed histogram-based face recognition system is tested on the HP and the FERET face databases, where there are 15 and 50 subjects with 10 varying poses and backgrounds. The colour faces in that dataset are converted from RGB into HSI and YCbCr and the dataset is divided into training and test sets. In this setup the training set contains *n* images per subject and the rest of the images (10-*n*) for the test set. The correct recognition rates in percentages of the HP and the FERET face databases are shown in **Tables 5** and **6**. Each result is the average of 100 runs, where we have randomly shuffled the faces in each class. It is important to note that the performance of each colour channel is different, which means that a person can be recognized in one channel, whereas the same person may fail to be recognized in another channel.

This observation prompts the idea of combining the results of different colour channels, which may increase the probability of making the correct recognition of a face image.

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	12.59	17.78	20.74	20.74	20.00	18.52
2	24.17	38.33	41.67	43.33	38.33	31.67
3	30.48	57.14	56.19	59.05	53.33	45.71
4	32.22	58.89	58.89	62.22	55.56	50.00
5	38.67	62.67	66.67	69.33	65.33	56.00

Table 1: *Performance of the PCA based system in H, S, I, Y, Cb and Cr colour channels of the HP face database*

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	36.89	48.67	49.11	47.33	49.78	49.11
2	41.50	54.75	53.50	52.50	58.25	57.75
3	52.86	62.86	56.29	56.57	67.71	64.00
4	58.00	69.00	64.67	66.00	73.67	70.33
5	62.40	74.80	69.60	72.80	77.60	74.80

Table 2: *Performance of the PCA based system in H, S, I, Y, Cb and Cr colour channels of the FERET face database*

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	57.78	61.48	54.07	59.26	53.33	54.84
2	75.00	70.83	57.50	71.67	68.33	71.67
3	83.81	82.86	75.24	81.90	81.90	84.67
4	87.78	84.44	84.00	83.33	85.33	85.56
5	90.67	85.33	86.67	86.67	85.55	88.00

Table 3: *Performance of the LBP based system in H, S, I, Y, Cb and Cr colour channels of the HP face database*

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	59.33	50.67	48.22	48.44	61.11	62.67
2	64.50	56.75	54.50	54.75	66.50	66.50
3	74.57	66.00	64.29	64.86	76.86	76.57
4	85.67	79.00	77.00	77.67	87.67	86.67
5	87.60	80.40	78.00	78.40	89.00	89.60

Table 4: *Performance of the LBP based system in H, S, I, Y, Cb and Cr colour channels of the FERET face database*

Majority Voting vs Feature Vector Fusion

The face recognition procedure explained earlier can be applied to different colour channels such as Y, Cb, Cr, H, S and I. Hence, a face image can be represented in these colour spaces with dedicated colour histograms for each channel. Here we have used all these six channels to represent and recognize a face image. As such, it is easy to assume that we can use all these six histogram at once for recognition purposes. This combination can be done by the use of MV or FVF. The main idea behind MV is to achieve increased recognition rate by combining decisions of the histogram-based face recognition procedures of different colour spaces. By considering the H, S, I, Y, Cb and Cr histograms separately and combining their results, and with the use of MV, the classification process becomes better.

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	74.22	65.19	57.45	57.93	59.26	68.29
2	85.17	80.00	74.17	81.33	73.50	75.17
3	89.33	85.52	84.57	83.81	76.95	84.76
4	91.33	85.87	88.89	87.11	77.33	90.22
5	92.27	86.93	90.10	89.07	80.80	92.27

Table 5: *Performance of the proposed histogram based face recognition system in H, S, I, Y, Cb and Cr colour channels of the HP face database*

# of face images in the training set	Recognition rates (%) of the Proposed Histogram based system					
	H	S	I	Y	Cb	Cr
1	71.00	60.89	57.44	58.89	61.78	63.33
2	79.88	73.50	66.63	68.00	66.25	70.25
3	89.88	77.71	72.71	73.14	68.29	76.00
4	91.67	80.83	81.50	78.00	74.67	79.67
5	92.20	84.00	86.40	81.20	78.40	81.20

Table 6: *Performance of the proposed histogram based face recognition system in H, S, I, Y, Cb and Crcolour channels of the FERET face database*

The MV procedure can be explained as follows. If $\{H_1, H_2 \ldots H_M\}_C$ is a set of training face images with different poses in colour channels C (*H, S, I, Y, Cb, Cr*), then for a given queried face image, the colour histograms H_{qC} (H_{qH}, H_{qS}, H_V, H_{qY}, H_{qCb} and H_{qCr}) can be used to calculate the correlation between H_{qC} and histograms of the images in the training samples as follows:

$$\chi_i = \max(H_{iC} \circ H_{qC}),\ i=H,S,I,Y,Cb,Cr \qquad (12)$$

Thus, the similarity of the i^{th} images in the training set and the query face can be reflected by χ_i, the maximum cross correlation coefficient. The image with the highest similarity measure in a channel, ε_i, (where *i=H, S, I, Y, Cb, or Cr*) is declared to be the vector representing the recognized subject. Given the decisions of each classifier in each colour space, the voted class *E* can be chosen as follows:

$$E = mv\{\varepsilon_H, \varepsilon_S, \varepsilon_I, \varepsilon_Y, \varepsilon_{Cb}, \varepsilon_{Cr}\} \qquad (13)$$

MV principle is employed to declare the most repeated class.

The proposed system using coloured histograms as the face feature vectors, maximum cross correlation coefficient as the histogram matching measure and MV principle as identifier, is tested on the HP and the FERET face databases, which contain 15 and 50 colour image subjects with 10 selected different poses. The face dataset is divided into training set of *n* images per subject and the rest of the images of a subject for the test set. The images used in the test set are not included in the training set. The correct recognition rates in percentages are included in **Table 7**. Each result is the average of 100 runs, where we have randomly shuffled the faces in each class. The results of the proposed system are encouraging, because with only having five images as training we can reach a recognition rate of 92.00% and 90.80% for the HP and the FERET face databases respectively.

MV is not the only way to combine the histograms of faces in different colour channels. Histogram vectors can also be simply concatenated with the FVF process which can be explained as follows.

Consider $\{H_1, H_2 \ldots H_M\}_C$ to be a set of training face images with different poses in colour channel *C (H, S, I, Y, Cb, Cr)*, then for a given query face image, the *fvf* is defined as a vector which is the combination of all colour histograms of the queried image (H_{qH}, H_{qS}, H_{qI}, H_{qY}, H_{qCb} and H_{qCr}):

$$fvf_q = [H_{qH}, H_{qS}, H_{qI}, H_{qY}, H_{qCb} \text{ and } H_{qCr}]_{1x1536} \qquad (14)$$

This histogram can be used to calculate the correlation between fvf_{qC} and histograms of the images in the training samples, as follows:

$$\chi_i = \max(fvf_{iC} \circ fvf_{qC}),\ i=H,S,I,Y,Cb,Cr \qquad (15)$$

Thus, the similarity of the i_{th} images in the training set and the queried face can be reflected by χ_i, which is the maximum cross correlation coefficient. The image with the highest similarity measure in a channel, ε_i, (*where i = H, S, I, Y, Cb, or Cr*) is declared to be the vector representing the recognized subject.

FVF has a more complex approach than MV as several histograms are being concatenated which increases the length of the vectors of the cross-correlation operation.

The suggested system using colour histograms as the face feature vectors, maximum cross-correlation coefficient as the histogram matching measure and FVF as identifier, is also tested on the HP face database. The correct recognition rates in percent are included in Table 7. Each result is the average of 100 runs, where we have randomly shuffled the faces in each class.

The results of the proposed system are outstanding because with only five images as a training set, 96.67% and 92.67% recognition rates are achieved for the HP and the FERET face databases respectively, compared with a recognition rates of 69.47% and 86.67% in the PCA-MV and LBP-MV based systems in case of using the HP face database, and 77.60% and 83.33% in the PCA-MV and LBP-MV based system in the case of using FERET face database. Among the proposed feature combination approaches, FVF outperforms the MV approach and gives 96.67% and 92.67% recognition rates for the HP and the FERET face

# of Training Images	HP Performance (%)				FERET Performance (%)			
	PCA-MV	LBP-MV	MV	FVF	PCA-MV	LBP-MV	MV	FVF
1	22.22	60.00	57.33	68.15	57.11	54.44	67.67	73.11
2	42.50	72.50	77.00	80.83	62.50	58.75	81.88	76.00
3	58.10	81.90	86.89	91.43	65.71	69.14	84.43	89.14
4	61.11	84.44	89.52	95.56	74.00	81.00	87.50	92.40
5	**69.47**	**86.67**	**92.00**	**96.67**	**77.60**	**83.20**	**90.80**	**92.67**

Table 7: *Performance of the proposed FVF and MV-based systems compared to PCA and LBP-based systems for 15 subjects from the HP face database*

Figure 9: *GUI main entrance window consisting of several interactive push buttons to train the system, do recognition and rest the system*

databases, where the MV method is 92% for the HP face database and 90.8% for the FERET face database. These results indicate the robustness of the proposed histogram-based approach. Obviously there is a limit of the angle of planer rotation where the face has been completely changed, e.g. if only back of head is in the image. It is important to notice that the performance of the proposed method with the use of MV and FVF has been improved significantly, compared to PCA-MV and LBP-MV.

Graphical User Interface

The whole software of this work has been written in MATLAB 7.5. To make the programme more user-friendly, the GUI toolbox of the software has been used.

In this section different windows of the programme will be introduced and discussed. The codes have been changed into an executable file were only a double click on the 'facerecog.exe' will start the programme. Creating an .exe, .obj and .dll file of the codes will make the programming of the robot in C/C++ much easier. These files can be called in in any high-level computer language.

Main Window

The main window is shown in **Figure 9**. This window consist of four push buttons, two figure location to show the input image and the matched image (in case of finding a matching image from the database) and a text box to announce the detection condition. The push buttons are:

- "Train the System": This will open the training window.
- "Recognize Me": This will start the recognition processes; no new window will be opened. All the procedure will be done while this window is open, the segmented face will be shown in the provided location for the input image and the detected or denied action (showing highly correlated image from the database in case of detection, or 'Access Denied' notice in case of rejection). The result of the recognition will be shown in the text box.
- "Initiate": This will initialize the whole programme. After this point the new training is necessary.
- "Help": This opens the help menu window.

Training Window

The training window, as shown in **Figure 10**, will be opened as soon as the 'Train the system' push button in the main menu has been pushed. This window consist of a push button to start taking photos from the person who is in front of the camera, a figure location to show the taken image and a text box to show the number of taken images.

After pressing the push button, 10 photos with the delay of 2 seconds will be taken. After taking a photo, the image will be processed and the face will be segmented out. Then the histogram of the segmented image will be obtained and be saved in the appropriate folder. When the 10 images have been taken, the "infomenu" window will be opened.

Infomenu Window

The infomenu window will be opened when 10 images for the training have been taken. It consists of four text boxes, where the user can enter the name, surname, ID number and the priority of the trained person. After entering the submit button, the information will be saved in a text file in a folder. There are three pre-set options for the priority: High, Medium and Low.

If a person with higher priority enters, a control signal will be sent to the robot by the computer that it should stop following the person.

High-Performance Face Recognition System

In this paper we introduced a high performance face recognition system based on colour histograms. A pre-processing section was employed to isolate the faces from the background and histogram matching is used to perform face recognition by cross correlating the histogram of a given face and the histograms of the faces in the database. Maximum cross correlation coefficient between the histogram of a given face and the histograms of the faces in the database was used to perform histogram matching. The combination of feature vectors in different colour channels, by using MV and FVF, improved the performance of the suggested system. The results show that the system is robust to pose changes due to the high correlation between the histograms of the pose varying faces. The performance under varying pose conditions reached 96.67% and 92.67% on the HP face database and the FERET face database using the FVF approach compared to 69.47% and 77.60% using the PCA-MV based eigenfaces approach.

Finally, a graphical user interface software has been developed to implement the proposed technique. ■

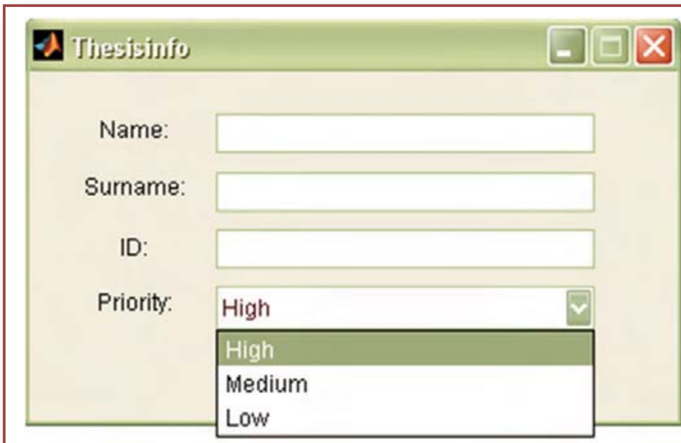

Figure 10: *Infomenu window where the user enters the information of the trained person*

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With estimates that up to 80% of a product's costs are set at the design stage, Envirowise believe businesses could unlock millions of pounds in potential savings by considering eco design principles during the product and packaging development cycle.

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PLC with PIC16F648A Microcontroller – Part 14

Associate Professor Dr Murat Uzam from Nigde University in Turkey presents a series of articles on a project that focuses on a microcontroller-based PLC. This is the fourteenth article where five examples are provided to show the use of multiplexer macros described in the last issue's article – Part 13

IN THIS ARTICLE we will consider five examples, namely UZAM_plc_8i8o_exN.asm, where N = 21, 22…25, to show the use of multiplexer macros described in the previous article. In order to test the example you can download the files from http://host.nigde.edu.tr/muzam/ and then open the UZAM_plc_8i8o_exN.asm program by MPLAB IDE and compile it.

Following that, with the PIC programmer software take the compiled file "UZAM_PLC_8i8o_exN.hex" and with the PIC programmer hardware send it to the program memory of PIC16F648A microcontroller within the UZAM_PLC. After loading the "UZAM_PLC_8i8o_exN.hex", switch the 4PDT in "RUN" and the power switch in the "ON" position. Finally, you are ready to test the example program.

To check the correctness of each program, you are referred to the related information for each multiplexer macro provided in Tables 1, 2…6 of the previous article. Note that, in some of these examples, we use some of the eight free running reference timing signals, namely T8, T9…T15 with the "T" timing periods 131.072ms, 262.144ms…16777.216ms respectively.

Example Programs

The first example program, "UZAM_plc_8i8o_ex21.asm", is shown in **Figure 1**. It shows the usage of two multiplexer macros "mux_2_1" and "mux_2_1_E". The schematic diagram of the user program of "UZAM_plc_8i8o_ex21.asm" shown in Figure 1, is depicted in **Figure 2**. In the first rung, the

```
;
        #include <definitions.inc>       ;basic PLC definitions, macros, etc.
        #include <cntct_mcr_def.inc>     ;Contact & Relay based macros
        #include <mux_mcr_def.inc>       ;Mux based macros
;
;--------------- user program starts here -----------------------
        mux_2_1      I0.0,I0.2,I0.1,Q0.0      ;rung 1

        ld           I0.6                     ;rung 2
        mux_2_1_E    I0.7,T11,T10,Q0.7
;--------------- user program ends here -------------------------
;
```

Figure 1: *The user program UZAM_plc_8i8o_ex21.asm*

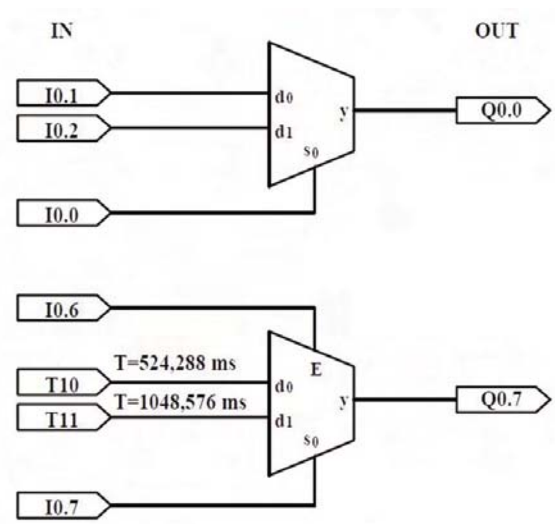


Figure 2: *Schematic diagram for the user program UZAM_plc_8i8o_ex21.asm*

```
;
        #include <definitions.inc>       ;basic PLC definitions, macros, etc.
        #include <cntct_mcr_def.inc>     ;Contact & Relay based macros
        #include <mux_mcr_def.inc>       ;Mux based macros
;
;--------------- user program starts here -----------------------

        mux_4_1      I0.0,I0.1,I0.5,I0.4,I0.3,I0.2,Q0.0  ;rung 1

;--------------- user program ends here -------------------------
;
```

Figure 3: *The user program UZAM_plc_8i8o_ex22.asm*

multiplexer macro "mux_2_1" (2×1 multiplexer) is used. In this multiplexer, the input signals are d0 = I0.1 and d1 = I0.2, while the output is y = Q0.0 and the select input is s0 = I0.0.

In the second rung, the macro "mux_2 _1_E" (2×1 multiplexer with active high enable input) is used. In this multiplexer, the input signals are d0 = T10 (524.288ms) and d1 = T11 (1048.576ms), while the output is y = Q0.7 and the select input is s0 = I0.7. In addition, the active high enable input E is defined to be E = I0.6.

The second example program, "UZAM_plc_8i8o_ex22.asm", is shown in **Figure 3**. It shows the usage of the multiplexer macro "mux_4_1". The schematic diagram of the user program of "UZAM_plc_8i8o_ex22.asm" shown in Figure 3, is depicted in **Figure 4**. In this example, the multiplexer macro "mux_4_1" (4×1 multiplexer) is used. In this multiplexer, the input signals d0, d1, d2 and d3 are defined to be I0.2, I0.3, I0.4 and I0.5 respectively, while the output is y = Q0.0 and the select inputs s0 and s1 are I0.1 and I0.0.

The third example program, "UZAM_plc_8i8o_ex23.asm", is shown in **Figure 5**. It shows the usage of the multiplexer macro "mux_4_1_E". The schematic diagram of the user program of "UZAM_plc_8i8o_ex23.asm" shown in Figure 5, is depicted in **Figure 6**. In this example, the multiplexer macro "mux_4_1_E" (4×1 multiplexer with active high enable input) is used. In this multiplexer, the input signals d0, d1, d2 and d3 are defined to be as free running reference timing signals T08, T09, T10 and

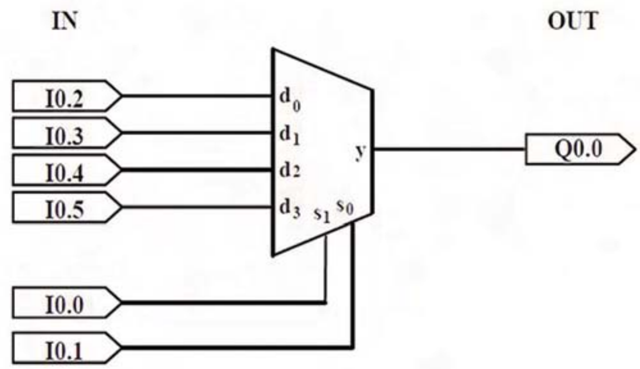


Figure 4: *Schematic diagram for the user program UZAM_plc_8i8o_ex22.asm*

```
#include <definitions.inc>    ;basic PLC definitions, macros, etc.
#include <cntct_mcr_def.inc>  ;Contact & Relay based macros
#include <mux_mcr_def.inc>    ;Mux based macros

;--------------- user program starts here -----------------------

    ld          I0.7                                    ;rung 1
    mux_4_1_E   I0.0,I0.1,T11,T10,T09,T08,Q0.0

;--------------- user program ends here -------------------------
```

Figure 5: *The user program UZAM_plc_8i8o_ex23.asm*

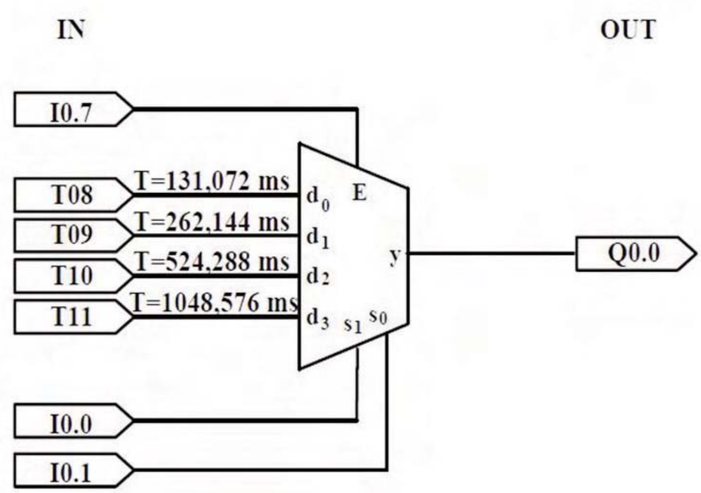


Figure 6: *Schematic diagram for the user program UZAM_plc_8i8o_ex23.asm*

```
#include <definitions.inc>    ;basic PLC definitions, macros, etc.
#include <cntct_mcr_def.inc>  ;Contact & Relay based macros
#include <mux_mcr_def.inc>    ;Mux based macros

;--------------- user program starts here -----------------------

  mux_8_1   I0.0,I0.1,I0.2,T15,T14,T13,T12,T11,T10,T09,T08,Q0.0  ;rung 1

;--------------- user program ends here -------------------------
```

Figure 7: *The user program UZAM_plc_8i8o_ex24.asm*

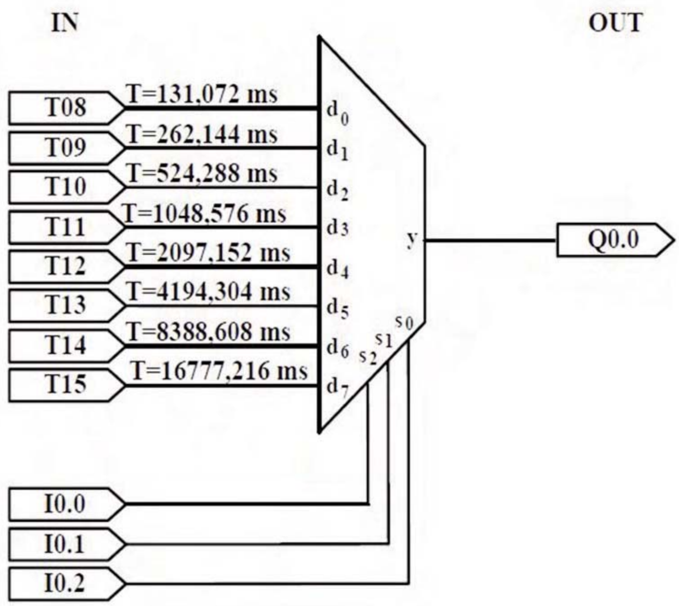


Figure 8: *Schematic diagram for the user program UZAM_plc_8i8o_ex24.asm*

```
;
        #include <definitions.inc>      ;basic PLC definitions, macros, etc.
        #include <cntct_mcr_def.inc>    ;Contact & Relay based macros
        #include <mux_mcr_def.inc>      ;Mux based macros
;
;--------------- user program starts here -----------------------
;
   ld   I0.7                                                    ;rung 1
   mux_8_1_E   I0.0,I0.1,I0.2,T15,T14,T13,T12,T11,T10,T09,T08,Q0.0
;
;--------------- user program ends here -------------------------
;
```

Figure 9: *The user program UZAM_plc_8i8o_ex25.asm*

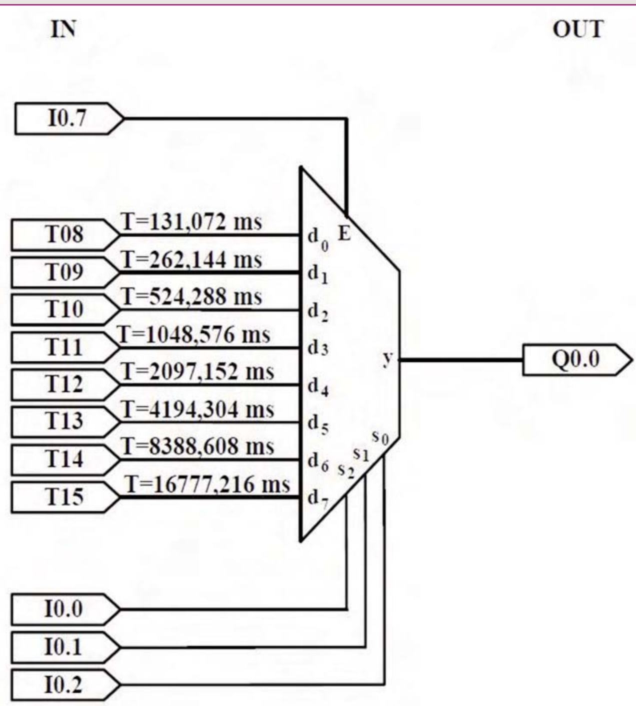


Figure 10: *Schematic diagram for the user program UZAM_plc_8i8o_ex25.asm*

T11 respectively, while the output is y = Q0.0 and the select inputs s0 and s1 are I0.1 and I0.0. In addition, the active high enable input E is defined to be E = I0.7.

The forth example program, "UZAM_plc_8i8o_ex24.asm", is shown in **Figure 7**. It shows the usage of the multiplexer macro "mux_8_1". The schematic diagram of the user program of "UZAM_plc_8i8o_ex24.asm" shown in Figure 7, is depicted in **Figure 8**. In this example, the multiplexer macro "mux_8_1" (8×1 multiplexer) is used. In this multiplexer, the input signals d0, d1, d2, d3, d4, d5, d6 and d7 are defined to be as free running reference timing signals T08, T09, T10, T11, T12, T13, T14 and T15 respectively, while the output is y = Q0.0 and the select inputs s0, s1 and s2 are I0.2, I0.1 and I0.0.

The fifth and last example program, "UZAM_plc_8i8o_ex25.asm", is shown in **Figure 9**. It shows the usage of the multiplexer macro "mux_8_1_E". The schematic diagram of the user program of "UZAM_plc_8i8o_ex24.asm" shown in Figure 9, is depicted in **Figure 10**. In this example, the multiplexer macro "mux_8_1_E" (8×1 multiplexer with active high enable input) is used. In this multiplexer, the input signals d0, d1, d2, d3, d4, d5, d6 and d7 are defined to be as free running reference timing signals T08, T09, T10, T11, T12, T13, T14 and T15 respectively, while the output is y = Q0.0 and the select inputs s0, s1 and s2 are I0.2, I0.1 and I0.0 respectively. In addition, the active high enable input E is defined to be E = I0.7. ■

If you've missed any of the previous articles in this series, you can now order it on line at ***www.electronicsworld.co.uk***

SIMPLE LDO WITH BIG SPECS

THIS LOW DROP-OUT (LDO) voltage regulator uses an N-Channel power MOSFET as series pass element of a power rating that suits your application.

N-Channel MOSFETs with very low ON-resistance and high current ratings can be used to achieve very low voltage drops. If a bipolar pass transistor were to be used, it would need substantial base current. By contrast, a MOSFET needs a much lower drive current on the gate.

This design inherently has no voltage overshoot during start-up and exhibits excellent load regulation. The difficulty with using a power MOSFET is how to charge and discharge the MOSFET's gate-source capacitance quickly. In this design Q2 ensures a very quick turn-off of the pass transistor. R1 controls how quickly the pass transistor turns on. Because turn-on is slower than turn-off, there is no voltage overshoot during power-up. When a load suddenly disconnects, the pass transistor is still 'on' and the previously established load current now dumps into the output capacitor C2. The output voltage increases slightly and the feedback loop turns off pass transistor Q1.

C2, therefore, needs to be large enough to absorb this short current spike and not build up much voltage overshoot.

Transistor Q3 acts as a current source. Therefore, the drain of Q3 allows fluctuations of drain voltage without much change in the drain current. This is why input voltage ripple has little effect on the output voltage. C2 is also needed to prevent oscillation.

C3 needs to be added to ensure stability at all load currents. Make C3 large enough to prevent oscillation but small enough to ensure quick response to load variations.

The circuit values given here work well for a transformer voltage around 8Vac at no load, and 1A of load current at 5V output. The circuit is presented here as a concept for your own design. You'll, therefore, need to find your own best values for the components.

Here are some guidelines.

The DC input voltage and ripple depend on the stiffness of the supply transformer and your choice for C1. The value for C1 strongly affects the ripple voltage on C1 for a given load current. Use this equation as a first approximation: voltage ripple = (I_{load} x T)/C, where T is the time between charge pulses (10ms for 50Hz operation). A good starting point for the value of C1 is 2000uF for every ampere of load current. Use the smallest value that will work to give the smallest size of C1 and lowest cost.

For C2, start with a value 100 times smaller than C1, and use a film capacitor of 0.1uF in parallel to C2 to decouple higher frequencies. Because this LDO is a linear regulator, the pass transistor will most likely need a heatsink to remain cool. Keep monitoring the temperature of the heatsink while you experiment.

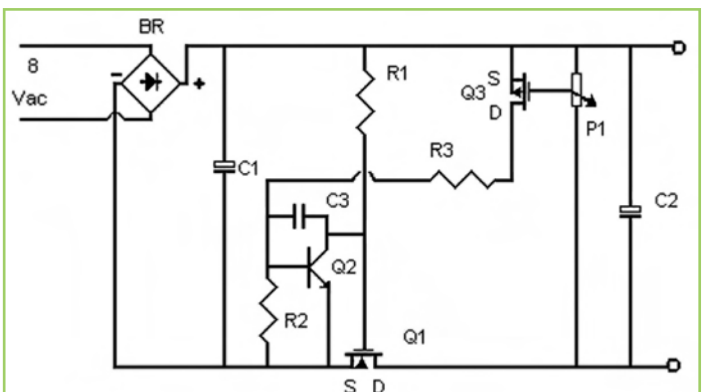


***Figure 1:** Low drop-out voltage regulator*

Component list for 8V AC input and 5V, 1A output:

Q1	**any N-Channel power MOSFET**
Q2	**2N3904 or other NPN small signal transistor**
Q3	**NDF0610 or any small P-Channel enhancement mode MOSFET**
R1	**1kOhm**
R2	**330Ohm**
R3	**1kOhm**
C1	**2200uF 25V**
C2	**22uF 16V or larger**
C3	**560pF or more, do experiment with it!**
P1	**Potentiometer 1kOhm to 100kOhm**

R3 is a current limit for times when Q3 is fully turned on. Select R3 to stay well below the maximum current limit allowed for the drain current of Q3 and the base current of Q2. Select R1 to limit the current into Q2 and to limit power dissipation in Q2.

A higher value for R3 gives more loop gain and, thus, better static load regulation but slows dynamic load regulation. Voltage feedback is accomplished with a potentiometric voltage divider in the form of a potentiometer P1. The value for the potentiometer is, therefore, not very critical. For Q3 a PNP transistor can also be used. ■

Robert Bliek
Canada

Figure 5: *Magnitudes of an ideal inductor and the inductances of Figures 3 and 4*

Figure 6: *Inductor simulator of R. L. Ford and F. E. J. Girling's*

Figure 7: *Proposed circuit for the one by R. L. Ford and F. E. J. Girlinguc*

Figure 8: *The magnitudes of an ideal inductor and the inductances in Figures 6 and 7*

KONTRON'S INDUSTRIAL SERVER KISS 4U KTQ45 OFFERS COMPREHENSIVE REMOTE MANAGEMENT FEATURES

Kontron's new, long-term available industrial server Kontron KISS4U KTQ45 offers highest level data processing for industrial applications thanks to its attractive Intel Core2 Quad processor Q9400 performance and wide range of management features.

Equipped with Intel AMT 5.0 for remote maintenance, Kontron PC Condition Monitoring (PCCM) for remote monitoring and an Intel Trusted Platform Module for the highest level of data security, the high-end industrial server is the ideal virtualization platform with long-term availability for the most demanding industrial applications.

Kontron's latest high-performance industrial server is based on Kontron's own embedded ATX motherboard Kontron KTQ45/ATXE with Intel 82Q45 Graphics and Memory Controller Hub, along with the Intel I/O Controller Hub 10 (Intel ICH10DO). With a front-side bus of up to 1.333MHz and up to 8GByte DDR3 SDRAM, the Kontron KISS 4U KTQ45 accelerates computing intensive embedded applications such as test and measurement, image processing, process control and video surveillance in energy, medical, automation or transportation market segments among others.

The Intel Trusted Platform Module (TPM 1.2) offers secure hardware encryption of all transferred data and with Kontron's PC Condition Monitoring solution (Kontron PCCM) preventive servicing can take place on an "on-demand" basis. Typical for the server range, the KISS server is equipped with an 80+ class 500 watt power supply unit, which can also be implemented as a redundant PSU.

www.kontron.com

HIGH-PERFORMANCE OSCILLOSCOPE FAMILY OFFERS EXTENSIVE ANALYSIS CAPABILITIES AND MIXED-SIGNAL MODELS

The Yokogawa DLM6000 mixed-signal oscilloscope is the flagship product in a new family of high-performance digital oscilloscopes featuring bandwidths up to 1.5GHz, memory of 6.25M points per channel, an intuitive graphical user interface and a number of advanced analysis features.

The new oscilloscope family consists of five 4-channel models: the two DLM6000 mixed-signal oscilloscopes with 16- or 32-bit logic capability and bandwidths of 500MHz and 1GHz; and three DL6000 versions with bandwidths of 500MHz, 1GHz and 1.5GHz, respectively. All the instruments have a maximum sampling rate of 5GS/s, apart from the DL6154 1.5GHz instrument with 10GS/s. The 16/32-bit logic inputs on the DLM6000 Series models have a maximum toggle rate of 250MHz or 100MHz, depending on the probe used.

The new range is priced very competitively, with the 16-bit DLM6054 with two 100MHz logic probes costing from 10,900 euros.

The new user interface, based on extensive market research, including detailed feedback from users, makes the instruments very easy to use. It incorporates a new physical layout with backlit buttons, new on-screen visual elements including graduated menus and innovative controls including a four-direction selector button and Yokogawa's unique 'jog shuttle' control.

In addition, the oscilloscopes offer an extensive range of capabilities for waveform characterisation, powerful tools for detecting glitches and anomalies, advanced signal enhancement and noise reduction technologies, and a range of options for serial-bus analysis and power measurement.

http://tmi.yokogawa.com/ea

DISTRIBUTOR LAUNCHES NEW PRODUCT FOR THE ELECTRONICS MARKET

contralube 770

Contralube770, a new product for electronic/electrical engineers is now launched, recommended for use with multi-pin connectors, spade, bullet and any other type of push fit electro mechanical contact area. It tackles issues such as fretting corrosion, oxidation, intermittent circuitry faults, moisture ingress and contamination issues.

This state of the art grease was created to help people that use electronic and data equipment that is exposed to the weather and/or circuit connections used alongside machinery, engines and any other equipment that vibrates. It was originally designed for the automotive industry as there is a constant battle with vibration corrosion and water ingress problems in their connectors and circuitry.

Now Contralube770 can be applied to electronic and data connectors in any industry and it will not degrade power levels, analogue or digital signals. The product is compatible with all noble and non-noble metals and most plastics.

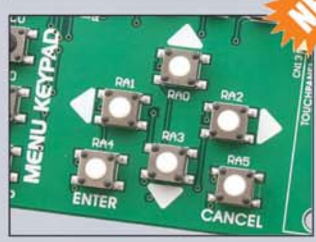
It is hailed as ideal preventative maintenance solution, stopping problems before they occur and less equipment down time.

Contralube770 is a higher spec and more permanent solution than the general purpose spray lubricants that are currently sold.

Contralube 770 is in stock and available to purchase from **www.rapidonline.com**

For more information, suggested uses and technical data on contralube770, visit

www.contralube.com



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Designed to suit your needs

Your development time can be considerably reduced, resulting in an early prototype design and fast time-to-market for your end product.

Thanks to many new features, you can start creating your own devices immediately. **EasyPIC6** supports **8-, 14-, 18-, 20-, 28- and 40- pin PIC** microcontrollers. The **mikroICD** (Hardware In-circuit Debugger) enables very efficient step by step debugging. Examples in **C**, **BASIC** and **Pascal** are provided with the board.

Hardware In-Circuit Debugger for step by step debugging at hardware level

On-Board 2x16 Serial LCD Display

Port Expander provides easy I/O expansion (2 additional ports) using serial interface

Keypad enables easy and fast data entry

Full-featured and user-friendly development board for PIC microcontrollers

High-Performance USB 2.0 On-Board Programmer

lead-free

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We deliver our products across the globe and our satisfied customers are the best proof of our first-rate service. The company is an official consultant on PIC microcontrollers and a Third Party Partner of Microchip Technology. We are also an official consultant and Third Party Partner of Cypress Semiconductors since 2002 and official consultants to Philips Electronics. All our products are RoHS compliant.

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CURRENT SENSOR FEATURES 120KHZ BANDWIDTH, HIGH-VOLTAGE ISOLATION AND INTEGRATED OVERCURRENT DETECTION

The new ACS710 from Allegro MicroSystems Europe is a 120kHz bandwidth, high-voltage isolation current sensor IC that provides an economical and precise means for current sensing applications in industrial, commercial and communications systems.

The new device consists of a precision linear Hall sensor integrated circuit with a copper conduction path located near the surface of the silicon die. Applied current flows through the copper conduction path, and the analogue output voltage from the Hall sensor IC linearly tracks the magnetic field generated by the applied current. This patented packaging configuration gives optimum accuracy because the Hall element is situated very close to the current being measured.

In addition, the current conduction path is electrically isolated from the low-voltage sensor inputs and outputs. This allows the ACS710 family to be used in applications requiring electrical isolation without the use of opto-isolators or other costly isolation techniques.

Allegro's proprietary integrated shield technology leads to a high degree of immunity to voltage variations and stray electric fields in the current conductor, guaranteeing low output ripple and low offset drift in high-side, high-voltage applications.

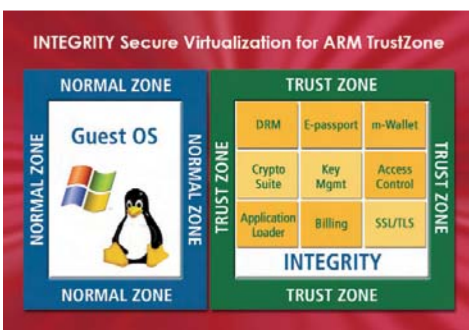
The voltage on the overcurrent input allows customers to define an overcurrent fault threshold for the device. When the current flowing through the copper conduction path exceeds this threshold, the open drain overcurrent fault pin will switch to a logic 'low' state.

www.allegromicro.com

A COMPREHENSIVE SOLUTION FOR ARM TRUSTZONE TECHNOLOGY-BASED DEVICES

Green Hills Software has announced INTEGRITY Secure Virtualization for ARM TrustZone technology. INTEGRITY Secure Virtualization (ISV) is built on Green Hills Software's INTEGRITY RTOS, the first and only operating system technology to be certified by the NSA to EAL6+ High Robustness, the highest level of security ever achieved for any software product.

INTEGRITY Secure Virtualization reduces costs through processor consolidation, improves system security and reliability, and reduces the certification burden in mission critical applications such as banking, industrial control, and medical systems. ISV supports TrustZone technology-enabled ARM cores, including the ARM1176JZ(F)-S, Cortex-A8 and Cortex-A9 processors, on leading system-on-chip (SoC) implementations from Texas Instruments Incorporated (TI), Freescale and other semiconductor partners.



INTEGRITY Secure Virtualization consists of the certified INTEGRITY operating system technology, coupled with facilities to execute one or more "guest" operating systems on TrustZone technology-enabled SoCs. INTEGRITY Secure Virtualization also provides a communications interface for the normal zone to request services of the secure zone. This architecture enables security and reliability-critical applications to safely execute with guaranteed memory, CPU time resources, and device access control, regardless of what is happening in the normal zone.

INTEGRITY Secure Virtualization runs on commercial, off-the-shelf platforms such as the Texas Instruments OMAP 35x and Freescale i.MX515.

www.ghs.com

RS232 BLUETOOTH SERIAL PRINTER ADAPTER FEATURES "OUT OF PAPER" MESSAGE

LM Technologies Ltd Bluetooth serial adapter now uniquely features an "Out of Paper" message from the LM048SPA adapter used with printers in the EPOS/general printer market.

The LM048SPA is the world's smallest RS232 Bluetooth serial adapter and is already an industry winner as the preferred non-protruding Bluetooth serial adapter solution. It is quickly and easily configured to any EPOS/POS system via the LM Technologies's user-friendly and intuitive software configuration utility.

The LM048SPA adapter provides the security of Bluetooth V2.0 +EDR with the additional options of auto connection authentication via pin code, and activation of the "Non Discoverable Mode" function.

Power-saving functions and 7-bit data are also available as options.

Star Micronics and Sharp are just two companies that have extensively trialled and now endorsed this solution. A number of other major manufacturers will soon be concluding trials.

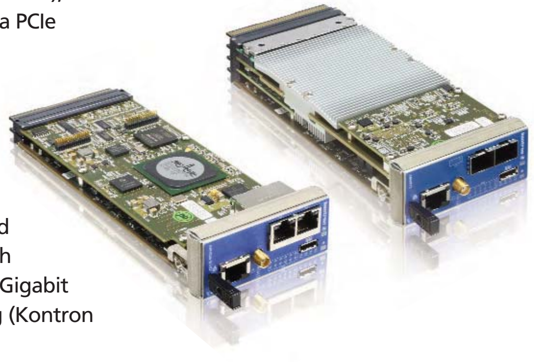
LM Technologies Ltd Business Division designs and manufactures a range of Bluetooth modules sold into industrial markets worldwide. LM Technologies offers a wireless integration path starting Bluetooth serial adapters migrating to bespoke module solutions. This enables clients to choose, dependant on volumes, the best solution to Bluetooth enable their products.

www.lm-technologies.com

HIGH-PERFORMANCE MICROTCA CARRIER HUB FAMILY SUPPORTS FOUR DIFFERENT HIGH-SPEED FABRICS

Kontron has supplemented its AdvancedMC/MicroTCA portfolio with the roll-out of the new modular Kontron AM4904/AM4910 MCHs (MicroTCA Carrier Hubs). To fulfill the management needs of nearly all data-intensive MicroTCA applications, with GbE, sRIO, PCIe or even 10 GbE fabrics, they support four different high-speed fabric variants.

The modular Kontron AM4904/AM4910 MCHs comes in four versions: a pure Gigabit Ethernet switching MCH (Kontron AM4904-BASE), in combination with a PCIe mezzanine (Kontron AM4904-PCIE) or with sRIO (Kontron AM4904-SRIO), and for high bandwidth demands, with 10 Gigabit Ethernet switching (Kontron AM4910).



All versions provide central management and data switching functionality on a full-size, single-width Advanced Mezzanine Card (AMC) form factor. Equipped with a powerful 600MHz PowerPC 405EX processor for MCMC functionality and switching management, they enable highly efficient, redundant system architectures with up to 12 AMCs, two cooling units and four power units. With a focus on high-performance applications, Kontron's latest MCHs are a perfect fit for data-intensive MicroTCA based communication applications in both telecoms, including 3G, LTE and network test equipment, and non-telecoms markets, such as military, medical, test and measurement, as well as image and video processing applications.

www.kontron.com

KONTRON UNVEILS NEW 1U MICROTCA PLATFORM FOR CARRIER ENVIRONMENTS

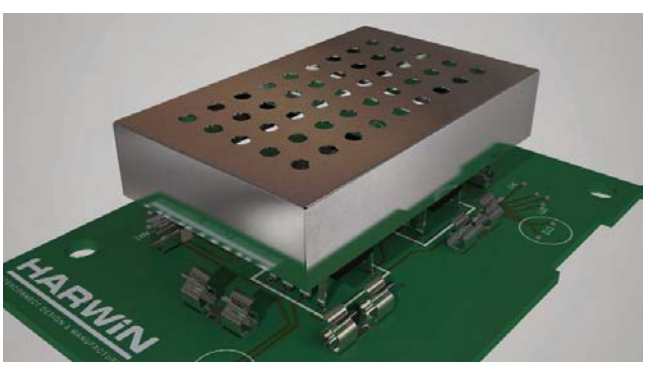
Kontron introduced the Kontron OM6061, a carrier grade 1U MicroTCA platform ideal for a wide assortment of Central Office and service aggregation point applications, ranging from SIP Servers, SSL Offload, and content-aware processing and QoS over Ethernet.

Featuring an 8-fan front-to-back cooling set-up, an integrated 360W -48V or -60V power supply, and a cost-optimized MCH module within in a 1U, 12.5 deep rack-mountable platform, the MicroTCA platform Kontron OM6061 offers six mid-size AdvancedMC slots for multiple system configurations. The carrier grade telecommunication platform is designed to meet NEBS and has been fully pre-tested with the cost optimized MicroTCA Carrier Hub Kontron AM4901, and the AdvancedMC modules for storage (Kontron AM4510), CPU (Kontron AM4010) and network processing (Kontron AM4204). The three remaining slots are optionally open for a redundant AMC configuration or additional AMC modules.

Also of interest to equipment vendors is that the Kontron OM6061 can be populated with up to six 10GbE Cavium-based AMC modules Kontron AM4220 for sustained traffic of up to 120Gb/s. Moreover, this carrier grade telecommunication platform features a high speed backplane and can already support next generation AdvancedMC modules with PCIe Gen2 and XAUI.

www.kontron.com

HARWIN ADDS LARGER SIZE SMT SHIELD CLIPS TO PORTFOLIO

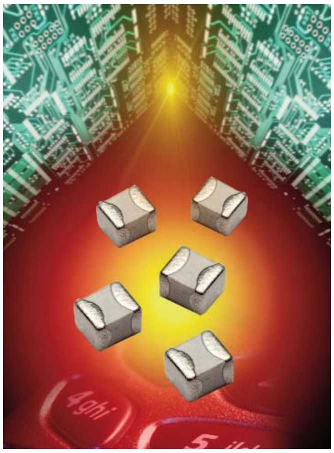


Harwin, a manufacturer of connectors and interconnect solutions for high-reliability applications, has broadened the range of surface mount EMI/RFI shield clips it offers with the launch of its Maxi Clip, which handles protective cans with wall thicknesses of between 0.7 and 1mm.

The use of clips reduces assembly time and simplifies rework. Rather than hand-soldering shielding cans to a PCB as a secondary operation, cans are simply be pressed into clips which have been placed during the main automated surface mount assembly processing. Not only does this save time, it also reduces errors and scrap, and enables sensitive components to be inspected and, if necessary, lifted and replaced much more easily.

The new Maxi clips compliment Harwin's Midi clips which handle cans with a wall thickness of 0.3mm and its Mini series devices which save PCB space and handle cans with a wall thickness of between 0.13 and 0.23mm. Supplied in industry-standard Tape & Reel packaging, Harwin's EMI/RFI clips are targeted at telecoms, datacoms, data processing and in-car entertainment systems. They have proved successful in cost-effectively helping to protect sensitive ICs used in diverse applications such as the Bowman Military radio project and automotive SatNav and DVD players.

www.harwin.com



AVX EXPANDS VOLTAGE RATING OF MLCC FOR HIGH POWER APPLICATIONS

AVX Corporation, a manufacturer of advanced passive components, has extended the working voltage range of its RF multilayer ceramic capacitor (MLCC) from 100VDC to 250VDC. Designated the SQCA Series, the high performance MLCC provides low ESR, high Q, high self-resonance, and a capacitance range of 0.1 pF to 100 pF in a 0605 case size.

Providing superior stability under the stresses of changing voltage, frequency and temperature, the SQCA Series MLCC is ideal for use in microwave RF/IF amplifier, low-noise amplifier, mixer, oscillator, filter network and MRI system applications.

The SQ Series MLCC chips feature a porcelain and ceramic dielectric material that provides high reliability, low ESR and up to 40% lower equivalent series resistance than previous versions, making the SQCA Series ideal for applications with high-current carrying capabilities and high quality factors.

The SQCA Series is RoHS compliant due to 100% tin terminations. Tin/Lead terminations are also available.

Pricing of the SQCA Series typically starts at $0.35 to $1.50 in quantity with a lead time of six weeks.

www.avx.com

CHROME IPOD DOCK ADDS A TOUCH OF CLASS TO IPOD OWNERS STEREO

For the music lover with almost everything, the engraved iPod, the Bang and Olufsen stereo is one accessory everyone should have: a chrome plated iPod dock.

Hastings-based electrical products manufacturer, Focus SB, has created the new piece in collaboration with in-house designer Terri McConnell.

Each dock, complete with remote control, has the designer's signature, and will be engraved with the number in the run limited to the first 50.

The docks come with a remote that allows users to change the volume or select a new song/album or artist. These limited edition docks are £125 each and can be purchased from Focus by telephoning their sales office on 01424 858060.

Focus SB is also offering customers the opportunity to have docks engraved with their own company name, logo or message for a one off £10 additional charge (minimum order 20 units).

www.focus-sb.co.uk

KONTRON NANO CLIENT 10.4": A NEW, ROBUST X86 ALTERNATIVE TO RISC PANELS

The new Kontron Nano Client 10.4" is a fanless Touch Panel PC with Intel Atom Z5xx Processor, which is protected by a fully-sealed IP66 stainless steel housing and is recommended for use as an HMI for harsh environments with high hygiene requirements. Its open x86 architecture also facilitates variant creation and offers a future-proof alternative to RISC-based solutions.

In contrast to RISC-based HMIs, which are clearly more application limited and dedicated to specific functions, the Kontron Nano Client 10.4" offers an open and therefore easily-scalable x86 platform. The ability to run the same software on a range of x86-based platforms means development efforts can be reduced. The hardware enables application-specific variations to be developed-OEM versions of the system are available through Kontron ODM Services (Original Design and Manufacturing).

The 10.4-inch touch panel PC is fully protected against dust and spray and meets the highest hygienic requirements. Its IP66-protected stainless steel (V2A) housing is free of any grooves and edges where dirt could accumulate.

The Kontron Nano Client 10.4 is equipped with a 45nm Intel Atom Z5xx processor up to 1.6GHz and the highly-integrated Intel System Controller Hub US15W, and a maximum of 1024 MB soldered RAM, allowing it to run demanding web-based visualizations.

www.kontron.com

WORLD'S FIRST LOW-VOLTAGE, HIGH-SPEED QUAD I/O SERIAL FLASH MEMORY

SST (Silicon Storage Technology) announced the industry's first 1.8V, high-speed quad-bit serial flash memory. Featuring an 80MHz operating frequency and a specialized instruction set, the new 26WF Series Serial Quad I/O (SQI) family of 4-bit multiplexed I/O serial interface devices enables execute-in-place (XIP) capability, allowing programs to be stored and executed directly from the flash memory without the need for code shadowing on an SRAM. The 26WF Series, the first high-performance 1.8V quad I/O serial flash product, marries the performance typically associated with parallel flash memory, with the low pin-count and space savings of serial flash memory, plus reduced power. This combination of high performance and low power consumption is ideal for mobile handsets, Bluetooth headsets, GPS devices and other small form factor, portable electronics.

Today's announcement builds on the success of SST's 3.0V SQI flash products, the 26VF Series, where all devices feature a novel read memory indexing which allows one of the fastest random access performances around.

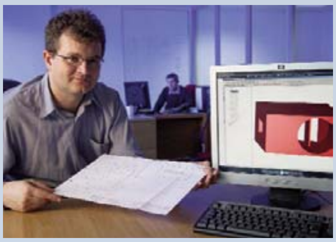
www.sst.com

LINK INTRODUCES DESIGN-AND-BUILD SERVICE

Link Microtek, the manufacturer and supplier of RF and microwave components and associated hardware, has introduced a new service for the design and manufacture of complete RF or microwave systems to meet customers' specific requirements.

Building on the company's long experience in the engineering of RF and microwave products, Link Microtek is able to provide a comprehensive system-level service, covering design, fabrication of parts using the company's in-house machine shop, system assembly, and testing of finished units at up to 40GHz in Link's own test and environmental laboratories.

The company's engineering team offers a genuine partnership and, working from a customer's initial concepts, can either design the system in its entirety or make recommendations for improvements in the original design.



Once the system design and specifications have been finalised, Link sources any standard components and then produces the necessary customised hardware in the machine shop at its premises in Basingstoke.

For further information, visit www.linkmicrotek.com.

ROBOT BUILDING MADE EASY

A new UK website – RobotBits.co.uk – launched for robot enthusiasts helps to make robot building easier by guaranteeing that mechanical and electronic components will work together.

"One of the concerns many robot builders face is having the confidence to know that the components they choose will actually fit and work together," said Jon Luke, founder of RobotBits.co.uk.

RobotBits has tackled this issue by including a 'works with' section on every product page of the website, helping customers choose components.

"As enthusiasts ourselves, we have suffered from similar frustrations, so have been careful to choose products from companies we trust and to fully test everything ourselves to ensure compatibility before adding them to our website," said Luke.

RobotBits features kits and components from many of the well-known robot brand names including Pololu, Solarbotics, POB and Arduino. RobotBits also offers free advice through its local-rate telephone line.

To find out more, visit www.RobotBits.co.uk or call on 0845 5 191282.

RobotBits.co.uk

WILSON PROCESS SYSTEMS WINS HEALTHCARE COWS CONTRACT

Sussex-based contract electronics manufacturer, Wilson Process Systems (WPS), has recently won a contract to deliver 1000 PCBAs to RDP Health Ltd, a UK manufacturers of medical computing products designed to operate in busy hospital environments.

RDP designs and produces COWS (Computer Carts on Wheels) for NHS Trusts. Usually, SFF PCs are mounted underneath a work-surface with either a low spec monitor for use on the wards, or a very high performance display for use in operating theatres. Power, battery and control functions are housed in the small footprint base under a covering.

Steve Knight, RDP's Operations Manager, said: "WPS was able to reduce costs with its component buying power and knowledge of alternative sources, and because they have some specialist equipment for mounting components such as LEDs etc. We supplied WPS with functional test gear so all boards are now supplied to us as 'known-good-products', and the first time we test units is for a full functioning test when the carts are assembled."

www.wps.co.uk

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Robot Bits

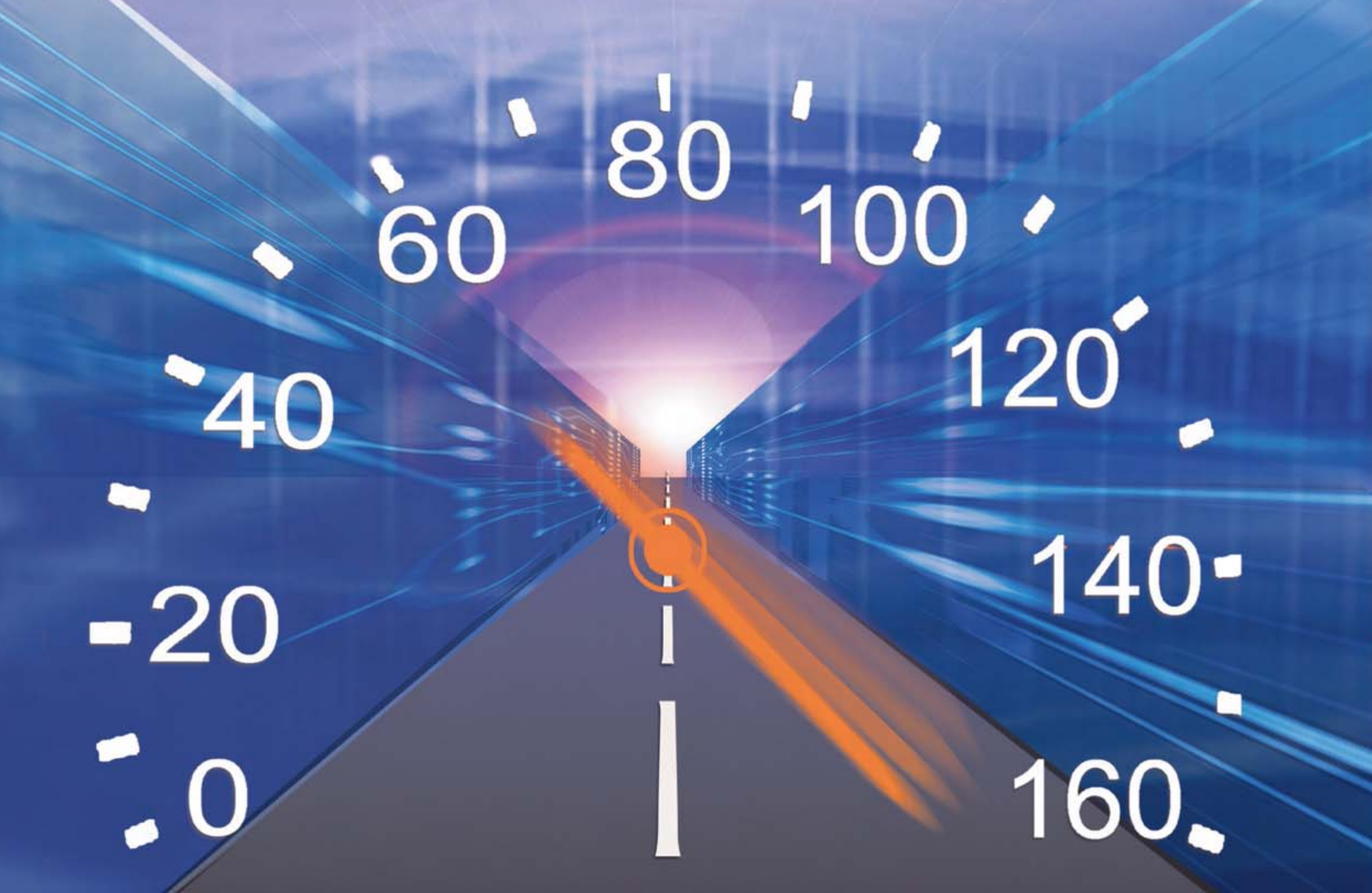
www.RobotBits.co.uk

New start-up makes Robot Building easier

A new UK website, launched for robot enthusiasts, helps to make robot building easier by guaranteeing that mechanical and electronic components will work together.

RobotBits.co.uk, a new start-up focused on providing robot kits and components to hobbyists, schools and universities, aims to make robot building easier and more accessible to enthusiasts of all ages by guaranteeing that the mechanical and electronic components purchased through their website will work together.

Colour screen shot and text is just £350 + vat for 12 months



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