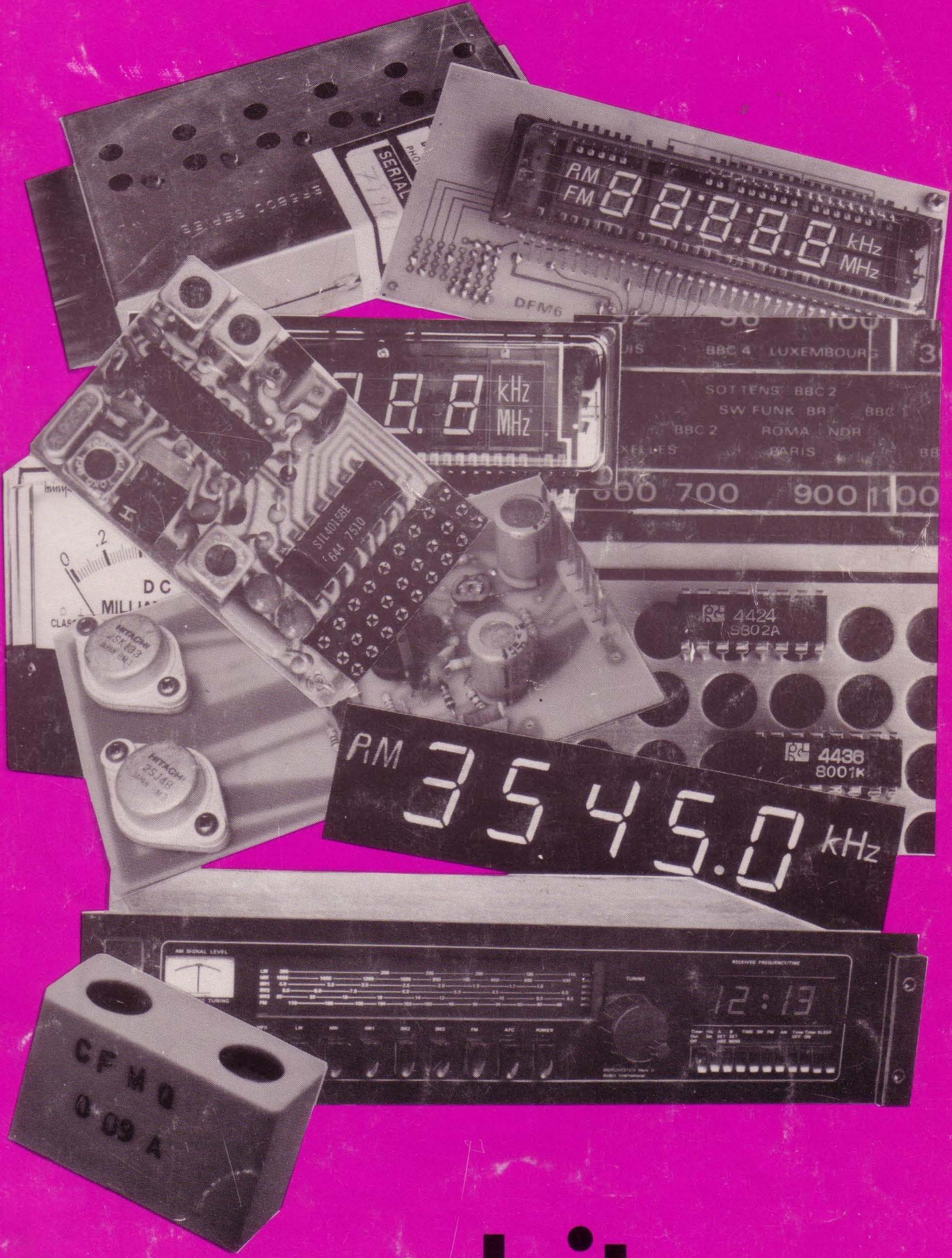
a communication on the subject of radio



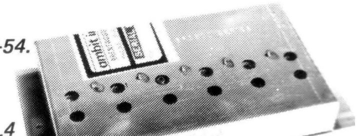
ambit international

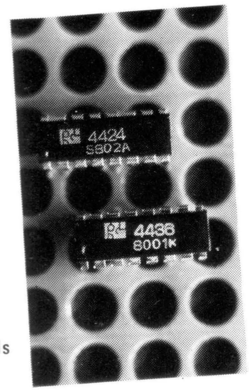
Tecknowledgey: number 3

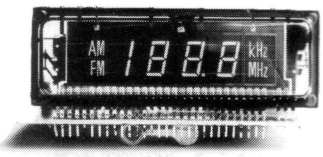
200 NORTH SERVICE ROAD, BRENTWOOD, ESSEX
Telephone: (0277) 230909 Telex: 995194

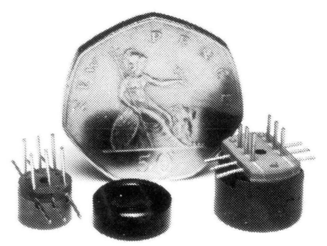
CONTENTS — an index combining parts 1, 2 and 3 of the catalogue

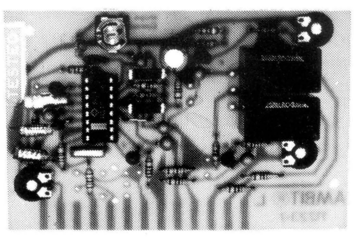
Items indicated 'PL', please refer to price list and newsletter(s) for details. Items marked CO are available to callers only.
Page numbers in BOLD type, indicate part 2 of the catalogue *Page numbers in italics refer to PART 3 of the catalogue.*

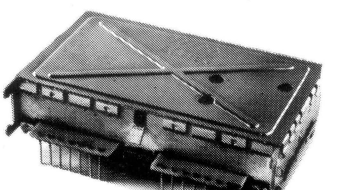


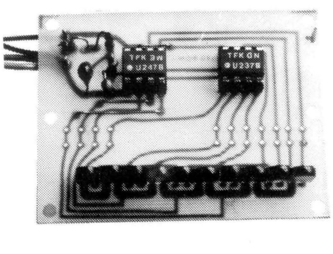












Abandon all hope ye who (etc)............

At the risk of appearing to be self indulgent, this page (as usual) sets out to offer a blend of the subjective and objective aspects of this edition in particular - and "things" in general. Firstly..........

"Things"

Ambit has evolved over the past few years into Europe's leading source of consumer radio components and ideas. We are now turning some of our attention to include communications ICs from Plessey - and a veritable plethora of frequency synthesisers of all shapes and sizes.

However, we have not entirely been able to dispel our image as retailers to the enthusiast market - and this still impedes the attitudes of some of our suppliers, who are simply too concerned with their own self esteem to be bothered to do business with the likes of a 'retailer'.

Those who are more enlightened have followed our progression into other markets, and we now enjoy a close working relationship with many major suppliers.

So if firms like Plessey, Ferranti, TI, RCA, Sanyo feel they are getting ignored - then we invite them to keep us informed of their contributions to the types of product areas we cover. We really don't have the time or inclination to spend hours phoning around to try and trace leads on new products.

Hopefully, now that we are at last customers for SL1600 Plessey will send round a representative one day. Until they do, we shall continue to wonder at their commercial practises.

If you think you have spotted some useful new parts that we should be aware of - then please let us know of the part and the source of the information, so that we may follow it through if suitable.

Ambit has always angled its approach to users/engineers rather than buyers, and as a result of this, we have taken a less than charitable view of some of those professional magazines that make excellent "lobby fodder" (to misquote a political expression). You know the sort of thing we mean, where the ad space reps always refer to their publication as either a "book" or a "journal" - and pass between titles within the publishing house such as "Foundation Garments News", "Brain Surgery Today", and "Nuclear Physics Review" like a bricklayer moving from one site to the next.

This slander is calculated on the basis that such persons wouldn't usually bother to turn past the cover of such an exciting and technically knowledgable publication as this - except perhaps to check the contents of page 3. Everybody duck !!

All this is leading up to the revelation that we are going to be seeking some advertizing from other electronics companies (in non-competetive fields, please.)

The ambit catalogue lasts longer than any of the present press - being not simply an annual thing, but something that is kept and treasured indefinately. We do not intend to supplant the functions of existing monthlies and weeklies, but merely provide an extra medium in the shape of a publication read and cherished by engineers. Accordingly, the advertizing will have to be "general" to take into account the life expectancy, and certainly not technically incompetent, like so much of the copy churned out by the brickies of the advertizing business.

What's New ?

The Ambit HQ premises for starters. If the site is tidy enough before this goes to press, then a picture may be on the back cover.

Within these sturdy walls, we hope to be able to continue to expand our general activities with more ease and general efficiency than under our previously crampt and confined conditions. In particular, we are extending our facilities concerned with technical publications to enable our kits to be made available to a much wider audience, through the preparation of very comprehensive instructions. Unless we can turn a project around in less than 6 months, there is a great danger that it will be completely out of date by the time it gets finished off. So a full in-house photography and printing facility is now added to our list of available services.

Component wise, frequency synthesis is the current boom interest for radio enthusiasts. After mauling the watch industry, digital electronics is at long last shaping up to decimate all the old approaches to the design and construction of radio equipment. We like to think we have been aware of most of what's going on - and believe our Hitachi FM/LW/MW/SW system is the first "complete to go" PLL/MPU set offered for the Euro-user. Following up that, we have done much work on the Mullard LN123/4 system - which will be revealed early in the new year as the companion system for or 5kHz to 30MHz AM/SSB/CW radio board described herein. Until Spring 1980, some of these projects will not be adequately documented for enthusiast customers, so until then fuller information and engineering advice is only available through our professional services schemes. ie Fee paying customers for the time being.

We have taken the opportunity that will be afforded by the easy access to versatile synthesiser systems to begin to include more communications parts - such as the SL1600 series, diode mixers, HF crystal filters etc. We are also able to supply these components ready assembled into sub-systems and modular designs - and apart from the gradual introduction of stock parts - we are pleased to quote for custom designs.

Radio control makes its debut in this issue, and we think we have introduced several parts likely to become "standards" in the next few years. The RCM&E series by Terry Platt has gathered a huge following of enthusiasts - and we offer a variation on the theme of the basic receiver, with a more repeatably designed front end using TOKO screened coils, and a 27MHz ceramic filter.

The TOKO one-chip transmitter and receiver ICs will find many applications in 4 channel systems- and the NE5044 will doubtless become the prime encoding device.

On a less glamourous note, we have extended the range of ferrite beads to include some very small types for insertion on transistor leads - and these can solve an amazing number of instability problems in RF amplifiers - as well as keeping Radio Moscow out of your moving coil cartridge pick-up amplifier.

If CB happens, we should be well placed to supply the bits for any frequency chosen. If the present racket of Euro CB on 27MHz is anything to go by, there really isn't much point in doing anything with that frequency except perhaps offer it to the Russian Embassy for diplomatic transmissions.

Freebies:

Despite our attempts to give away free money to enterprising enthusiasts - we still get shamefully little reaction from budding authors. Those brave souls who try and suggest an idea to pursue frequently find that by the time it's all ready, then technology has swept past to the next stage. A brief look at recent editions of Radio Communication affords an example.

The most useful work the enthusiast can do, is to try and do something different with an apparently straightforward IC. The RCME RC receiver is an excellent example of this type of thinking at work - where a use is evolved that the designer of the IC would never have dreampt of.

Most radio ICs have alter egos waiting to be unleashed - but we are still waiting to hear from enthusiasts willing to have a go under our sponsered parts scheme. Maybe schools/colleges would like to apply ?

SL1600 series

Despite discouragement at the hands of Plessey's commercial operatives, we have at last included this useful range of ICs

Introduction

Despite the impressive capabilities of the SL1600 range, and the undoubted excellence of the engineering and thought therein, we have still not yet come to terms with the commercial aspects of dealing with Plessey Semiconductors. Since catalogue space costs money, we will restrict this gripe to a bare minimum, but we would like to see a Plessey salesperson now and again.

The SL1600 series is simply the time honoured SL600 series of TO5 packaged devices tranferred to plastic DIL, with a more bearable price tag attached.
The range is described in some useful detail in the Plessey Radio Communications Handbook - written and edited by the ever helpful James Bryant. With such excellent technical and engineering support, it is a joy to use and develop with these ICs - and we strongly suggest that the investment in a copy of the handbook is well worth the money. (See our current PL for price details.)

Background

The 1600/600 series are basically built up around the need for a set of SSB communications devices. Most of the block functions of an SSB receiver are included in these excellent ICs:

SL1610 RF amplifier with Gv of 10, and 140MHz BW
SL1611 RF amplifier with Gv of 20, and 100MHz BW
SL1612 IF amplifier with Gv of 50, and 15MHz BW

These RF/IF amps all include provision for AGC from the SL1621/1625 and have been designed with RF very much in mind.

The SL1613 is a limiting amplifier, intended for FM use with applications in limited SSB RF clipping, and as an AGC generator. The BW of 5-150MHz enables the device to be used in a wide variety of applications, including direct conversion systems.

SL1620 Voice derived AGC system for SSB ALC
SL1621 Voice derived AGC for the SL1610/1/2

The basic action of these two is identical, namely to provide audio derived AGC for both transmit and receive systems. The SL1620 is designed for use in conjunction with the SL 1630 audio amplifier, and the SL1620 for use in conjunction with the voltage levels of the SL1610/1/2 - but both devices provide fast attack, programmable delay with a speech pause 'memory'. Apart from uses in communications, they make excellent AGC systems for TV sets in houses where there is a danger of waking the baby every time the audience bursts into fits of hilarity.

SL1623 AM detector, SSB demodulator and AGC detector.

This device is a low level AM detector (ie no high level IF signals to encourage instability) and provides carrier relatea AGC for the 1610/1/2 ICs. The SSB detector is of the differential variety.

SL624 AM/FM/SSB/CW detector system to 30MHz

The SL624 is one of the gems of the range, with a very versatile array of functions for a communications receiver. It employs a synchronous AM detector, quadrature FM demodulator - and a self oscillating (with external resonator) SSB/CW demodulator. A built in audio amplifier with DC control is also provided.

SL1625 AM detector, and AGC generator

The SL1623, but without the SSB detector - ie a low level AM and carrier related AGC detector.

SL1626 VOGAD mic amplifier system

A comprehensive device for providing a constant audio level in systems where a wide range of different inputs may be encountered. Mic amps in all forms of communications are the main uses, although the device may be used in the audio output of receivers to provide additional AGC. (Otherwise known as a compressor)

SL1630 200mW audio amplifier with DC control

A headphone/small speaker amplifier, with DC control facility. It is suited for use with the SL1620 AGC IC mentioned above.

SL1640 Double balanced mixer for generator use
SL1641 DBM for receiver stages

The SL1640/1 are refined versions of the long lived DBM series including the MC1496 etc. The 1640 is designed for low leakage, with IM of -45dB typ with 60mV RMS input. It has a useful 75MHz bandwidth - and both types are provided with trim facilities..

SL1664 - now SL6640 FM IF + Audio amplifier stage

The NBFM communications parts have now been given a "66XX" number - although the devices themselves are unchanged.
The SL6640 is described in part two of the catalogue series - briefly, it is a 10.7MHz input NBFM IF amplifier, detector, AF amplifier with DC volume control and adjustable muting. Apart from the obvious applications, the device may be used in variety of applications, including remote carrier intercoms, IR communication systems etc etc.

SL1669 - now SL6690 Ultra low power NBFM IF amplifier and detector

This part is designed for use in very low power and low voltage applications such as pagers and low power telemetry systems. The IC includes a 455kHz IF amplifier and detector, followed by a LF audio amplifier, and schmitt trigger. The trigger circuit may be used as either a pulse shaper (in digital/tone coded systems) or as an element of a S/N derived squelch system.

SL6600 Double conversion receiver with PLL detector

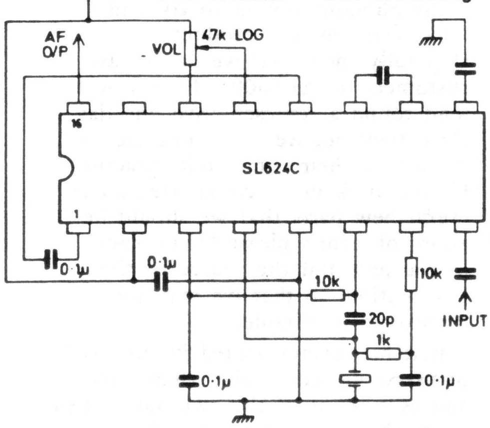
The top of the NBFM range from Plessey at present, the 6600 is a very sensitive device containing :
A limiting IF amplifier (good to 25MHz), followed by a double balanced mixer, crystal oscillator, 455kHz IF amplifier and a PLL NBFM detector system with squelch derived from sensing the state of the lock of the detector.
With a supply of 1.5mA at 7v, the SL6600 is easily accomodated in many apllications - both as original equipment, and as an add on NBFM stage. The typical 3 to 5uV sensitivity for 20dB S/N is quite exceptional in this type of device, and presently represents the state of this particular art.

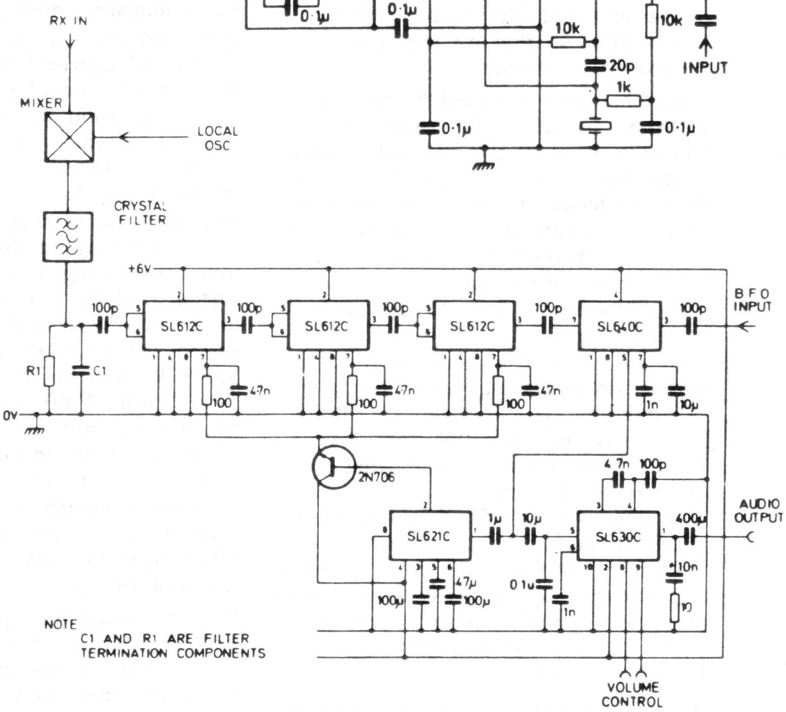
Using the SL1600/6600 series

As mentioned earlier, the devices are well supported with application information in the Plessey "Radio Communication Handbook".
As well as an individual analysis of each of the devices discussed, information is supplied on the designs of SSB HF transceiver systems, providing a wide variety of sophisticated functions. A new design for the SL1600 series is presently under preparation, and will be published shortly.

A few brief abstracts:

SSB detector — self-oscillating





SSB receiver circuit diagram

TDA1072 : Versatile AM radio system

General

Every now and again, it is pleasing to see that the Japanese do not have the total monopoly of consumer microelectronics. The Mullard TDA1072 is currently the state of the art in multi band AM radio systems - and it specifically designed with a level controlled oscillator for use with varicap diode tuning.

The oscillator is a single pin device, requiring very simplified bandswitching over multiple feedback types. The general device is full of the best features: Multiplier mixer, low distortion synchronous detector, 100dB range tuning indicator, buffered oscillator output - and a specified range to 30MHz.

Block circuit

The circuit alongside is based on a general MW application - two fuller notes cover

1) An electronically tuned and switched LW/MW set (16 pages of data)
2) A LW/MW/SW high performance tuner with MOSFET input (27 pages)

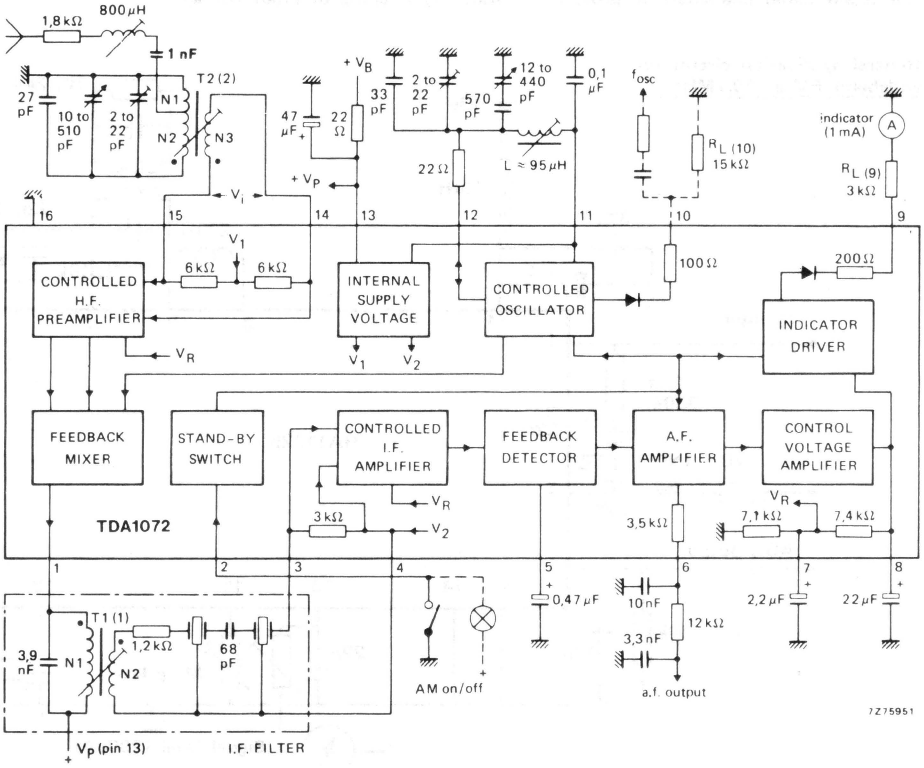
We have left the original Mullard coil specs in the diagram details, but have also established the following standard parts are suitable:

T1 IF filter:
CFM2 with input matching transformer. With DC blocking, the output transformer also useable for better spec.
Or almost any of the rest of our ceramic ladder series using the CFM2 matching arrangements.

T2 Antenna coil
MWC ferrite rod antenna, or RWR331208 coils etc.

L osc
Simple - simply chose from the ranges in the coil listings.

See catalogue part one for details of stock coils and filters - plus additional information in this catalogue on new filters etc.



(1) T1 : N1/N2 = 34/9; Q_o = 65; Q_L = 60; Z_{21} = 700 Ω at $R_{L(3)}$ = 3 kΩ; Z_{11} = 5,2 kΩ.
(2) T2 : N1/N2/N3 = 14/67/17; L = 175 μH; Q_o = 145; Q_L = 50 (f = 1 MHz); V_i/V_G = –6 dB.

Electrical specifications

	min	typ	max	unit
Supply voltage	7.5	15	23	volts
Supply current	15	22	30	mA
Bias voltages pins 14/15		2.75		volts
input Z Vin less than 300uV		6k/6pF		
Vin less tha 10mV		9k/2.5pF		
Output Z at pin 1	200k/4pF			
Conversion conductance		5.5		mA/V
Max PP voltage out at pin 1		2.8		v p-p
Output current at pin 1		1		mA
Control range of preamp		30		dB
Max RF input (pins 14/15)		2.8		v p-p
Oscillator				
Frequency range	0.6		31	MHz
Impedance at pin 12	1		200	k ohms
Controlled amplitude		140	200	mV
DC output voltage at pin 11		Vcc-1.3		volts
AM switchoff voltage at pin 2	3.5	2.5	Vcc	volts

	min	typ	max	unit
DC Osc output load	0		15	mA
DC Output resistance		7		ohms
Osc output voltage (pin 10)		200		mV
Osc output resistance		150		ohms
Permissable current output			2	mA peak
IF and AF stages				
DC input voltages at pins 3/4		2		volts
Input Z pin 3	2.4	3	3.9	k ohms
Control range for 6dB		62		dB
AF out /2mV in at pin3		350		mV
AF output impedance (pin 6)		3k5		ohms
Field strength indicator				
No input, 2k7 load on 9		0	140	mV
500mV input	2.5	2.8	3.1	V
Output resistance		250		ohms
With AM off (pin 2 high)		6		volts

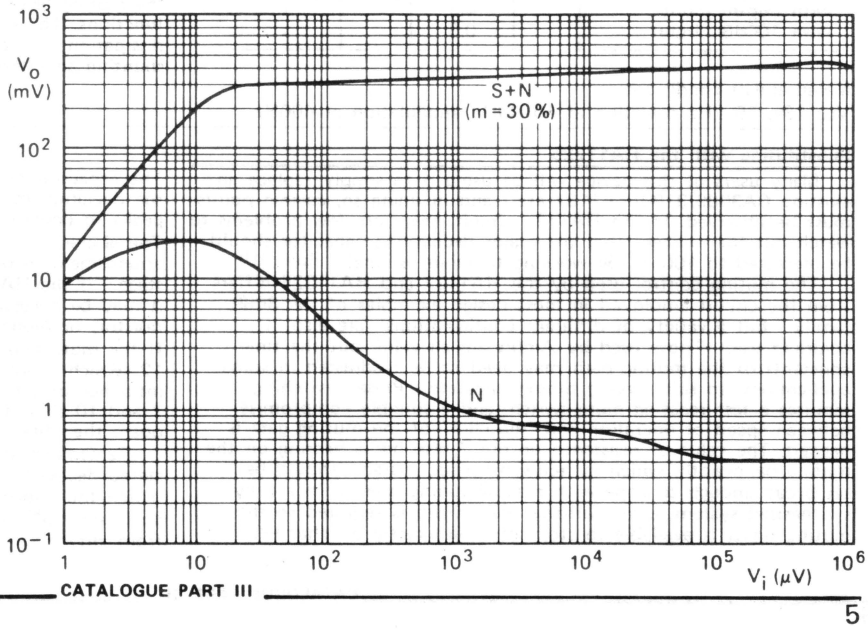
Performance data

Vcc 15v, f in 1MHz
f mod 400Hz at 30% AM
unless indicated otherwise

Sensitivity and S/N - see graph alongside

Input for AGC threshold	14uV
AGC range for 6dB output change	91dB
Maximum input voltages for	
80% mod 3% THD	650mV
30% mod 3% THD	900mV
30% mod 10% THD	1300mV
THD for	
2mV input, 80% mod	0.5%
500mV input, 80% mod	1.8% typ

See elsewhere in this issue for details of our new TDA1072 3 band DC controlled tuner module.

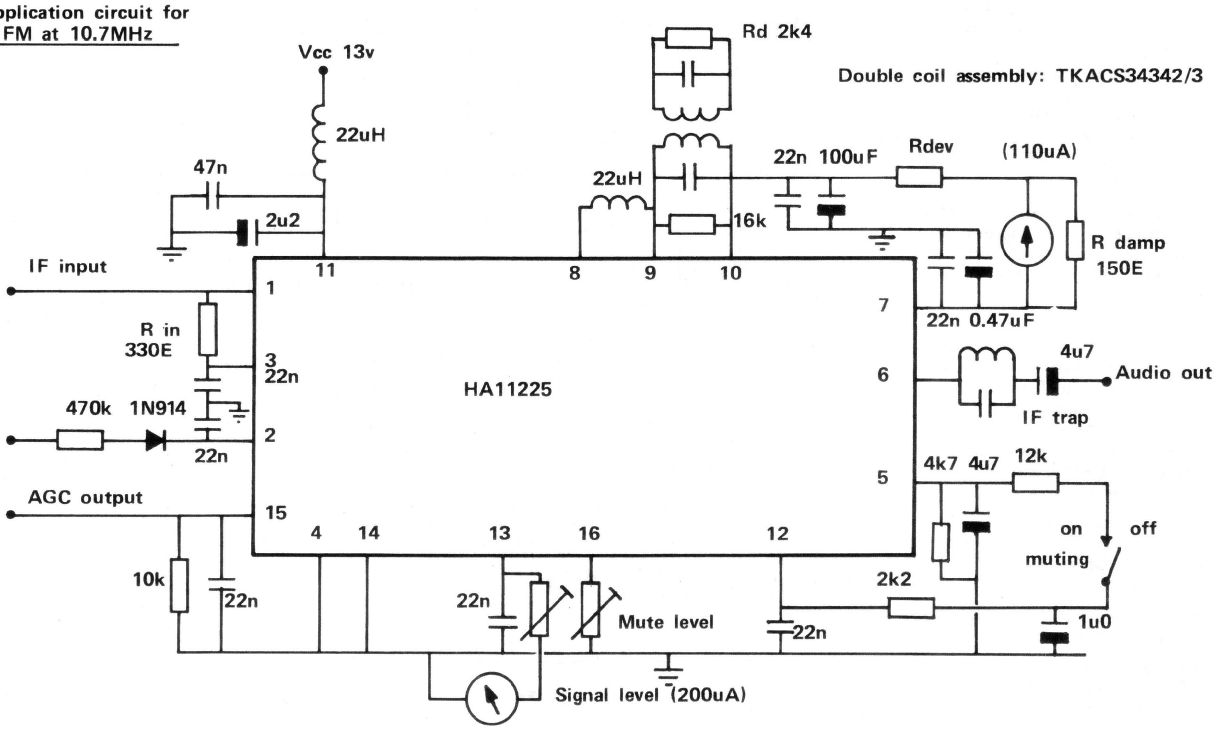


ambit INTERNATIONAL

HA11225 : the latest in the CA3089E saga

* **Pin compatible with CA3089/HA1137 families (also CA3189E with the omission of two components)**
* **Best S/N yet achieved (84dB typ., 78dB guaranteed)**
* **High audio output**
* **Lower distortion (typ. 0.03%), and low distortion maintained over a wide range of detuning and input levels**
* **Consistant meter characteristic provides accurate log tracking of input voltage**

General application circuit for wideband FM at 10.7MHz



Electrical specifications

Parameter	min	typ	max	notes
Supply voltage	12	13	14	volts
Supply current at 13v	16	25	33	mA
Input limiting voltage		31	37	dBu
AF output	265	380	510	mV rms
THD		0.03	0.1	%
Signal to noise ratio	78	84		dB
AM rejection	45	54		dB
Muting attenutaion	70	85		dB
Muting bandwidth pin 12 at 1.4v	55	105	145	kHz
Muting sensitivity pin 16 open circuit	36	44	60	dBu
Maximum muting level watch power dissipation does not exceed 590mW	75			dBu
Signal meter voltage				
with 0dBu input		0		v
with 70dBu input	0.9	1.45		v
with 100dBu input	4.7	5.2		v
AGC voltage				
with 86dBu input		4.3		v
with 100dBu input		0.3		v

Measurement conditions:
Vcc 13v, 75 kHz deviation at 10.7MHz, modulation at 400Hz

General information

The HA11225 is the current state of the art in FM IF amps. Fortunately, it is pin compatible with most of the previous members of the CA3089 families - although it now uses pin 16 for adjustment of the threshold of the muting level onset.

Adjustment at pin 16 does not affect the deviation muting operation, which is still a function of the resistance of the path between pins 7 and 10. (See the HA1137 in part 2 cat)

One of the major features enhancing the performance of the HA11225 is the use of 100uF at pin 10 to decouple the internal zener stabilizer. In the earlier 3089 types, this zener was buried in an inaccessible part of the circuit - and the very indirect path to the decoupling led to incurable noise problems. The CA3189E and the HA11225 take this zener directly to pin 10, where zener noise is virtually illiminated. The use of Hitachi's refined low noise techniques in the design of the detector and audio amplification stages further reduced noise until the level attained is now rather better than many other links in the chain from the studio to the listener.

Since stereo S/N ratios are inherently worse than those of mono transmissions, the availability of a muting level threshold adjustment (at pin 16) enables this IC to be used with the muting level switched in accordance with the mono/stereo functions. The level of adjustment is limited by the power dissipation of the IC, since the resistance at pin 16 is basically shunting an internal current. Take care not to exceed the ratings.

Substituting with the HA11225

Many users of this device will be using it to simply replace an existing CA3089/3189/HA1137 etc - and in doing so, certain points must be observed. In the HA1137 circuit, there are few problems to watch out for - except that the major decoupling capacitor should be increased to 100uF (6v working tantalum is fine). (pin 10)

The muting control used by the HA1137 and HA11225 differs from the circuit employed in most design examples of the 3089 family - but a variety of different configurations are also in common use. Those used by Ambit are all valid, but the time constants of the muting path may need to be readjusted in some applications, to suit varying tuning rates etc. Preset switching requires a fast reacting mute switch, and a suitable compromise between speed of operation and 'the click' of the muting gate is needed. The value of resistance on pin 7 also has a bearing on the muting bandwidth, which in non-synthesised systems should be set broad enough to cope with manual tuning speeds. In a fully synthesised system, a far narrower bandwidth is required so the muting signal is available to stop scanning exactly on channel.

The capacitor on pin 7 will also affect the speed of operation of the deviation mute - too little capacitance will cause modulation peaks to operate the deviation muting gate.

The above circuit uses a double tuned detector, and since this arrangement provides larger audio output than a single tuned stage - if the HA11225 is to be used with a single tuned detector, it may be necessary to lower the value of the primary damping resistor, or reduce the choke between 8/9 to about 15 or 18uH.

If the mute fails to open at all, then check the voltage at pin 12 (which should go low as the station is tuned). To disable the influence of the deviation muting, the reistance between pins 7 and 10 should be reduced to about 3k9. If the path is left open, then the mute will fail to operate under all input levels.

Remember, that once the IF has been substituted, then the stereo decoder and audio preamp stages may not be able to match the improved performance. Early 1310 decoders, and particulary SN76115s should be replaced with more recent types from Hitachi or TOKO. (HA12003/KB4400/KB4409 etc.)

TDA1005A : FDM stereo decoder

General information

It wouldn't be an Ambit catalogue if we didn't have another example of the the State of the Art in stereo decoders to offer. Not wishing to blemish this record, we give you the new TDA1005A from Mullard - an improved version of the original TDA1005. (It needed to be !)

First and foremost, the system operates in the FDM mode. Frequency division multiplex is generally regarded as a better means of decoding stereo signals to the almost universal TDM (time division multiplex) system. The basic difference is that in the TDM system, the decoder operates by switching in time, and FDM systems switches in frequency. In the FDM configuration, the 38kHz information is taken from the main channel (L+R) signal in a tuned circuit, and fed to the matrix decoder section separately.

ie TDM implies multiplexing by dividing up time slots and FDM multiplexes by dividing up the frequency spectrum

What it all boils down to in practical terms, is that in the FDM mode, the TDA1005A produces very few IM products even with high amplitude, high frequency audio inputs. The generally softer multiplexing of FDM operation leads to a reduction in the various ultrasonic spurii associated with the decoding process switching products. The IM products are typically −60dB (0.1%).

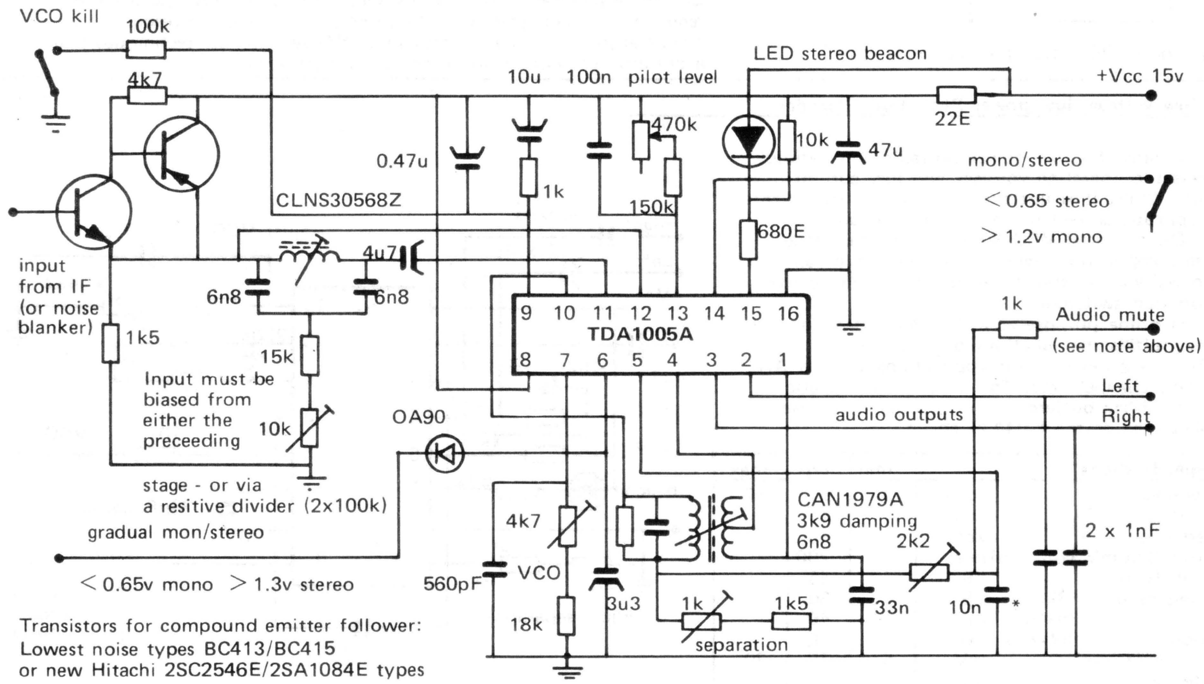
Much work has been done on applications of this device by Mullard/Philips, and this is available in the form of a report of some 23 pages. It contains much information on the use of filters, soft mono/stereo switching and audio muting, together with comparative specifications for the TDM and FDM modes of operation.

Additional features of the TDA1005A are a smooth mono/stereo takeover function, achieved by shifting pin 6 towards ground - in this way gradually reducing the decoder efficiency until zero separation.

Audio muting may be provided by injecting current into pins 1/4. In TDM mode, 10v max via 10k will produce 30dB of output muting. In FDM mode, the application of 5.2 v directly (as in the FDM mode, pin 1 is decoupled) will produce 64dB of muting effect at the output.

Because the pilot tone and composite channels may be separated, it is possible to notch the 19kHz at the input. The 19kHz is routed to pin 12 (omitting decoupling), before the notch filter takes effect. The availability of the main channel signal independent of pilot tone considerations is also available on many other types of PLL decoder - and once the separation has occured, the composite minus 19kHz may be used in a sliding de-emphasis system, where a capacitor is used to narrow the bandwidth via an FET controlled 'VDR', driven from the signal output level. In this way, the noise on weak stereo transmissions may be suppressed (with loss of separation).

Specifications	notes	min	typ	max	units
Supply voltage		8	15	18	volts
Supply current	ex lamp		22		mA
Supply for lamp	not exc. P tot		100		mA
Switching voltages	wrt pin 16				
to mono	pin 14	1.2		Vcc	volts
to stereo	pin 14			0.65	volts
hysteresis	pin 14		0.2		volts
Separation at 1kHz	TDM	40	50		dB
	FDM	50	55		dB
FM IF rolloff correction	TDM	48		70	kHz
Input (comp) for .35% THD			1		volt pp
Input impedance		35	50		k ohms
Voltage gain per channel	TDM		7		dB
	FDM		10		dB
Balance		1	0	1	dB
Output voltage	TDM	0.61			Volts rms
	FDM	0.97			Volts rms
Total harmonic dist.					
at 1kHz	TDM		0.2	.35	%
	FDM		0.1	.35	%
at 300Hz - 10kHz	TDM		0.2		%
	FDM		0.1		%
Ultrasonic rejection					
at 19kHz	TDM		36		dB
	FDM		46		dB
at 38kHz	TDM		40		dB
	FDM		38		dB
IM distortion at 12kHz mod					
	TDM		40		dB
	FDM		60		dB
Adjacent channel beat rejection					
at 114kHz	TDM		52		dB
at 190kHz	TDM		55		dB
at 114kHz	FDM		70		dB
at 190kHz	FDM		74		dB
VCO frequency			76		dB
VCO temp coeff			450		ppm/°C



*de-emphasis in the FDM mode is applied before the decoder action.

The TDA1005A in FDM mode, with 19kHz pre-filter stage

KB4436 : IMPULSE NOISE CANCELLING IC

General

The KB4436 is very similar in most respects to the KB4423 (see p36 of Catalogue part 1). It is intended to interpose between an FM detector stage, and a stereo decoder - and by sensing noise spikes in the audio signal, a blanking period is inserted to prevent a loud click occuring in the output of the tuner.

To do this, the audio is split into two bands - one below 55kHz, and one above 55kHz - where the noise spike is sensed and detected to interrupt the base band path for a period determined by the RC constant on pin 11. In FM stereo, this is nominally set to provide 25uSec blanking pulse.

The C/R at pin 13 determines the sensitivity of the threshold of noise detection - and by using a transistor as a variable resistor-cum-switch driven from the meter putput, it is possible to adjust the circuit such that noise cancelling operation only occurs during a signal that is fully limited. Signals including inherent noise (as a result of low signal level) will cause the noise blanker to be triggered continuously which adds to the overall noise - rather than reducing it.

Signals on the margin of noiseyness - ie signals with noise that causes breakup on modulation peaks - may be 'cleaned up' somewhat by the KB4436 (and KB4423).

The basic difference with the KB4436, is that pins 7 and 8 are used in a switching circuit, driven from the stereo decoder beacon output, such that the pilot tone is switched off when pin 8 goes low (ie when the beacon is lit during mpx reception.

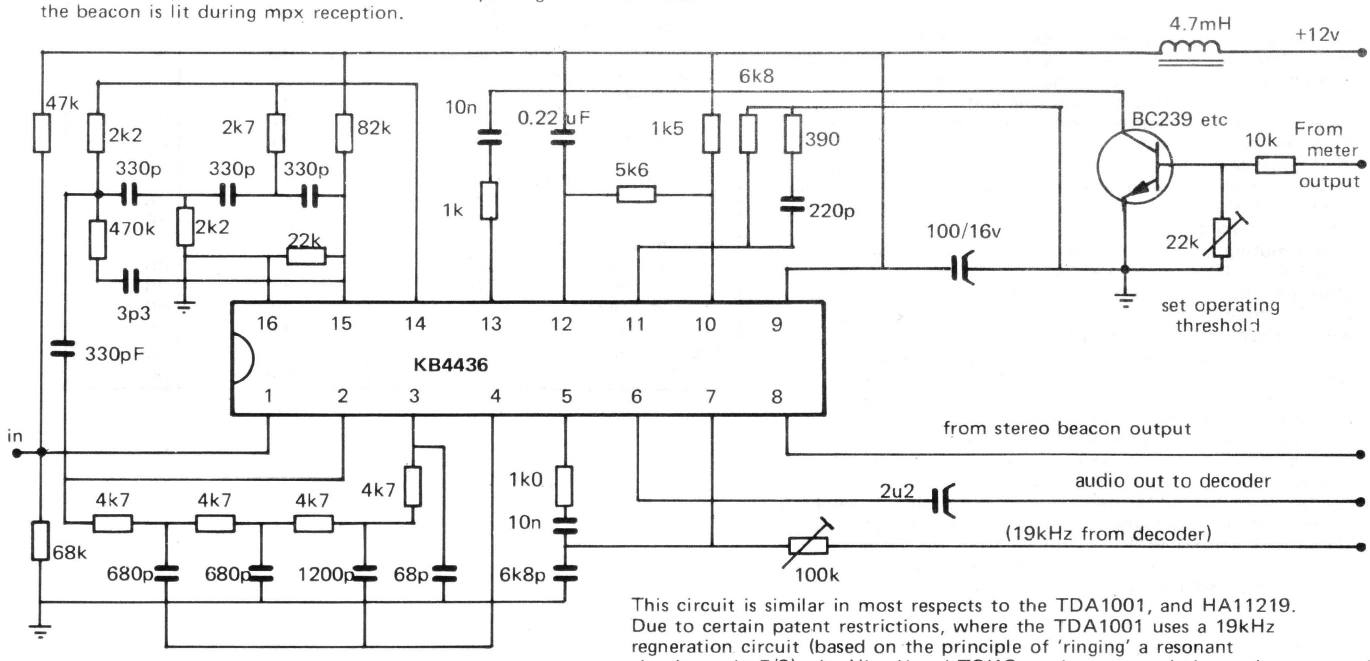
(KB4424 low power mpx decoder IC)

The pilot tone is introduced into the output of the noise blanker, via the buffered output that appears on the 1310 families - in this way total continuity of the pilot tone is maintained during period of the signal blanking. However, like the KB4423, it doesn't really seem to make any difference whether or not this feature is used, since the blanking period is so short that the PLL doesn't have time to lose lock for any audible disturbance.

It is possible these are various devices simply to circumnavigate a patent on noise blanking in FM stereo radio

Electrical specifications		min	typ	max
Operating voltage	V	8	12	16
Current consumption	mA			28
Voltage gain	dB	-1	0	+1
Max output	vrms			1.5
Input impedance	kohm		30	
Distortion	%			0.1
Gate time	uSec		25	
Noise input sensitivity (tr. ON)	mV	40		
19kHz switched on - pin 8.........	v			0.2
off	v	11.5		

Full spec 11 pages of A4



This circuit is similar in most respects to the TDA1001, and HA11219. Due to certain patent restrictions, where the TDA1001 uses a 19kHz regneration circuit (based on the principle of 'ringing' a resonant circuit at pin 7/8), the Hitachi and TOKO versions use techniques that avoid infringement, although achieve similar ends.

NB input may require DC blocking capacitor (2-10uF)

KB4424 : low voltage, low power PLL stereo decoder

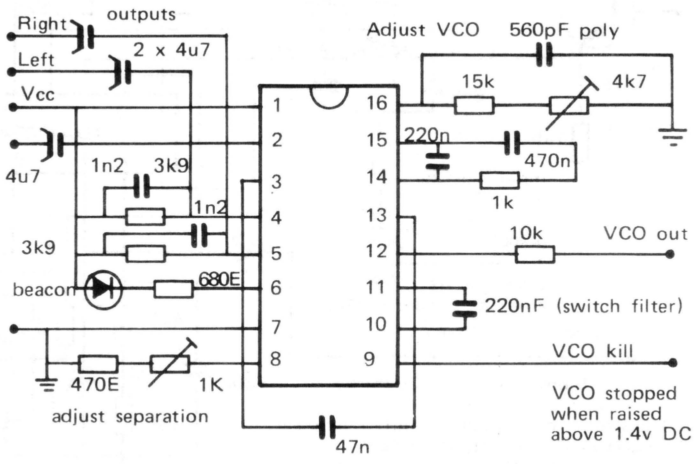
General

This IC is intended for low power portable applications, requiring versatile operating voltages and low current consumption. Whilst it is not the last word in HiFi, the intended applications will be more than adequately fulfilled. A DC operated VCO killer is available, since most portable applications will require the use of an AM section fed via the stereo decoder to minimize the need for function switching.

As well as its prime purpose as a PLL MPX decoder, the KB4424 also fulfills the functions of a low power low cost PLL tone detector for applications such as ARI, Carfax, "Silent channel" etc. The VCO output is also available as a 19kHz buffered output, suitable for use in the applications of the KB4436 above.

Electrical specifications			min	typ	max
Operating voltage		volts	4	6	12
Operating current at 6v		mA		15	
Input impedance		kohm		40	
Audio output (200mV comp in)		mV		200	
Separation at 1kHz		dB	40	45	
Ultrasonic rejection	19kHz	dB		30	
	38kHz	dB		40	
SCA rejection	(57kHz)	dB		80	
Stereo 'on' 19kHz level		mV	5	12	20
Stereo hysteresis		dB		3	
VCO kill voltage at pin 9		v		1.4	

Standard test conditions:
Vcc 6v, 25°C, L+R 200mVrms, F=1kHz, 10% pilot



General application circuit of the KB4424

NB Power supply line decoupling is not shown illustrated here
Pilot tone filtering may be added (see Ambit 91310 for details)

TDA1028 : AC signal switching with DC

General

The TDA1028 is a quad op-amp, connected as an impedance converter. Each amplifier has two switchable inputs, with clamp protection diodes. The input currents are independent of the switch position - and the outputs are short circuit protected. The device provides electronic selection of four channel signal sources in audio amplifiers, where its prime application is as a mono/stereo switch, and monitor input selector.
It is designed for use with the TDA1029 input selector, and the TDA1074 electronic potentiometer system for volume and tone control functions.

Electrical specifications	min	typ	max	
Supply voltage	6	20	23	V
Supply current (unloaded)		2.5		mA
Input voltage			5	Vrms
Input current			20	mA
Switch control current			50	mA
Total dissipation			800	mW
Operating temp	-30	20	+80	°C
Input offset		2	10	mV
		20	200	nA
Input bias current		250		nA
Cap between adj inputs		0.5		pF
DC input			Vcc	v
Supply rejection		100		uV/V
Equiv input noise voltage (Rin less than 1k, 20-20kHz)		3.5		uV rms
Equiv input noise current		0.05		nA
Crosstalk between switched on/off inputs 1k ohm/1kHz		100		dB
Voltage gain when on		1		
Current gain		10^5		
Output resistance		400		ohms
Output current		5		mA
Freq response: Vin 1v pp, R in <1k. R_L 10M, C_L 10pF		1.3		MHz
Slew rate		2		v/uSec
Control voltages:				
High	3.3			V
Low			2.1	V
Input current High			1	uA
Low		100	250	uA

Switch control

On inputs	Channel controlled	control voltage V1-16	V8-16
I-1, II-1	2 to 4, 15 to 13	H	-
1-2, II-2	3 to 4, 14 to 13	L	
III-1, IV-1	7 to 5, 10 to 12	-	H
III-2, IV -2	6 to 5, 11 to 12	-	L

AC performance characteristics

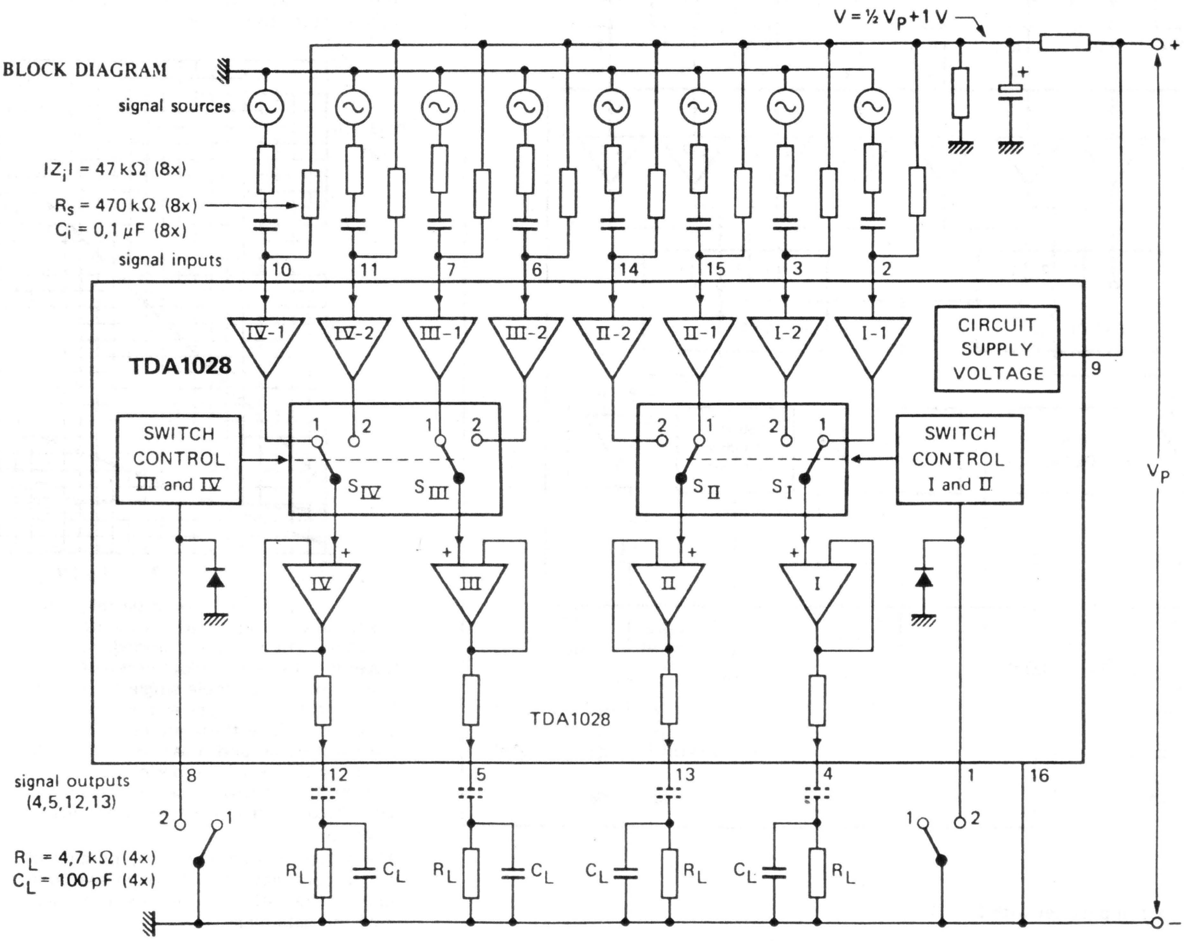
Measured in the circuit below, under the following conditions:
V supply 20v, T_{amb} 25°C, Ci=0.1uF, Rs =470k, R_L = 4k7, C_L = 100pF (unless otherwise specified)

		min	typ	max
Voltage gain	dB		-1.5	
DC output voltage variation when inputs are being switched	mV		10	100
Distortion at Vin 5v, f 1kHz	%		0.02	0.1
at Vin 5v, f from 20Hz to 20kHz	%		0.04	
Noise output voltage (unweighted) from 20Hz to 20kHz	uV		5	
(Weighted) According to DIN45405	uV		12	
Amplitude response Vin 5v, f 20Hz to 20kHz	dB		0.1	
Crosstalk between switched on and switched off inputs (1kHz)	dB		75	
Crosstalk between on input, and output of other channels at 1kHz	dB		90	

Notes

Like the TDA1029 companion device, the TDA1028 is a highly accurate analogue switch, and may be used to transfer DC as well as AC information. The AC performance available is consistent with the requirements of HiFi amplifiers and preamplifiers.
The cross talk is closely associated with the source impedance, and this fact should be noted when designing in applications where the crosstalk factor is a prime concern.
For logic directed audio systems, the TDA1028/1029 are very significantly superior to CMOS analogue switch arrangements.
Full data on the TDA1028/1029 is available in a 30 page booklet.

BLOCK DIAGRAM



TDA1029 : 4 input stereo switching with DC

General

The TDA1029 is the companion to the TDA1028, and in many respects it is electrically similar.
The TDA1029 is a dual operational amplifier, connector as an impedance converter, each amplifier having 4 mutually switched inputs. Its prime function is in signal source switching in stereo audio amplifiers.

Electrical specifications

See those for the TDA1028, and

	min	typ	max
Bias voltage			
DC output at pin 10		10.2-11.8	V
Output impedance		8k2	ohms

Switch control

On inputs	Channel controlled	control voltages pin 11	12	13
I-1, II-1	1 to 15, 5 to 9	H	H	H
I-2, II-2	2 to 15, 6 to 9	H	H	L
I-3, II-3	3 to 15, 7 to 9	H	L	H
I-4. II-4	4 to 15, 8 to 9	L	H	H
I-4, II-4	4 to 15, 8 to 9	L	L	H
I-4, II-4	4 to 15, 8 to 9	L	H	L
I-4, II-4	4 to 15, 8 to 9	L	L	L
I-3, II-3	3 to 15, 7 to 9	H	L	L

An internal blocking circuit ensures that not more than one input will be switched at any one time.

AC performance characteristics	typical	worst case	
Voltage gain	-1.5		dB
Output voltage change during input switching	10	100	mV

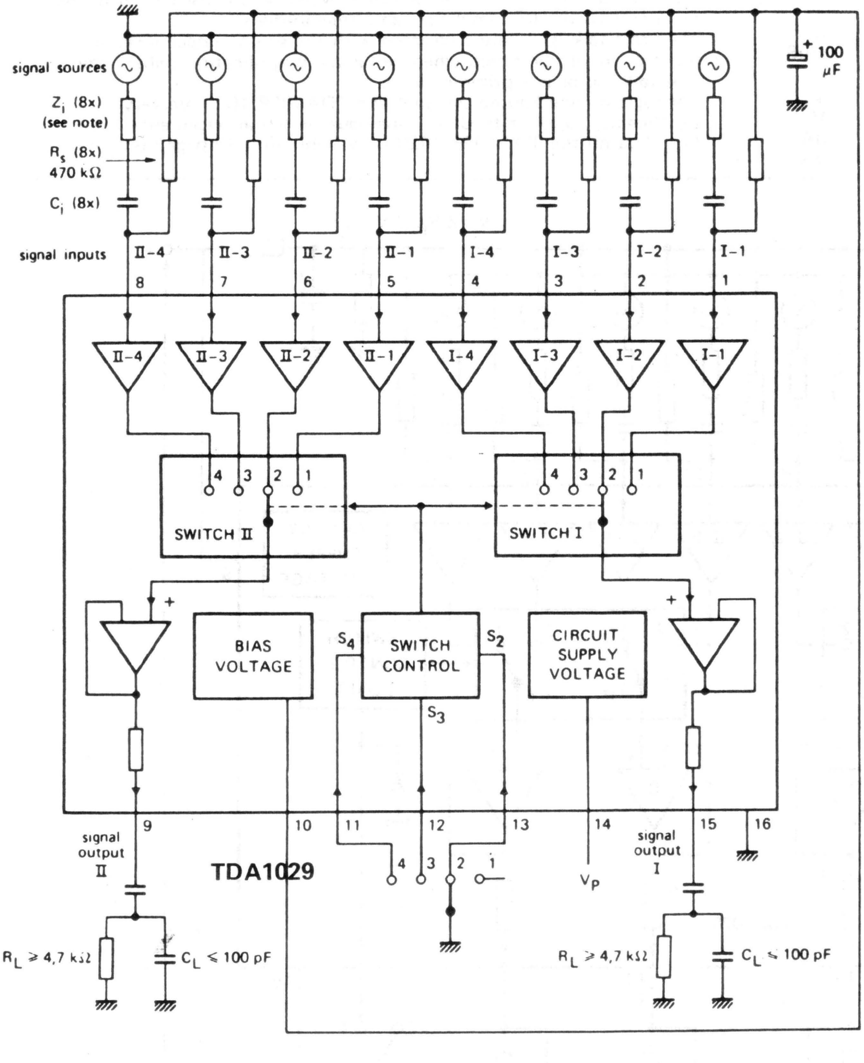
All other parameters as per TDA1028

Applications information

DC switching - and particularly switching where the rail voltages are the basic logic references - is especially suited to all forms of digital signal processing. In the particular case of the TDA1029, the ability to be able to switch low level inputs - such as the phono/tape inputs etc - with the use of a remote DC connection - frees the designer from the considerations of stray pickup of hum/noise, since the switching is performed within a few cm of the actual socket. All input sockets in HiFi equipment tend to run from the back connection panel, to the front switching assembly - and then back into the signal processing circuitry - which is a considerable distance if stray pickup is to be eliminated. The only alternative to PCB mounted connections, is to used screened cable, which is tedious to prepare, and most uneconomic in production and test.
Apart from interfacing into MPU operated systems, DC control is a necessary feature of most IR/US remote controll systems. It is also of great benefit in the design of all types of test equipment - such as signal generators - where it provides an easy interface to computer bus terminations in applications such as stepped attenuators etc.
Commutating filters are readily made from these devices - and the speed of operation also permits the ICs to be used in signal and noise blanking applications at IF levels up to about 1MHz.

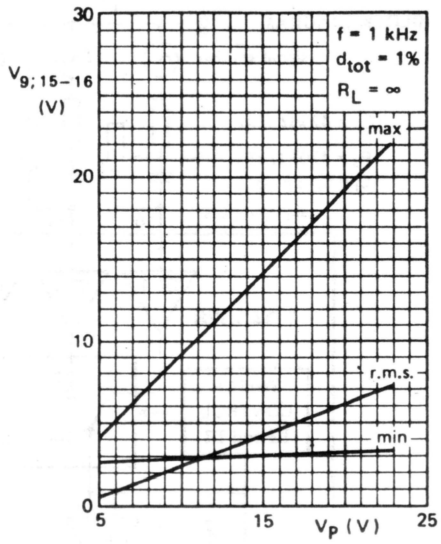
Since the device is basically configured as a unity gain op amp, it is possible to build active filter circuitry around these op amps, and provide the system with an integral in/out switching facility.

A very extensive data booklet of 30 pages is available covering these and various other aspects of DC switching with the TDA1028/1029.



Note: $|Z_i|$ = 47 kΩ in parallel with 200 pF.

The dynamic range of the TDA1028/1029 is primarily a function of the supply voltage (V_P) available. The following graph shows the detail at the 1% THD point, though for more exacting performance, the rms output level should be about 20% below the graph value for operation in the 0.04% distortion regions.



care should be taken to ensure unused inputs are not simply left to float. The lower the impedance, the lower will be the residual crosstalk - but beware of using simple single transistor emitter follower stages as impedance level converters, since these are going to introduce distortion and noise that will degrade the possibilities of the basic ICs.
The emitter follower itself possesses the same high input impedance problems anyway.
The switching operation may be carried out in conjunction with touch switching- such as the SAS6610 described in part 2 of the catalogue

TDA1074 : Dual DC controlled electronic potentiometer

General

Combined with the TDA1028 and TDA1029, the TDA1074 provides the means for total DC control of all HiFi specification preamplifier control functions.
The TDA1074 may be used as either a stereo volume control (with integral balance feature), or as a bass and treble control providing full tone control facilities with the applications of a simple control potential.
The stereo matching of the device is excellent - far better than most conventional mechanical potentiometers. The operation of the circuit is virtually noiseless, since the wiper noise of the control pot is very effectively decoupled.
Earlier attempts at producing DC volume and tone control have proven to be rather less than HiFi, mainly through excessive noise. The TDA1074 has avoided all these problems, and is now available with a considerable amount of back up applications data. (30 pages)

Electrical specifications

Parameter	units	min	typ	max
Supply voltage	v	7.5	20	23
Supply current	mA		20	
Input signal	v rms		1	6
Output signal	v rms			6
Total distortion	%		0.05	
Output noise	uv		50	
Adjustment range	dB		110	
Channel sep	dB		80	
Channel balance	dB		0.5	
Control voltages	V_{9-8}			1
	V_{10-8}			1
	$-V_{9-8}$			1
	$-V_{10-8}$			1
Op temp range	°C	−30	20	+80

Performance in tone control configuration

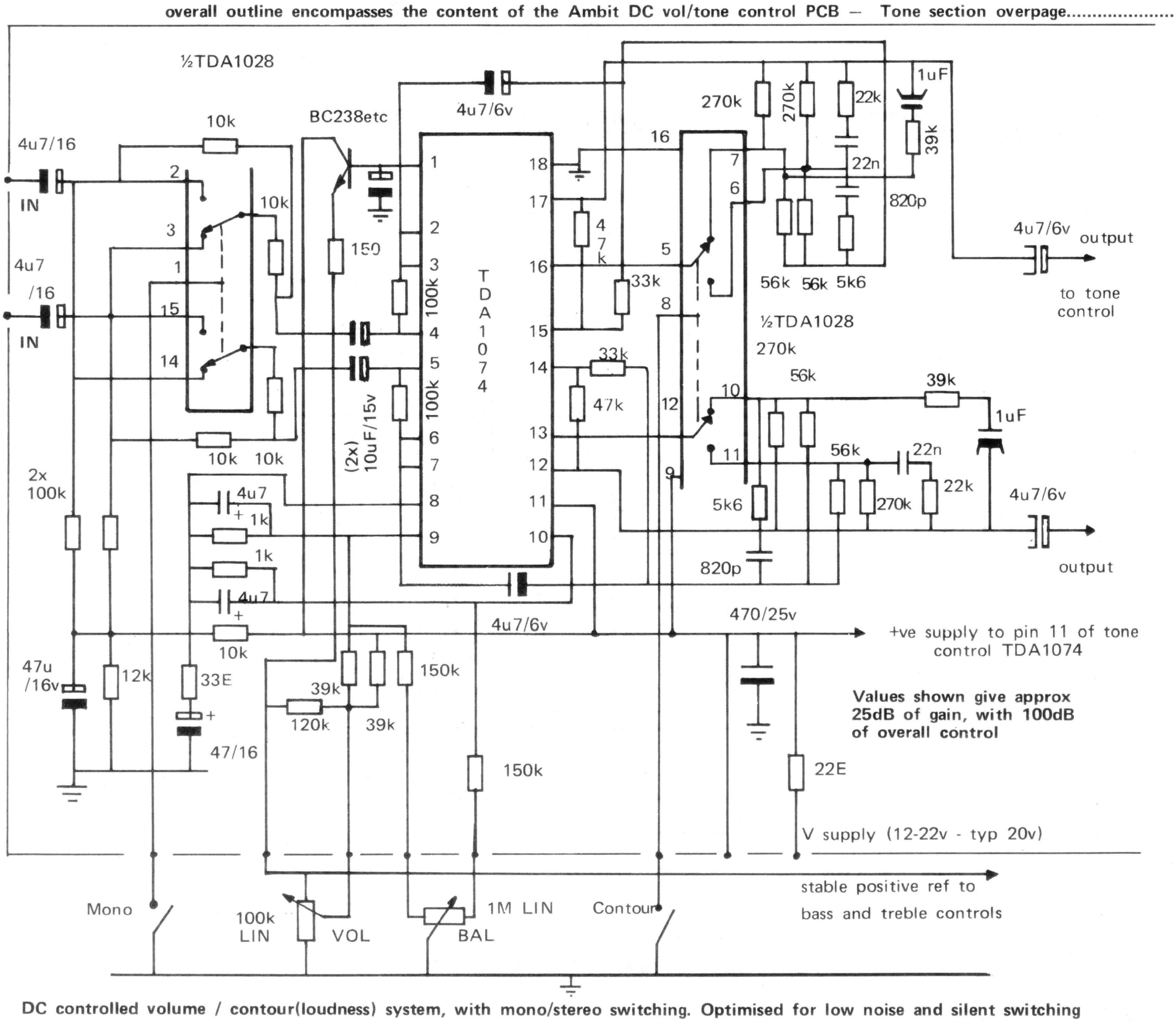
Conditons
V supply 20v, 60 ohm generator source impedance,
Load greater than 4k7, less than 30pF.
Measurements at 1kHz, unless otherwise indicated

Supply current	typ 20mA
Frequency response (−1dB) (V_c=0)	10Hz to 20kHz
Voltage gain at linear f response (V_c=0)	0dB
Maximum gain variation at Max boost/cut	±1.5dB
Bass boost at 40Hz (Vc +120mV)	17dB
Basscut (Vc -120mV)	−17dB
Treble boost at 16kHz (Vc +120mV)	16dB
Treble cut (Vc -120mV)	−16dB
Total distortion at Vin 5V rms, Vc=0	
(at linear frequency response) f = 1kHz	0.03%
At f= 40Hz to 16kHz	0.07%
Channel separation at Vin 5v rms, Vc=0	80dB
Output noise voltage Vc=0, f=20Hz to 20kHz	75uV unweighted
According to DIN45405 - peak value	170uV
Signal level for Total dist. of 1%, Vc=0	6v
Hum suppression for f = 100Hz, with 200mV or less ripple on supply voltage	46dB

General dynamic specs

Open loop voltage gain	66dB
Slew rate	3V/uSec
Equiv input noise (Vc=0, f=20/20000Hz)	10uV / 150pA
Control input resistance	1 Mohms
Signal input resistance	10Mohms
Signal output resistance (internally limited current)	0.5ohm (15mA)

overall outline encompasses the content of the Ambit DC vol/tone control PCB — Tone section overpage....................



DC controlled volume / contour(loudness) system, with mono/stereo switching. Optimised for low noise and silent switching

TDA1074: basic circuit applications

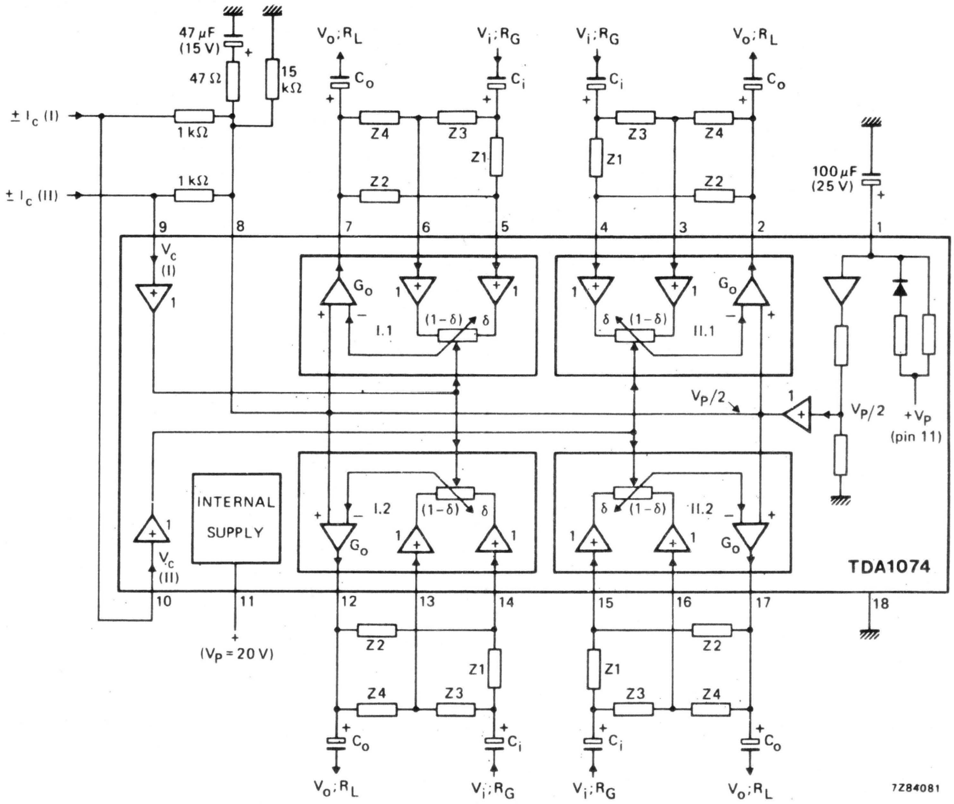


Fig. 1 Block diagram and external components; $I_{c(I)}$, $I_{c(II)}$, $V_{c(I)} = V_{9\text{-}8}$, $V_{c(II)} = V_{10\text{-}8}$ are control input currents and voltages; Z1 = Z2 = Z3 = Z4 = 22 kΩ; R_G = 60 Ω; R_L = 4,7 kΩ; C_i = 2,2 μF; C_o = 10 μF.

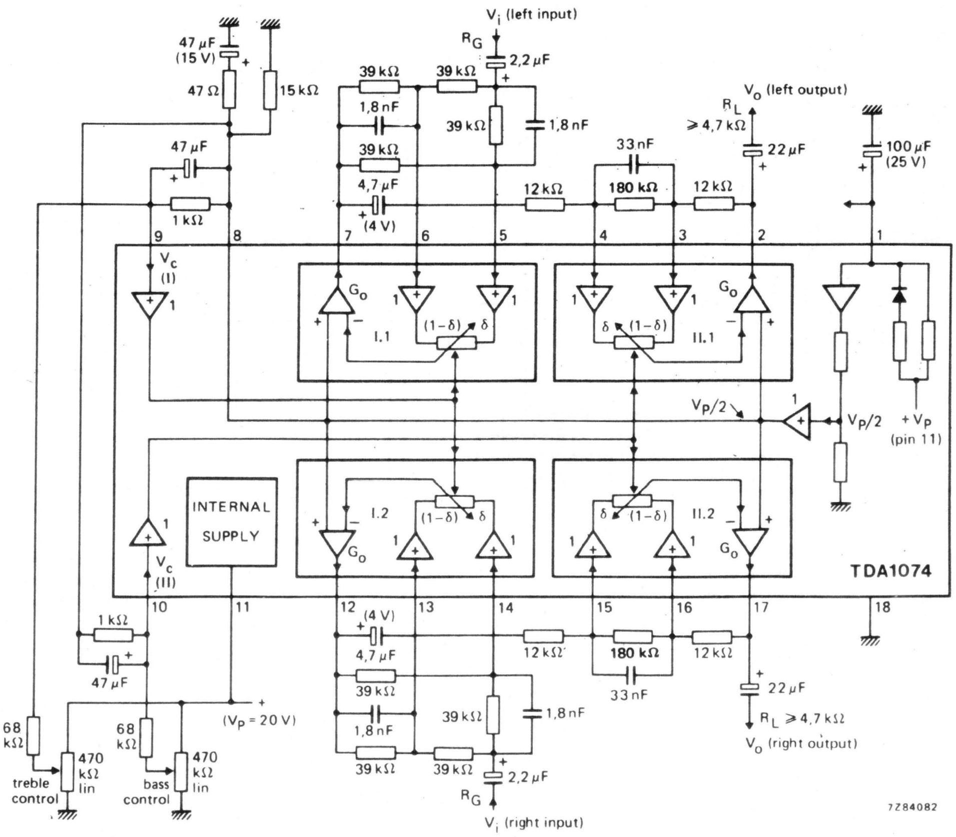


Fig. 2 Application diagram for treble and bass control.

KV series

15:1 capacity ratio tuning diodes

General

AM tuning varicaps - or any varicap with a reliable and repeatable 17:1 tuning range - have only been available at very high cost until the past couple of years.

Historically, Motorola were first with MVAM1 and the MVAM2 series. Both these required 25v tuning bias for correct operation, and even the follow up series, the MVAM115 needed 14v to allow sufficient margin.

It was argued at that time that to reduce the bias range to the more convenient 1-9v would result in poor diode characteristics, and unrepeatable matching. Any drift in tuning voltage would be very severe, since AM radios do not possess AFC (as a rule) - and the much narrower bandwidth would tend to underline shortcomings with frequency stability.

Perhaps the single factor that limited the AM tuning diode above all others was the accurate matching of sets for multiple tracking. The MVAM115/125 series were a step in the right direction by providing a set of individually matched diodes in separate packages - but the costs of matching were considerable.

Multiple diodes in a single package were beginning to lose favour, since the interaction of the various tuned circuits using the single package are not inconsiderable with the increased demands for signal handling. PCB design is also restricted by the need to feed all tuned circuits from a central point.

For a double diode, this isn't too much of a problem. But in some countries, regulations relating to the reradiation of local oscillator are sufficiently severe as to rule out the multiple diode approach altogether.

Against this background, TOKO have been working on a new range of diodes designed to answer these basic objections - and provide the necessary solutions for the rapidly increasing trend towards synthesised radio in consumer radio. The result is the KV1210/1220 series.

Major Features

KV series diodes are available in combined, or discrete packages. However, the discrete packages are supplied in a unique style of 'snap-apart' packaging, that ensures accurate matching of diode characteristics whilst retaining the flexibility of individual devices.

The 9v series of diodes is suitable for all types of applications in consumer equipment, ranging from car radio to synthesised tuner systems. In some communications enviroments, the use of 25v bias tuning diodes may be advantageous, as the effects of strong signal overloads decrease with the increasing bias voltage used.

Both the 9v and 25v series exhibit excellent 'Q' characteristics - which get no worse than a low of 200 (typ 500) at 1v bias. As the bias is increased, the Q rapidly climbs to 500-1000.

The KV series are suitable for use in shortwave applications to 30MHz - a typical measured Q of 80 in conjunction with the 154AN7A6441EK short wave oscillator coil. The measured Q of this coil with an air gang capacitor on a Q meter showed about 82 - indicating the varicap has negligable effect at the frequency of 12MHz.

The KV series are useable in oscillator circuits (especially as back-to-back configurations) up to 120MHz, making them an excellent low cost alternative to expensive hyperabrupts frequently employed in communications equipment.

Main applications

Apart from the obvious radio applications already covered, the varicaps may be used in a variety of useful ways:

Remotely tuned antenna receiving systems:
 Both signal and tuning bias may be fed down a single coax connection

Tracking audio filters

Variable coupling elements in adjustable bandwidth filter systems

Automatically tuned preselectors in receivers having a high first IF (so that there is no danger of complication from image responses)

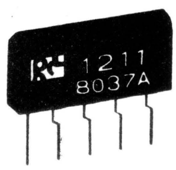
Audio frequency synthesisers using a mixing technique

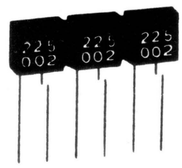
Metal locator systems - both BFO, IB and Phase Angle systems.

Electrical data and specifications

SELECTION GUIDE

STRUCTURE	9v series	25v series
Triplet array	KV1210	KV1220
Double diode	KV1211	KV1221
Triple discrete	KV1215	KV1225
Double discrete	KV1216	KV1226
MAXIMUM RATINGS		
Reverse voltage	Vr 20v	Vr 30v
Forward current	If 50mA	If 50mA
Power dissipation	Pd 100mW	Pd 100mW
Operating temp.	-55 to +80°C	-55 to +80°C
Storage temp	-55 to +125°C	-55 to +125°C





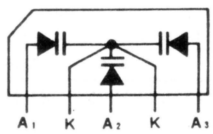
DYNAMIC RATINGS : 9v series

Item	Symbol	min	typ	max	units	notes
Reverse V	Vr			20	V	Ir 10uA
Leakage	Ir			100	nA	Vr 15v
Capacitance	C_{1v}	440	500	560	pF	Vr 1v
	C_{9v}			35	pF	Vr 9v
Cap ratio	$C_{1/9}$	15	17			
Q	Q	200	500			Vr 1v
Temp coef	TC		500		ppm°	Vr 5v
All tests carried out at 1MHz						

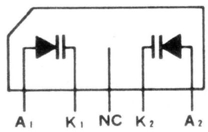
25v series

Item	Symbol	min	typ	max	units	notes
Reverse V	Vr			30	V	Ir 10uA
Leakage	Ir			100	nA	Vr 25v
Capacitance	C_{1v}	510		620	pF	Vr 1v
	C_{25v}	16	21	26	pF	Vr 25v
Cap ratio	$C_{1/25}$	20	24			
Q	Q	200	500			Vr 1v
Temp coef	TC		130		ppm°	Vr 13v
All tests carried out at 1MHz						

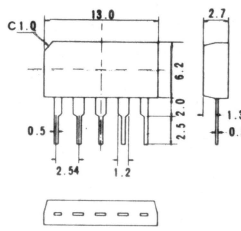
Package styles and dimensions (mm)



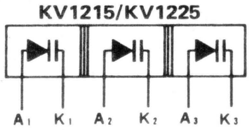
KV1210/KV1220



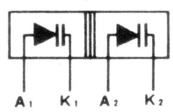
KV1211/KV1221

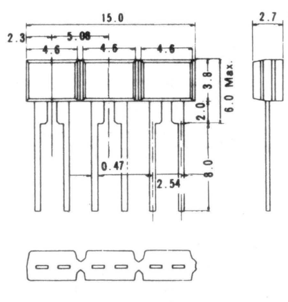


KV1215/KV1225

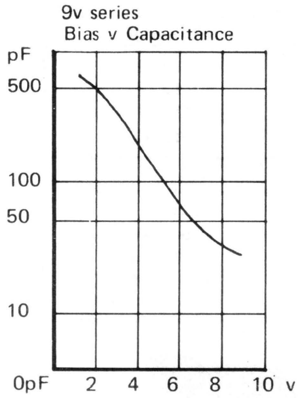


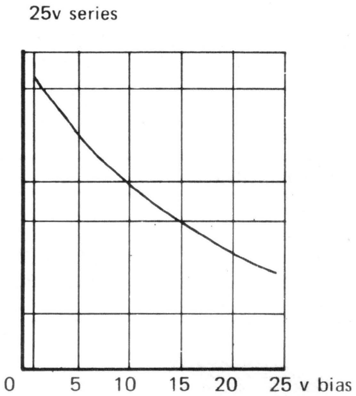
KV1216/KV1226

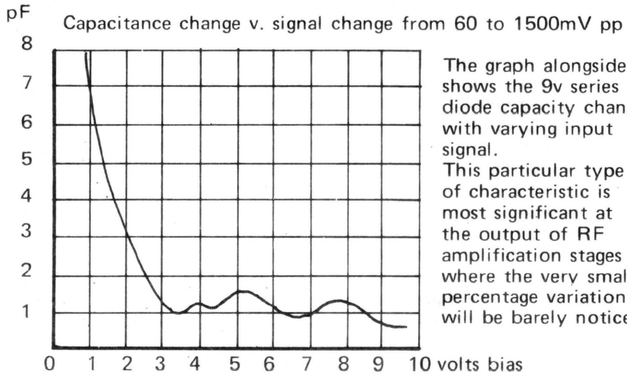




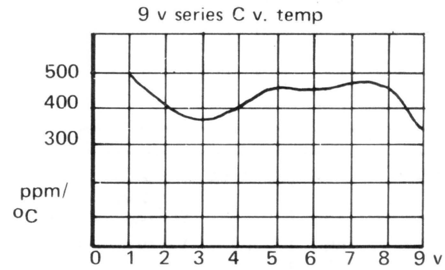
KV series varicap tuning diodes : Graphs and further information

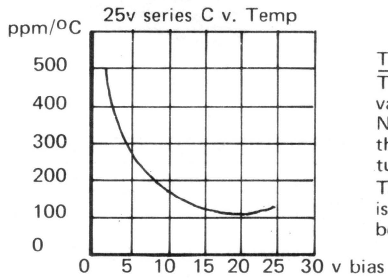




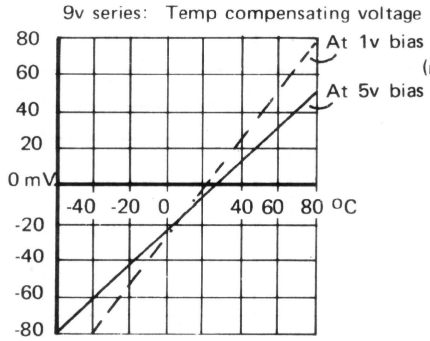


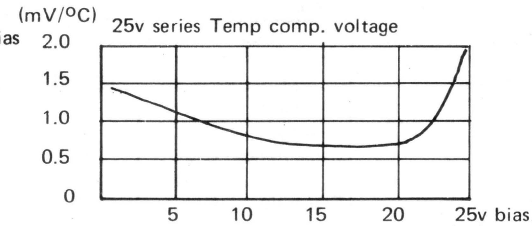
The graph alongside shows the 9v series diode capacity change with varying input signal.
This particular type of characteristic is most significant at the output of RF amplification stages where the very small percentage variation will be barely noticed





Temperature coefficient and varicap diodes
The determination of the correct compensation for a varicap tuning diode is one of the black arts of electronics. Not only does the TC vary with ambient and bias level - there is the inductor TC to consider as well. And the tuning voltage power supply. And the tuning pot itself. Taking the last point first, the tuning potentiometer is relatively easy, since it acts a divider, the TC ought to be self eliminating. But special law pots will have a variable TC according to the nature of track TC.
A wirewound linear ten turn pot is probably the safest means of deriving the tuning bias, albeit a non ideal law.





A further consideration for the tuning pot is the fact that the reverse leakage current will tend to double every 10°C or so - in the case of silicon junctions. The value of the tuning pot should thus be kept to 50k or less, otherwise use in rigourous enviroments where temperatures in excess of 50°C may not be satisfactory.

Ambit's long experience of DC tuned AM radio systems is accessible to OEM customers through our commercial/professional services scheme. All types of custom work is undertaken in connection with radio/audio/allied topics. Please apply for further details stating the nature of your requirements as concisely as possible.

SELECTION and GROUPING

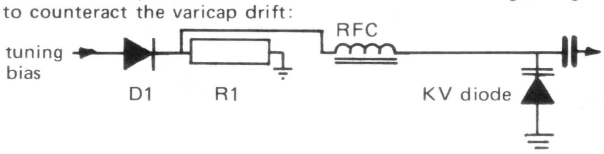
KV series diodes are batched according to the typical capacitance at a bias level of 1v. The grouping are indicated on the package according to the following tables. Bulk orders should be accompanied with details of four groupings preferred, and every effort will be made to ensure these are supplied. Individual requirements cannot be met without additional cost.

CAPACITY AT 9v	24.0 - 30.5pF		28.5 - 35pF		CAPACITY AT 25v 16.0 - 26.0pF
CAPACITY AT 1v	KV1210, KV1211	KV1215	KV1210, KV1211	KV1215	KV1221, KV1225, KV1226
465-494.5pF	13	D	33	N	
475.5-506pF	14	E	34	O	
487-518pF	15	F	35	P	
499-530.5pF	16	G	36	Q	
510.5-542.5	17	H	37	R	
522.5-555	18	I	38	S	
510-580					X
550-620					Y

In the case of the snap apart packages, the group marking is placed on the diode to be used in the local oscillator tuned circuit.

USING VARICAP DIODES

In most VHF applications, the relatively small capacity ratio required makes the compensation of TC drift less critical. But in high ratio devices, some form of correction is very desirable. Using a low TC power supply, one of the most widely used methods is to provide a diode in series with the tuning voltage to counteract the varicap drift:



The 25v series lend themselves easily to this correction device, since the requirement is a relatively flat 1mV. Checking the specifications of various cheap signal/rectifier diodes will reveal a useful solution with a IN4148 providing about 2mV correction with the appropriate R1 value (check individual manufacturers data - but about 100k). This overcorrection for the varicap allows room for the coil TC - which is also positive. Then, of course, don't forget the TC of R1 itself.

It isn't really surprising therefore, that the preferred method of correcting TC in tuning ciruicts is to make the basic circuit using components of good stability, and then simply use empirical techniques to accomplish the fine tuning. There are so many different factors to take into account, (and the TC of the coupling capacitor hasn't been mentioned yet), that it is nearly impossible to accurately predict the behaviour of the final product where the range of varicap tuning is so broad.

Of course, if you use frequency synthesis techniques, the PLL is the ideal form of compensating feedback - and you can really simply forget about the TCs - except insofar as worst case typical performance parameters can be accomodated without the system running out of range at either tuning extreme.

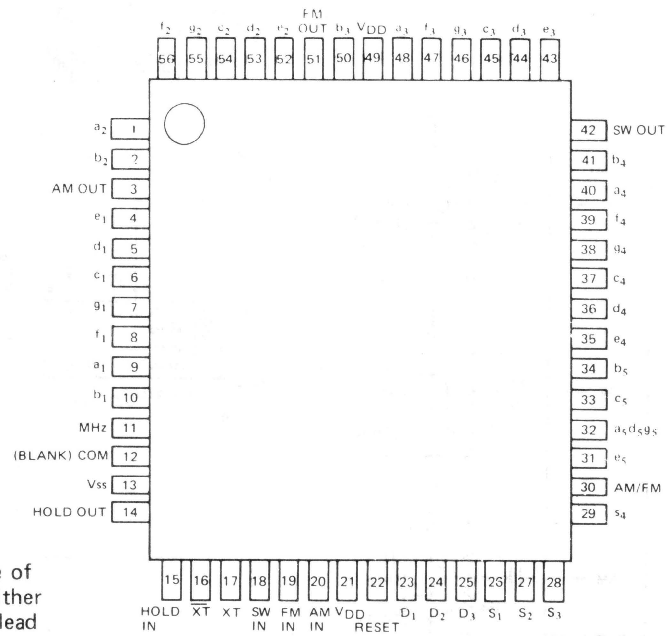
Anyone discovering a repeatable and reliable approach to the compensation of 'open loop' varicap diode tuned circuits is invited to let us know about it !!

OKI electric co. MSM 5527

DIGITAL FREQUENCY DISPLAY LSI : MSM5527

- Direct frequency readout for consumer and communications receivers
- 4½ digit resolution: 'AM' to 3999.9 kHz; 'SW' to 39.999 MHz; 'FM' to 399.99 MHz
- Resolution to limits of the stability of non-synthesised local oscillators: 'AM' 100Hz; 'SW' 1kHz; 'FM' 10kHz
- Direct count facility with no offset
- Event counting with no gate period
- Display hold feature, reduces power consumption by freezing reading, and switching off the prescalar
- All common AM and FM IF offset values are available with diode programming, including 10.7MHz for SW dual conversion receivers
- Current consumption only 4-5mA for LCD version
- Part no. MSM55271 for vacuum fluorescent displays
  Part no. MSM5527 for LCD displays

(Top View) 56 Lead Plastic Flat Package



(Note) Pin 12 : COM for MSM5527RS and BLANK for MSM55271RS.

General

The MSM5527 is the most recent addition to OKI's unique range of frequency display CMOS LSI. It provides a static display with either LCD or fluorescent type displays - and in order to do this, a 56 lead flatpak package is used with 1mm lead spacing.

Accordingly, great precision and care is called for in the use of this type of package.

100Hz resolution is available on the direct AM input facility, which enables the IC to be used in precision marine RDF systems that are designed for use in the 300kHz region.

The 10kHz resolution available at the FM input, via a divide by 100 prescaler, enables this device to provide readouts through the aircraft, amateur and public service bands. The 10kHz step is the limit of tuning stability for a non-synthesised local oscillator, and so further resolution to the 5kHz channel points would not be feasible.

A particularly useful feature is incorporated, whereby the display may be frozen using the Hold In and Hold out terminals.

The application of a positive voltage to hold in, will disable the prescaler 300mSec later. Thus freezing the display instantaneously, and preventing any 'glitch' disturbance as the pre-scaler is turned off 300mSec later.

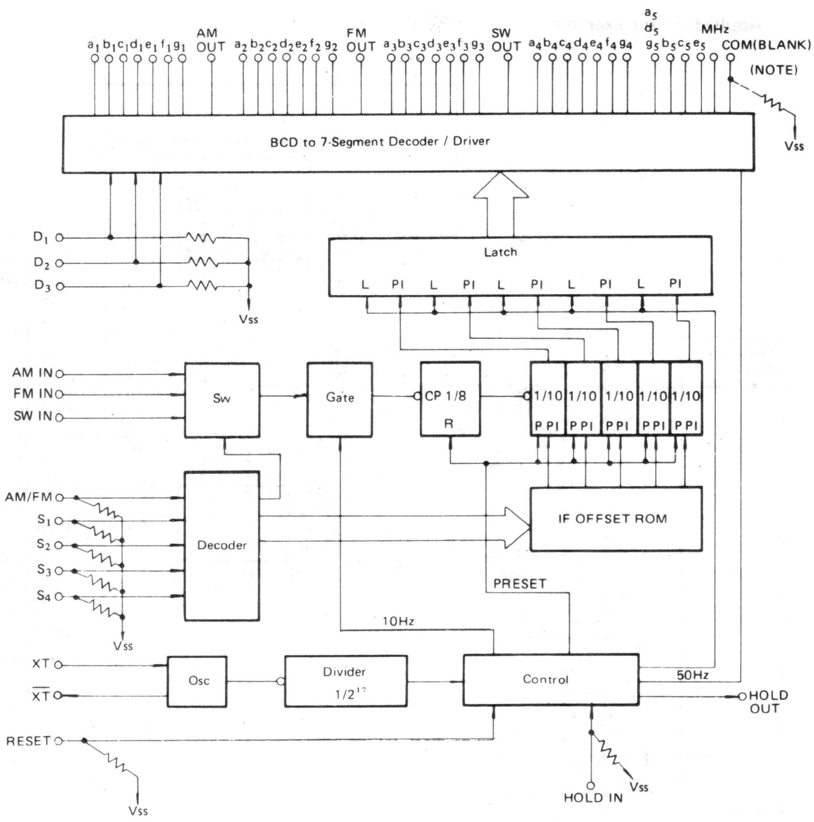
In varicap tuned sets, the necessary hold voltage may be derived from an integrator sensing the tuning voltage line, so that if there is any change detected - the prescalar may be re-enabled to update the count.

In purely mechanically tuned systems, the hold function may be disabled by touch sensing of the tuning control - or a switch specifically for display hold.

The event counter function may be used in conjunction with an external gate system to permit the measurement of period.

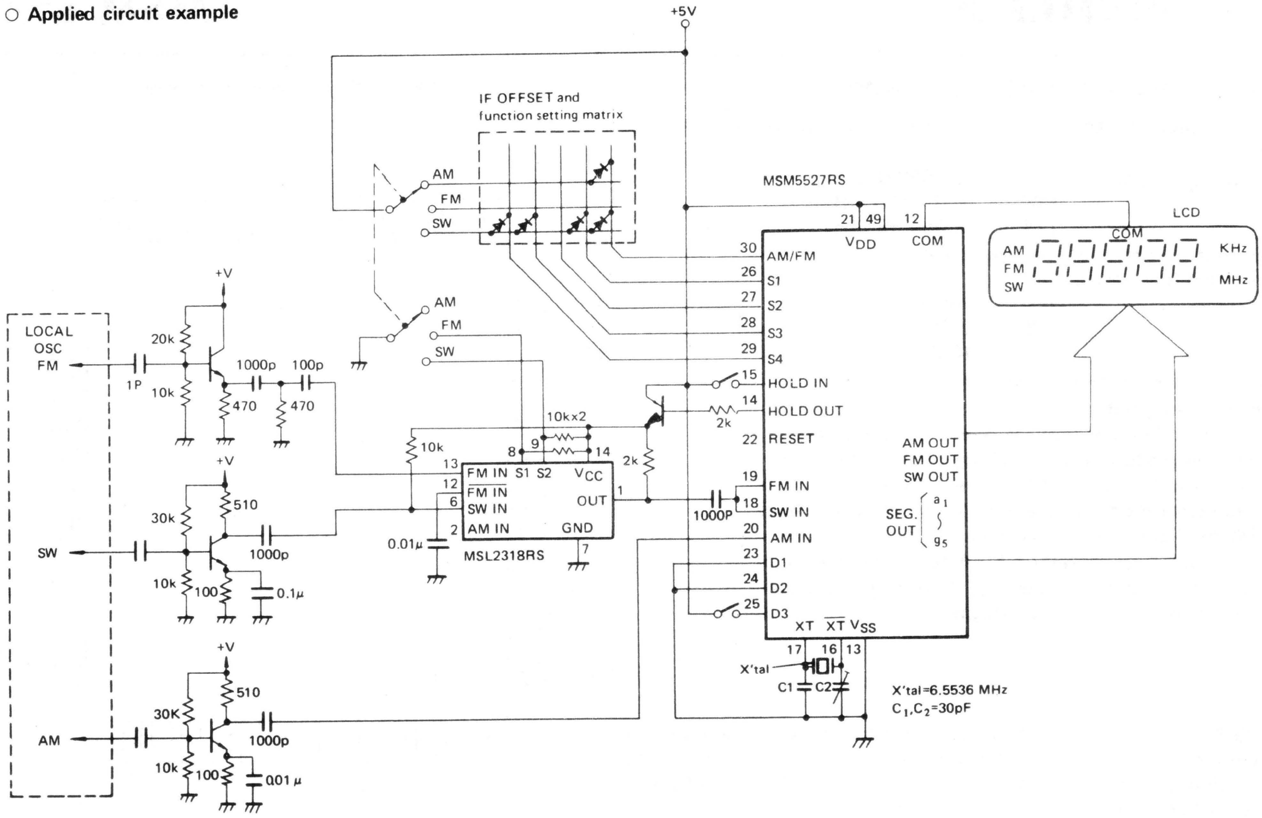
One particularly useful feature is the 10.7MHz IF offset made available in the SW mode, thus enabling use with dual conversion HF receivers using widely available 10.7MHz crystal filters. There are no plans for the introduction of a display system to accomodate the 9MHz offset frequently encountered in amateur communications equipment.

Details of suppliers of 10.7MHz SSB filters are available on application. (TOYO etc)

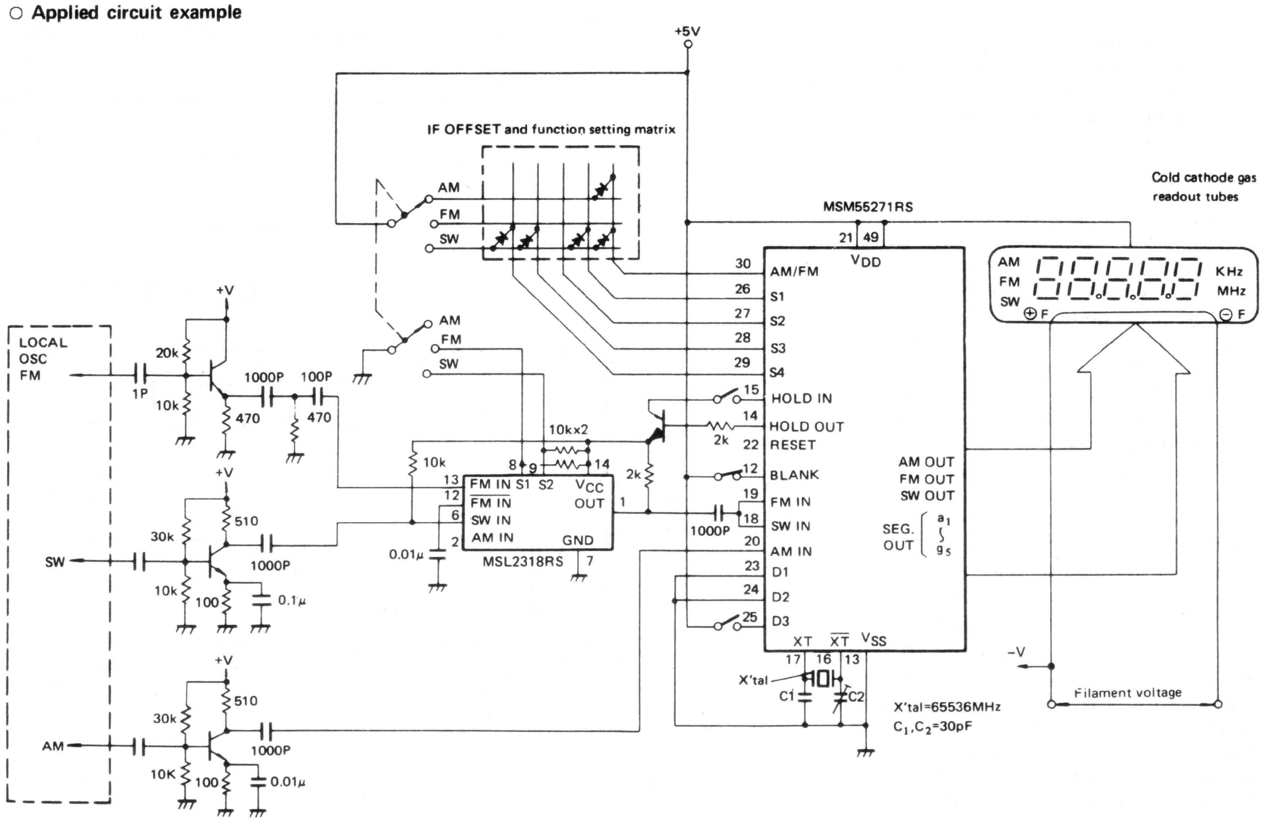


(Note) The dash line - - - - - and (BLANK) denote the case of the MSM55271RS.

○ **Applied circuit example**



○ **Applied circuit example**



Pin-by-pin

Crystal oscillator

A 30pF/parallel 6.5536MHz crystal is used to provide the timing and gating references

AM,FM,SW input

A built in CMOS amplifier at each input enables operation from 1v p-p local oscillator drive (or via prescaler). Capacitor coupling should be used where the signal is less than 3.6v p-p. The OKI MSL2318 should be used for prescaling.

Reset (with built in pull-down)

1) When operating in AM,FM or SW modes, if the reset pin is taken to V_{DD}, the contents of the IF OFFSET rom is displayed for checking programming.
2) When operating in the FC, or event Counter modes, the counter is reset when the Reset pin is taken to V_{DD}, and the data in the counter is immediately loaded into the 18 bit latch.

AM/FM, S1,S2,S3,S4 (with built in pull down)

These pins determine the operating mode and the IF offset ($H = V_{DD}$, $L = V_{SS}$ or open circuit)

SELECT INPUT AM/FM	S1	S2	S3	S4	DISPLAY	IF OFFSET VALUE
H	L	L	L	L	AM	-455kHz
H	L	H	L	L		-260kHz
H	L	L	H	L		-450kHz
H	L	H	H	L		-261kHz
H	L	L	L	H		-468kHz
H	L	H	L	H		-470kHz
L	L	L	L	L	FM	+10.7MHz
L	H	L	L	L		+10.63MHz
L	L	H	L	L		-10.7MHz
L	H	H	L	L		+10.66MHz
L	L	L	H	L		+10.74MHz
L	H	L	H	L		+10.77MHz
L	L	H	H	L		-10.63MHz
L	H	H	H	L		-10.65MHz
L	L	L	L	H		-10.66MHz
L	H	L	L	H		-10.67MHz
L	L	H	L	H		-10.68MHz
L	H	H	L	H		-10.71MHz
L	L	L	H	H		-10.74MHz
L	H	L	H	H		-10.75MHz
L	L	H	H	H		-10.77MHz
L	H	H	H	H		-10.78MHz
H	H	L	L	L	SW	-455kHz
H	H	H	L	L		-468kHz
H	H	L	H	L		-2.0MHz
H	H	H	H	L		-10.7MHz
H	L	L	H	H	FC	NONE
H	L	H	H	H	Counter	NONE

Fig. 3 shows a setting example of AM, FM, SW, and S_1–S_4.

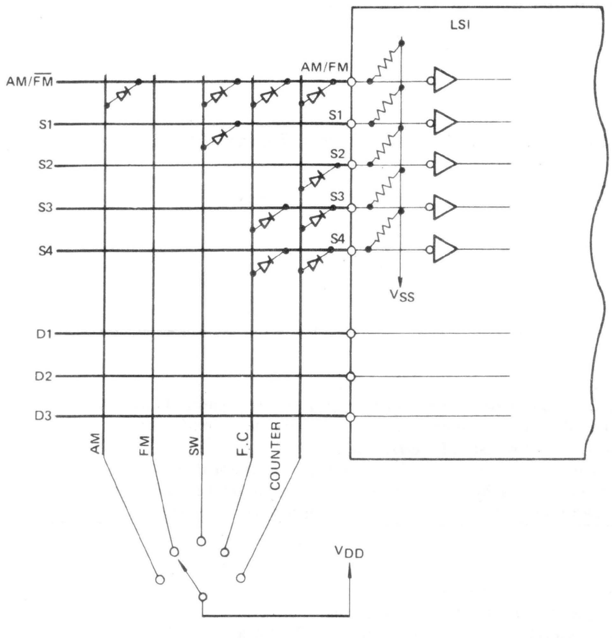


Fig. 3

D1,D2,D3 (with built in pull down)

Programming these pins determines the display 'round up' and number of digits to be used (4 or 5)

(S1,S2,S3 contd)

SELECT pin D1	D2	D3	Display digits	Display contents
L	L	L	5 digit	Normal display
H	L	L		0 displayed for first 4 digits on AM & SW 0 displayed for last digit on FM, with 1,3,5,7,9 for the penultimate digit.
L	H	L	4 digit	Last digit is 0
I	H	H		Normal display
H	L	H		0 displayed on last digit in AM mode 1,3,5,7,9 for final FM digit

Hold in/out

The hold in terminal freezes the display when taken to V_{DD}, and 300mSec later drives the Hold out terminal low, thus enabling the prescaler to be switched off to conserve power.

DISPLAY FORMAT

Display selected	4 digit	5 digit	Unit
AM band	0-3999	0.0-3999.9	kHz
FM band	0.0-399.9	0.00-399.99	MHz
SW band	0.00-39.99	0.000-39.999	MHz
FC	0-3999	0.0-3999.9	kHz
Counter	0-3999	0-3999.9	(events)

AM,FM,SW out

For connection to band indicators on LCD or fluorescent displays, activated when the appropriate input selected.

MHz

Frequency unit display indicator when FM or SW selected

Common (MSM5527)

The LCD backplane 50Hz, 50% duty cycle

BLANK (MSM55271)

Display is unaffected while Blank is at V_{DD}, though outputs float when taken to V_{SS}, enabling the display tube to be used in conjunction with other LSI, such as a clock timer.

DEVICE RATINGS

1 MSM5527 for LCD	Min	Typ	Max	Notes
Power supply voltage , V_{DD}	-0.3	5	7	Volts
Input voltage	-0.3		V_{DD}	Volts
Storage temperature	-55	20	+150	°C
Operating range	-30	20	+85	°C
Fosc		6.5536		MHz
H input voltage (DC connection)	3.6			V
L input voltage			0,8	V
H output voltage at Iout = -15uA	4.2			V
L output voltage at Io=15uA			0.2	V
Output current, Vout=0v/Vout=5v	-0.2/0.2			mA
Hold out output current, Vo2.5v/0.4V	-0.2/1.6			mA
Hold out output voltage H, Iout -40uA	4.2			V
Hold out output voltage L, Iout=1.6mA			0.4	V
Current consumption, no load			4	mA
Operating frequency, Vin 1v p-p AC	5			MHz
2 MSM55271 for fluorescent display				
Power supply voltage	-0.3	5	7	V
Input voltage	V_{SS}-0.3		V_{DD}+0.3	V
Output voltage	V_{DD}+0.9		V_{DD} -20	V
Storage temperature	-55	20	+150	°C
Operating range	-30	20	+85	°C
Fosc		6.5536		MHz
H input voltage	3.6			V
L input voltage			0.8	V
H output current, Vo = V_{DD} -1v	-0.2			mA
Output leakage current, Vo = V_{DD}-20v			-10	uA
Hold out as MSM5527				
Operating frequency, as MSM5527				

FURTHER INFORMATION

See the information relating to the DFM3 and DFM6 units, described elsewhere in this issue.
The MSM5527 and MSM55271 respectively, are used with both LCD and vacuum fluorecent displays to provide a user programmable direct frequency readout based on the features described herein.

DISCRETE DEVICE UPDATE

General

This page is a general supplement to previous issues, covering various discrete transistors, diodes, varicaps and misc. passive components.

Transistors

AUDIO	Complement	type	Base	P_{tot}	I_c -mA	V_{ceo}	ft MHz	h_{fe} at I_c and V_{ceo}	NF dB at	f Hz
2SC2546	2SA1084	NPN	1	400	100	90	90	400-800 2mA 12v	0.2	1kHz
2SA1084	2SC2546	PNP	1	400	100	90	90	400-800 2mA 12v	0.2	1kHz

The above devices are (at the time of writing) the world's lowest noise audio devices, measured at 10k ohm, 6v and 0.2mA with a 1kHz reference. They offer exceptional voltage capability, and should easily substitute older types of silicon transistor if you want to update older audio equipment. They are ideally suited to moving coil pickup preamps, and similar demanding applications.

	type	Base	P_{tot}	I_c -mA	V_{ceo}	ft MHz	h_{fe} at I_c and V_{ceo}	NF dB at	f Hz
BF362	NPN	2	120	20	20	800	20min 3mA 10v	4.5	800MHz
BF479	PNP	2	160	50	25	1800	20min 10mA 10v	4.5	800MHz
BF679	PNP	2	160	30	30	900	25min 3mA 10v	2.8	800MHz
BFW92	NPN	3	130	25	15	1600	25min 25mA 1v	4.0	500MHz
BFY90	NPN	4	200	25	15	1300	25min 2mA 1v	5.0	500MHz

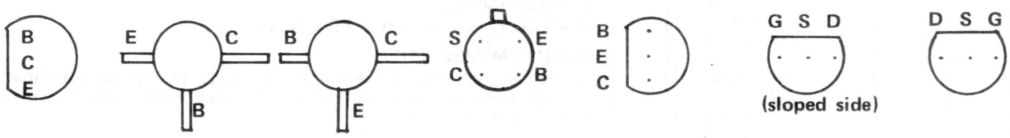
The above are mainly VHF/UHF RF, OSC and Mixer devices, designed for a variety of applications in TV tunerheads, CATV systems and communications applications. The very high Ft makes them suitable for a variety of wideband and instrumentation applications.

	type	Base	P_{tot}	I_c -mA	V_{ceo}	ft MHz	h_{fe} at I_c and V_{ceo}
BF440	PNP	5	300	25	40	250	60/120 1mA 10v
BF441	PNP	5	300	25	40	250	30/125 1mA 10v

The above pair have been adopted for use as Ambit's standard PNP RF transistors, for applications such as oscillators, multipliers etc.

JFET		type	Base	Ptot	IDSS	VGS(off)	y fsmmho	GaindB at MHz		NFdB	Notes
2SK168	CASCODE	n-ch	6	150	10/20	−3.0 v	10	27	100	1.7	3 pin cascode type
2SK55		n-ch	6	200	3/7mA	−5.5v	8	18	100	2.0	Low current type
J310		n-ch	7	625	24/60mA	−6v	12	16	100	1.5	Hi gain/lo noise

The above JFETs are introduced into our range to provide a low current selected type, for various battery applications, such as radio control systems etc. The J310 is included as this offers excellent VHF performance, with 11dB gain at 450MHz, and 2,7dB NF.



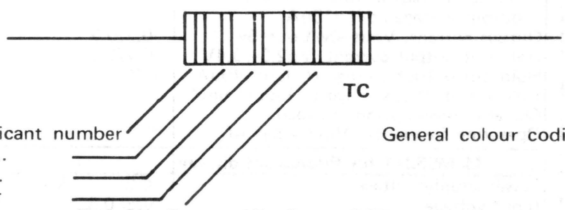
Base 1 2 3 4 5 6 7

METAL FILM RESISTORS

AEG type 0207

Stock range E12 series 1ohm to 1Mohm Temp coeff +/− 50 ppm Tolerance 1% 0.35W rating 250v max

In order to take advantage of the new low noise stereo decoders, IFs and general audio semiconductor improvements, Ambit is using these new metal film resistors in critical applications. These resistors are marked in a 6 ring colour code, in accordance with the usual practise for this type of resistor. However, since many users will be unfamiliar with this code, it is reproduced here for reference:



Ring 1 1st significant number
2 2nd ..
3 3rd ..
4 Multiplier
5 Tolerance :Red=2%, Brown=1%, Green=0.5%
6 Temp coeff :Black=200ppm, Brown=100ppm, Red=50ppm

General colour coding is the same as for 5% carbon film types etc.

The range stocked by Ambit is the SMA0207 50ppm/1% - so bands 5 and 6 are Brown and Red respectively. The 5th/6th ring are separated slightly further apart then the rest of the bands.

EXAMPLE

Band 1	Yellow	4
Band 2	Mauve	7
Band 3	Black	0
Band 4	Brown	x10
Band 5	Brown	1%
Band 6	Red	50ppm

= 4700 ohms

FERRITE BEADS

FX1115 et alia:

As well as the oft cited, and much loved FX1115 ferrite parasitic suppressor bead, we offer three other beads in the same ferrite for a wide variety of applications requiring the decoupling/isolating effects afforded by threading them over conductors/base leads etc.

FB1 (FX1115)	5mm long	2mm hole	4mm wide
FB2	5mm long	1.6mm hole	3.5mm wide
FB3	4mm long	1.2mm hole	3mm wide
FB4	2mm long	0.6mm hole	2mm wide

General notes on the use of ferrite beads
The beads used here are made from a high permeability, low frequency material - whose losses increase considerably with an increasing frequency, thus effectively damping HF instabilities.

MICROMETALS TOROIDS and TOROID MOUNTS

The following toroids have been included as stock items:
EMI types T 94-40, T 37-40, T 25-40 Resonant types T20-2, T20-6, T20-12

A range of toroid mounts, with multipin bases is also now stocked for ease of assembly using smaller cores:
TM200-4, TM401-6, TM421-3, TM501-6 Details of these and all the cores are available in the Micrometals Catalogue **No.3.**

A new 16 page catalogue detailing the EMI filter cores, with particular reference to SMPS filter applications is available in addiditon to the basic catalogue. (Also includes details for pulse transformer cores.) - **ORDER AS Micrometals Catalogue No.4**

OTHER DISCRETE COMPONENTS

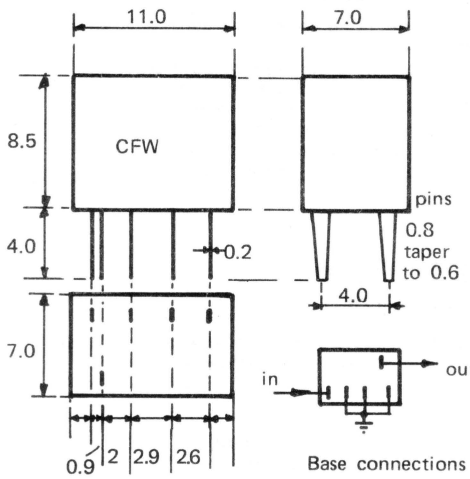
Now that Ambit has moved to considerably larger premises, we will be expanding our ranges of passive components. Specifically, a range of ceramic capacitors, tantalum and electrolytics types will be included in a future edition of the price list

CERAMIC FILTER UPDATE: CFG/CFW/CRM

The following filters are additions to our current ranges of MURATA and TOKO filters, and are stock items. Other filters in the series are available to special order. Further stock types will be phased in during the next year.

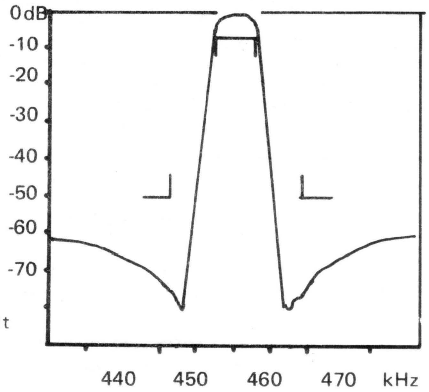
CFW455HT -multi element, non-hermetically sealed ceramic ladder filter for 455kHz

Dimensions in mm (drawn approx x2)



Base connections

Typical bandpass
markers indicate spec limits



Electrical specifications

Type	6dB BW	50dB BW	Z in/out
455HT	±3kHz	±9kHz	2K

Insertion loss 6dB max

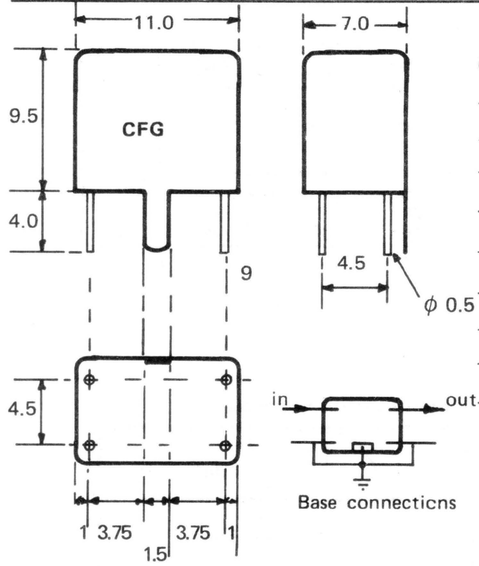
Notes on CFW series

The Murata CFW series (NTK LF-HS) are the next step up from the CFU range, with an extra two resonators.

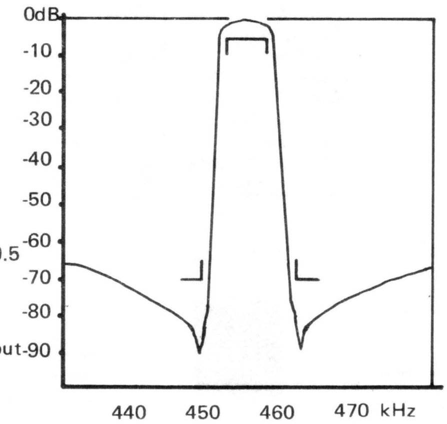
Since the original specs were drawn up, the typical performance of the filters now far exceeds the specification - as the graph alongside shows.

The CFW455HT is a higher spec than the basic CFW455H, possessing a spurious response level of −60dB or better. It is ideally suited to radio control (AM or FM) and all types of communication and high quality broadcast AM radio receiver.

CFG series - miniature multi element, hermetically sealed (metal case) ceramic ladder filter for 455kHz



Base connections



Electrical specifications

Item	CFG455H	CFG455I
6dB BW	±3kHz	±2kHz min
70dB BW	±7.5kHz	±5kHz max
Loss	7dB	8dB max
Ripple	3dB	3dB max
Spurii	-50dB	-50dB ±100kHz
Impedance	2K	2K ohms

Notes on the CFG series

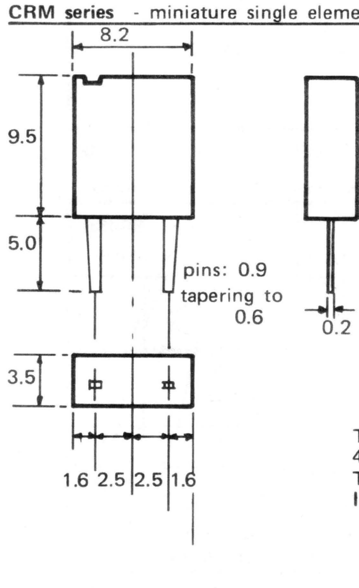
The CFG series are the smallest types of ceramic ladder filter available in metal encapsulation. The spec is basically the same as the larger CFR455 ranges.

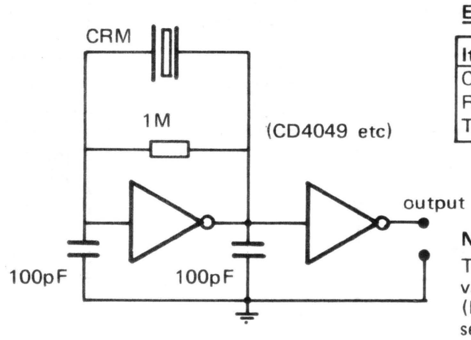
The extreme selectivity available in the small encapsulation can only be achieved provided precautions are taken to avoid stray input/output coupling.

In view of the small size of the CFG series, these filters are readily accomodated in equipment requiring improved selectivity. The FRG7 filter may usefully be replaced by the CFG455H (for AM and broadcast), or with the CFG455I if better SSB performance is desired. The CFG455I is still quite useable for AM, although some of the higher frequency audio will be lost.

The filter supplied with the FRG7 is an NTK LFC6 (Murata CFM455H), so a useful improvement is noticed when it is replaced with a CFG series unit. There is a fair amount of space around the filter in the FRG7, so it is not inconceivable that a diode switched supplementary PCB could be devised to slot in.

CRM series - miniature single element ceramic resonator





TOKO CRM elements are available in the range 400 to 600kHz to special order.
The CRM readily performs the function of IF traps in decoupling and bypass applications

Electrical specifications

Item	CRM455A
Centre frequency	455kHz ± 2kHz
Resonant resistance	20 ohms maximum
Temp. stability	± 0.3% max (F/F)

Notes on CRM series

The TOKO CRM series are equivalent to various other single element resonators (Murata CSB etc). As well as providing selectivity as an emitter bypass device in tuned amplifiers.

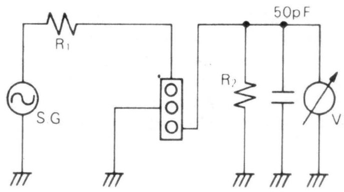
It can be used for various applications, specifically as a clock oscillator for MPUs and as a stable BFO element.

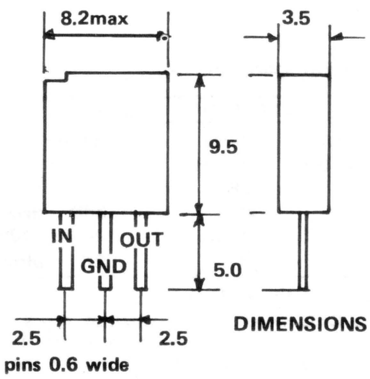
It is also useable as a quadrature element for NBFM, although work on this applications is not yet completed.

CFM2 MECHANICAL IF FILTERS for 455-470kHz (- INCLUDING CFMA and CFMQ types)

Low cost block filters for the region 455-470kHz are usually made from ceramic technologies. And whilst ceramic IF filters can be made to very exacting specifications, there are certain inherent problems associated with their use.
For example, the low cost ceramic IF filter has a particularly poor rejection of spurii at higher frequencies, due to the inter-electrode capacitance effects - comparing them with the excellent response of the CFM2 at higher frequencies revealing the first major advantage of the mechanical coupling technique.
The long term stability of TOKO's mechanical filter is also superior, along with all physical stress tests including temperature, humidity, and vibration.
These basic advantages make the CFM2 series ideally suited to communications equipment, and car radio, where the harsh enviroment usually encountered in the operation of such equipment rules out ceramic filter applications.
Combinations of the CFM2 with low cost ceramic filters - such as Murata CFU455 series - exploit the advantages of both types of filter - namely the excellent shape factor of a ceramic filter network, coupled with the excellent stopband characteristics of the CFM2.
Various applications with matching transformers, for the input and output, are also described in this note - and it should be noted that the CFMA series of mechanical filters, is simply a single package version of the CFM2455, with external input matching IFT.
Likewise, the CFMQ is a single package assembly that includes both input, and output matching transformers. In the interests of stock holding capacity, the input/output matching transformers are stocked separately, along with all ranks of the basic 455kHz CFM2 element.

Test Circuit





DIMENSIONS (mm)

PART NUMBERING SYSTEM: CFM2 series

[1] H CFM2 [2][3][4][5]

1 Centre frequency tolerance : A = +/– 1kHz (special order only)
NIL = +/– 2kHz (standard)

2 Centre frequency in kHz

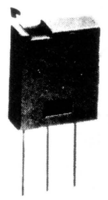
3 Bandwidth (see table)

4 Input impedance conditions: L = for matching IFT
NIL = resistive

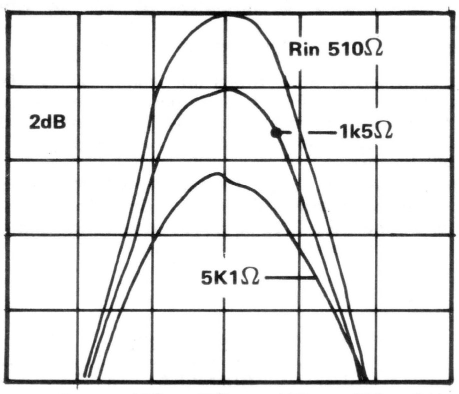
5 Output impedance conditions: as 4

Parameter	CFM2 455A	455B	455C	455D	455E	
Center freq			455±2			kHz
Bandwidth	4min	6min	8min	10min	12min	kHz
Selectivity ±9kHz	>18	>16	>12	>9	>6	dB
Ripple			2 max			dB
Insertion loss			6 max			dB
Stop band			30 min			dB
Input impedance	1k	1k5	2k	2k	2k	Ω
Output imp	1k5	2k	2k	2k	2k5	Ω

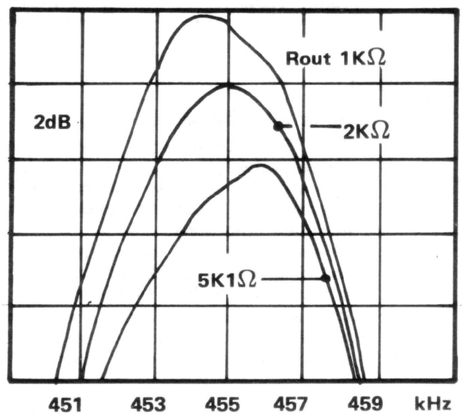
CFM2-455



Output impedance fixed at 2KΩ



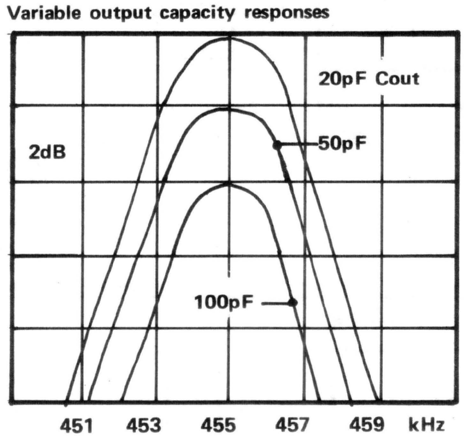
Input impedance fixed at 1K5Ω



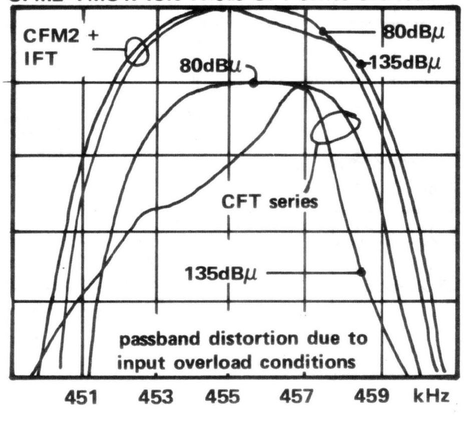
CFMQ=
CFM2 +
in/out
matching IFT

CFMA =
CFM2 +
input
matching IFT

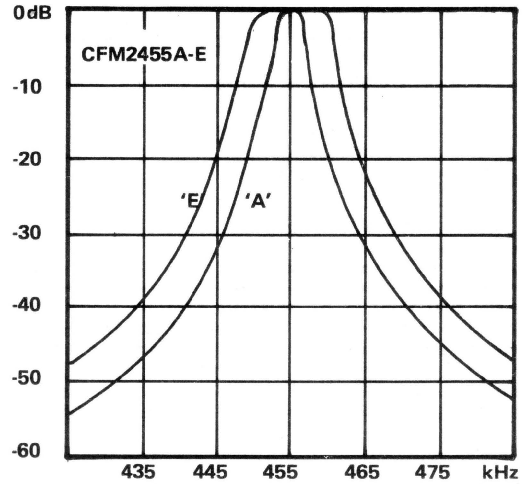
Variable output capacity responses

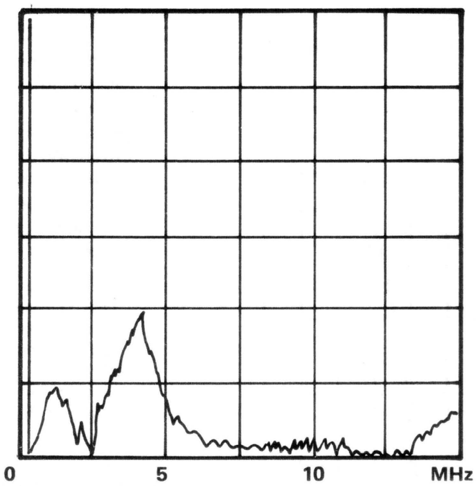


CFM2+7MC4718N versus CFT series overload

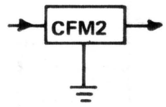


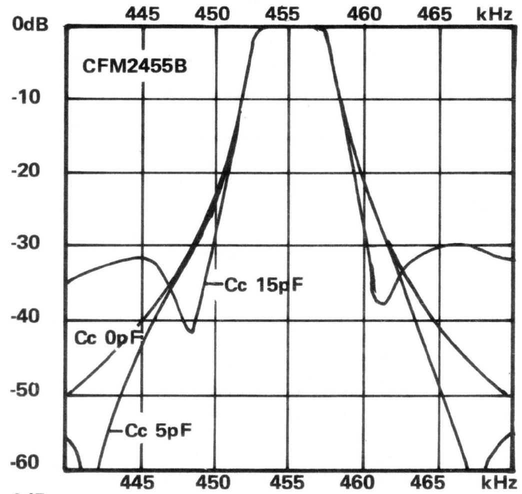
BANDPASS CURVES

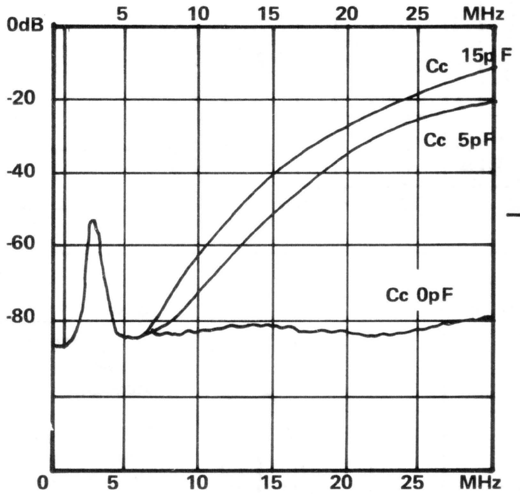




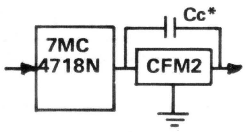
Single CFM2 filter element response

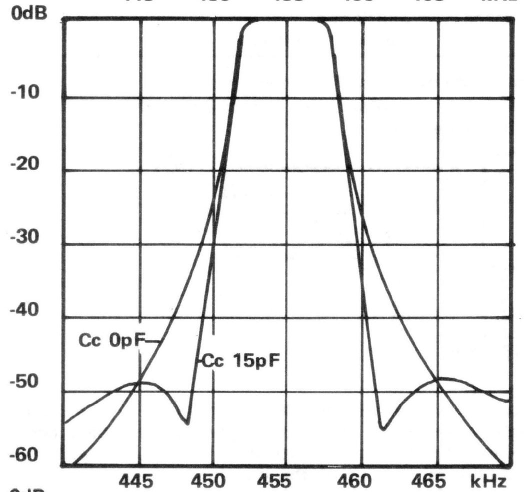


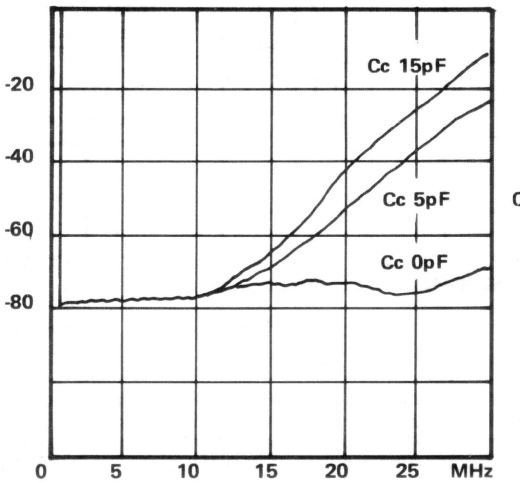




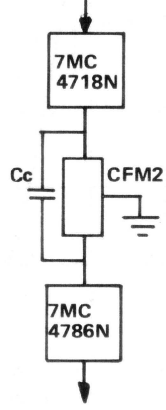
CFM2 filter with input matching transformer and various values of coupling cap.

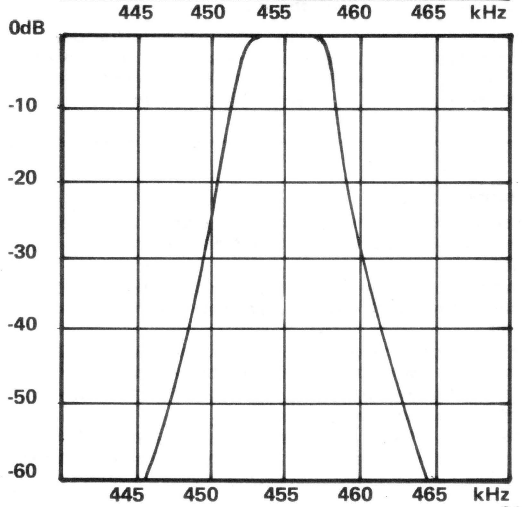


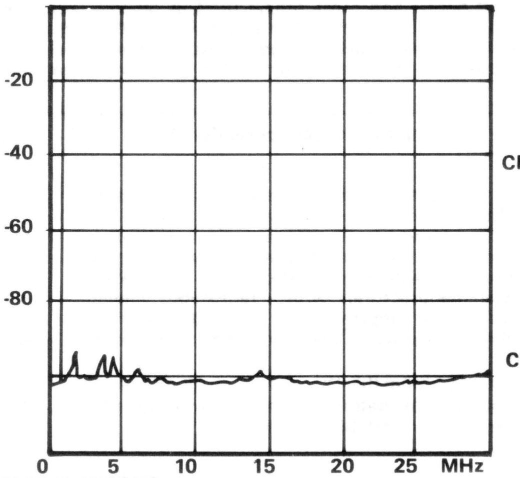


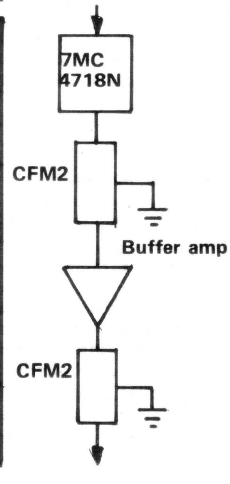


With input/output matching tranformer and various Cc







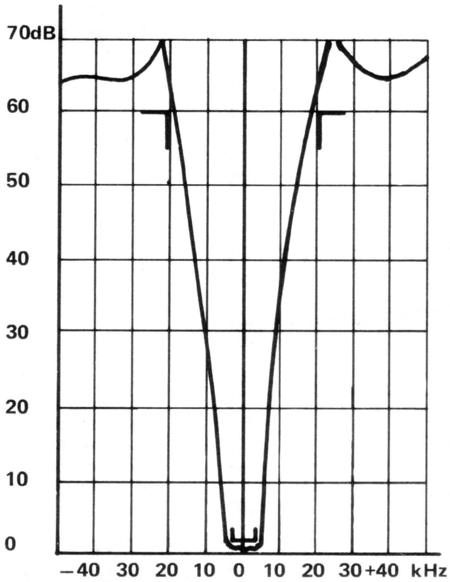


CRYSTAL FILTERS 35.4MHz / 10.7MHz

Cathodeon BP4772 (or sim)

A 35.4MHz crystal filter for use in the first IF of HF receivers

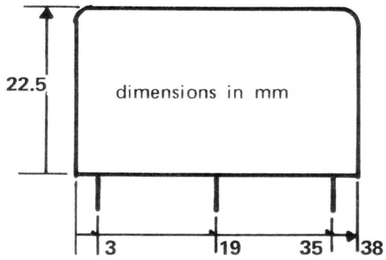
Curve of typical filter, markers indicate spec limits

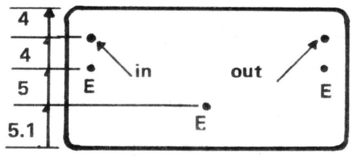


Specifications

Centre frequency	35.4MHz
Passband	± 4.25kHz at −3dB
Stopband	± 20kHz at −60dB
Passband ripple	2.0dB max
Insertion loss	3.0dB max
Ultimate attenuation	60dB min ± 5MHz
Out of band spurii	30dB min
Termination	50 ohm in/out
Maximum input power	**20mW**
Operating temperature	−10°C to +65°C

This is intended as the first filter in HF receivers covering 10kHz to 30MHz (with LO of 35.4 to 65.4 MHz). The terminations are made to suit common 50 ohm systems, although it must be noted that diode DBMs are best fed with pure resistance - and so the output of the first mixer (if of the popular DBM types) should be isolated in a buffer stage. If fed directly, 3rd order IM suffers - an FET stage is best.

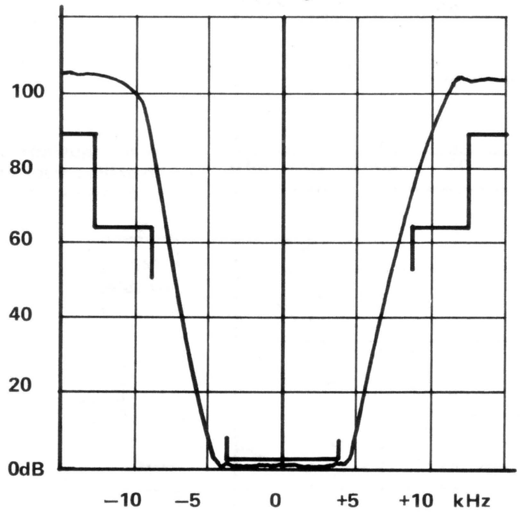




underside pins

TOYO H44F02 (or sim)

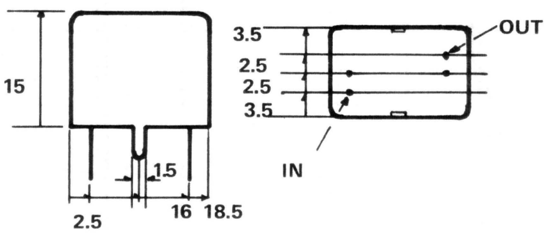
10.7MHz 12.5kHz channel spacing



Specifications

Centre frequency	10.7MHz
Passband	± 3.75kHz at −3dB
Stopband	± 8.75kHz at −65dB
	± 12.5kHz at −90dB
Passband ripple	2.0dB max
Insertion loss	3.5dB max
Ultimate attenuation	90dB ± 12.5kHz to ± 300kHz
Input/output impedance	3k3/2.5pF
Operational temp	−40°C to +80°C

The H44FO2 is the 12.5kHz channel spacing counterpart of our 10M4B1. It is an 8 pole monolithic filter, meeting all MPT requirements in suitable designs.



NB

The H44F02 (like other monolithic 10.7MHz filters of this type) requires input/output matching with a suitable IFT. The same types described for use in conjunction with the 10M4B1 are suitable, although experimental work has shown that the very best results are obtained with a 10.7MHz tuned circuit directly across the input/output terminations - with an FET input/output isolating buffer stage. Passband ripple has been reduced to below 0.3dB by careful adjustment of the matching reistances.

Both the filters described here - and most others used in quality radio equipment must have carefully screened input/output circuitry, or much of the available stopband attenuation will be wasted. This phenomenon increases with frequency, and so the 35.4MHz filter must have a screen shield placed across the underside of the PCB for best results. Linear circuit layout is a great help in achieving best results, and correct choice of termination impedances is essential if ripple free operation is to result.

Crystal filters : general notes and considerations

Historical background

The discovery of the piezo electric effect in natural quartz led to W.G.Cady developing the first successful 'crystal' oscillator in 1912. Although natural quartz is abundant, its occurance in nature is random (although the actual lattice is of course consistent) - and so processes for the forced growth of synthetic quartz have been refined to provide crystal manufacturers with a repeatably aligned configuration. This leads to vast improvements in yield, due to fewer flaws, and the ability to process the quartz bar with automatic cutting equipment.

The growth of the crystal is along the 'Y' axis in most instances, as this offers the best number of wafers for the construction of both frequency elements, and crystal filter elements.

Orientations

The various blanks are cut from the crystal bar according to long established practises, based on the frequency range required.

The most popular cut is the AT - and the best yield is obtained from the 'Y-bar' - as you can see from the diagrams.

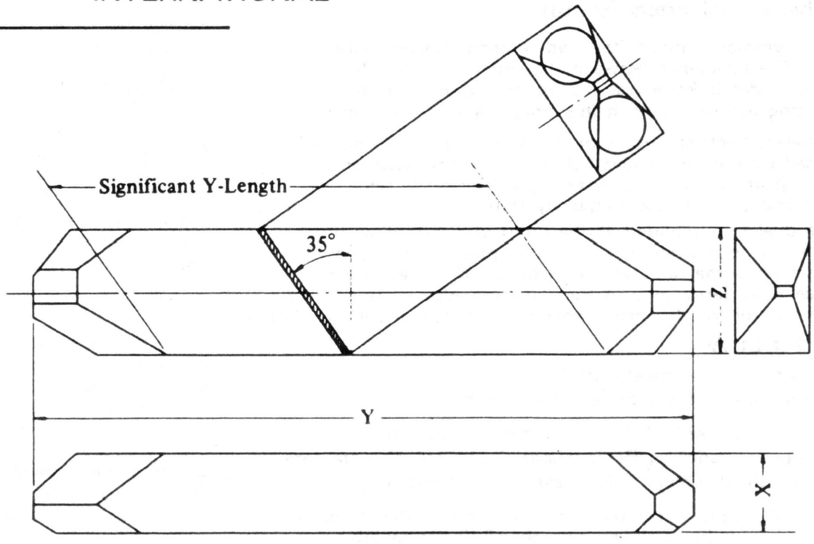
Whilst it is not anticipated that many of you will be trying to grind your own blanks, information on the actual detail used for manufacturing is included in the table accompanying the illustrations.

The monolithic filter is discussed at some length in this note - and most of these are based on AT quartz blanks.

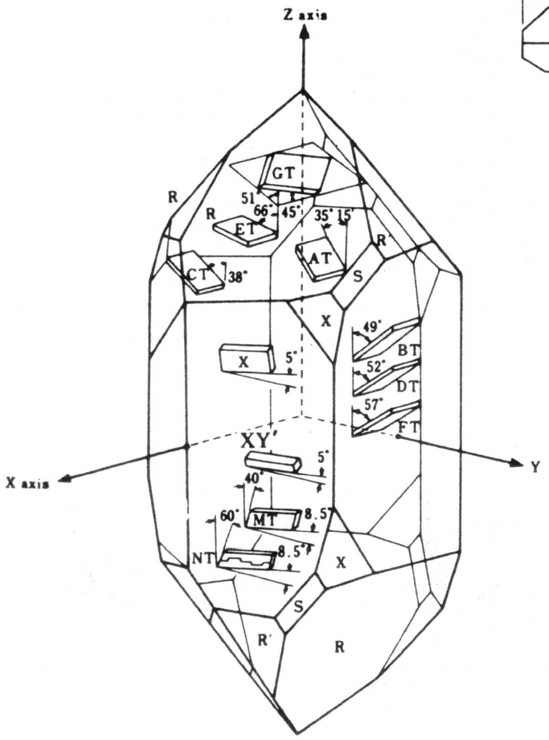
Quatrz crystals : Cutting details

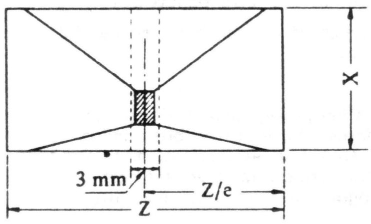
The quartz bar is produced by a process of first 'seeding' the growth with a correctly positioned slice, and then forcing the crystals to growth at high temperature and under pressure of about 1500kg/cm^2

The resultant Y bars are 160-230mm in length.



Planes of orientation referred to in the 'cut'



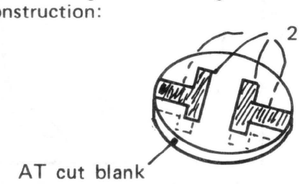


Cut type	Frequency span in MHz	Mode	formula
AT	0.5 - 25	thickness shear	1670/t (kHz/mm)
AT	10 - 350	thickness shear	1670.n/t (n=3,5, or7)*
BT	0.001 - 35	thickness shear	2560/t
BT	20 - 75	thickness shear	2560.n/t (n=3)*
CT	0.25 - 1.0	face shear	3080/L
DT	60kHz to 500kHz	face shear	2070/L
MT	50kHz to 500kHz	longitudinal	2700/L
NT	4kHz to 100kHz	length/width flex	5700x W/L^2
+5° X	60kHz to 250kHz	longitudinal	2830/L
+5° X	10kHz to 50kHz	length/width flex	5200x W/L^2
XY'	1kHz to 35kHz	length/width flex	5700x W/L^2
X	350kHz to 20MHz	thickness extens.	2970/t
Y	500kHz to 25MHz	thickness shear	1980/t

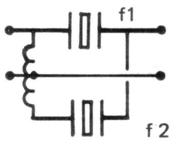
*n = mechanical overtone mode
t = thickness, L = length, W = width

Monolithic filters

Monolithic crystal filters are filters constructed by fixing two resonant regions in a single wafer, by the use of the following construction:



The classical half lattice crystal filter requires two separate crystal blanks to perform the same function - and then these two separate elements (poles) matched with a hybrid transformer:

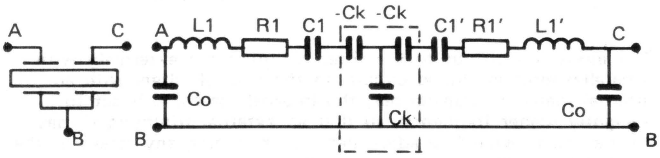


Half lattice filter with two discrete crystals

Construction is thus far more straightforward with monolithic techniques - as first practically demonstrated at the Bell telephones labs by R.A.Sykes in 1965. The breakthrough came about when it was recognized that the monolithic wafer was basically formed of two acoustically coupled resonators - which lead to the analysis of the network in its LC equivalent, and thus to repeatable formulae which could be used in manufacturing.

Since that time, up to 8 electrode pairs have been deposited on a single wafer - although for practical purposes, the monolithic 'dual' is the stock element used in multiple configurations.

The electrical equivalent of the monolithic dual is



Both resonantors consist of the terms L1, C1, R1 and L1', C1', R1'. Co and Co' are the electrode capacitances. The 4 pole network involving Ck and —Ck represents the mechanical coupling between the resonators. This factor, k, is given by

$$k = C1/Ck \quad (1)$$

Usually, between 10^{-4} and 10^{-3} for fundamental duals.

The resonant frequencies of both resonator pairs are either equal, or very close, given by

$$f1' = f1 = \frac{1}{2\pi\sqrt{L1C1}} \quad (2)$$

In the context of the monli dual, two extra resonances now occur. Firstly when the input/out are paralled, the symmetric frequency occurs:

$$f\ sym = f1\sqrt{1-k} \quad (3)$$

The crystal vibration is in phase.

When connected in series, the antisymmetric response occurs,

$$f\ asym = f1\sqrt{1+k} \text{ or approx } fsym(1+k) \quad (4)$$

Monolithic crystal filters (contd)......

In the antisymmetric mode, the two resonant systems vibrate in antiphase, The difference between the symmetric and the anti-symmetric modes is known as the 'mode spacing' for the dual. Mode spacing increases with higher coupling (k) as shown in (4).

The equivalent electrical circuit of the monolithic dual may be transformed into an equivalent half lattice bridge, with the hybrid transformer. The two systems then have the same amplitude and phase versus frequency responses.

When the dual is terminated at the input and output with an impedance Z in = Zout, the basic two pole filter response results - which is the same as the response of the equivalent half lattice filter terminated in the same impedance. The formula for the derivation of the characteristic impedance of such a filter is:

$Ro = 2 \pi L1 \Delta f$

Δf = half the filter bandwidth

L1 = motional inductance (see equiv. circuit)

Matching the filter with different impedances leads to increased passband ripple - reducing Ro increases ripple, but also steepens the passband slope. Classic filter wave theory leads to

$\frac{R}{Ro}$ = .8, ripple in passband is 0.1dB, whilst the stopband width is three times the -3dB bandwidth.

$\frac{R}{Ro}$ = .5, ripple in passband 0.22dB, whilst stopband width is reduced to 2.8times the -3dB bandwidth

These basic criteria hold for all types of crystal filter, so careful selection of the basic parameters of

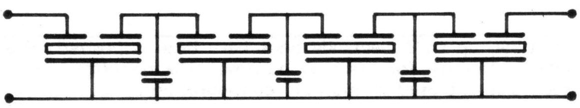
a) termination impedance
b) characteristic frequency

all filter types (Bessel, Gaussian, Chebyshev, Legendre) can be fabricated. Just as a capacitor across the input/output can be used to steepen the slope of the CFM2 series mechanical filter, the same effect is observed in monolithic dual crystal filters - which are in several aspects analogous to the mechanical filter technique of the CFM series.

Single monolithic duals are usually presented in the form of three lead HC18U crystal, and are used widely in dual conversion receivers as a prefilter at 10.7MHz. The effect of a single filter as a bandpass element is not significant, but the narrowing of noise bandwidth assists in design of subsequent amplifier stages. Some of the newer generation ceramic IF filters for 10.7MHz manage to achieve good stopbands (-50dB or so), and so the combined effect of the two is a viable low cost method of further improving the roofing characteristic of the pre-filter.

Multipole monolithics

Although is it theoretically feasible to spread the selectivity of the IF throughout the gain distribution with separately matched interstage filters, this technique is very tedious to design accurately, - and holds no advantages (in most applications) over the more direct approach of using a single multipole filter array. The internal circuit of the TOYO (and most other monolithic 8 pole filters) is :



The input and output of the filter is terminated externally with a parallel resonant tuned circuit. In the case of filters with an internal matching transformer, the internal matching is set to a slightly higher frequency, so that an external trimming C may be used to provide fine adjustment for matching any strays in the surrounding circuit. Filters with internal matching usually present a DC path to ground - and should not be expected to pass more than 5mA or so of DC.

Overtone MCFs are also possible, but the relative bandwidth of the system shrinks by a factor of $1/n^2$ - where n is the overtone mode used.

The loss of the monlithic crystal filter is very small when it is compared to L/C and ceramic techniques, largely due to the enormously higher 'Q' factors of quartz.

Multipole filters (each dual counts as two poles) are intermatched by a shunt capacitance - used to absorb the Co factor of the equivalent electrical circuit for the MCF. They vary inversely with the bandwidth of the filter - until the point is reached where no capacitance is used at all. At 10.7MHz, this about 35kHz, and so wider filters have to use internal inductance to tune out the effects of the electrode capacities. Considering the comment relating to the use of capacitance across the input and output terminations - you will see that even without external coupling, the internal coupling effects will be present.

Normalized to the centre frequency, the minimum bandwidth of a stacked filter is

$$\frac{BW\ min}{fo} = \frac{6n}{QL}$$

Fo is the centre frequency, n the number of resonators and Q is the quality factor of the resonators. L is the insertion loss expressed in dB. Qs from 25,000 to 100,000s are frequently possible in the MCF.

The intrinsic Q for quartz is in fact 16 million - divided by the operating frequency in MHz.

The maximum bandwidth of the filter is given by

$$\frac{BW\ max}{fo} = 0.0035/N^2$$

where N is the overtone order.

By using an inductive termination, the static capacitance may be tuned out, extending the possible bandwidth of the filter to as much as 2.5% of the centre frequency (When using the terminal inductances as resonators.)

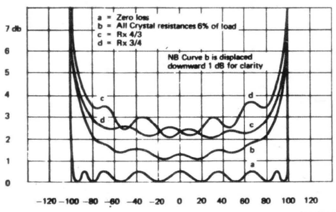
Limits of mistermination

In the ideal world, filters and terminations would be as one. But in practical circuits, a degree of mismatch is inevitable. Just how much mismatch may be judged from analysis of the effects. The mismatch of the filter tends to leave a characteristic asymmetry to the bandpass - which, if you have a sweep generator - presents a useful clue at to what's wrong.

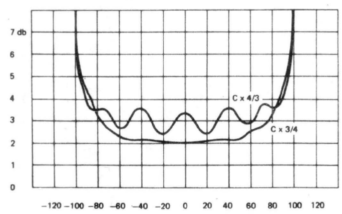
The information is presented in normalized form, that is to say it applies equally for any filter of n poles - that has been designed for a similar ripple characteristic. The same applies to both the resistance and reactance, though for the matching capacitance, the true termination relates to the filter design - eg Chebyshev - 0.5dB ripple - reactance = –1.05 term. resistance For 0.25dB and 1.0dB, the figures are –1.235 and –0.857 respectively. Generally, it is permissable to take R and C as equal.

The effects of mistermination are reduced with increasing numbers of filter poles

Effect of termination error, resistance. n = 8 0.5 dB ripple.

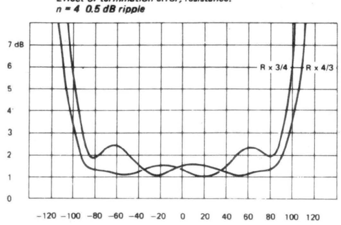


Effect of termination error, capacitance. n = 8 0.5 dB ripple

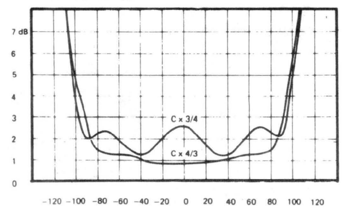


% of 3 dB design bandwidth

Effect of termination error, resistance. n = 4 0.5 dB ripple

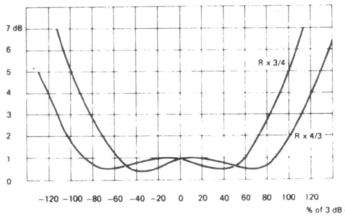


Effect of termination error, capacitance n = 4 0.5 dB ripple

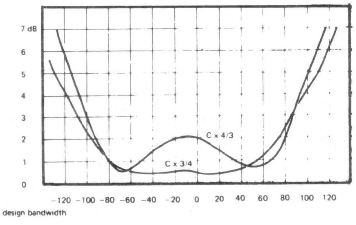


% of 3 dB design bandwidth

Effect of termination error, resistance. n = 2 0.5 dB ripple.



Effect of termination error, capacitance n = 2 0.5 dB ripple

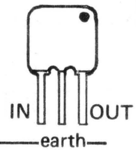


% of 3 dB design bandwidth

ambit INTERNATIONAL

TOKO's new FM ceramic IF filters : 'CFSH'

* Greatly reduced temperature coefficient
* Low insertion loss
* Excellent longterm stability
* Range of bandwidths for mono/stereo



General

The trend towards synthesised car radio in Japan has highlighted the need for low temperature coefficient IF filters - since the tuning remains spot on, the problem of IF filter drift remains. TOKO's CFSH series is designed to provide low drift, coupled with good phase linearity and good spurious rejection compared to existing types.

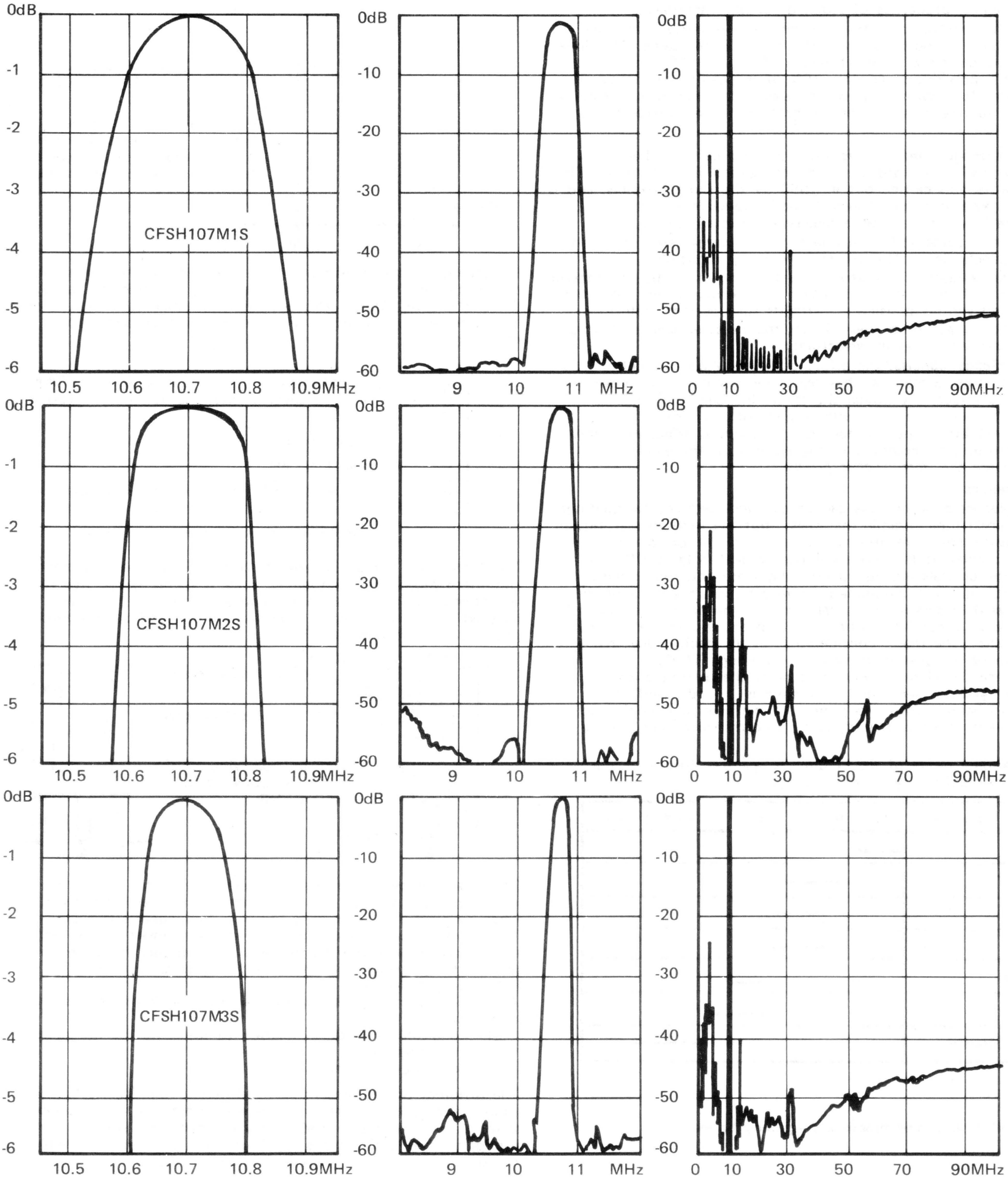
The range is presented in three bandwidths:

280kHz for stereo where maximum bandwidth is permissable

230kHz a minor compromise of quality/bandwidth where the channel spacing is tight

180kHz tight stereo bandwidth (if used in a synthesised tuner, it may be assumed that detuning tolerance is not required). Also switched in to provide narrow IF facilities in some high class tuner systems.

Specifications	M1S	M2S	M3S	
In/out impedance	330	330	330	ohms resistive
Centre frequency	Rank A	10.70	± 30kHz	none
	Rank B	10.67	± 30kHz	blue
	Rank C	10.73	± 30kHz	orange
	Rank D	10.64	± 30kHz	black
	Rank E	10.76	± 30kHz	white
-3dB bandwidth	280	230	180	± 50kHz
-20dB bandwidth	600	550	500	kHz MAX
Temp coefficient	0 ± 50 ppm/°C typ 20			
Insertion loss	6	6	7	dB MAX
Spurii (9-12MHz)	-30	-40	-40	dB MIN
Ripple	0.5	0.5	0.5	dB MAX
Maximum terminal voltage	50	50	50	volts DC
Output capacity (external circuit)	15	15	15	pF MAX
Longterm stability	0.5	0.5	0.5	% / 10 years
Operating temp	-20 to +80°C			



CFSH/ continued — Ceramic filter notes

Effects of incorrect termination impedances:

As a general point, it is worth noting that the effects of incorrect matching on ceramic filters is much the same as for crystal filters. Crystal filters - being made of elements with higher 'Q' tend to respond more drastically to mismatch conditions, but the general shape and nature of the distortions are all derived from the same basic theory.

For those of you seeking a deeper insight into the nature of this type of filter, and with £25 odd to spare, "Modern Filter Theory and Design" by Temes & Mitre (Wiley Interscience - NY) is a mine of useful information. "A" level further maths is really a prerequisite of this particular science - although the empirical approach to the problem is quite enlightening for the purposes of all but actual filter designers.

FM ceramic filters

Murata were probably the first to produce really viable low cost FM ceramic IF filters - as long ago as 1962 in fact, Vernitron may wish to claim some proprietory rights to the general idea, but they really no longer figure as serious volume suppliers in the consumer market. Ceramic filter technology is only a short step from ceramic capacitor technology.

The ceramic filter operates on the piezo electric principle, deriving specific characteristics from a controlled combination of the electric and mechanical properties of selected ceramic materials.

It is smaller, cheaper and more readily mass produced than the crystal filter - although much of the theory is the same. The limit of the practical range of the filters is 10kHz to 100MHz, with a minimum bandwidth of approx. 10% of the centre frequency. The thermal and long term stability of ceramic material does not make it feasible (at the present time) to produce filters outside this scope.

Basics

The piezoelectric effect occurs when a potential is applied a crystal lattice to produce a distortion and polarization in that lattice. The original work was carried out in 1880 by the Curie brothers, and is sometimes known as the "Curie Effect" - [beware of the other Curie discoveries in connection with magnetism and radiation].

Thus stress is converted from one form of energy (electricity) to another (mechanical). The process is entirely reversible.

The actual extreme of the stress effect is known as the spontaneous polarization of the material. It varies in accordance with potential applied, and various physical constraints such as temperature.

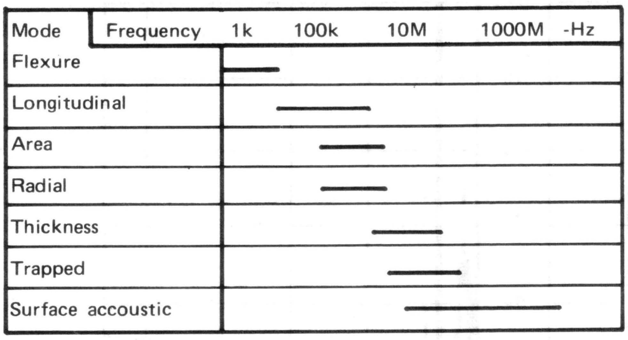
Greek scholars will have guessed that the stress effects of thermal gradients are known as *pyroelectricity.*

To be perfectly concise, the effect occuring when an electric field is applied across a material that reacts mechanically, is known as "Lippman's effect". Although the term "Piezoelectric" is used widely. We haven't finished with terminology yet, since certain materials exhibiting the spontaneous polarization effect can show a reversible effect, according to the polarity of the electric field, and these are the *ferroelectric groups.*

Piezo ceramics

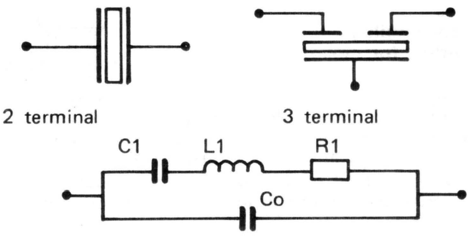
Certain piezoceramic crystals can be *calcined* (reduced by melting) into polycrystalline ceramic materials, but the random distribution of the crystal structure produces a self cancelling piezo effect overall. However, if the process is carried with a large DC field applied, the crystals are pulled into alignment and thus polarized. Barium titanate, lead zirconate-titanate etc form the basis of the material used in ceramic filter fabrication. The process, along with additives etc., is carefully controlled to produce a stable thermal characteristic - the major consideration in this technology that remains to be completely conquered - although TOKO's new FM filters in the CFSH series answer most of the present critisisms. From this, you will see that the ceramic is 'synthesised', as opposed to being grown and ground in the case of crystals - so they are easily mass produced, and stamped into a wide variety of shapes.

The vibrational mode of the ceramic is again analogous to the orientations used in crystal terminology



Vibrational modes, and practical frequency bands

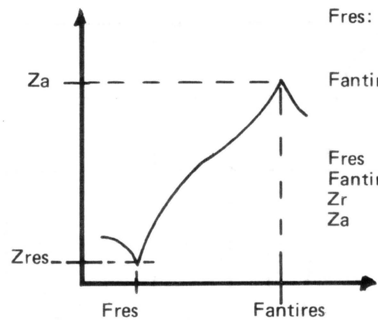
Symbols and equivalent electrical circuit



2 terminal equivalent circuit

C1 = Equivalent compliance
L1 = Equivalent mass
R1 = Equivalent resistance
Co = Parallel equivalent capacity

The impedance/frequency characteristics of the piezoceramic are again the in same form as the quartz crystal:



Fres: $\frac{1}{2\pi . \; L1.C1}$

Fantires = $\frac{1}{2\pi \left[\frac{L1.C1.Co}{Co+C1}\right]}$

Fres = resonant frequency
Fantires = anti-resonance freq
Zr = impedance at resonance
Za = antiresonance impedance

Depending on the deposition of the electrodes on the material, one of the various modes of vibration may be selected. Taking the example of TOKO's CFM2 series of mechanical filters, it may be illustrated in association with the principle of mechanical coupling:

Longitudinal vibration is used at 455-470kHz, and the specific resonant frequency (Fres) is given by

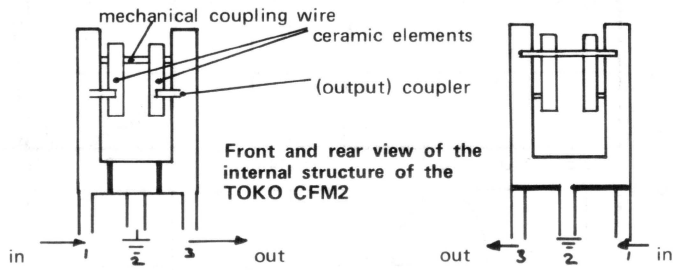
Fres = $n/2L\sqrt{E/\rho}$

n is 1,2,3 etc

the term $\sqrt{E/\rho}$ is the transmission speed of vibration for a given material

(At 455kHz L = 3.8mm in the CFM2)

The CFM2 is a unique system of coupling in low cost ceramic-cum-mechanical filter technology, since it comprises of two of the resonators described in the above analysis, mechanically coupled at resonance, using a carefully chosen and dimensioned phosphor bronze wire coupler.



Front and rear view of the internal structure of the TOKO CFM2

The filter operates in an easily understood manner - since the first resonator will be seen to commence mechanical flexure when an electrical signal at is resonant frequency is applied. This vibration is mechanically coupled (as opposed to the more usual capacitor method) to the output resonator, which then reconverts the energy into an electrical signal at the specific impedance of the filter.

As in all piezo filters, this gives rise to certain possible spurious resonances:

1 Longitudinal mode in relation to the width of the element
2 Thickness mode
3 Overtone operation

The different modes present vastly different impedances to the circuit, and so correct termination will effectively keep unwanted spurii to a minimum. Preceding any ceramic filter stage with a bandpass tuned L/C filter is really essential to direct the element to operate ONLY on its design frequency mode.

Ceramic filter notes/.....continued

FM filters

Bearing in mind the table of mode of operation versus the frequency, it seems fair to assume that 10.7MHz filters will be operating in the trapped mode, or the thickness shear mode. This mode of operation is analyzed in reasonably easily understood terms:

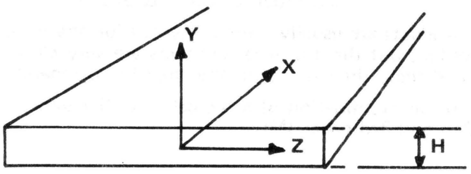


Figure a Section of piezo ceramic plate

Considered the following case:
Polarization is in the X axis
Y axis is open
Waves in the Z axis are then displaced by a factor "U" in the X axis, where U is proportional to

$$\cos\left[\frac{n.\pi.Y}{H}\right] \exp j(\omega t \pm \beta.z) \quad .. \ 1$$

Where H = Piezo medium thickness
ω = Angular frequency of the propagating wave
β = Propagation constant in the Z axis

$$= \frac{n.\pi}{H}\sqrt{\left(\frac{f}{n.f'o}\right)^2 - 1} \quad .. \ 2$$

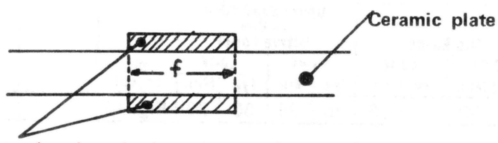
Where fo' is the fundamental frequency of the mode of vibration (thickness shear) given an infinite width
n is the harmonic number for the plate

The when n=1

$$\beta = \frac{\pi}{H}\sqrt{\left(\frac{f}{fo'}\right)^2 - 1} \quad .. \ 3$$

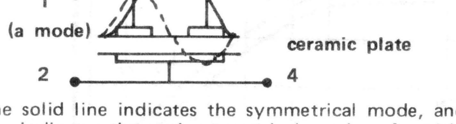
β is a real number if f > fo'

Where f > fo', then β is a pure imaginary number, and fo' becomes the cut-off frequency for waves that are propagating along the Z Axis.
ie Waves above frequency fo' are freely propagated, whilst below fo', the amplitude is attenuated exponentially.

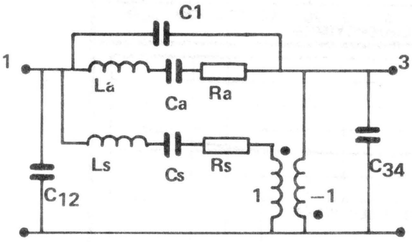


With electrodes deposited as shown, the area between has a lower frequency (fo) than fo'. This is attributable to the mass load effect on the piezo effect - leading to the fact that the wave energy between the elctrodes *does not propagate outside this area* - because the cut off frequency (Fo') adjacent to the electrodes, is higher than f. Thus the frequency f, which is between fo and Fo' is trapped in the area between the electrodes leading to the resonator technique used most widely for the 10.7MHz series of ceramic IF filters.

This theory is readily moved on to considerations surrounding the multielement filter. Different resonances are trapped under each electrode pair, The distribution of the two vibrations is illustrated

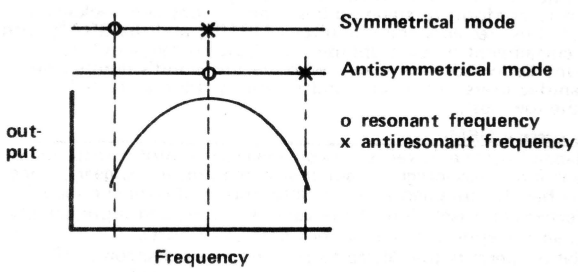


The solid line indicates the symmetrical mode, and the broken line indicates the anti-symmetrical mode. As with monolithic quartz filter units, this uniwafer filter is analyzed:



Equivalent circuit of the multicoupling mode

The passband resulting from the uniwafer construction:

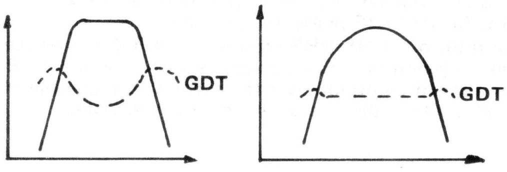


Group delay in FM filters

Without going into detail, the group delay of the filter is one of the features determining factors such as stereo separation and distortion at the edges of the passband. It may be quantified in the term "Group Delay" (GDT). This is the error of phase linearity as the passband curve is plotted for a given filter.

$$TD = \frac{d\phi}{d\omega} \quad .. \ 4$$

Basic filter shapes are



Butterworth Gaussian

Butterworth filters have flat tops, and bad GDT, and thus are not ideal FM filters. The Gaussian filter corresponds to a half sinewave in the areas of passband interest, and thus may be shown to possess an excellent GDT.
These characteristics lead to certain features which are used in conjunction with some applications. The Flat top Butterworth filter shows little bandwidth change with changing input levels, but the rounded Gaussian shape means that the bandwidth becomes effectively narrower with decreasing input amplitude. In communications enviroments, the Butterworth filter is more widely used (where HF distortion is not a problem). In low distortion HiFi FM, the gaussian filter is essential - although it may effectively impair the sensitivity compared to the Butterworth form.

A brief word on SAWs

The nature of the surface accoustic wave filter is another complete section in itself, but is should be noted here that the SAW is fabricated in such a way that group delay and amplitude characteristics may be individually optimized. However, they still suffer enormous (20dB) insertion losses, and overall do not justify their additional cost (yet.).

CFSE/CFSH series filters

CFSE filters are characterized for stereo reception, thus using the basic Gaussian characteristic, the GDT is good, but the bandwidth is not as tight as some applications might require. In the UK, it is rare that narrower filtering is needed, and it is perhaps better .to provide a narrow selectivity option by switching. The new CSFH 280kHz bandwidth filter tends very slightly towards a Butterworth characteristic, and offers generally tighter selectivity. As mentioned earlier, the need for a broad tolerance towards detuning is getter less with the onset of the synthesiser, which guarantees exact tuning conditions.
Murata SFE10.7ML are the closest approach yet to a Gaussian linear phase filter in low cost ceramic technology, but they are rather wide in some applications, and require additional gain to overcome some extra 3dB or so loss. (per unit).

Mismatch conditions

In any AC electric theory, the maximum power theorum is a centrepiece of mathematical elegance. What it boils down to is that for the best transfer of AC energy from one system to another, the load impedance of one stage, must be matched by the source impedance of the next. Failure to observe this results in power losses in the system, and reflection of energy. The reflected energy front interacts with the forward energy, and the result is interference effects that cause:

1 increased passband ripple
2 additional insertion loss
3 distorted GDT

The characteristic "fingerprint" of certain mismatch conditions for a given filter configuration can lead to easy analysis of the impedance problem. TOKO CFSH filters, in common with most other filters, are specified with regard to reactive and resistive impedance errors. (1 data sheet per device) This information will enable problems to be tracked down very readily - as long as a wobulator/ spectrum analyser is available to use.

SBL1 - MD108 etc : Low cost DBMs

General

Low cost hot carrier diode mixers are amongst the few components that seem to conform to standard base connections and packaging. The SBL1 thus replaces the ubiquitous MD108, and the SRA1, with a minor curtailment of LF response. The SBL1 is made by the Minicircuits Laboratory company, who are the world's number one DBM manufacturers - so quality and reliability are maintained despite the low cost.

Why use passive DBMs

The wideband passive mixer is almost universal in professional radio equipment (and increasingly in amateur communications gear), since the prime benefit obtained is the greatly improved dynamic range. They offerexcellent isolation of the various inputs, and a predictably flat frequency response at the various entry/exit ports.
This isolation permits the device to be used as an electronic RF switch or attenuator - however applications of this nature are not the prime purpose of the DBM, so the information provided here relates generally to the use in radio receiver mixer stages.
The first point to note is that the device is a 'standard level' unit, requiring a +7dBm LO drive. This sort of power is not the sort of drive you get from a single transistor oscillator - and if you try, the thermal drift as it warms up will be hopeless. Amplifiers on the LO output are possible, but not easy or very satisfactory where a broad range of frequency coverage is called for. The best solution is to use a synthesised local oscillator, where the loop feedback stabilizes the LO over the entire range of operation. In fact, since the DBM IF is DC coupled, the DBM itself makes an excellent phase detector for HF/VHF phase locked loops.
Placed at the front of an HF/VHF receiver, the DBM will reduce the 3rd order IM products brought about by high level inputs. But since there is a lot of confusion over the exact nature of what is a 3rd order IM product, the rest of this note will explain:

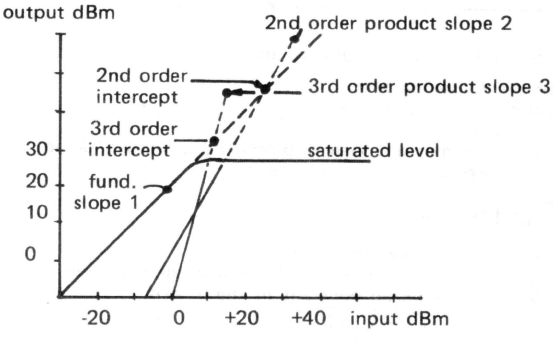
3rd Order IM intercept

Intercept points are terms used to define the performance of RF systems, by defining the relationship existing between a pair of fundamental signals, their second harmonics, and the IM products.

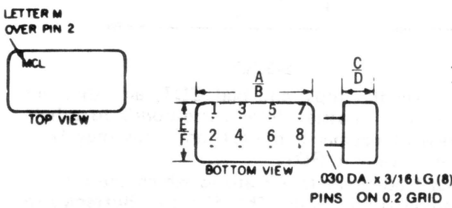
Fundamental F1 and F2 — 2nd order F1±F2, 2F1, & 2F2
3rd order 2F1±F2, & 2F2±F1

The 2nd order products are usually a long way outside the band-pass of the amplifier, but the 3rd order products are very close in to F1 and F2, and these then appear in the amplifier passband.

This then leads to an examination of a typical amplifier under conditions of 2nd and 3rd order IM:



(continued overpage)

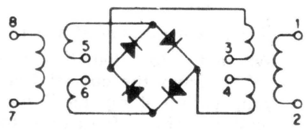


	A	B	C	D	E	F
INCHES	.770	.800	.305	.285	.370	.400
MM	19.5	20.3	7.7	7.2	9.3	10.1

PIN LAYOUTS

	Fig. 16
LO	8
RF	1
IF	3, 4
Ground	2, 5, 6, 7

NOTE:
PINS 3 AND 4 MUST BE CONNECTED TOGETHER.

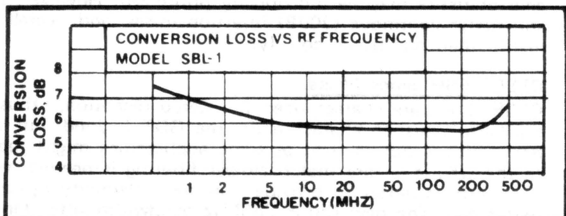


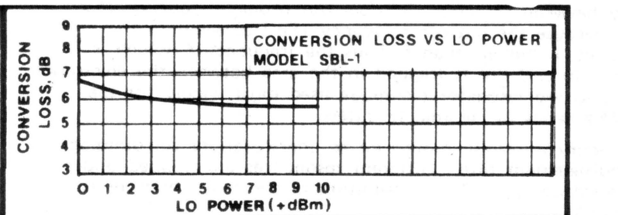
SPECIFICATIONS Series SBL-1

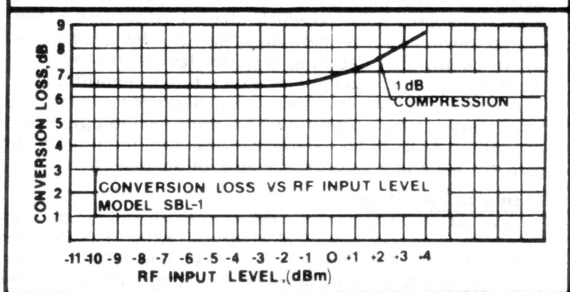
Model No.	Frequency Range MHz			Conversion Loss, dB				Isolation, dB												Pin Layout
				One Octave From Band Edge		Total Range		Lower Band Edge To One Decade Higher				Mid Range				Upper Band Edge To One Octave Lower				
								LO-RF		LO-IF		LO-RF		LO-IF		LO-RF		LO-IF		
	LO	RF	IF	Typ.	Max.	Typ.	Max.	Typ.	Min.	Typ.	Min.	Typ.	Min.	Typ.	Min.	Typ.	Min.	Typ.	Min.	
SBL-1	1-500	1-500	DC-500	5.5	7.5	6.5	8.5	50	35	45	30	45	30	40	25	35	25	30	20	Fig. 16

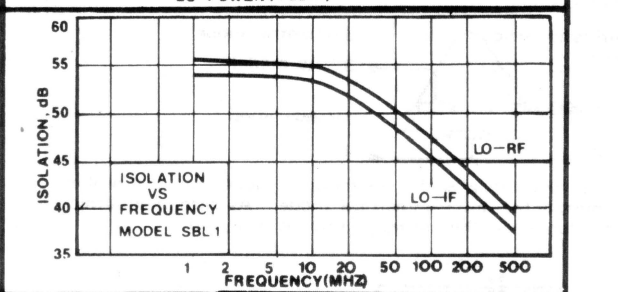
COMMON SPECIFICATIONS

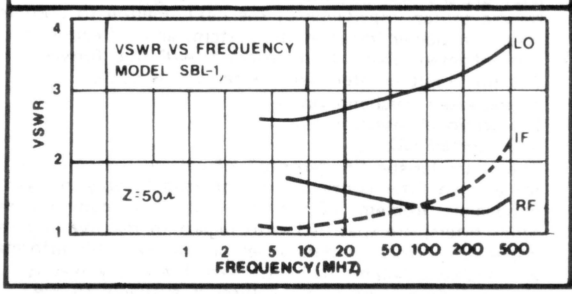
Impedance, all ports, 50 ohms. Input power: 50 mW. Peak IF current: 40 mA. Operating and Storage Temperature: −20°C to +85°C. Pin Temperature (10 sec): 510°F.

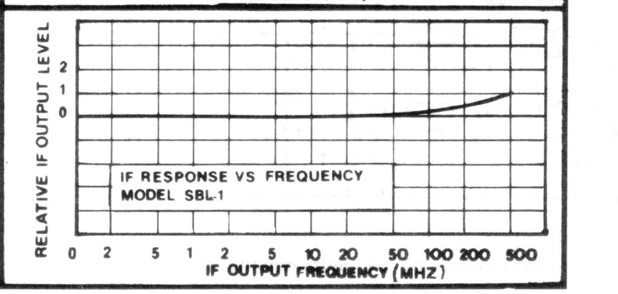






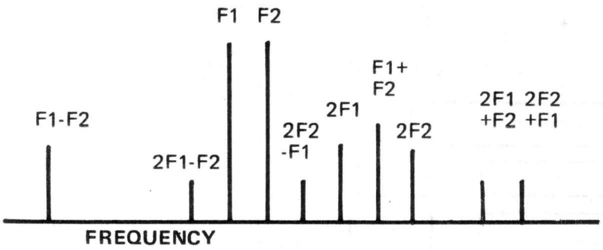






Using DBMS : IM intercepts

Typical intermodulation products:



In the linear region of an amplifier, the plot of input v. output has a slope of 1 (by definition). The slope of the 2nd order products will be 2, and the third order products will be 3. From the graph on the previous page, you will see that the intercept point is defined as the output level where the extensions of the linear portion of the respective plots will intersect with the extension of the fundamental plot.
These products are derived mathematically by extracting the appropriate terms from a power series expansion of the collector current with 2 fundamental signals present. Thus the 2nd and 3rd order intercepts ought theoretically to occur at the same point. However, power series is an approximation of the actual circuit and its behaviour, so this case is rarely met in practise. Also, push pull designs will cancel even order products.
Thus armed with information of the nature of the 3rd order IM, a designer can calculate the worst case spurious levels.

The output fundamental levels of two equal input signals are related to the output spurii by

2nd order IM2 = 1P2 - P
3rd order IM3 = 2(1P3 -P)

Where P is the output level in dBm of the two fundamental signals.
IP is the respective intercept point in dBm
IM is the respective spurious level in dB down from the fundamental signals.

EXAMPLE
IP2 = 30dB, IP3 = 20dB
determine the relative spurious levels of two fundamentals with equal -10dBm output level:
IM2 = 30 -(-10) = 40dB
IM3 = 2[20-(-10)] = 60dB (down from fundamental)

The real world seldom presents two equal signals for the problem, so the following rule is used to convert:

Take the larger of the two signals, and subtract it from one third of the difference between the two signals. The result is an amplitude that essentially equalizes the two signals.
The worst case third order spurious level at this new equivalent fundamental level will thus be the same as for the two original unequal values.

Back to the DBM.........

The output of the DBM is proportional to the cube of the input voltage. So if two equal input signals, F1 and F2 were to be applied, the output would be proportional to $F1^3$. So changing F1 by 10dB, changes the output by 30dB. This means that there is a 2:1 improvement in the 3rd order response as the inputs are reduced - ie there is a net improvement of 20dB.

Comparing performance on the basis of this approach may be slightly confusing - since a -100dB m two tone 3rd order response, from a -30dBm input, is the same as saying a -70dBm response for an input of -20dBm.

Using the 3rd order intercept concept again, the mixer may be evaluated, since at 0dBm input, the 3rd order IM product occurs at -20dBm. So the point at which the 3rd order product equals the fundamental output will occur at about +10dBm input (10mW of signal). Thus the intercept for the SBL1 occurs at +10dBm. Since many DBM data sheets do not include details of the 3rd order IM, an approximate result may be obtained using the 1dB compression point. (the reference point chosen to indicate the linear performance of the mixer is collapsing). The IM point is then about 10-15dB above the 1dB compression point - with the 15dB point being at the lower end of the mixer range (in the case of the SBL1, it is in the region below 100MHz).

10dBm is a lot of signal in many simple receiver applications, but with the power levels used in modern amateur and also commercial transmitters, coupled with the desire to have high gain antenna systems on the receiver input, it is not inconceivable that it may be encountered in HF these days. The main danger is then beginning to be directed toward the costly (at least, it should be) first filter. By the time the mixer output reaches 10mW, a PIN AGC attenuator is well able to clamp the input.

Dynamic range

The main point of using the DBM is to achieve better dynamic range, by which is meant the range of signal levels that can be handled in a given amplification system (be it radio or whatever). The lower end of the receiver dynamic range is defined as the noise floor of the system, at which input level, the S/N ratio is unity. The laws of the universe have defined the practical limit of the floor as being:

-114dBm/MHz + 10log Bandwidth +NF (MDS)

Where -144dBm/MHz is the KTB noise power of source impedance
NF is the amplifier noise figure in dB
BW is the noise bandwidth in MHz, either of the amplifier itself, or some external device such as a filter..

The top end of the range of an amplifier is the previously discussed 1dB compression point, where

Power in = Power out — Gain (dB) +1dB

Thus we have an expression for the dynamic range, using the MDS term above (minimum discernible signal):
DR = Pin — MDS (dB)

Taking the example of the Telefunken 1-900MHz broadband amplifier using the low cost BFT95 transistors —
G = 23dB
NF = 4dB
Po = approx -15dBm
The MDS is then = -114 + 10log900 +4 = -107dBm, and the DR = -15 -(-107) = 92dB.

A further extension here is to consider the spurious free dynamic range, where the IM products are kept below the lower limit (MDS) of the amplifier. This factor is referred to as the DRS, and for two signals of equal amplitude:

DRS = 2/3 (IP3in — MDS)

Where IP3in = (IP3 - G), the third order intercept point as related to the input of the amplifier.
The IP3 of the TFK amplifier is not stated, but works ot to be about 10dB in practise, so

IP3in = -20dBm, and the DRS = 2/3[-20-(-107)] = 84.5dB

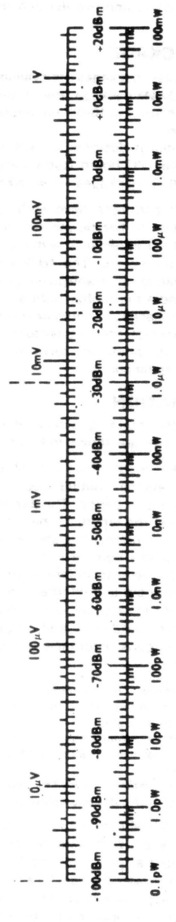
So, the maximum input signal for which the 3rd order IM products will equal the MDS becomes

(-107 + 84.5)dB = -22.5dBm - which means that from the noise floor to input signals of -22.5dBm, the amplifier output will be free from spurious responses that might otherwise distort the transition of signal information through the system.

The chart alongside relates the terms of
volts
dBm
microwatts
in a 50 ohm system

The increasing use of the dBm in specification standards has lead to some confusion over the various relationships

Whilst reference to ohms law leads to the answer, a quick look-up table is far quicker. The ranges covered here are basically those encountered in radio receiver work



The term "dBu" is becoming widely used in international specification standards.
This relates to dB with respect to 1 microvolt and should thus not be confused with dBuW.
0dBu = 1uV
60dBu = 1mV

(for "u" read μ - the typesetting gets to be too tedious when the typefaces have to be changed too often !)

SERVO AMPLIFIER 544

TYPICAL CONNECTION OF NE544AA FOR LINEAR ONE SHOT TIMING

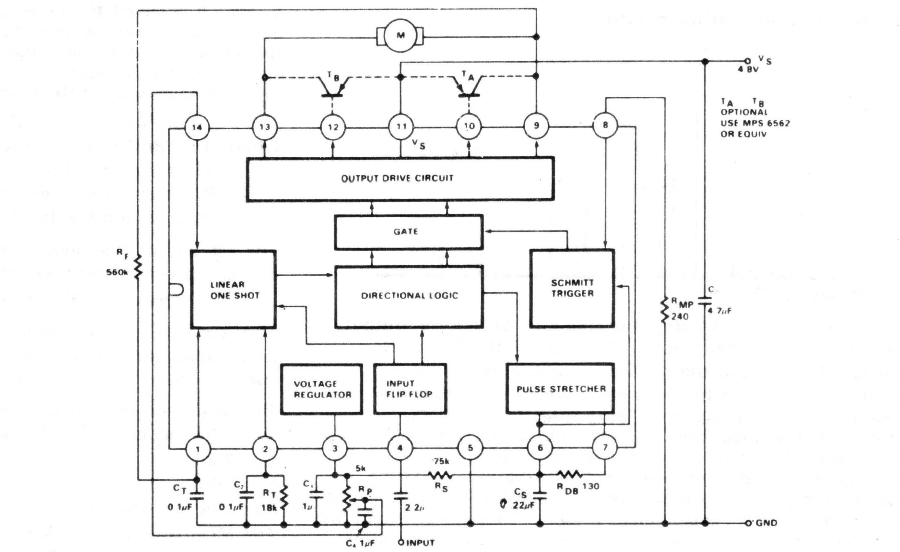


Figure 2

DESCRIPTION

The NE544 is a new servo amplifier design for digital proportional RC systems which incorporates the latest state-of-the-art in integrated circuit technology. The basic systems concept was developed in close cooperation with a number of leading manufacturers of radio control equipment.

The design philosophy behind the NE544 was to provide the RC servo systems designer with maximum flexibility in adapting the amplifier performance characteristic to his particular servo system and at the same time to keep the external components count low. To achieve this goal, all the basic servo amplifier functions, such as motor drive, deadband and minimum output pulse, are integrated into the IC, but can be modified over a wide range by using external transistors or padding resistors respectively. This makes it possible to use the IC for extremely low cost applications as well as for the most sophisticated RC servo systems. Additional features of the circuit are very low standby power drain (typically less than 6mA), an internal voltage regulator for improved power supply rejection and a highly accurate monostable multivibrator. This circuit may be used in 2 different charging modes: linear and exponential. In the linear charging mode, the internally generated charging current is programmable over a wide range with a resistor to ground. Usable currents range from below 10μA to above 1mA. In the exponential charging mode, the internal current source is simply bypassed with an external resistor from pin 1 to the regulator output.

The bidirectional power output stage can supply load currents up to 500mA (NE544N and NE644N package only). Output drive pins for external PNP transistor provide the user with the option of increasing the motor drive by bypassing the internal compound PNP transistors.

The NE544 also provides external pins to adjust deadband and to vary the hysteresis of the Schmitt trigger. This gives the user maximum flexibility in adapting the servo amplifier to a large variety of servo motor and gear train combinations. A dynamic brake integrated into the output stage serves to suppress inductive noise spikes and helps to improve the dynamic performance.

IC PACKAGE

Three basic packages are available: the 14-pin NE544N package, the 16-pin NE644N package and the 16-pin miniature NE644W package.

The pinout is designed in such a way that very small PC board layouts are possible with all 3 packages. The NE544N and NE644N packages have sufficient power dissipation to handle motors with a minimum of 8Ω impedance with the integrated power transistors. The NE644N package has separate signal and power grounds, and can handle the higher output power. The NE644W package should be used primarily with external PNP transistors and is intended for miniature servo and high accuracy applications.

OPERATION

The basic building blocks of the NE544 servo driver are shown in Figure 1.

A positive input signal applied to the input pin (4) sets the input flip-flop and starts the one shot time period. The directional logic compares the length of the input pulse to that of the internal one shot and stores the result of this comparison in a directional flip-flop. The exact difference in pulse width between input and internal one shot pulse, called the error pulse, is also fed to a pulse stretcher, deadband and trigger circuit. These circuits determine 3 important parameters:

1. *Deadband*—The minimum difference between input pulse and internally generated pulse to turn on the output
2. *Minimum output pulse*—The smallest output pulse that can be generated from the trigger circuit
3. *Pulse stretcher gain*—The relationship between error pulse and output pulse

Proper adjustment of these parameters can be achieved with external resistors and capacitors at pins 6, 7 and 8. The trigger circuit activates the gate for a precise length of time to provide drive to the bridge output circuitry in proportion to the length of the error pulse.

DC ELECTRICAL CHARACTERISTICS $T_A = 25°C$, $V_S = 4.8V$, Figure 1 unless otherwise noted

	PARAMETER	TEST CONDITIONS	LIMITS Min	Typ	Max	UNIT
V_{CC}	Supply voltage		3.2	4.8	6	V
I_{CC}	Supply current	Quiescent	4.2	5.5	7.3	mA
V_{TH}	Input threshold					V
	On			1.5		
	Off			1.4		
Z_{IN}	Input resistance			2.2		kΩ
	Output voltage					V
V_{OL}	Low	Pin 9 or 13, I_L = 400mA		0.3		
V_{OH}	High			3.9		
V_R	Reference voltage		2.4	2.5	2.7	V
PSRR	Power supply rejection	$3.5V \leqslant V_S \leqslant 6V$		.01		%/V
	Minimum dead band	RDB = 0		1		μs
•	One shot temperature coefficient			.01		%/°C
	Standby power			27		mW
	PNP drive current	Pin 10 and 12		20		mA

Mini servo driver PCB type 9544N now available from Ambit

TYPICAL APPLICATION AS A LINEAR SERVO AMPLIFIER

Figure 2 shows a typical connection of the NE544 as a high performance servo amplifier for remote control servo applications using the 14-pin dual in-line package. The input pulse may be dc coupled if a reset is used in the receiver decoder. Output drive to the servo motor is applied through pins 9 and 13 with PNP transistors T_A and T_B optional for high performance applications. The wiper of potentiometer R_P is mechanically coupled to the servo control surface providing positional feedback. The internal one shot in this application is operating in the linear charging code.

LINEAR ONE SHOT TIMING

In contrast to most conventional servo drivers which use exponential one shots, the NE544 uses a linear one shot. This makes it possible to design servo systems with very high positional accuracy and linear pulse width to position transfer fuctions. The timing of the linear one shot can best be explained with the help of Figure 3.

The timing cycle starts after the input pulse sets the input flip-flop and releases the reset transistor T_R. This allows current I_C to charge up capacitor C_T in a linear fashion. Current I_R is programmed by resistor R_T. The op amp serves as a linear voltage to current converter, with the current through R_T and C_T matched identically. The inverting input of the op amp is internally referenced to 1.8 volts so that the current I_R is given by this equation.

Equation 1

$$I_R = \frac{V_I}{R_T} = \frac{1.8V}{R_T}$$

The timing period of the internal one shot is complete when the voltage ramp at pin 1 reaches the threshold set at pin 14. This time is given by this equation.

Equation 2

$$T = \frac{C_T V_{14}}{I_R}$$

If we substitute the typical values given in Figure 2 we obtain this equation.

Equation 3

$$T = \frac{(0.1 \times 10^{-6}F)\ (1.5V)}{0.1 \times 10^{-3}A} = 1.5 \times 10^{-3}sec$$

When the internal one shot has timed out, the input flip-flop is reset. The reset transistor T_R is clamped to ground as soon as the input pulse goes to zero. Figure 4 shows the relationship of the input pulse, the internal one shot pulse, the ramp at pin 1 and the error pulse for a condition where the input pulse is longer than the internal pulse.

In contrast to most conventional designs, the total value of the feedback pot R_P is no longer important, since it serves only as a voltage divider. A reasonable lower limit is 1.5kΩ to keep power consumption low and to prevent loading of the voltage regulator. In the typical application a 5K pot is used.

SIMPLIFIED CIRCUIT DIAGRAM OF LINEAR ONE SHOT

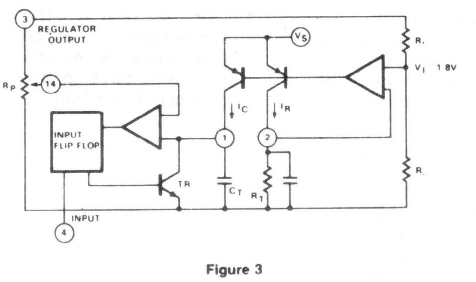
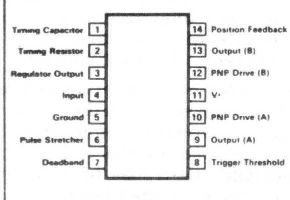


Figure 3

N PACKAGE



Order part no. NE544N

NE5044 3 to 7 channel parallel to serial encoder

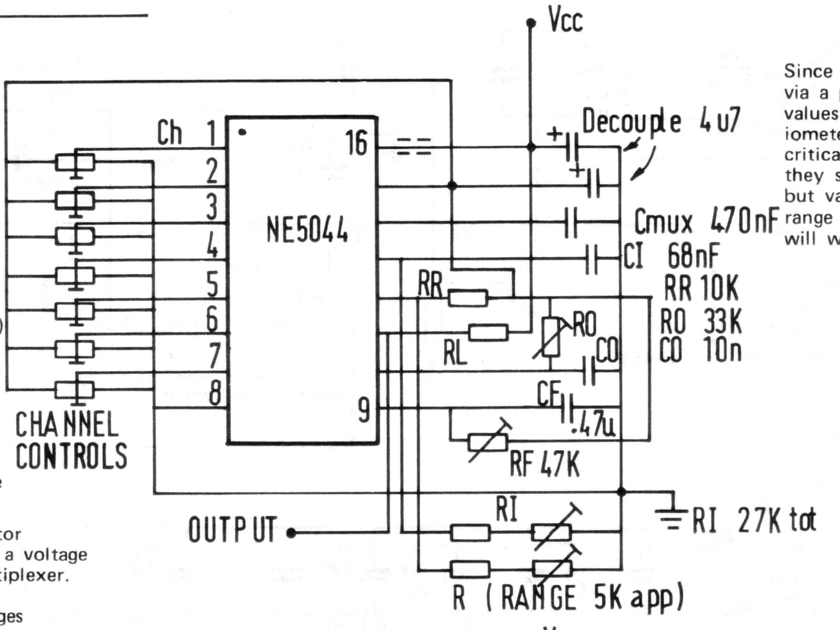
*Ideal RC encoder
*Linearity better than 0.3%
*Internal voltage regulator for the input range 6-16v
*Potentiometer encoding for easy mixing and versatile encoding

* 18 pages of data and applications available

CMOS encoders have been 'standard' in radio control for some time past, but the new NE5044 offers some very appreciable advantages in terms of simplicity, versatility and flexibility.
The use of potentiometer connected control inputs means that the input is easily decoupled from RF, and is far less sensitive to control noise, than with most conventional encoders.

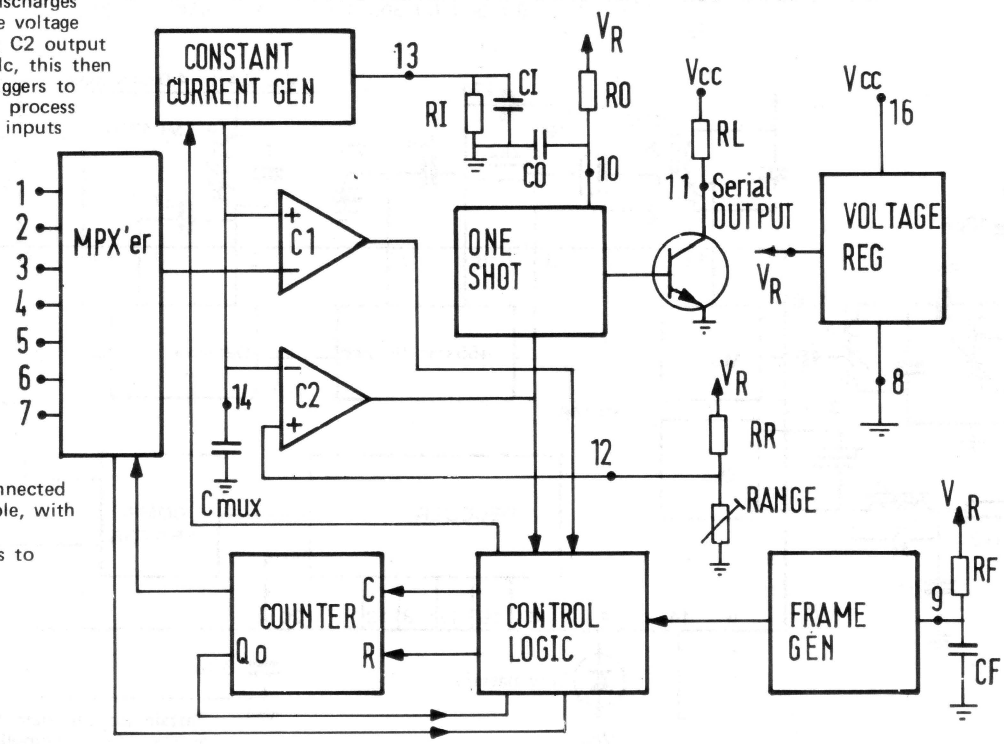
Operation

Each input to the multiplexer is fed to C1 as a voltage in a stobed sequence. When the input is inactive, it appears as an open circuit - and when it is being read, it still presents greater than 1M ohm to the control.
Channels 4,5,6,&7 may be used to select the total number of channels per frame, by simply grounding the channel input which 1 greater than the total required. (ie ground ch.4 to achieve a three channel encoder)
The constant current generator alternately charges and discharges Cmux to provide a very linear ramp for C1 and C2 comparators.
C1 and C2 then compare the voltage on Cmux with the range input voltage.



Since the encoding is via a potential, the values of the potentiometers is not very critical. Obviously, they should be linear, but values in the range 5k to 100K will work.

When Ic is positive - the capacitor charges linearly, until it reaches a voltage equal to the output of the multiplexer. C1 now goes high, reversing the direction of Ic as Cmux discharges linearly until it reaches the voltage on pin 12 (V range). Now C2 output goes high, again reversing Ic, this then clocks the counter, and triggers to output one shot. Thus the process repeats for the number of inputs selected.
When Cmux reaches the final positive peak in the frame, it reverses and discharges to Vrange.
The counter reaches the point where Qo is high when V Cmux = Vrange
Cmux then charges again, but since Qo is high, the output of C1 is ignored.
The voltages continues to charge to Vclamp, and stays there until a pulse from the frame generator is received.
The frame generator is connected in this diagram as an astable, with period 0.66 x RFCF.
The frame pulse thus resets to the start of the sequence.



Specifications	MIN	TYP	MAX	UNIT
Supply voltage	6		16	V
Supply current (Vcc 10v)		11	15	mA
Regulator output	4.5	5.0	5.5	V
Regulator current output			-20	mA
Regulation against supply		.005	.02	V/V
Multiplexer input current		50	200	nA
Input voltage range	1.5		5	V
Channel crosstalk		1	5	uSec
Output pulse				
Position	1350	1500	1650	uSec
Position linearity error		5		uSec
Pos. TC		.15		uSec/^{o}C
Pos. PSR		.5	1	uS/V
Width RoCo = 300uSec	255	300	345	uSec
Saturation voltage Io =25mA		.5	1	V
Frame time with RFCF30mS	17	20	23	mS

The above measured at 25^{o}C and Vcc 10v

The device features internal clamping, that limits the range of control voltage inputs to 1.5 - 3.5v, thus preventing interaction between controls if one channel should happen to go open circuit.

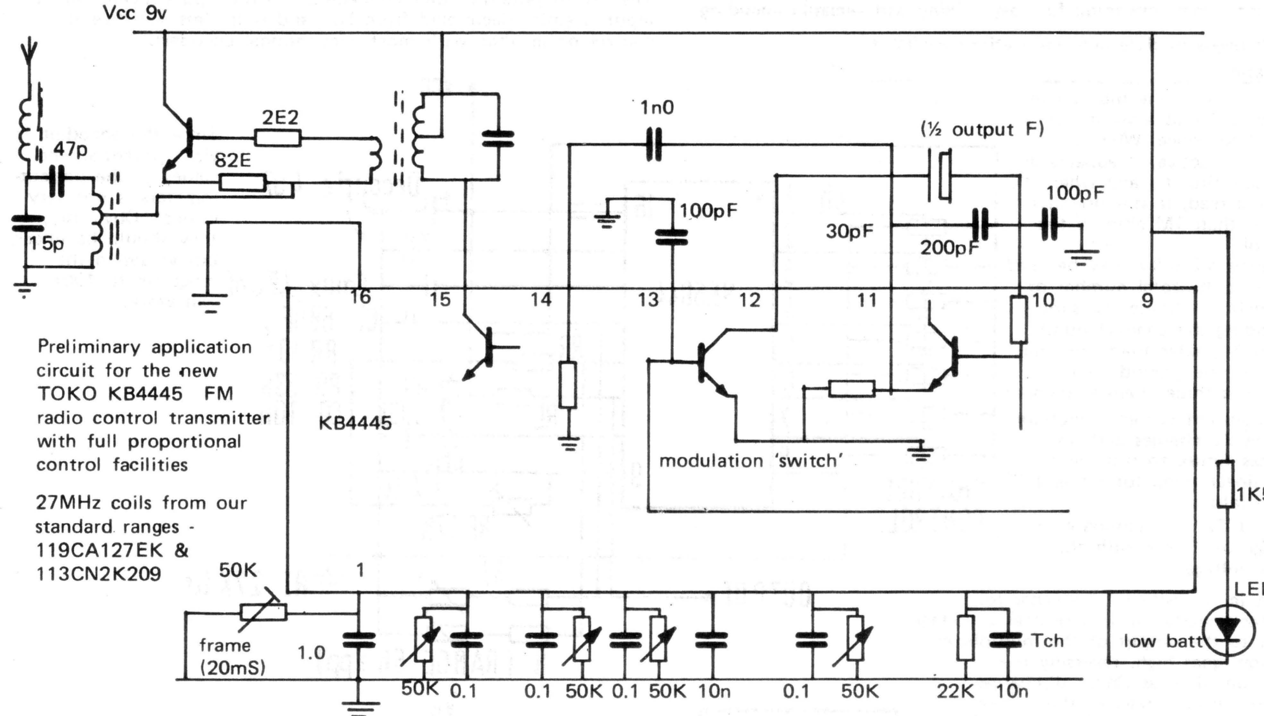
Due to the very versatile nature of the NE5044, design work should be carried out with reference to the manufacturers data.
The values on the top circuit diagram are for the usual 1.5mSec neutral system used in enthusiast RC systems.

The encoder is reasonably proof against the effects of RFI, and it should be noted that placing any semiconductor in close proximity to intense RF fields is likely to result in problems. Ferrite bead decoupling (FX1115 etc) will be an effective deterrent in most instances, - particularly on the leads from the various control sticks/switches. Intelligent decoupling with ceramic capacitors is essential - but always remember that the effects of the RF field will be dramatically changed according to the antenna loading of the average RC transmitter, Check the effects on channel stability with the antenna at all positions.

KB4445 and KB4446 FM digital proportional Radio Control ICs

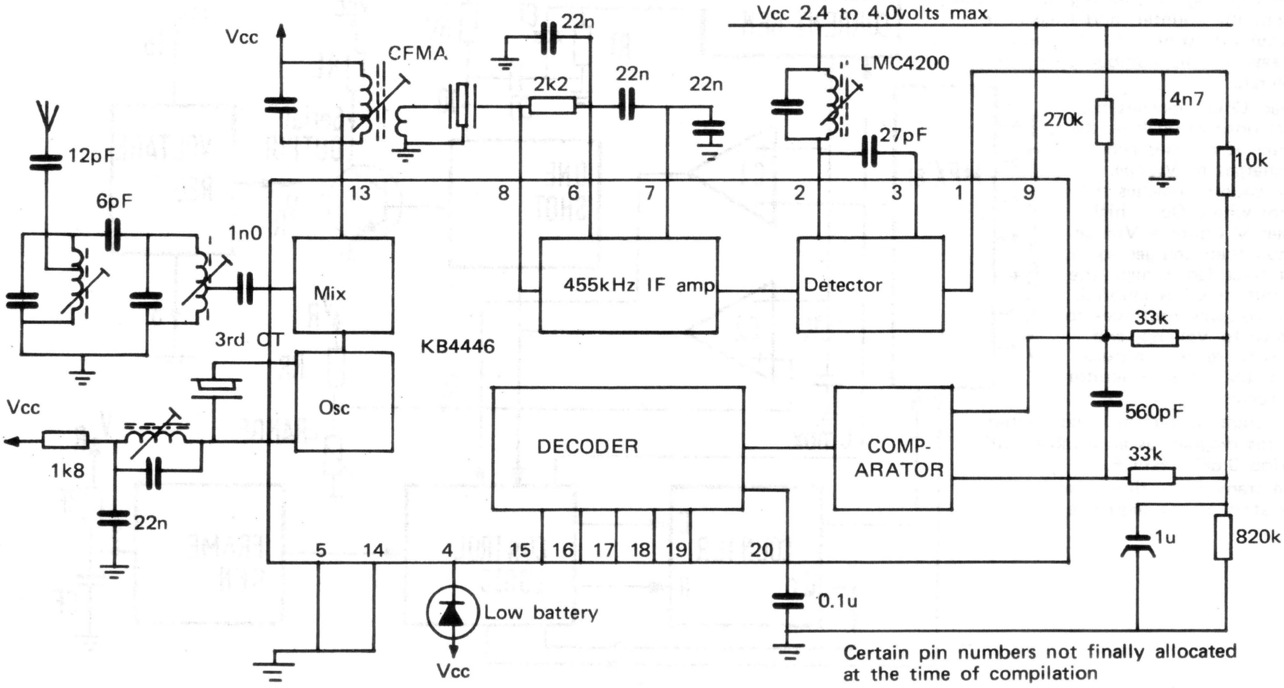
Preliminary details

- ***Complete RF/Encoder/decoder systems**
- ***Low voltage operation with battery condition indicator**
- ***All the attributes of the FM system at VLF to 50MHz**



Preliminary application circuit for the new TOKO KB4445 FM radio control transmitter with full proportional control facilities

27MHz coils from our standard ranges - 119CA127EK & 113CN2K209



Certain pin numbers not finally allocated at the time of compilation

Notes

The details given here are preliminary information, intended to preview these interesting new developments by TOKO.

The KB4445 is a 4 channel digital proportional radio control transmitter IC, with all the necessary encode logic combined with the crystal oscillator modulator and output driver stages. For low power applications, the internal output stage may be used on its own - or for higher power systems, it may be used to drive an external amplifier stage.

The modulator on the KB4445 uses a transistor that switches a fixed (or variable) 30pF capacitor across a crystal in a fundamental oscillator mode. Thus pin 12 may also be used as a source of the encoded signal for other purposes.

Low battery indication is provided by a flashing LED when the voltage drops below 4.4v.

The KB4446 receiver, may be used in conjunction with the KB4445 or any other similarly encoded FM transmission. It decodes up to 5 channels to drive standard servos - and features very low operating voltage, noise immunity through use of a comparator circuit, and hardly any external components.

Preliminary specifications (full specs by Feb 80)

Parameter	KB4445	KB4446
Supply voltage	4 to 12v	2 to 4.0v
Supply current	20-30mA	12-15mA
Channels	4 prop	5 prop
Channel pulse w	1.5mS (nom)	
Servo drive		2mA
RX sensitivity		26dBu for 20dBS/N
Deviation	adjustable	2-4kHz typ
Frequency range	8-50MHz (for xtal operation)	20kHz-50MHz Xtal : 8-50MHz

The RX sections of the KB4445 are very similar in most respects to the MC3357P - yeilding almost identical specs over the common operating ranges. The crystal oscillator is specifically a 3rd OT type.

For IR and US operation, the IF may be used directly in a TRF mode.

Samples of these devices are expected to be freely available by February 1980. Meantime we regret we can only service OEM enquiries for further information and demonstrations.

MOVING COIL LINEAR METERS

ML52 series — 2% typ linearity

Previous issues of the AMBIT catalogue have detailed many different low cost 'indicator' meters. The ML52 series are offered as LINEAR movement types, in a wide range of stock FSD readings.

Range	Divisions	Int.resistance
0-100uA	50	1k4 ohms
0-1A	50	100 mohms
0-3A	30	30 mohms
0-10A	50	10 mohms
0-15V	50	1k per volt
0-30V	30	1k per volt
0-100V	50	1k per volt
0-1mA	50	120 ohms

The 0-1mA meter is frequently used as the basis for various special scale applications.
For OEM customers, we have a wide range of other types of low cost precision meters, with face areas ranging from (44x44)mm to (100x120)mm. These are available with a standard range of voltmeter and ammeter scales, or with fully customized scales

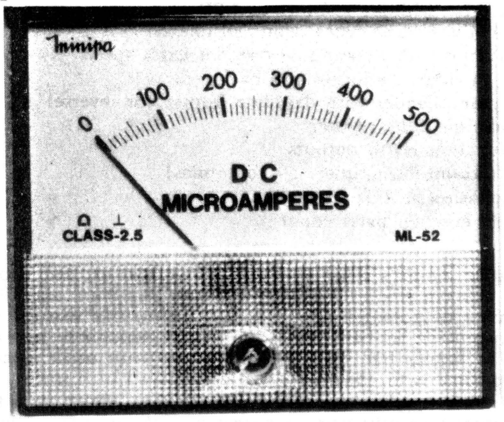
DV M 176M — 200mV (199.9) FSD

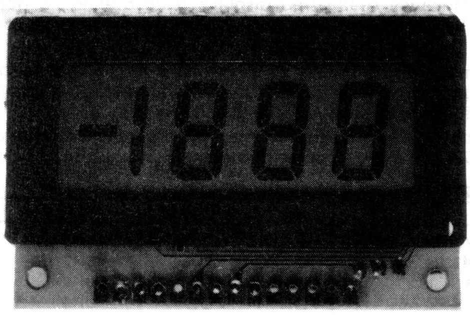
In recognition of the need for a really neat panel DVM, we offer a unit based on the ICM7106 functions, but packaged and presented in such a way as to occupy minimal panel area.

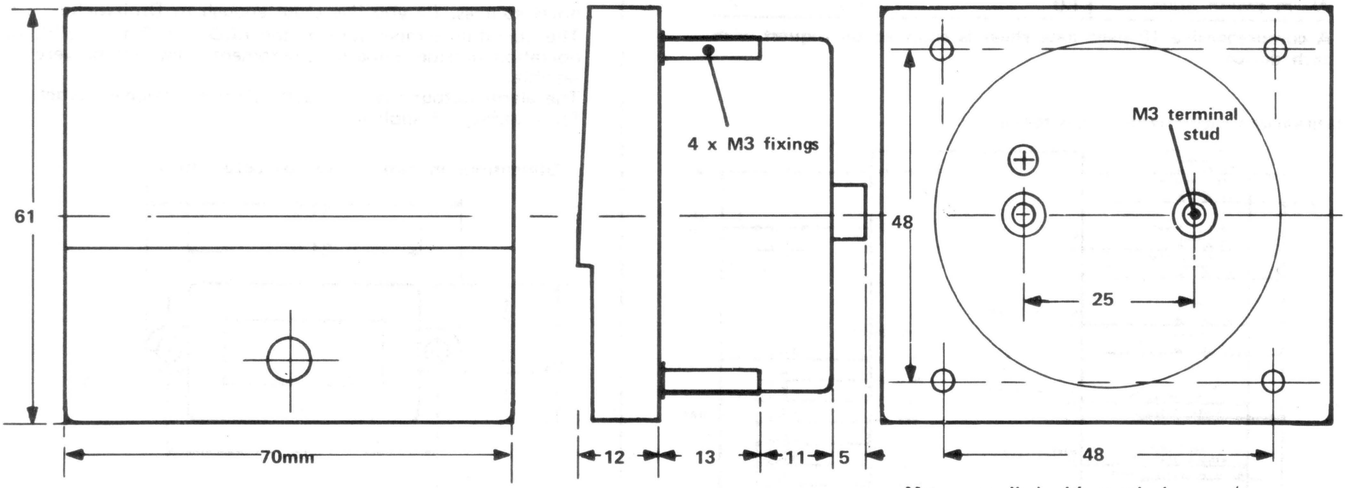
Feature	unit	min	typ	max
Supply voltage	volts	4.5	9.0	12
Supply current	mA		1.0	2
Input leakage	pA		1	10
Linearity	counts	−1	±0.2	+1

Full data sheet with applications available

COMPACT LCD 3½ digit DVM

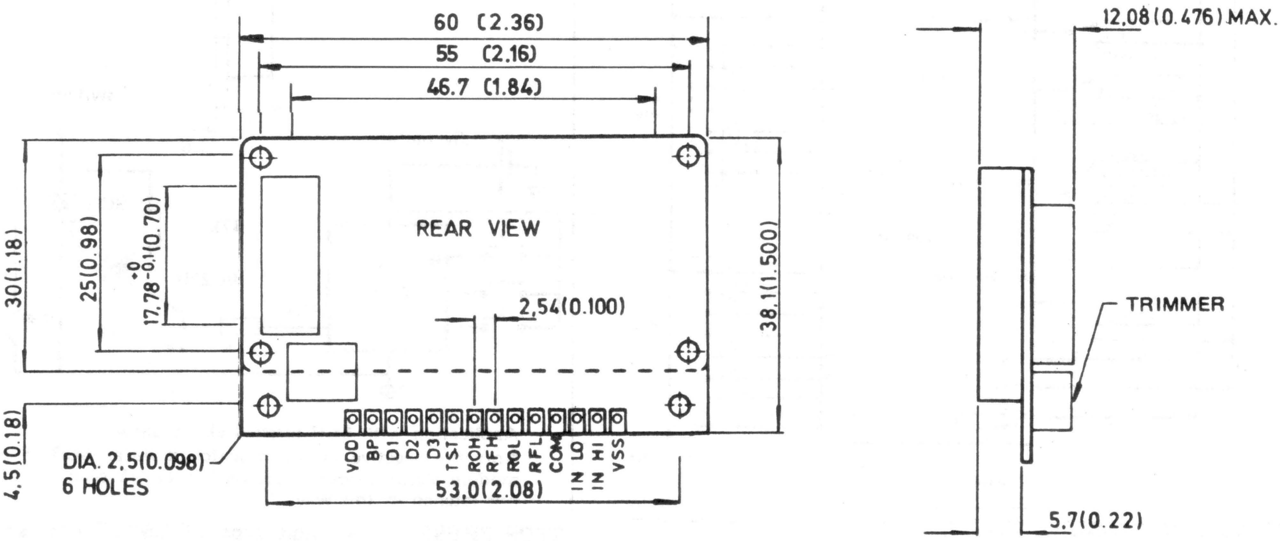






ML52 series meter dimensions in mm

Meters supplied with terminal screws/tags



MK50366N Non multiplexed Clock timer IC

Features

* Direct drive without resistors to LED or vacuum fluorescent tubes. Non-multiplexed
* Fast setting with forwards/reverse for extra speed
* 12 or 24 hour, 24Hour alarm cycle
* Four year calander with day/date format (or reverse)
* Second time zone register
* Multifunction alarm outputs
* Sleep (countdown timer: 59 to 1 mins)
* Colon flashes at 1Hz
* Minimal external parts count

General

Compared to Ambit's usual range of unique and exciting products, you may think a clock IC is a bit old hat. But there are still a large number of potential applications - and so this one has been chosen to provide as many functions as can reasonably be expected, and also be totally free from RFI associated with other clock ICs (including some with apparently static displays.)

10mA of current drive is provided per segment, which with the improvements currently being achieved in LED efficiency, will drive a very bright common cathode 7 segment display. Using the very high efficiency orange/red displays from Stanley, 5 point LEDs may be driven in series per segment to form large format characters.

Perhaps a more direct means of achieving a larger and brighter type of display is to use Futaba vacuum fluorescent display (see part two of the catalogue for the 5LT02 display).

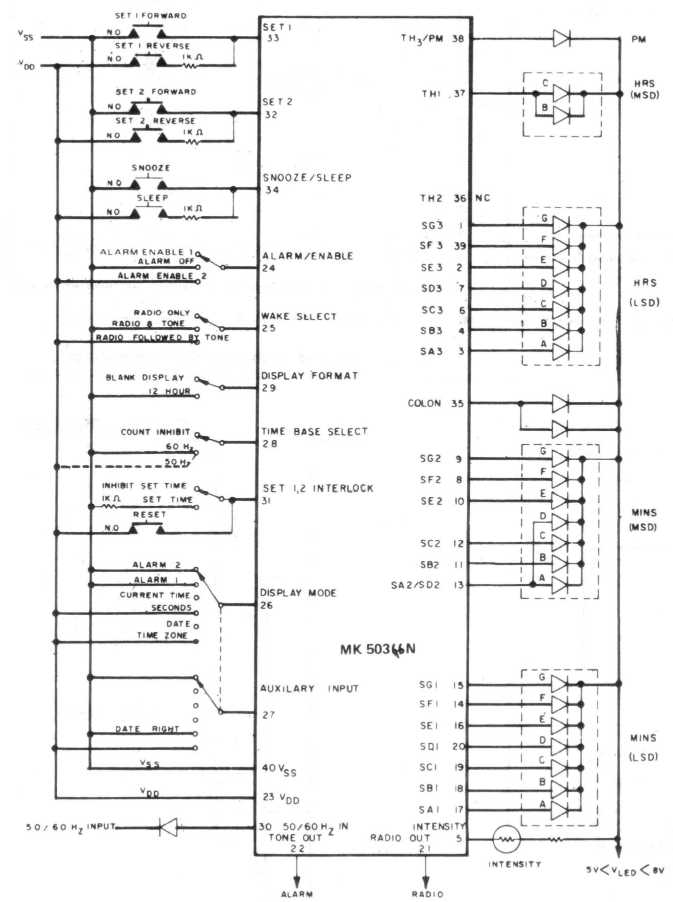
Intensity control for the display may be provided automatically with an LDR, or using a potentiometer.

Electrical specification (abridged)

Parameter	min	max	units
Operating voltage VDD	−12	−16	volts dc
Standby voltage	−6	−10	volts dc
Logic levels "1"	VSS	VSS−1	volts
"0"	VDD+1	VDD	volts
Intensity control	3	30	kohms
Segment outputs		−26	volts
Segment drive	10	16	mA
Alarm switch drive	1.0		mA

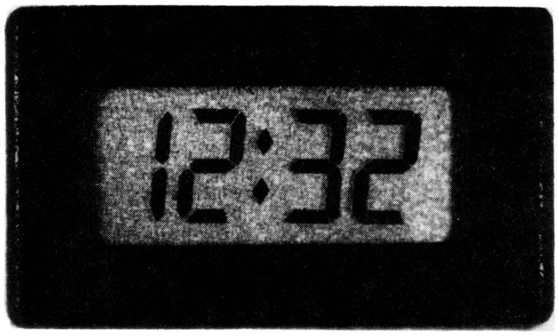
A comprehensive 10 page data sheet is supplied on request with each device

GENERALIZED CIRCUIT (12 HR. DISPLAY FORMAT)



NOTE Any input may be tied to VDD or VSS or allowed to "FLOAT." Floating is intended to be an open position on the switch. The input may not be biased with external voltage to simulate floating conditions.

LC-CM161 (PCIM161) - Clock/calendar module



Illustrated approx. twice full size

Features

* 6 timekeeping functions, 24 hour cycle alarm, 4 year calander with alphanumeric day/date
* 12/24 hour and month date reversal options
* 6.4mm LCD characters, with backlight lamp
* Mounting lugs included
* Crystal controlled, 6uA consumption

General

With only three setting switches, and armed with the table of setting procedures, the PCIM161A performs most of the timekeeping functions anyone could need in such a unit.

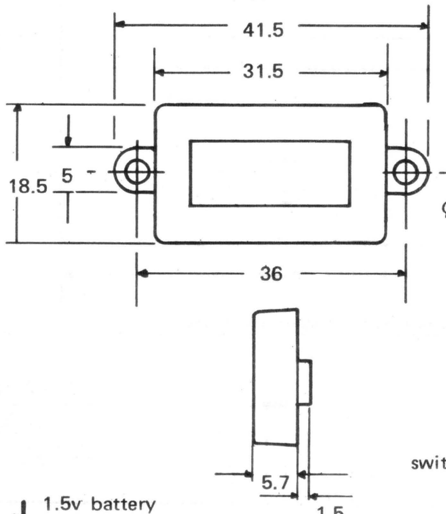
Being small and easily mounted, with a small 1.5v battery providing literally years of operation, the PCIM161A can be fitted into a host of applications requiring a self contained unit of this type.

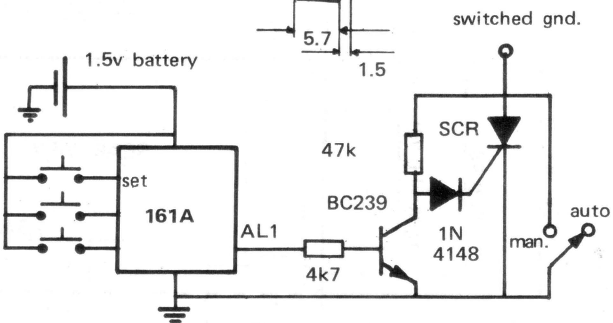
A fourth press button activates the incandescent back light which draws 12.4mA - so this should be used sparingly if battery life is important. However, the general consumption is so low that the unit is easily solar powered (with a single NiCad backup), and may be powered from a number of novel sources. (If you live close enough to Droitwich.........)

The operating temperature of the LCD is +5°C to +45°C, so operation outside domestic enviroments may not be very feasible.

The alarm output may be used to drive a latched switch for a variety of applications.

Dimensions in mm (approx actual size)



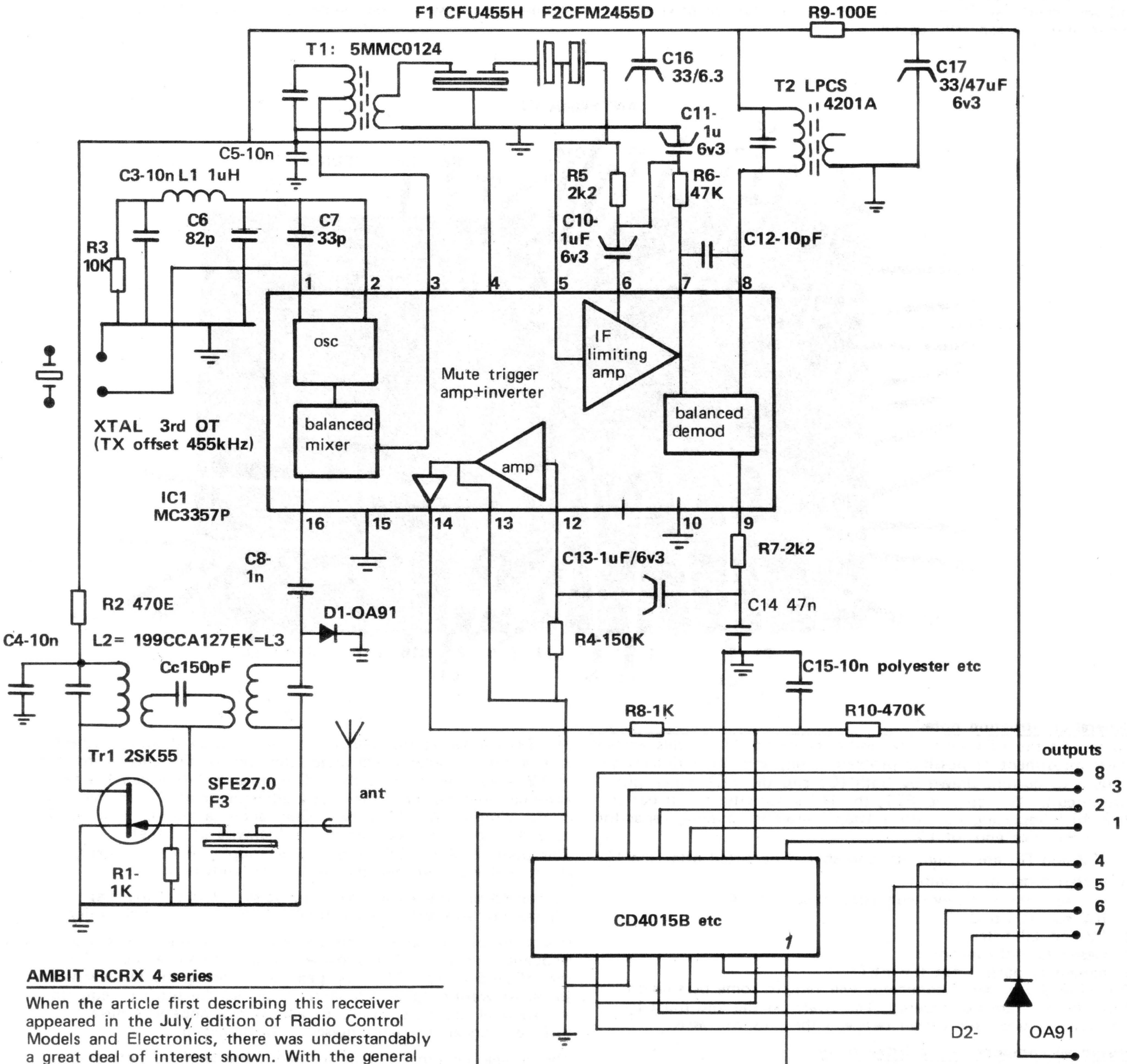


Application with SCR latched alarm output

Note the negative supply is switched via an SCR. The module alarm output resets itself if 15 seconds if not latched in this way.

STOP PRESS : ½" digit type LC-CM172 now stocked

8 channel digital proportional radio control receiver - based on the July 79 RCM&E series



AMBIT RCRX 4 series

When the article first describing this recceiver appeared in the July edition of Radio Control Models and Electronics, there was understandably a great deal of interest shown. With the general interest in enthusiast RC growing quickly, a new FM constructor feature is bound to draw attention.

The original design by Terry Platt was quickly examined closely, and despite efforts to make a dramatic improvement and major alterations to the circuit - the original circuit is really refined to the optimum in most respects. All that remains is for a little fine tuning of some of the possible shortcomings, namely:

The original uses an untuned input to the FET. This provides an easy match to various antennas, but it also provides easy access to strong out of band transmissions to block the FET. There isn't much room on the PCB, so we have used one of the new 27MHz ceramic filters from Murata instead. The extra insertion loss of this stage is made up by using a low IDss FET in common source mode. The 2SK54/2SK55 are ideal candidates. The input/output impedance of the filter is in fact 270E, but using the higher value on the output as shown slightly reduces the passband without serious side effects.

L1 and L2, which were originally open wound coils - have been replaced by TOKO 7KC coils, made especially for the purpose. These are not only reliable and repeatable - but the coupling is better controlled via Cc, further improving the RF selectivity of the circuit, until the end result is the same as when using 4 RF tuned stages of Q approx. 80-90.

L1 is now a simple 1uH molded choke, again reducing some of the scope for misconstruction through home-wound parts. C6 has been adjusted to suit the changed inductance.

It should be noted that the FET choice is critical, since too much current in this part of the circuit will pull down the whole supply to IC1, due to the decoupling effects of R9.

If the voltage on the left of R9 is pulled down more than a volt, then the FET is probably drawing too much current, and this problem has been noticed in some instances. R9 may be reduced to about 68E, but this reduces the effectiveness of the decoupling of the receiver from the heavy current drawn in the servos. The whole receiver draws about 4-6mA total.

AMBIT RCRX4 : SPECIFICATION

Measured at 5v input	typ	unit
Antenna sensitivity	1	uV
Current consumed	4.3	mA
Outputs	8	channels
Required deviation	1.5	kHz
Adjacent channel (10kHz)	-90	dB*
Stopband (60-1MHz)	-90	dB*
Stop voltage	3	v
Change in decoded pulse from 10uV to 100mV input range	10	uSec

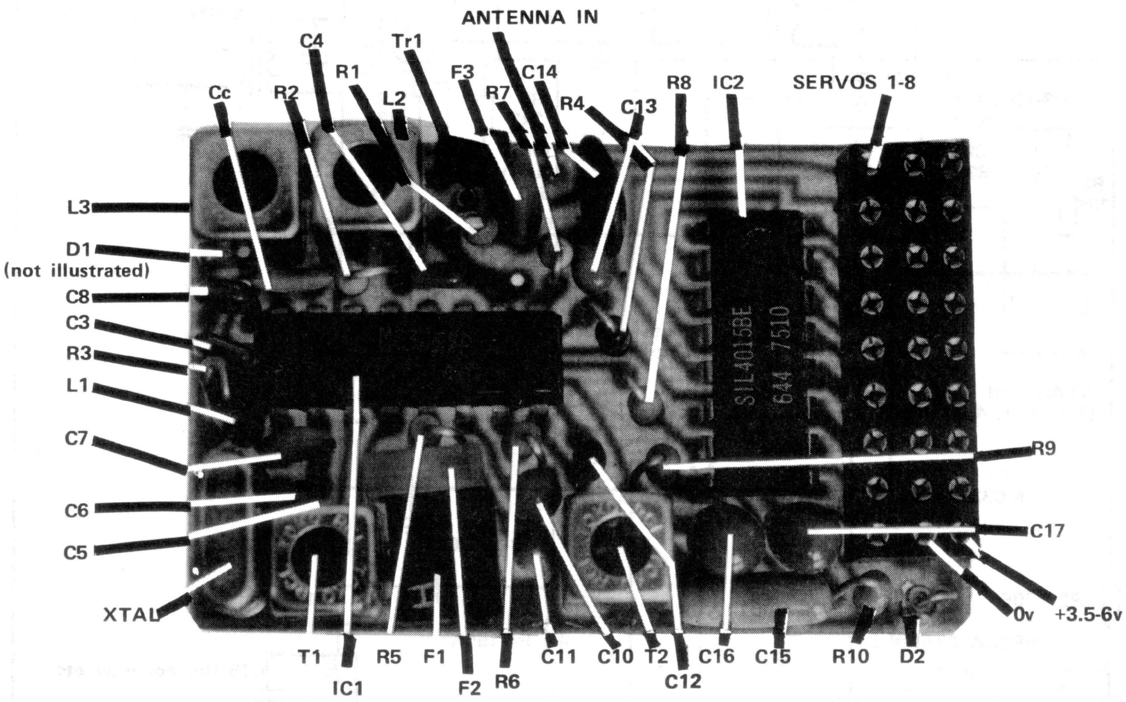
*Adjacent channel measured with a correctly encoded pulse train used to modulate the generator. Although audio effects may not be -90dB down, the decoder effectively ignores these interference signals.
NE5044 used for encode source at 50 frame rate.
*RF stopband ±1.5MHz

AMBIT RCRX4

Component overlay

This approach to instructional information is a slight departure from our usual approach, and we would be interested to hear and we would like to hear if you think this style of general presentation is generally useful.

T1 on the illustration is shown as a 7E style coil, but in order fit the SLM box for the RCME receiver, we have now changed T1 to a miniature 5M series coil which permits easier fitting



General construction notes

Do not fit the FET until after initial checking. (Tr1), this enables easier assessment of possible problem areas, and will indicate the amount of current drawn by both IC1 and the FET when put into circuit. Without the FET, the RX sensitivity is still better than 4uV (replacing F3 with a 100pF antenna coupling capacitor to the RF 'hot end' of L2.

F1, F2 and R5 are a tight fit, and should be fitted after the IC.

Suggested order of assembly:

1 IC1 soldered in , servo connector block fitted
2 IC2 socket fitted
3 L2 and L3 fitted
4 Resistors and capacitors
5 remaining parts -not the FET

The SLM 9 way servo connector will require some persuasion, since the pins are not tapered. The swarf on the end must be removed - a small file drawn between the pins will assist.

Ensure all components are fitted tidily

The crystal socket may be supplied in the form of gold plated cage jacks - these should be fitted onto the pins of a suitable crystal to enable accurate alignment during insertion and soldering.

Testing and setting up

Examine all soldering carefully, check for bridged connections dry joints. Clean up the board underside with flux remover or a similar suitable solvent. (Don't use one that eats plastic).

Since the receiver should draw only about 4mA-5mA, put a resistor in series with 5v power source to limit current if a short exists. Use a 100E resistor for the inital current check, with a multimeter with 25mA range also in series, This will only drop 0.4v if all is well - but will limit the current to 50mA if a dead short exists. Assuming all is OK, remove the limiting resistor.

Fit the FET, and check the voltage on both sides of R9 before and after the FET is in circuit. It should not drop more than 0.2 to 0.3 volt with the FET in circuit.

Listen to the recovered audio signal by placing a high impedance crystal earpiece on the junction of R7/C14. There will be a crackling/hissing sound, which changes character as T2 is adjusted. Plug in a crystal (if you haven't already done so - use 3rd OT 455kHz IF types).

A high impedance audio amplifier is almost as good as the earpiece - but this may introduce extra 'earth' which is not going to be strictly representative of the conditions encountered in the real world of battery operation only.

The DC voltage at the detector output (pin 9) should be checked with a high impedance voltmeter (on approx. 5v range). The core of T2 is adjusted so that this DC voltage sits between the two extremes that can be achieved by adjustment of T2 (in the middle of the discriminator 'S' curve.) Only about a quarter turn will be needed to bring the coil on-tune (clockwise as a rule).

This point is about 2.5v with a 5-6v supply, in other words, about half way between the supply rail voltage.

T1 should be tuned for maximum noise - it is a fairly flat adjustment, tolerant of substantial innacuracy in initial tuning.

If you have a suitable transmitter ready, or a signal generator and frequency meter - then provide just enough signal signal for the RF (and IF coil T1) to be peaked. Flooding the receiver with an overpowering signal will tend to distort the settings required at low level inputs, so don't fit an antenna once you have sufficient signal to enable noises to be tuned in.

The voltage on pin 9 should be about the level set with no input - though remember that if you set up with an antenna fitted, then it is possible that CB interference could have upset the initial settings.

The tuning of the RX to the TX frequency is essential if pulse width errors are to be avoided, so if you have a servo connected to the output, - make certain that with the transmitter in an idle position, the servo is approx. centered up.

Fine tuning is accomplished by progressively reducing the input signal and retrimming the cores of the RF stage and T1. When you think you are ready, check that the servo doesn't change position dramatically when the transmitter is brought close by. If it does, the usual trouble will be the actual frequency setting of the transmitter, and this should be checked and trimmed until the effects diminish.

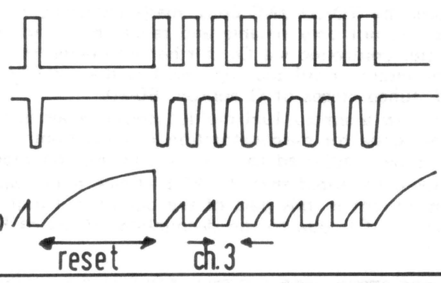
It is anticipated that those of you with access to RF test equipment will be familiar in their use and application.

Waveforms:

TX output from encoder

RX at R7/C14

IC2 reset (pin 15)



FREQUENCY SYNTHESIS TECHNIQUES

Introduction

In case you hadn't noticed, digital frequency synthesisers using the PLL technique are beginning to appear in vast numbers both in semi - professional, and consumer radio equipment. The PLL synthesiser has shrunk from a package count of some 15 or 20 ICs, down to a one or two IC solution - and the prices are now sufficiently attractive to enable very widespread use of the technique. The basic PLL synthesiser system is shown below in fig one.

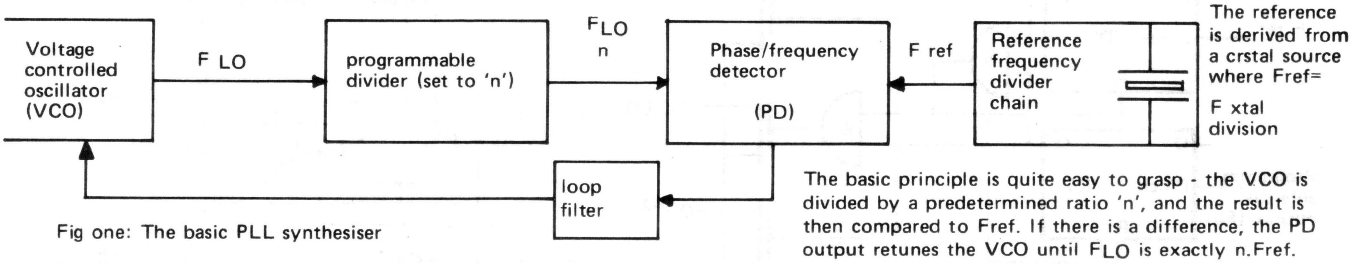


Fig one: The basic PLL synthesiser

The basic principle is quite easy to grasp - the VCO is divided by a predetermined ratio 'n', and the result is then compared to Fref. If there is a difference, the PD output retunes the VCO until F_{LO} is exactly n.Fref.

This technique is widely used - and perhaps best demonstrated by reference to the CD4046 one chip PLL from the CD4000 series. The speed of this device limits the operation to about 1MHz - and of course, most tuning sysnthesisers for radio require 120-150MHz capability.
One way is simply to 'prescale' the input from the VCO with a high speed divider (10 or 100) - but in doing this, the reference frequency must also be lowered in order to maintain the same output channel spacing at the VCO itself. One of the classic loop problems concerns getting the reference frequency out of the loop tuning voltage - done by the loop filter described in fig one - but as the reference is reduced in frequency, the filter time constants are greatly increased, and the speed of operation of the loop itself is slowed down considerably.

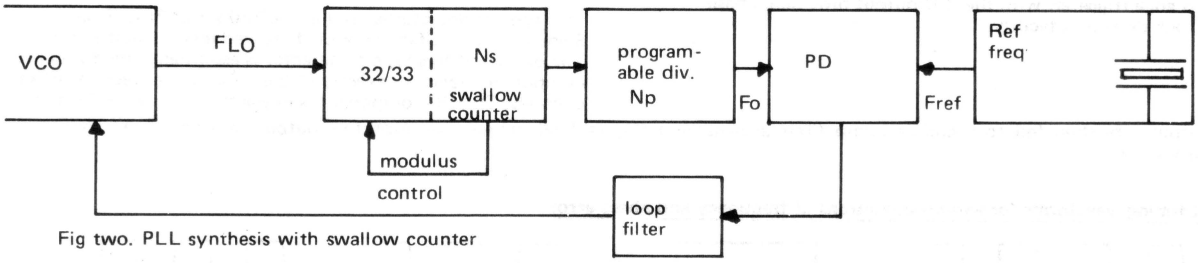


Fig two. PLL synthesis with swallow counter

In order that the Fref may be maintained at the highest possible frequency (namely the actual output channel spacing), a technique known as 'swallow counting' is used. Nothing to do with ornithology - this technique relies on the use of a dual modulus high speed prescalar. In the example of fig 2, division is programmable by either 32 or 33. 10/11 is frequently used, but 32/33 takes a typical band two FM LO and brings it straight down to a speed that can be handled in CMOS.
The basic technique of swallow counting may appear a little hard to grasp at first sight, and various analogies have been used to try and make the point clearer - but stories of bricklaying and the like are of dubious benefit, and so a worked example will be used to try to show the principles involved.
The swallow counter, like the programmable divider, is preset with information concerning the required division. This value is Ns, the value in the programmable counter (PC) being Np.
The system initially counts with the prescalar set to 33, and each output pulse decreases the swallow counter preset value by 1 - until it reaches zero, when the modulus control line switches state to 32. The swallow counter then ceases counting. Meanwhile, the main PC carries on counting down till it too reaches zero. The system then resets throughout, and the cycle repeats. (Ns and Np being reloaded)
Thus in arriving at the PD, F_{LO} has been divided by:

33 . Ns (the number of times the prescalar was in the 33 mode)
plus 32 . (Np -Ns) - the number of times the prescalar was in the 32 mode,)

(1) thus N = 33Ns + 32(Np-Ns) for this system - diagram three details the actual pulse timing:

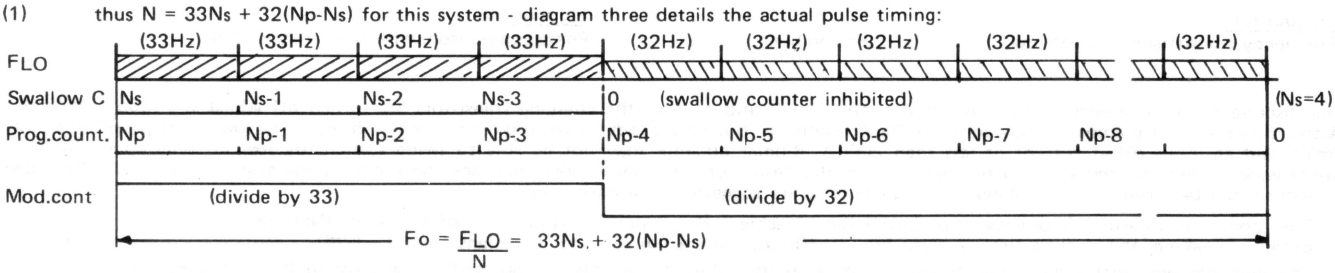


Figure three. Pulse swallowing example showing the timing of one complete cycle of the count sequence

For this technique to work, Np must be equal to or greater than Ns, and the expression, (1), simplifies to N = Ns + 32Np (2)
To define all possible integers of this division ratio, Ns needs to be stepped from 0 to 31 before changing the value of Np, but to satisfy the condition that Np is at all times equal to, or greater than, Ns - the minimum value for Np is 31.
In a system where Np is a 10 bit ($2^{10}-1$ = 1023) counter, the maximum and minimum divisions ratios are then

Nmax = Ns max + 32 Np max = 31 + (32x1023) = 32767 (3)
Nmin = Ns min + 32Np min = 0 + (32x31) = 992 (4)

Remember that the values of 32/33 are arbitrary, and may be substitued by 10/11 or whatever. It just happens that several of the one chip synthesis systems are designed around this basic ratio - using a 5 bit swallow counter, and 10 bit programme counter.
Relating this to a typical application, the LO of an FM set may be required at 99.8MHz, with 25kHz reference this is an N of 99.8/0.025 = 3992, which is well within range. Reducing the reference to 5kHz channel spacing, N = 99.8/0.005 = 19960, still in range of the system capacity. To achieve 5kHz resolution with fixed prescalaing of say 100, the reference would only be 50Hz, which presents a stiff task to the loop filter design.

Too much filter

The loop filter is the crucial part of the system. Too much filter time constant and the reaction time of the loop is very slow. Rather like a pendulum, the correction voltage builds up such inertia it goes straight through the desired channel and out the other side - if the overshoot is too severe, then the loop either takes a very long time to settle, or simply oscillates about the desired channel in a rather slow and ponderous fashion. If the loop damping is too inadequate, then although lock will be rapidly aquired, the jitter brought about by the Fref in the tuning voltage will cause the oscillator to be unstable and noisey about the desired channel, since the rapid changes in the phase detector output state are fed to the oscillator with inadequate filtering.
The ideal system is one where lock is aquired very rapidly, and once achieved, provides an output that disables the digital edge detector, and enables a phase detector based on analogue techniques that provide high gain over a narrow range of degrees of error.

FREQUENCY SYNTHESIS TECHNIQUES/2

Elements of the system

Having started with the principles of pulse swallowing/swallow counting, the rest of the number processing of the system is relatively easy. Assuming the correct division ratios for both the reference counter and the programmable counter chain have been set up, the phase/frequency detector comes next. A popular method is the digital detector shown in figure four. This provides signals when the input frequency from the programmable divider is either, high, low or phase locked .

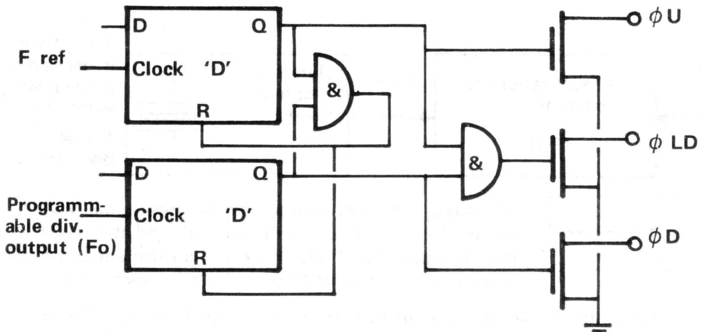


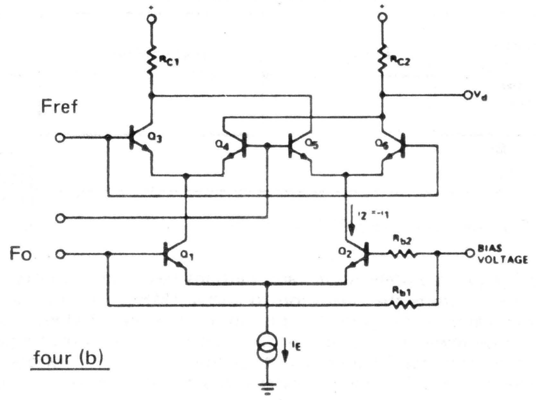
Fig four (a)

A digital type edge frequency/phase detector (from Plessey NJ8811)

This type of detector system is fairly universally employed, since it offers fast transition, and low jitter and noise when locked up. Variants of this theme are found CD4046, MC4044 etc.

The system is edge triggered, with the LD output providing phase pulses and lock detection facility.

INTEGRATED PHASE DETECTOR



four (b)

This type of detector is used in the 560 series PLLs from Signetics - it is in fact a double-balanced mixer (multiplier) The output of this type of detector is essentially the difference in input frequencies - exactly in the same way as receiver mixer operates. The DC component is extracted to provide the error.

These digital outputs are then fed to a charge pump filter arrangement (figure five), where the respective outputs are combined to drive the tuning voltage.

Relative output timing waveforms for various conditions of frequency and phase error

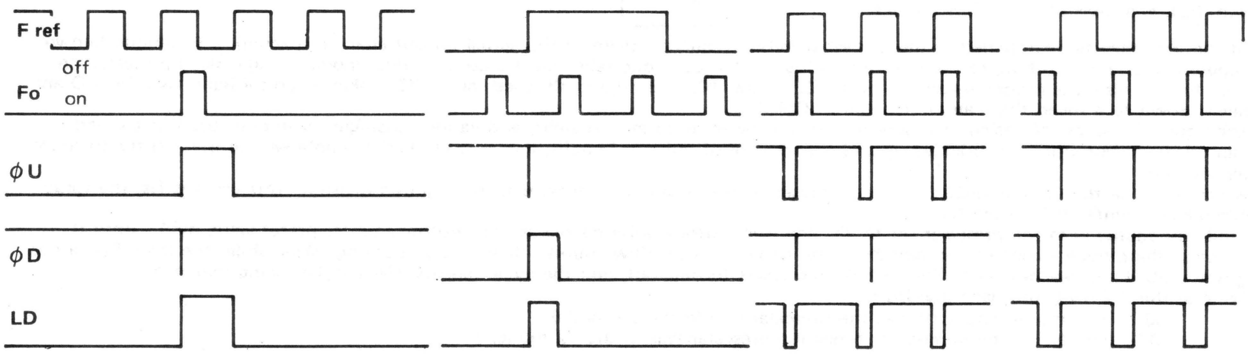


Fig four (c)
Frequency comparison . Fo too low

Fo too high

Fig four (d)
Phase comparison, with frequency locked

The discrete circuit is especially suitable for low noise applications, since the charging transistors are turned off whilst the loop is locked. Active filters using Op-amps can produce excellent results to, but care needs to be taken to chose a device with low input offset (less than 2mV), and an input offset current of less than 200nA. Bipolar op-amps may not be able to swing sufficiently low in some applications where wide frequency ranges are to be spanned. In the design of the filter - more than anywhere else in the system - care must be taken to achieve the best results. The following design criteria are suggested as a basic guide:

The loop filter's natural frequency must be chosen to achieve the required settling time of the loop. (Nat freq = ωn. Broadly speaking, the settling time is given by $10/\omega n$, where ωn is given in radians/second. $(2\pi f)$

The loop damping factor must give adequate loop stability, whilst being able to cope with large steps in the loop frequency, without the overshoot previously described. A value of 1.0 gives approx. 10% overshoot.

In the calculation of values for both types of filter, determine the following:

Kv = VCO gain (radians/sec/volt)
N = Variable division number
= VCO frequency/reference frequency

then R1 = between 4k and 200 ohms for the first type of filter
= between 30K and 6k for the second type of filter

$$C1 = \frac{Kv}{2\pi .\ N.\ \omega n^2 .\ R1} \quad \text{.. .. filter type a}$$

and $$C1 = \frac{2.3Kv}{\pi .\ N.\ \omega n^2 .\ R1} \quad \text{.. .. filter type b}$$

$$R2 = \frac{2}{\omega n.\ C1}$$

$$C2 = \frac{1}{15.\ \omega n.\ R2}$$

It isn't the purpose of this article to analyse the basic operating principles of the phase locked loop - one of the most concise and useful description of the general theory and practise of the PLL has been published by Signetics, in conjunction with their range of monolithic PLLs. The 14 relevant pages are available from our photocopy service if required.

As an educational and experimental exercise, the 560 series phase locked loop offer an excellent inisght into the operation and behaviour of PLLs. Familiarity obtained through experience with 560 series will be invaluable in the development of synthesiser based PLLs. The CD4046 CMOS PLL likewise provides a method of getting to grips with PLL techniques, and the benefit of at least some 'hands on' experience of the PLL is invaluable before tacking the sterner requirements of the types of synthesisers about to be described. Although calculations are generally valid, practical considerations usually have the last word in the system development.

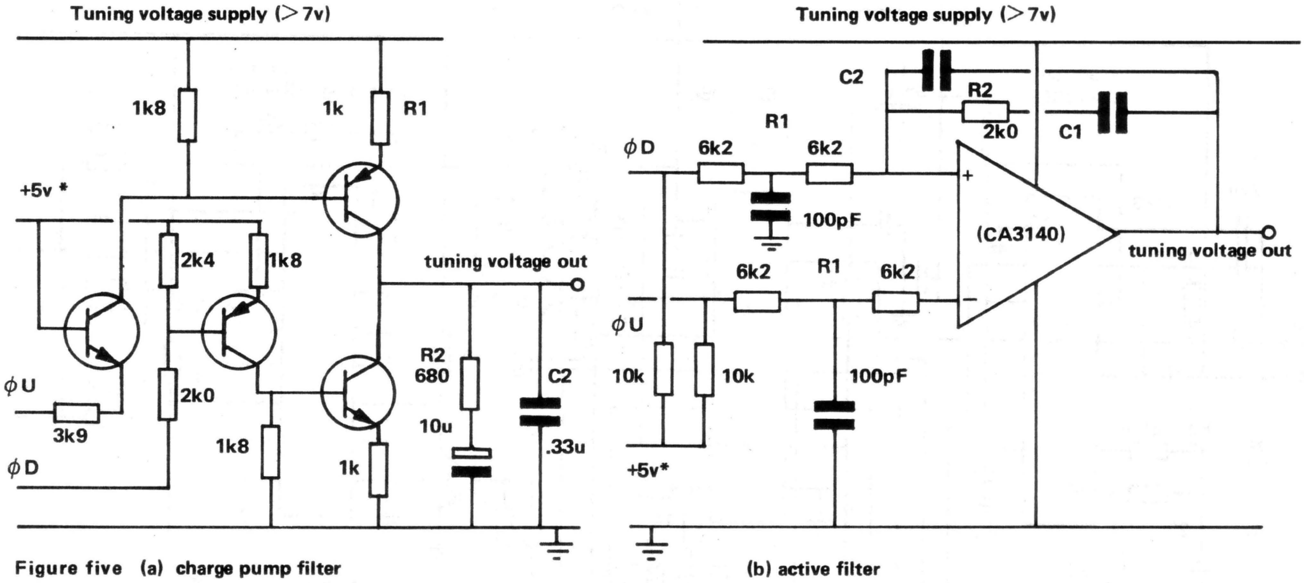


Figure five (a) charge pump filter **(b) active filter**

* or value chosen to suit the phase detector IC used - values for R1/R2/C1/C2 are based on a general wideband application

Summary of the PLL so far:

The heart of the the synthesiser is the PLL, and in particular the phase detector and filter. A brief glossary of the terms used in PLL analysis is provided here for reference:

Loop gain	**(Kv)**	The products of all the DC gains of the loop elements in units per second
	(Kd)	The conversion gain of the phase detector in volts per radian (2π f)
	(Kθ)	The VCO conversion gain in radians per volt per second
	(A)	The voltage gain of the amplifier following the phase detector
thus	**Kv = (Kd.Kθ.A)**	

Additional reading

Phased Locked Loops	**Signetics**	**(now Mullard)**	**pp809-824 Philips analogue circuit databook**
ICAN-6101	**RCA**	**(CD4046 appcns.)**	**pp598-601 CMOS data book**
MC4044 data sheet	**Motorola**		
Radio communications Handbook	**Plessey**		**pp various**
PLL design fundamentals	**Motorola**	**(may be out of print)**	**AN535**

Current synthesesis devices and techniques

The present state of the art is in a state of flux, with many major manufacturers contributing parts for use in consumer radio and TV applications. Specialists sources, such as Plessey, Hughes AC, and Mullard, have also designed communications orientated parts - plus there are many building blocks available in basic logic families from TTL/ECL/CMOS etc. However, the purpose here is to try and summarize the dedicated ICs, and these are:

Mullard Mullard have two families, the LN123/LN124 are primarily a high class parallel data entry set, designed for various communications applications from VLF through UHF and beyond. The system is split into two sections, one IC, the LN123 (now LN1231) is a reference oscillator divider chain, with various programmable division ratios. The LN124 (now LN1241) forms the basis for the programmable counter/swallow counter, with 4½ decade range. With three dual modulus prescalars - the LN1241 has a maximum capability of 6½ decades with an input frequency of 4.5GHz. LN1231 also includes a special form of phase detector, whereby lock is aquired very rapidly through an edge triggered phase/frequency detector - but once aquired, it is held by switching over to an exceptionally high gain phase detector based on sample and hold techniques, with a gain of several thousand volts per degree of phase error. The second phase detector is only enabled when the first detector has brought the signal within its range of control.

The LN1241 may be 'slaved' to a second LN1241, to add further resolution to the system (an extra two decades per additional LN1241). In this way, a LN1231 and two LN1241 devices can provide an HF synthesiser that covers the range 30-60MHz (for up conversion to a 30MHz first IF) in 10Hz steps. The programmable dividers also include a subtractor facility, that enables the IF offset to be removed for direct readout on BCD programming switches.

The system is also fully compatible with various digital and MPU control techniques, and is suitably stable and 'pure' to meet various requirements concerning enviromental and electrical specifications for commercial and military users.

Preliminary information package : 34 pages

The second Mullard system is based on a serial data control synthesiser IC, the SAA1056. The device is programmed with a 17 bit serial data stream - primarily derived from a custom programmed one-chip MPU. The system uses a 5 bit swallow counter, and 10 bit programmable counter (as per the example at the start of this article).

The remaining two data bits are used to set the values of the internal reference frequency, and to provide a degree of parity check through the transmission of a leading zero. The device uses an external prescaler.

Control of this system is easily achieved using a small home computer, with the appropriate serial output interface. Programming with BASIC is advantageous, but the derivation of the basic data word is very simple, since all that is needed is to convert the required division ratio (accounting for the IF) into a binary number for the IC is :

Example

Assume 89.1MHz is required to be received with a 10.8MHz (osc high) offset. The LO is thus 99.8MHz, and with a 25kHz reference frequency, the required division ratio will be 3992 in 15 bit this looks like

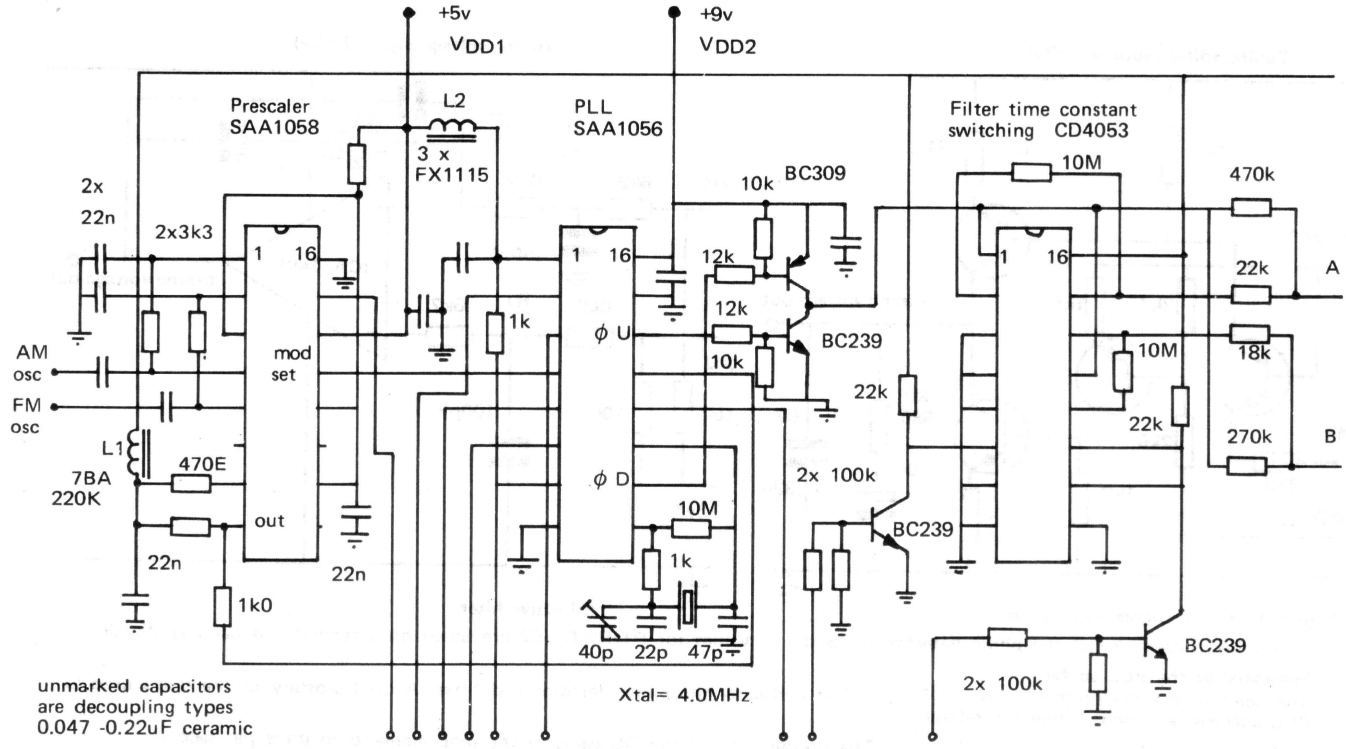
2^{14}	2^{13}	2^{12}	2^{11}	2^{10}	2^9	2^8	2^7	2^6	2^5	2^4	2^3	2^2	2^1	2^0	
16384	8192	4096	2048	1024	512	256	128	64	32	16	8	4	2	1	
0	0	0	1	1	1	1	1	0	0	1	1	0	0	0	binary number

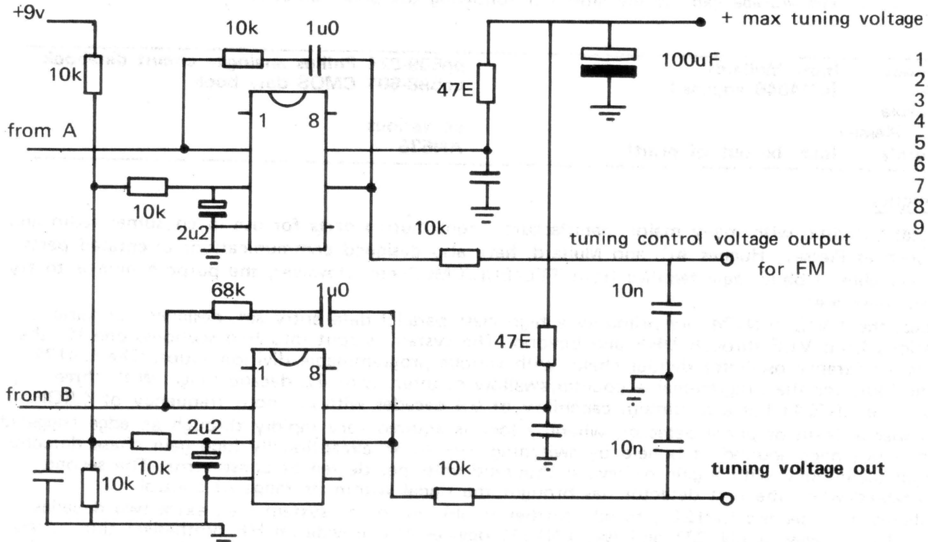
Now comes the cunning bit, since the formula for loading the swallow and programme counters is (the "remainder")

N = Ns + 32.Np, and to divide by 32 in binary, simply shift the binary number 5 places to the right..

Thus the swallow counter number is derived from the last five bits of the 15 bit division ratio, (11000), and the value for Np is simply what remains, since the effect of removing the last five numbers is to produce division by 32 in binary arithmetic. Np is thus (0001111100). In decimal terms, Ns = 24, and Np = 124.

Frequency synthesis techniques /4 **A versatile programmable synthesiser control system**





CA3140E etc (output of the op amp must be able to supply lowest tuning voltage required)

1 Prescaler reset (NC)
2 Clock
3 Data enable
4 Serial data (17 bit format as discussed)
5 Clock output (unbuffered)
6 Select reference in conjunction with bit 16
7 Lock detect output (high when in lock)
8 Time constant switch inhibit
9 Switch select:
With logic low on 9, pins 13/14 are linked and 1/15 are linked
With logic high pins 12/14 are linked and 2/15 are linked
With 8 held high, all interconnections are valid
Taking 8 low inhibits all interconnection

Thus with logic high at point 9, the time constant is longest. Taking point 9 low bypasses part of the time constant R so fast aquisition of lock may be achieved. Once the lock indication is available at output 7, this may be used to switch pin 9 to provide long time constants for minimum noise whilst the loop is locked.

FIGURE SIX : Ambit's synthesiser control PCB - based on the MULLARD RTS design

This PLL synthesiser loop system is suitable for frequencies in the range 50kHz to 200MHz. Whilst primarily intended for operation with serial data supplied via an MPU, simpler programming may be achieved with CMOS logic arrays, or even switches with a suitable debounce circuit. The reference frequency with a 4MHz crystal, is set according to:

Control bit 16	Reference select (point 6)	Reference division	Fref with 4MHz Xtal
1	1	160	25kHz
1	0	400	10kHz
0	1	800	5kHz
0	0	8000	0.5kHz

Although the values alongside are suitable for use in conjunction with all know broadcast channel standards, a different crystal will enable various alternatives.

The maximum résolution of the output channel will be limited by the maximum division ratio of the loop system (32,767 with a 32/33 prescaler) The clock output is available unbuffered, and the Ambit PCB includes provision for the use of an additional CMOS buffer stage if this is required in the overall system. (Parts not supplied as standard.)

As an alternative to the two stage time constant system, the receiver muting output can be used to prevent the passage of noises that are associated with stepping frequency in the loop. After the phase detector, two alternative filter routes are provided for the AM and FM tuning lines, so that the filter components for each respective Fref may be optimized without further switching complexities.

Remember that additional decoupling on the tuning control line (in the actual radio circuitry) will add to the time constant of the system quite considerably, and should be taken into account when developing the system.

Input sensitivity	5-10mV (from 50 ohm source)	from 500kHz to 125MHz. Input impedance approx 1k at 1MHz
Current consumption	100 -125mA at 5v, 1-4mA at 9v	
Logic levels on 2,3,4	Low 0 to 0.2V$_{DD1}$, High 0.8V$_{DD1}$ to V$_{DD1}$.	

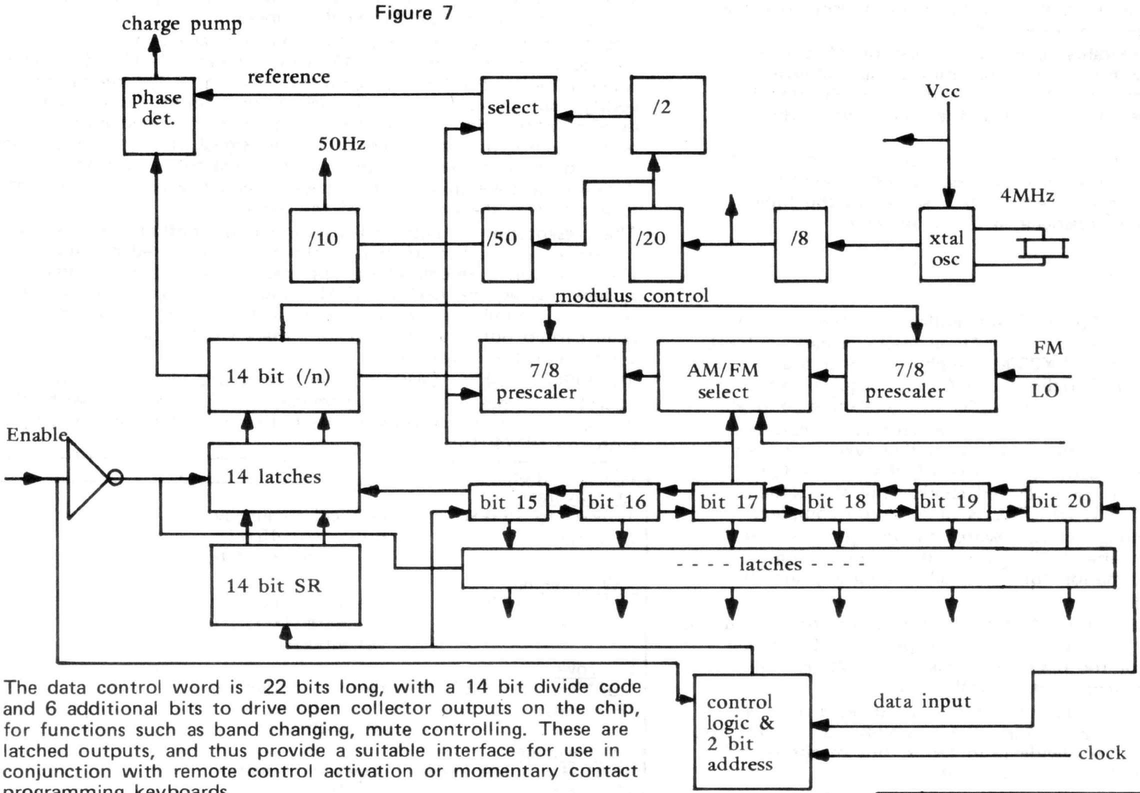
Each module/kit is supplied with manufacturers data on the various ICs employed to enable development of the basic system in conjunction with various applications.

FREQUENCY SYNTHESIS TECHNIQUES / 5

DS8906 National Semiconductor's entrant

The DS8906 from National is an all-in-one approach, exactly on the same basic lines as the Mullard approach. A single IC includes the prescaler functions, (in ECL technology), and all the remaining programmable dividers, phase detector, charge pump etc are included in an I^2L process.

Figure 7



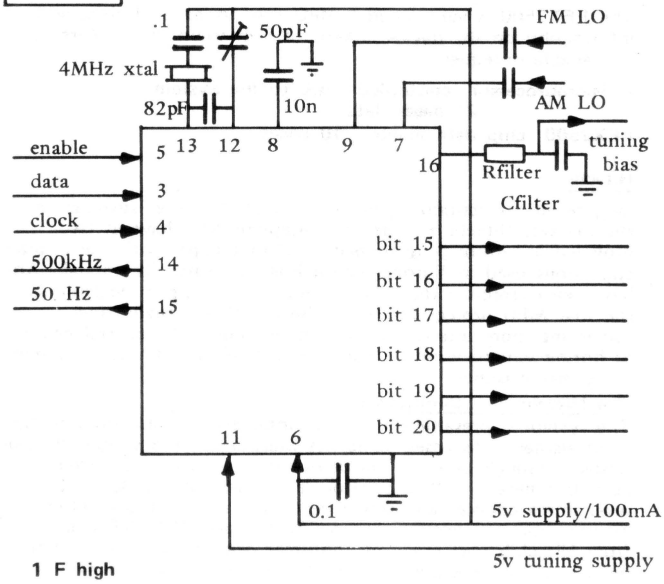
The data control word is 22 bits long, with a 14 bit divide code and 6 additional bits to drive open collector outputs on the chip, for functions such as band changing, mute controlling. These are latched outputs, and thus provide a suitable interface for use in conjunction with remote control activation or momentary contact programming keyboards.

The remaining two bits are used to set the address - since other devices may be connected to the same data bus (the display decoder driver - for example).

The 8906 is derived from the 8907, which is the equivalent part for use in US and similar channel spacing systems (200kHz FM, 10kHz AM). The 8906 system resolution is 12.5kHz for VHF up to a guaranteed minimum of 120MHz oscillator (examples have shown typical operation to 150MHz+). Which is suitable for any channel allocation, including those encountered in amateur and some commercial systems.

The AM input operates with frequencies up to 15MHz minimum, and provides 500Hz channel spacing. All the various cautions and considerations that have been mentioned in connection with other PLL synthesis systems apply in this approach, and whilst 500Hz channel spacing on AM may seem to be suitable for shortwave operation, the loop will need to be designed carefully to ensure sufficient gain and damping is available.

The IF offsets employed are purely determined by the programme used to feed the PLL.



DS8906/7 **National synthesiser PLL 12 pages of data available (as at 9/79)**
8906 evaluation controller PCB type 98906 available (inc facilities for loop filter as per figure 6.

MM5439 National Semiconductor's TV synthesiser part

The presentation of this part is slanted very heavily towards TV tuning systems, and until fuller information is available, it is going to be difficult to assess its potential outside these areas.

Notwithstanding that, the device employs a technique which is a very clever solution to the problems associated with jumping in large frequency steps with almost no overshoot. The technique is currently the subject of some unceratinty with regard to patent, and so the following must be considered with that qualification clearly understood.

The counter side of the system is bus controlled by an MPU, just as with most of the systems in consumer applications, but the phase detector system is the bit that really interests.

It operates in two modes, by checking to see if the phase error between the outputs of the programmable and reference dividers is more or less than 0.72°.

For errors of more than 0.72°, , the counter that provides the first output is held in its initial condition - by means of preset or reset pulses respectively, until the second counter provides an output.

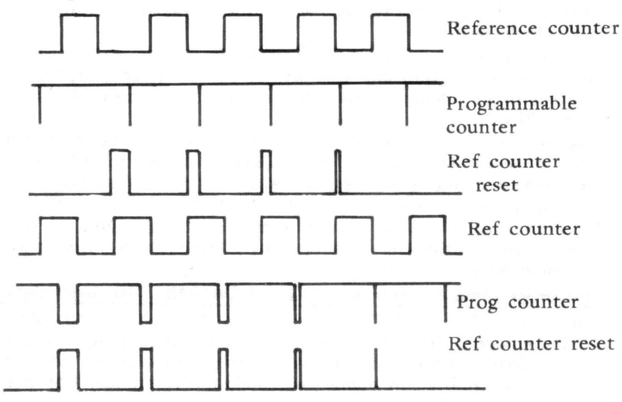


Figure 8

FREQUENCY SYNTHESISER TECHNIQUES /6

MM5439 description of phase detector operation (contd)

Thus both counters start counting simultaneously, to keep their phase difference to a minimum. (see fig. 8)

When the phase error is less than 0.72°, (2uS), the counters are no longer fed with reset pulses, and the circuit continues to function as a classical phase locked loop, using an UP and DOWN phase comparator output to provide correction pulses driving a standard op-amp integrator filter circuit.

This technique considerably speeds the aquisition of lock after large frequency steps have been programmed - and almost enables frequency multiplex tuning - so that the teledata of one channel, may be gathered whilst the picture of another is being viewed.

As mentioned, the logic diagram of this type of detector is the subject of uncertain 'ownership' at present, and so whilst it may be possible for us to release details soon - at the time of compilation of this catalogue, it is not possible.

FAIRCHILD

The system offered by Fairchild has tended to suffer from the penalties of being first in the field. Or at least, an early entrant. The main PLL IC is the FEX2500, which is another multi-purpose system, for both radio and TV - and requiring the services of a custom programmed MPU to make sense of it all.

The chip itself incorporates many of the same basic features as used in the Mullard/Philips serial controlled system, except that instead of being a serial control word, the data is fed in four by four format, plus ID and clock. Thus the actual connection between MPU and PLL is rather more complex.

The method of deriving the data words for this process is much the same as for the other MPU controlled systems, being a combination of information for the swallow counter, and the programmable divider.

The FEX2500 operates from 100kHz to 4MHz directly, and up to 1GHz, with the use of a suitable external prescaler. Selectable references provide for the basic 1kHz, 5kHz and 25kHz units of the reference frequency with a 4MHz crystal.

The total counter is 16 bits long, and is sub divided into two sections, - either 13bit programme/3 bit swallow, or 11 bit prog with 5 bits swallow. (In divide ratio terms, this means that either 8192/8, or 2048/32 may be used).

The FEX2500 is very much a MPU user orientated chip, and so information on the device is kept to a minmum here. More info is available onrequest.

A Microprocessor controlled PLL tuning System
21 pages data
FEX2500 chip data sheets 10 pages

Hitachi

Despite all the tantalizing possibilities of the other systems on the market, Hitachi are the only suppliers who have at last provided a custom programmed MPU that copes with the channel allocations used in European broadcasting. Naturally enough, Ambit has been examining the system, and will making it available as the first MPU project for the enthusiast that actually does something more interesting than programming the central heating or boring you to within a hair's breadth of insanity with tedious TV games routines.

The Hitachi PLL Synthesiser

The system is presented as a fully integrated combination of the main elements discussed so far. A prescaler , with various division ratios (although only the divide-by-ten is used in the system described here) - a PLL control device, with phase detetector, programme counter and reference divider system (no swallow counter you will notice), and a CMOS mask ROM MPU device from Hitachi's vast range of consumer orientated range of 4 bit products. The program includes the necessary software as a controller for a MW/LW/SW/FM broadcast receiver, combined with clock facilities.

The display is controlled by a further device, which can drive either LEDs (via limiting resistors, or a Fluorescent display in a direct mode. Being latched, the static display provides little or no RFI.)

Channel spacing is in accordance with agreed standards, namely 9kHz channels for LW/MW, and 50kHz channels for FM. For the few remaining FM transmissions using 25kHz increments, the error in tuning is negligable except with the narrowest of IFs.

Since this system is the first that may be considered to be a viable proposition for development - the practical aspects will be discussed at some length.

The programme counter and PLL : HD44015 (10 pages data)

The HD44015 is a CMOS PLL, capable of taking up to 12.5MHz sinewave input. Thus without any need for prescaling, AM and up to 10MHz approx of SW may readily be presented for the programme counter to divide to match the reference. For the AM bands, 9kHZ is required, and for SW it is 5kHz.

The speed of this PLL means that FM may be accomplished without resorting to more than a 10 division in the prescaler. This process avoids the need for swallow counting complexity, since a reference of 5kHz (50kHz prescaled by 10) is still suitable for use in second order loops. The arrangements of the HD44015 is not binary, but BCD, with programmable decade counters. The data is fed in 4x4 format (akin to the FEX2500), and then latched.

MPU information also conveys the information required to set the reference frequency for the band in operation.

The phase detector output contains an integral charge pump configuration, and the HD44015 even includes a MOSFET on chip, which provides the function of a level translator for the tuning voltage, and an active filter. A lock detector output is included to provide information for the MPU scanning system and muting outputs, since the system uses a muting signal during all operations that require a frequency change. Thus overshoot effects are silenced during large transitions (from a preset frequency at one of the band, to a preset at the other end of the band).

The presentation of data in BCD form is in conflict with most of the other approaches adopted in the serial controlled consumer ICs. However, it resembles the approach of the Philips communications system devices, the LN1231/1241 - and so may lend itself more readily to thumbwheel programming. No work on this aspect has been carried out at Ambit yet, but there would appear to be some scope for investigation.

The MPU is a 4 bit CMOS system, consuming virtually no power in operation. It has been preprogrammed at the mask stage with all the necessary software to provide the following features in conjunction with the HD44015:

AM operation:

LW coverage	
146kHz to 353kHz	received frequency
IF offsets	Either (A) 457kHz
	Or (B) 466kHz
MW coverage	
513kHz to 1620kHz	received frequency
IF offsets	(A) 459kHz
	(B) 468kHz
SW coverage	
5.95MHz to 9.775MHz	received frequency
IF offsets	(A) 460kHz
	(B) 470kHz
FM coverage	
87.5MHz to 108.1MHz	received frequency
IF offset	10.7MHz

Notes The offsets vary according to the band selected in the AM mode. Whilst this is not an ideal solution, it is the result of the restrictions imposed on the routines within 4 bit architecture and the relatively small amount of ROM. Assuming the MW offset is the value actually used, then the errors on the LW and SW functions will be 2kHz or less (A) - which is within both the tolerances of standard IF filter systems, and the actual bandwidths used. 6 or 8 kHz is commonly employed in AM broadcasting, since the shape factors of commonly available low cost filters (CFM2 series, CFU455 series) do not require such a narrow nose bandwidth as many electronics engineers have come to associate with earlier coil-only designs - in order to obtain the same skirt and stopband performances. The channel spacings comply with the broadcast standards of the LW/MW and SW, and do not pose any problems within the context of a broadcast receiver.

Channel scanning

On all bands, the MPU will provide data for the PLL to scan across the band (either from LF to HF, or vice-versa) and either:

1 Pause for 5 seconds on signals detected that provide a 'Stop Scan' signal to the MPU ...or

2 Remains on the station first accepted, until instructed to continue its scan.

Notes During the pause or hold sequence, the data may be enabled, so that the station band and frequency may be assigned to one of eight memory locations.

Skip scanning

The device may be arranged so that stations are only accepted according to a given parameter (stereo transmission) that may be detected and fed to the MPU controlling the scan sequence.

Manual UP/DOWN

Apart from auto scanning, manual channel increments may be arranged by push button control. This operates in two modes:

1 Momentary contact increments the channel by one

2 Holding down the button cause the station increment to clock at 80mSec per channel until released

Memory

Eight memories are available in RAM (volatile), which locate both the required station frequency, and the band. Write protect is provided with the aid of an interlock function, in the shape of an 'Enable' button, and write enable LED indicator. The station memory number is available in BCD for either 7 segment, or point

FREQUENCY SYNTHESISER TECHNIQUES/7

Hitachi synthesiser system (contd)

..source LEDs.
Thus stations on either LW/MW/SW or FM may be stored with instant access from any tuning location available.

Pseudo memory is available in the form of a last station memory-which means that if the last station frequency on FM was 89.1MHz - regardless of whether or not this was stored in one of the 8 station memory locations - returning to FM via the band select button will recall 89.1MHz automatically.

When the radio is turned off via the Radio/clock switch (ie not disconnecting the power), switching the radio back on will call up the last displayed frequency.

Clock functions

A 24 hour clock function is available when the radio is turned off, or during the operation of the radio. The tuned frequency then may be recalled by operating any button associated with radio functions. 5 seconds thereafter, the display reverts to time without disturbing the radio operation during the entire sequence of operations.

Summary

Whilst some of the functions may provide messers. King & Co. with grounds to gripe about the odd kHz or two IF offset, the overall package is clearly thought out, and above all else, presented as a ready to run system.
It represents the state of the art in first generation MPU operated tuning synthesisers, and whilst it is not all things to all men, it performs functions associated with both consumer and professional audio systems, that have hitherto been very expensive to realize.
The overall cost of this package is low. Less than some one-chip MPUs alone. By the time this publication is released, it is very possible that Ambit will be able demonstrate the Ambit system, and provide first examples to manufacturing customers.
Full documentation is available now

HD10551 ECL prescaler	**2 pages**
HD44015 CMOS PLL	**10 pages**
HD44752 CMOS MPU	**15 pages**
HA12009 I²L display driver	**7 pages**

Other goings on:

Synthesis systems have been announced by the following:

General Instrument, whose handywork appears inside the new Telefunken HiFi receiver. Since GI are jittery about releasing details of their approach, claiming all sorts of secret features. it is difficult to take it seriously as a general market prospect. The main claim to fame appears to be 38 programme memories, but if you can remember much past the contents of ten presets, this is exceptional.

NEC, a fairly straightforward consumer package, but with no visible signs of support for UK customers. Maybe when NEC have got their interests in MPU parts and memories straightened out, they will show signs of life in departments responsible for this type of product.

Siemens offer an interesting range of parts, with a 'hard wired' parallel data system, as well as a few serial systems based around MPUs. Maybe details of these will be recieved and digested in time to be included here, otherwise, photocopied data will be available when this is published.

Toshiba seem to be in much the same sort of state as NEC.

OKI have systems not unlike the other Japanese approaches in overall concept, but with a combined MPU/PLL combination. But although we have established lines of communication with OKI, they seem to be as difficult to deal with as most Japanese sources of semiconductors. Whether or not this is a genuine communication problem, or whether it is a reflection of general Japanese contempt for the scope of the UK market, isn't too clear. Perhaps they will be able to supply more details when they get a Japanese/english dictionary.

If anyone connected with any company making parts for use in synthesis systems - not described here - would care to get in touch so that we might add information to our bulging library, we would be most pleased to hear from them. But be prepared for an honest appraisal of your system.

Notes to students/enthusaists

In common with our usual practise in attempting to get a little innovation and thinking going where new parts are concerned, we are particularly interested to be able to support work being carried out with systems such as PET, NASCOM etc to provide interesting and useful programmes for driving our serial controlled systems - particularly the Philips and National schemes.
Serious proposals for the production of suitable software will be considered favourably as regards preferential pricing on hardware required - and we will sell your program details to other interested users for an agreed commission acting as software brokers.
Please address enquiries to 'Synthesis Support Programme'

Whatever next ?

It would be foolish to pretend that this brief resume of the state of frequency synthesis could provide comprehensive analysis of such a volatile technology. It is intended as a form of introduction to the many enthusiasts and professionals who have missed out on the pace of current advance. Most people haven't yet assimilated the series of digital tuning indicators on offer, let alone given serious thought to the possibility of using MPU synthesis.

Various sections have included reference to additional data and some further reading on offer via our own data library - and we like to think we have gathered a fairly unique set of comparative systems and experiences to be able supply usefully objective comments.

It is absolutely plain that radio set design will be synthesised in the same way as digital techniques have overtaken analogue slide rules. Those who persist in ignoring these developments will either be going bankrupt, or deaf from the sand clogging their ears.

The first generation of approaches are on the whole fairly unimaginative radio wise, and are plainly developments by engineers who are primarily concerned concerned with MPUs, rather than radio designs, since the universal approach in the consumer application has been simply to turn existing designs into 'crystal phase locked loop quartz synthesisers' in accordance with the wishes of the marketing departments of the consumer electronic manufacturers.

No preprogrammed system has taken advantage of many of the features afforded by synthesis, except perhaps in terms of the preset facilities, and rather obvious things like scanning. The basic radio design is untouched.

However, the use of synthesised radio means that some rather more basic techniques are now feasible. Continuously tuned - rather bandswitched radio is possible. The stable first oscillator required for up-conversion is just as easy (or complicated) as the synthesised oscillator for a bandswitched design.
Up conversion to an IF, of say 10.7MHz, is quite simple to achieve. Filters are available in low cost ceramic (with loss of sensitivity towards the lower reaches of LF) or for a few pounds more, in excellent crystal multipole arrays. Perhaps the introduction of 8 pole crystal filters to the mass consumer market would bring about a drastic cutting of the price.
The cost of bandswitching, and tracking the tuning of a LW/MW/SW set must surely begin to approach the cost of a consumer grade crystal filter soon.
The note on the Ambit 5kHz to 10MHz /30MHz receiver chasis elsewhere in this issue of the catalogue may provide a little much needed food for thought. We do not claim to have invented the principle (after all, **expensive** communication receivers have been using the approach for years) but we would like to try and impress the need for investigation of the idea on manufacturers of less exotic radio.

At this point in the proceedings, and without wishing to sound too philosophical, it is worth noting that the British public haven't yet really caught up with technological advances of even five years ago. Technologists tend to drift along in a world of their own, craving for their next fix of innovation long before humble Joe Public has even begun to come to grips with the innovation before last. In fact, most of the world isn't really aware of what hit it when LSI began to appear in everyday applications. About 9 in 10 owners of a digital watch are not competent in the programming and setting procedures required.
Watch makers provide an interesting warning for radio makers, since most Swiss watch makers smugly stated that electronic watches would not replace their centuries old craftsmanship - claiming that electronic watches were too expensive and (LED) impractical. Rather like the noises made by some radio makers in connection with early examples of the synthesised tuner.

Going on at Ambit

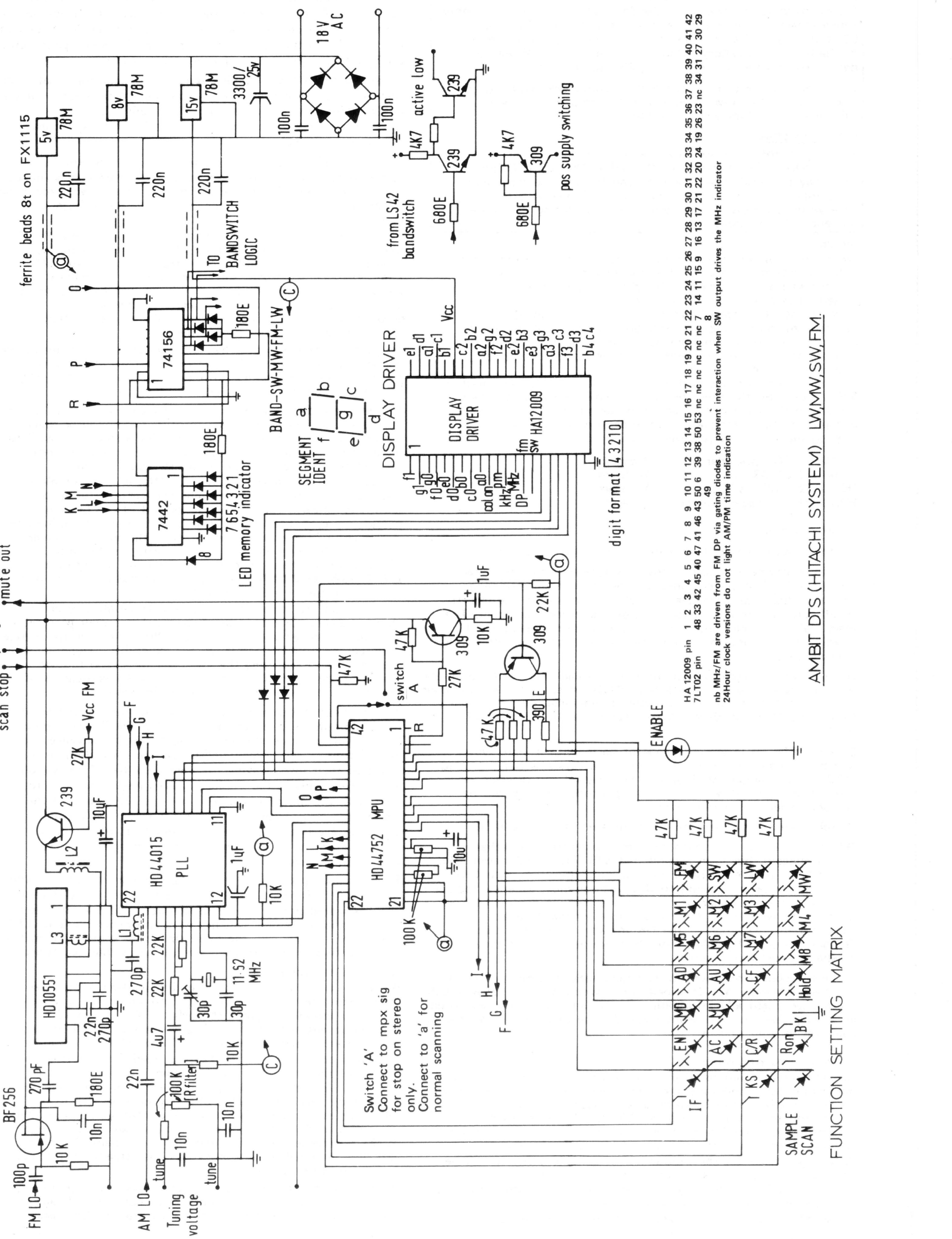
We are basically refining our second generation approaches with custom feature MPU drive. A new range of DC control and tuner modules is going through the first stages of production engineering - including systems intended for some of the more imaginative methods of control outlined above.
A system centred on the LN123/4 is ready for release, when pricing of the constituent parts is more tolerable.
However, since this whole approach has been conceived as a distinctly speciality feature of our general service in the 80's - we may not be able to provide information to enthusiast customers in the same way as we do for our general range, as the synthesiser project is basically a professional service, and thus intended to be supplied as consultancy. ie for money. Lots of it, with any luck.

We hope to be encouraging some enthusiast participation in the subject - and would remind you to check the details in the column alongside headed 'Notes to Students/enthusiasts'.

FREQUENCY SYNTHESIS /8

The Hitachi 44752 system

Ambit's digital tuning system (DTS) is based on the Hitachi system described earlier. The boards contain all the parts shown here, plus a Futaba 7LT02 vacuum fluorescent display tube. The components situated around the PLL tuning voltage outputs are essentially part of the PLL, and must be considered in connection with any other time constants attached to the tuning voltage lines. When the system is commercially available, it will be supplied complete with data relating to the various component ICs, and overall application information.



HA 12009 pin 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32 33 34 35 36 37 38 39 40 41 42
7LT02 pin 48 33 42 45 40 47 41 46 43 50 6 39 38 50 53 nc nc nc nc nc 7 14 11 15 9 16 13 17 21 22 20 24 19 26 23 nc 34 31 27 30 29
49 8

nb MHz/FM are driven from FM DP via gating diodes to prevent interaction when SW output drives the MHz indicator
24Hour clock versions do not light AM/PM time indication

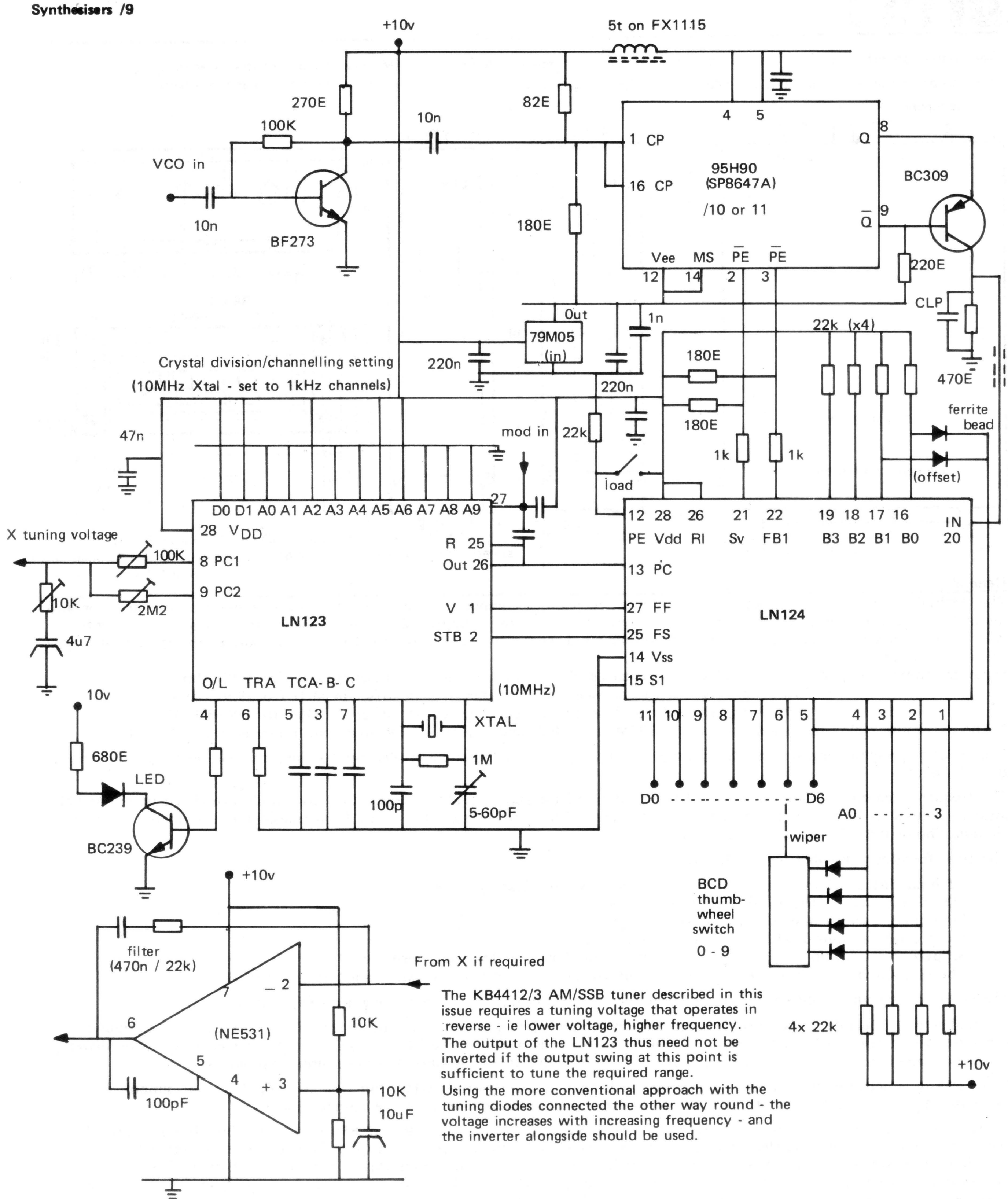
FUNCTION SETTING MATRIX

AMBIT DTS (HITACHI SYSTEM) LW,MW,SW,FM.

Parallel data controlled synthesiser

Mullard LN123/LN124 based unit

Synthesisers /9



The KB4412/3 AM/SSB tuner described in this issue requires a tuning voltage that operates in reverse - ie lower voltage, higher frequency. The output of the LN123 thus need not be inverted if the output swing at this point is sufficient to tune the required range.
Using the more conventional approach with the tuning diodes connected the other way round - the voltage increases with increasing frequency - and the inverter alongside should be used.

Notes

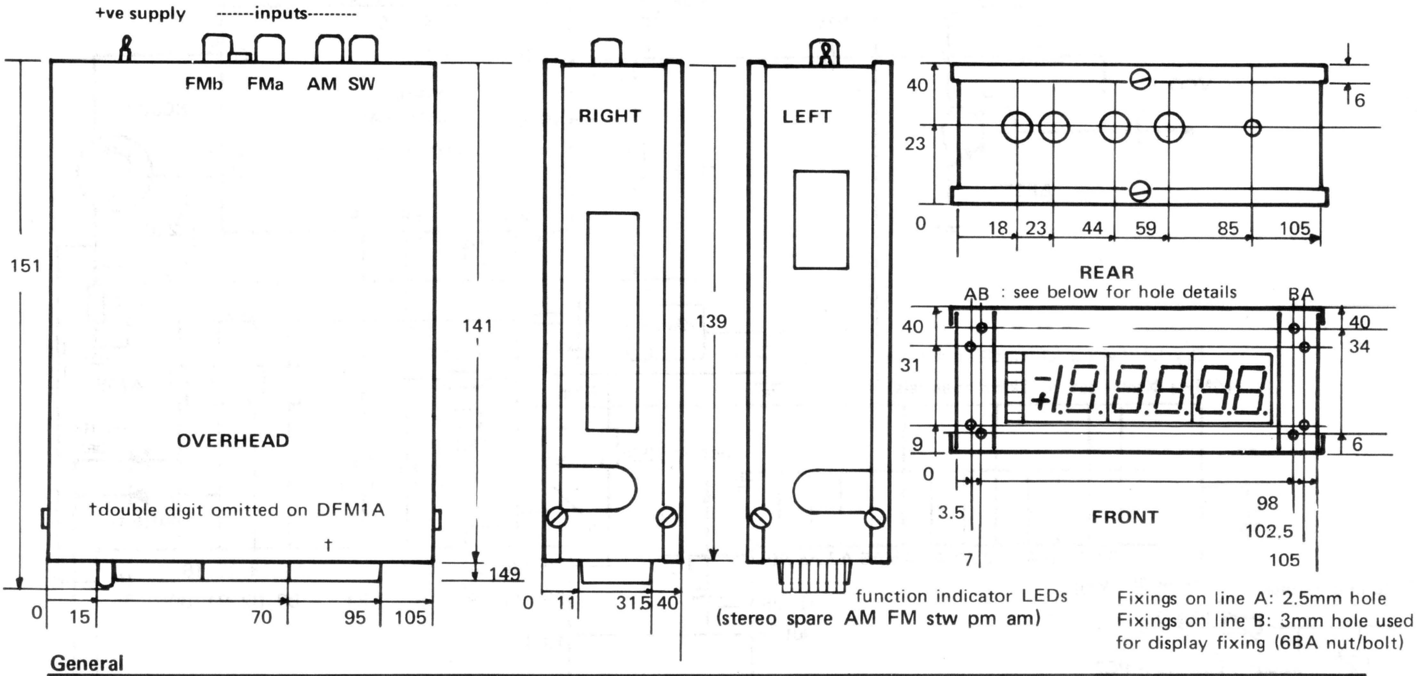
Despite the availability of these exceptionally versatile and pure LSI synthesiser builing blocks, circuits are still appearing using some very long winded CMOS techniques that simply should not be wasting space in the publications concerned. The above has been included as an example of the applications of the forthcoming units available from AMBIT, to forestall any thoughts you may have about doing it the hard way. The above system can use an offset of either 10.7MHz or 32.5MHz to provide 1kHz steps in HF with SSB purity for either receive or transmit purposes. The circuit diagram is offered as an example of the sort of simplicity involved, and must not be taken as the basis for any 'final' circuit, as the various considerations of loop time constants etc. will need more careful attention with regard to the overall system.
Much the same circuit is involved for a 12.5kHz VHF synthesiser with the main changes simply relating to the diode programming and selection of crystal/channelling etc.
Further details of Ambit's universal frequency synthesiser module will be released in due course.

DFM 1

Mechanical specification — **all dimensions shown in mm +/– 1mm**

Top and bottom plate are identical, and held in place with three ST screws.

Material: 1mm mild steel, electroplated
Inputs : via standard phono sockets



General

The AMBIT DFM1 is a multifunction digital frequency readout combined with various time/timer functions. The unit contains internal voltage stabilizers, and runs from a DC source of 7 to 16v at approx 250 - 300mA with the display lit. The entire unit has been designed and constructed with RFI in mind, and may thus be operated in close proximity to virtually any type of radio receiver circuitry.

Separate inputs exist for "AM" - display reads local oscillator frequency less IF offset in kHz (to 3999kHz)
"SW" - display reads local oscillator frequency less IF offset in MHz (to 39.999MHz)
"FM" - display reads local oscillator frequency less IF offset in MHz (to 399.9MHz)*

* The maximum count frequency is typically 175MHz on FM, and 45MHz on SW using the prescalar type MSL2318P

The versatile programming system also permits direct frequency operation, prescaled by 10 using the SW input, and 100 with the FM input.

The frequency counter inputs are buffered by input transistor stages, enabling sensitivities commonly associated with AM/FM local oscillators to be read without further amplification. This also isolates the circuit from transient effects in the counter circuits. Phono type sockets are used.

The time functions available on the DFM1 are

12 hour display clock, with AM and PM indicators, and flashing colon
"On timer" set to desired on time (hours/mins)
"Off timer" set to desired off time (hours/mins)
Sleep timer, a countdown from 1 to 59 Minutes max
Stopwatch mode, up to 12 hours, with either hours/mins, or Mins/secs displayed
The stopwatch mode may be retained running whilst the display is indicating another function.

The status of the unit is displayed by a series of function indicator LEDs on the main display panel.

A signal is also available, which may be internally programmed to provide either an hourly "bleep", or various continuous tones for use in connection with the alarm timers. Provision is made on the PCB for an RS type of reed relay to be driven from the timer circuitry. The connections are basically uncommitted to enable maximum user flexibility.

IF offsetting is achieved by internal diode links, according to the IF offset table; function switching may be simplified by using the Ambit DFM1 Switch PC2, which includes the diode matrix shown on the circuit diagram.

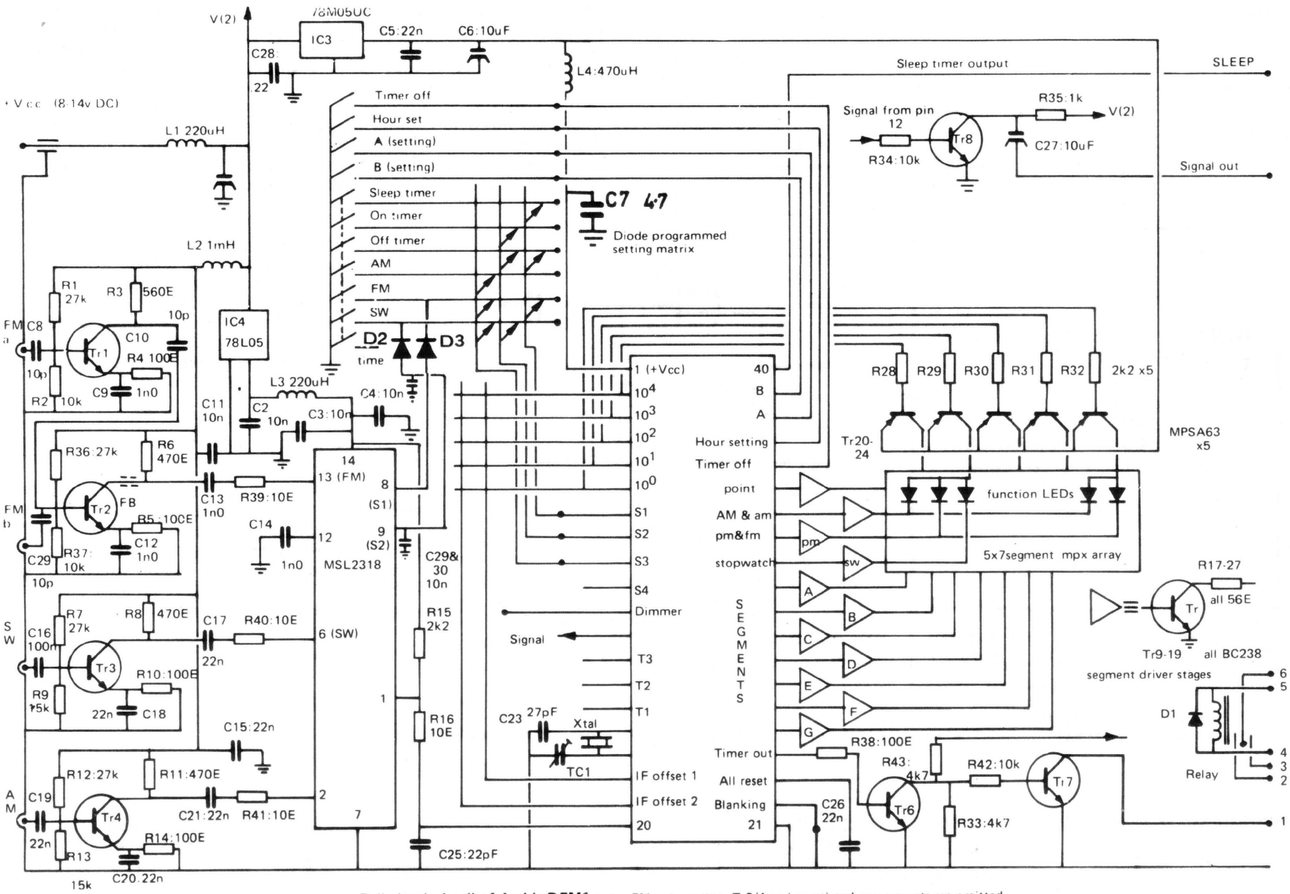
A local oscillator FET buffer board, type BF1, is also available to interface AM and FM local oscillators. The AM and SW oscillator inputs may be paralleled in certain circumstances, and although the SW display format is fully functional down to below longwave, it is sometimes desireable to switch to the AM format, since this is a direct count method, not using the prescalar divider at all, and thereby eliminating a possible source of RF hash at sensitive lower frequencies. The DFM1 screening is sufficient to avoid this problem in all but the most sensitive applications.

The LED displays employ high brightness Telefunken ½" orangered 7 segment displays, and are clearly visible over 30 feet+

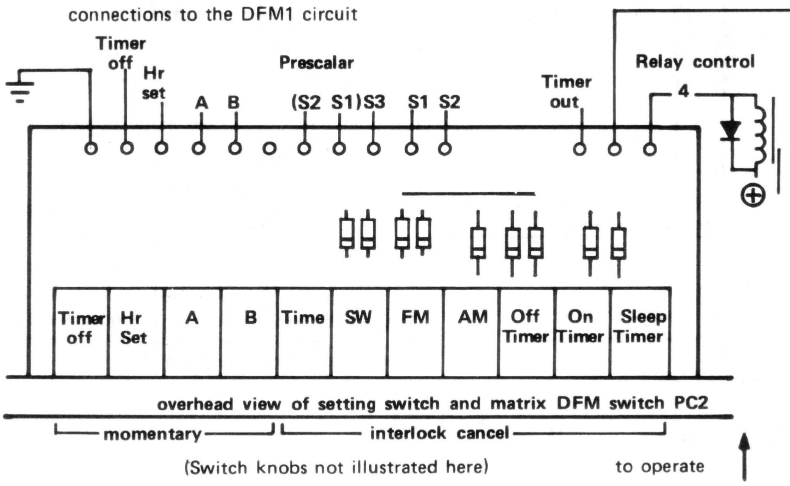
This unit is based on the MSM5524 LSI from OKI, further details of which appear on page 17-20 of the second part of the AMBIT catalogue.

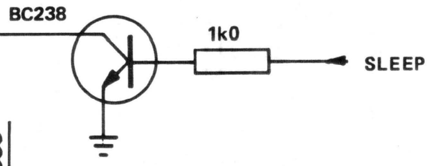
A version type DFM1A, is available less the AM and SW facilities, for FM only tuners. The display is arranged so that this unit is not readily converted to full SW display facilities. The clock and timer functions remain unaffected.

A dimmer facility is also provided, which may be either switched or used in conjunction with a light sensitive resistor to dim the display under low ambient lighting conditions. A further terminal for fully blanking the display is also accessible, for use in applications such as car radio/clocks, where the display 'on' current could be unacceptable.



Full circuit detail of Ambit DFM1 : for FM only option, Tr3/4 and associated components are omitted





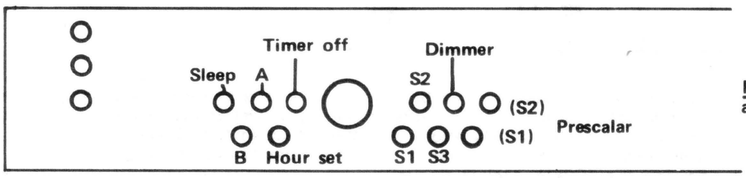
Notes

The various timer and relay functions are left fully open and uncommitted, so that virtually any load configuration may be driven with the appropriate interface circuitry.

The board has provision for a reed relay, either single pole (RS 348-970), or double pole (RS349-355) - although most 6-12v relays up to 100mA, can be driven via Tr7.

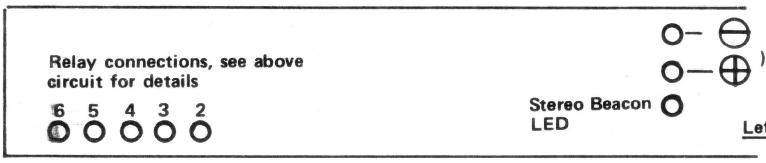
To operate mains powered equipment via the timer, it is essential that a relay with corrected rated contacts be used- and this must be mounted outside the DFM1 case.

Depressing the 'Sleep Timer' switch overrides the ON/OFF timer control of the relay control.



Right hand side output and setting terminations

Depending on the application, connections (S1) and (S2) must be taken via 1000pF feedthrough terminals. Although desireable in some instances, connections other than these need not use capacitive feedthroughs unless RF interference is apparently being radiated from the switch connection lines.



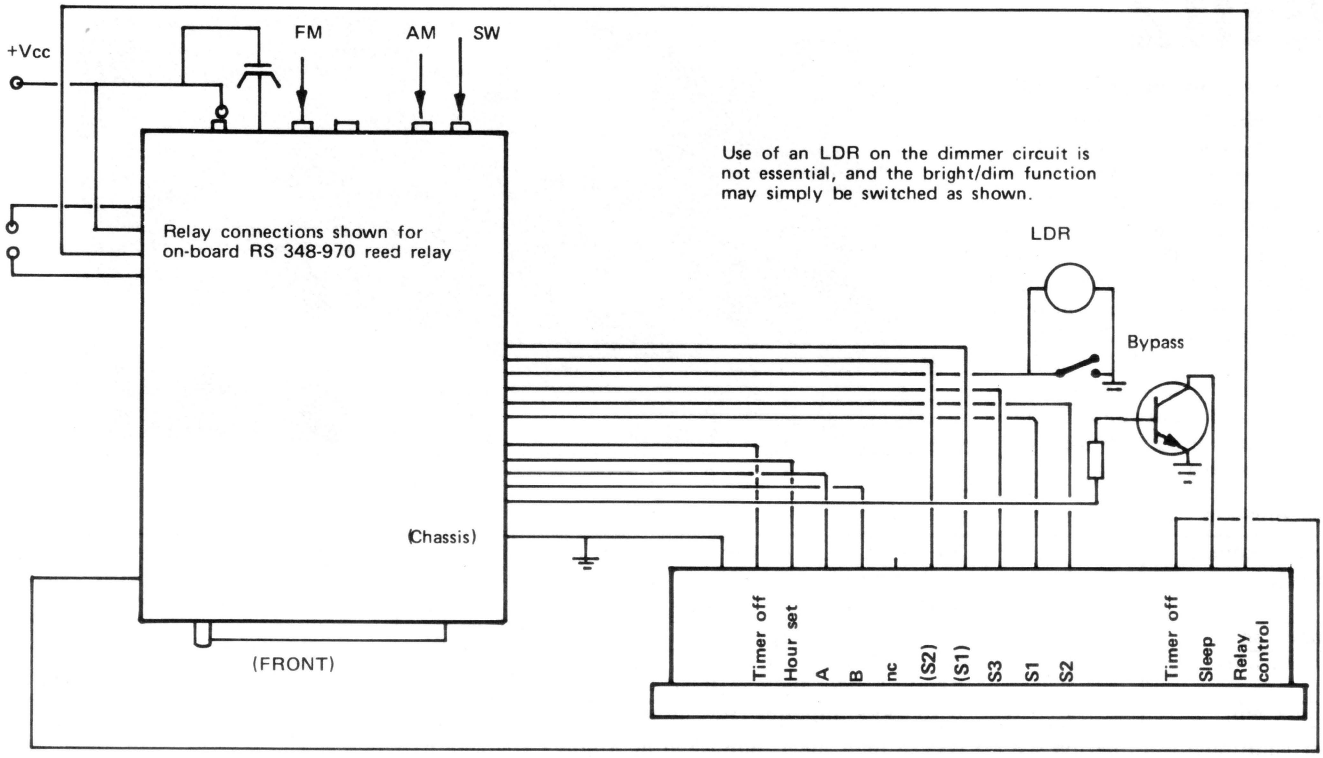
) cathodes of +/– indicators

Left hand side

The fine tune +/- indicators are connected to the common anode of the first digit in DFM1A systems, but may be isolated in applications of the DFM1 where the first half digit is not employed in the numeric readout.

Spare termination holes are provided to enable access to provide alternative switching systems for the timer/audio signal facilities, by those users sufficiently confident of their abilities to interpret circuit diagrams and the information herein.

In keeping with our policy of continual development, we reserve the right to amend and alter any specifications without notice.



Overhead view of the connections to the DFM1 and DFM switch PC2

FUNCTION SETTING AND PROGRAMMING FOR THE DFM1

Reference to the MSM5523/4 data will give full information regarding the logic tables for function selection and IF offset selection. Where the full function setting switch is not used (eg FM only applications) , simpler programming may be acheived using the S1,S2,S3 lines with 'hard wired' connections.
The following tables are intended as user operational instructions for the above system: X = switch depressed, 0 = switch not depressed

DISPLAY SELECTED	Switch A	Switch B	Operation
CLOCK	X	X	Reset clock counter to AM 1.00
	0	X	Set minutes (hours held)
	X	0	Set hours (minutes held)
	0	0	Normal operation of clock function
ON TIMER	X	X	Reset ON timer to AM 0.00
	0	X	Set minutes (hours held)
	X	0	Set hours (minutes held)
	0	0	Retain set time
OFF TIMER	X	X	Reset OFF timer to AM 0.00
	0	X	Set minutes (hours held)
	X	0	Set hours (minutes held)
	0	0	Retain set time
SLEEP TIMER	X	X	Reset sleep timer to 00 minutes
	0	X	Countdown 00 to 59 - 58 -57 etc to 01 minutes
	X	0	Hold setting time
	0	0	Normal operation with display counting down

STOPWATCH FEATURE

The DFM1 timer features include a stopwatch function, where hours, minutes and seconds (up to 12hr59min59sec) may be timed. The display size means that either hours and minutes, or minutes and seconds may displayed - although when the hours/minutes display is used, the seconds register is counting and may be displayed at any time by reverting to the minutes/seconds display. This does not affect the hours count, which is retained unaffected.

SELECT DISPLAY	Switch A	Switch B	Hour set switch	Result
ON TIMER	0	0	Single momentary depression (X')	Hrs/Mins stopwatch displayed
OFF TIMER	0	0	Single momentary depression (X')	Mins/secs stopwatch displayed
(Stopwatch mode)	0	0	Second momentary depression (X'')	Return to timer selected
(Stopwatch mode)	X'	0	0	Commence count
(Stopwatch mode)	X''	0	0	Stop count
(Stopwatch mode)	0	X	0	Reset stopwatch register

The stopwatch count can be left running whilst returning the DFM1 to any of its other funtions - and the fact that this is going on will be shown on the stopwatch function LED indicator - which remains lit as long as the stopwatch mode is running. To reaccess the stopwatch, simply press the appropriate timer select button.

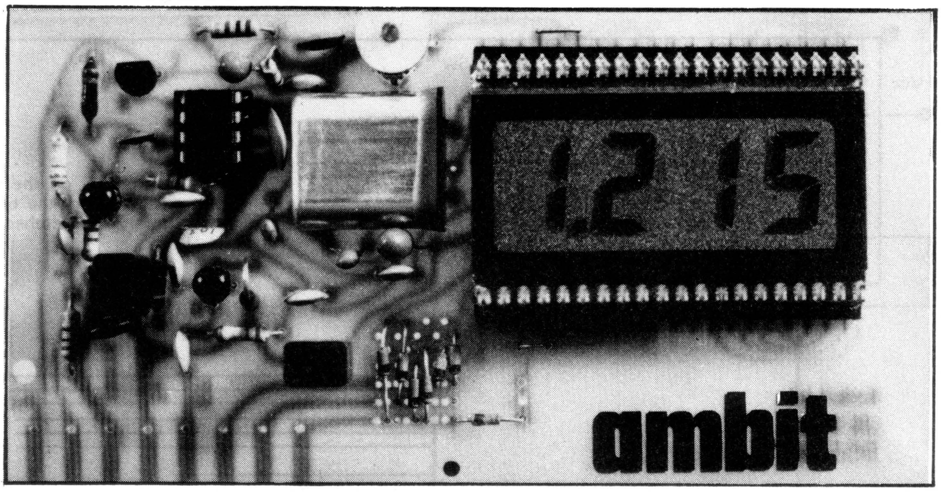
MISC NOTES ON OPERATION

A direct frequency count may be acheived by taking S1, S2 and S3 to ground. (Using the AM input). The display reads 0-3999kHz directly. A prescaled reading may be obtained by using the SW (x10) input, or the FM (x100) input - and taking the appropriate prescalar switch line to ground (S1), or (S2).

DFM2

Direct readout for LW/DF/MW and FM using low power LCD

Using the OKI MSM5526, the DFM2 gives reading in the AM band to a maximum of 2.999MHz (1kHz res.), and 299.9MHz in the FM mode. (100kHz res.) The upper frequency limit for the FM mode is limited by the prescalar maximum input frequency, typically 200MHz.
The direct count facility reads kHz directly when AM is selected and the AM input used. If the FM input is used, the display is then prescaled (x100).



1 2 3 4 5 6 7 8

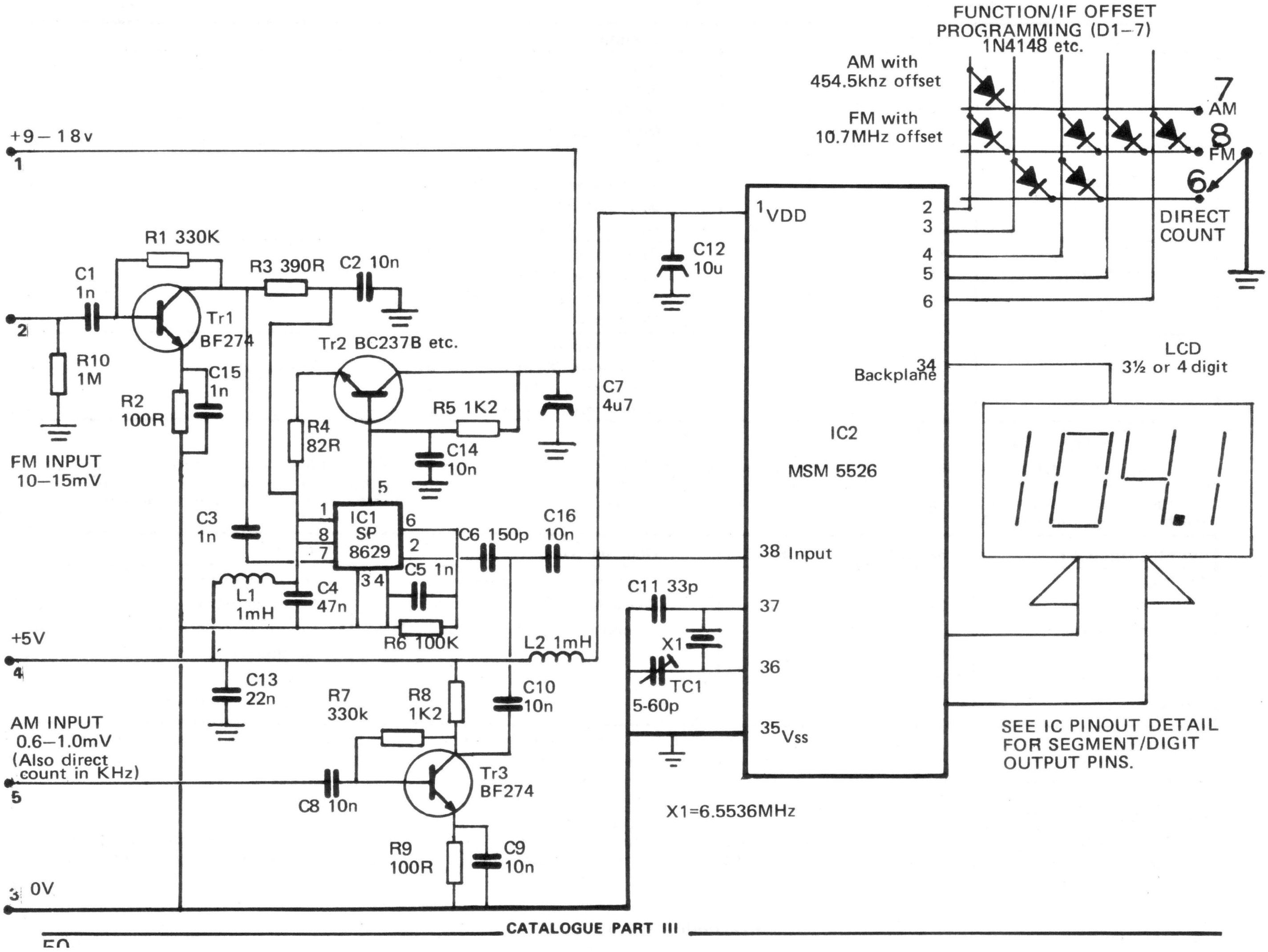
Specification

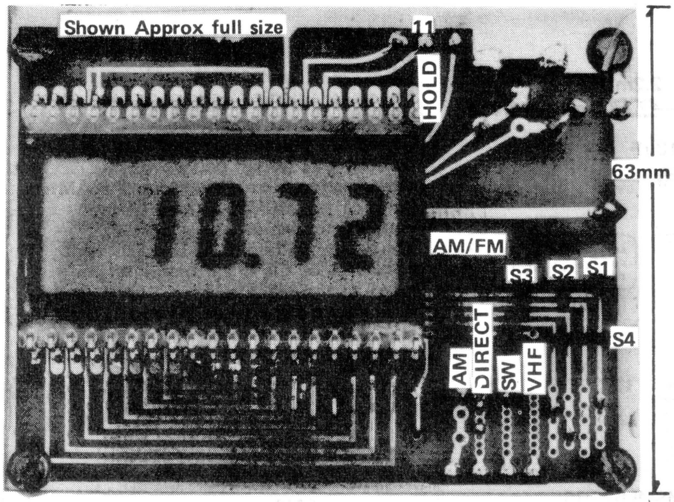
Power required		15-30mA at 5v to 18v (using built in regulator)	Including prescalar
Sensitivity	AM	1mV typically at 1MHz	
	FM	10mV typically at 100MHz	from 50 ohm source
Timebase		6.5536MHz crystal	
Dimensions		63 x 120 x 15mm overall 18mm height digits (reflective display)	
Unit count		0 to 2999 impulses (AM input)	

For additional information see MSM5526 data sheet, published in Ambit Catalogue part two.

Further information on module

The LCD means low power consumption, and low radiation of display noise. The unit need not be screened, but must be mounted at least 5cms from ferrite rod antennas. The PCB is supplied with parts positioning ledgend, and set for standard IF offsets of 454.5kHz, and 10.7MHz. The PCB diode programming area will accomodate any offset chosen from the table shown in the IC data sheet.





Fixing centres 54mm x 75mm

DFM 3

* DIRECT READOUT OF RECEIVED FREQUENCY IN THE MW/LW/RDF BAND with 100Hz resolution
  MW/LW/SW BAND with 1kHz resolution
  VHF FM BAND with 10kHz resolution
* LOW POWER OPERATION WITH LCD, AND DISPLAY LATCH SYSTEM TO DISABLE POWER TO INPUT CIRCUITRY WHEN NOT TUNING
* ALL STANDARD IF OFFSETS AVAILABLE PLUS 2.0MHz and 10.7MHz IN SW MODE
* COVERAGE:
  AM 0 to 3999.9kHz
  SW 0 to 39.999MHz
  VHF 0 to 399.99MHz (with correct prescalar - DFM3 F max is typically 230MHz)
  DIRECT:
  Using appropriate prescaled inputs, the direct count range is as above - but without IF offset

GENERAL

The AMBIT DFM3 is the 'State of the Art' in LSI received frequency display systems. It combines low power, with good resolution (up to the capability of most non-synthesised local oscillators), and the convenience of a static display - with a unique freeze facility. RFI is thus reduced to negligable proportions.

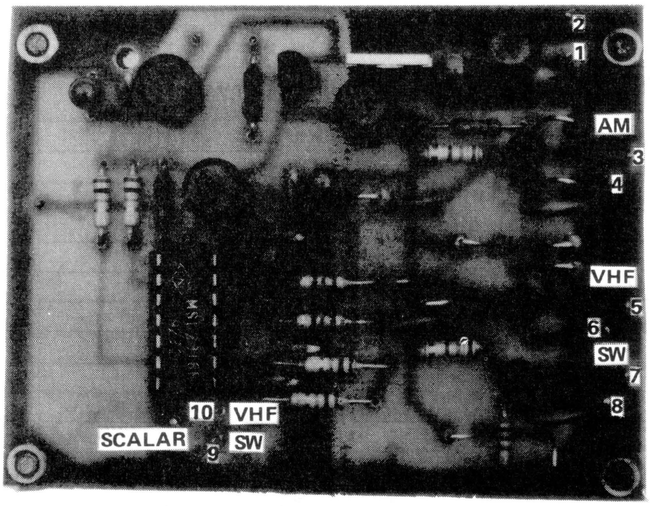
For applications requiring only the AM facility (Marine RDF), the display board is all that is needed - consuming 3-5mA. AM only may also be used in conjunction with the second tuneable IF of multi conversion receivers, providing 100Hz resolution.

The second board includes all necessary input buffers and prescalar functions for operation in the SW and VHF ranges.

IF offset options are preset via diode matrix programming to the following standard values:

AM 455kHz
SW 455kHz
VHF 10.7MHz

Other values available (see over) to special order - or user may reprogramme as desired.

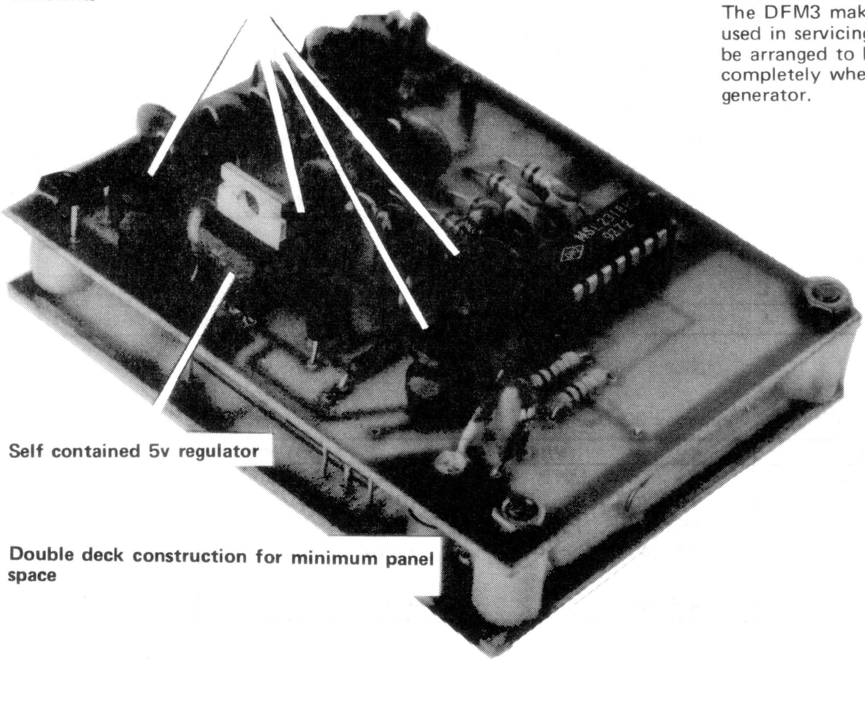


Input buffer/amps for all ranges

Applications

The DFM3 resolution represents the maximum practical degree that can be used with non-synthesised local oscillators. It is suited to all types of broadcast receiver, communications receivers etc.
The 10.7MHz offset in the SW range is particularly interesting, since this permits the use of 10.7MHz up-conversion types of IF in low cost broadcast receivers, where the lack of a suitable type of display medium has previously been the inhibiting factor.
The DFM3 makes an ideal readout for test equipment used in servicing radio equipment, since the offsets may be arranged to be switched externally - or removed completely when used in conjunction with a signal generator.

Choke decoupling of prescalar switching transients



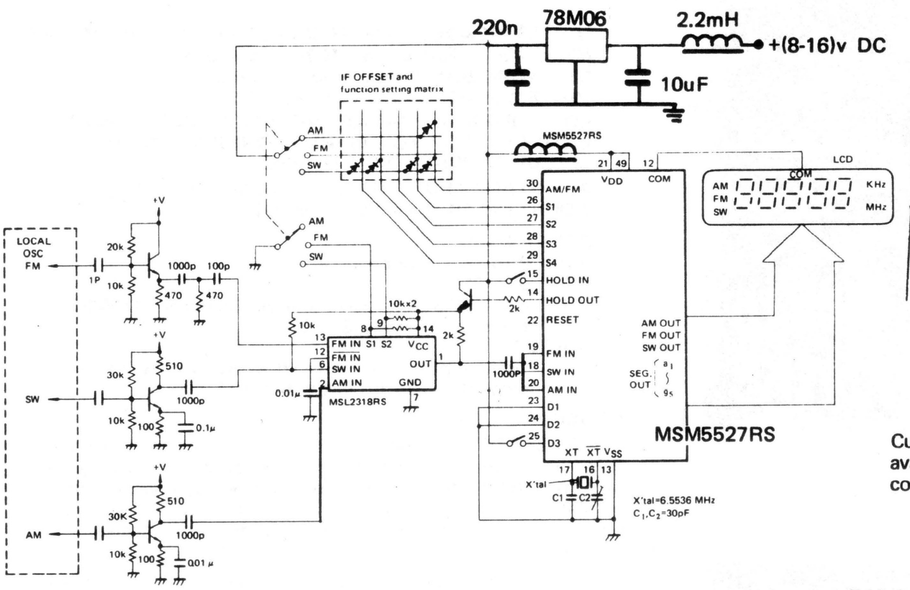
Self contained 5v regulator

Double deck construction for minimum panel space

1	12v supply
2	Power supply ground
3	AM range input
4	AM input ground
5	VHF range input
6	VHF input ground
7	SW range input
8	SW input ground
9	Select SW prescalar
10	Select VHF prescalar
11	Hold select

Complete circuit diagram

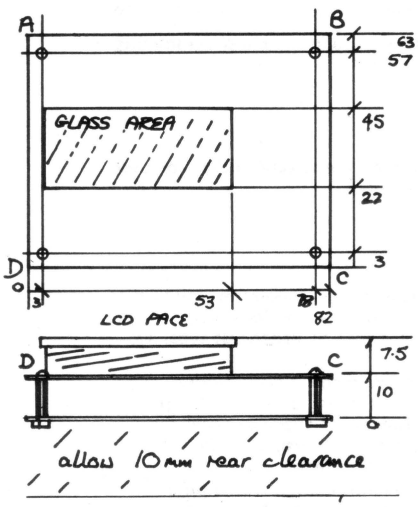
(LCD connections are shown grouped for convenience)



Notes
The basic circuit used incorporates the AM function of the prescalar IC. Where continuously updated display readout is required, and power consumption is at a premium, the AM input of the MSM5527 may be fed with a sinewave directly. (pin 20)
The AM must be AC coupled to take advantage of the linear biasing at the input to the MSM5527.
A sinewave of between 50 and 100mV RMS is sufficient to drive the counter in this fashion.

Customized DFM3 adaptations are available on special request - at additional cost.

Dimensions in mm, and mounting details



The unit is supplied with mounting spacers

The DFM6 version is a fluorescent display DFM with similar specification to the DFM3 - other than the additional requirements for powering the display. (Futaba 7LT02).
3vAC is required at 60-80mA for the filament, together with –12 to –20 v for the illuminating voltage, at approx. 20-30mA max (all lit).
A DC–DC/AC converter will be made available for single rail DC operation.

Table 1

H: V_{DD}
L: Open or V_{SS}

SELECT INPUT					DISPLAY SELECT	IF OFFSET VALUE
AM/FM	S_1	S_2	S_3	S_4		
H	L	L	L	L	AM	–455 KHz
H	L	H	L	L		–260 KHz
H	L	L	H	L		–450 KHz
H	L	H	H	L		–261 KHz
H	L	L	L	H		–468 KHz
H	L	H	L	H		–470 KHz
L	L	L	L	L	FM	+10.7 MHz
L	H	L	L	L		+10.63 MHz
L	L	H	L	L		–10.7 MHz
L	H	H	L	L		+10.66 MHz
L	L	L	H	L		+10.74 MHz
L	H	L	H	L		+10.77 MHz
L	L	H	H	L		–10.63 MHz
L	H	H	H	L		–10.65 MHz
L	L	L	L	H		–10.66 MHz
L	H	L	L	H		–10.67 MHz
L	L	H	L	H		–10.68 MHz
L	H	H	L	H		–10.71 MHz
L	L	L	H	H		–10.74 MHz
L	H	L	H	H		–10.75 MHz
L	L	H	H	H		–10.77 MHz
L	H	H	H	H		–10.78 MHz
H	H	L	L	L	SW	–455 KHz
H	H	H	L	L		–468 KHz
H	H	L	H	L		–2.0 MHz
H	H	H	H	L		–10.7 MHz
H	L	L	H	H	F.C.	No IF OFFSET
H	L	H	H	H	COUNTER	No IF OFFSET. Internal presetting inhibited.

General DFM3 specifications (Vcc 12v - Ta 25°C. Vcc max 30vDC)

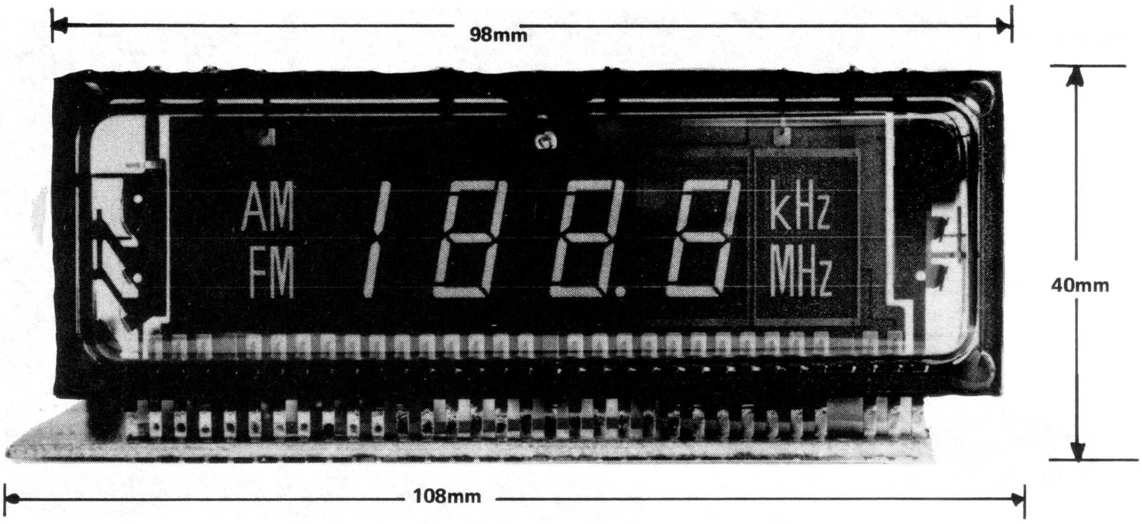
FUNCTION	AM	SW	• VHF
Frequency range	0 - 3999.9 kHz	0 - 39.999 MHz	0 - 230.00MHz min
Sensitivity (50 ohm)	10-20mV PD	10-20mV PD	10-30mV PD
Power consumption (12v)	65mA	65mA	65mA
Display frozen	7.5mA	7.5mA	7.5mA
AM only operation	4.2mA		
Input voltage range	4.5v pp	4.5v pp	4.5v pp

ambit
INTERNATIONAL

DFM 4

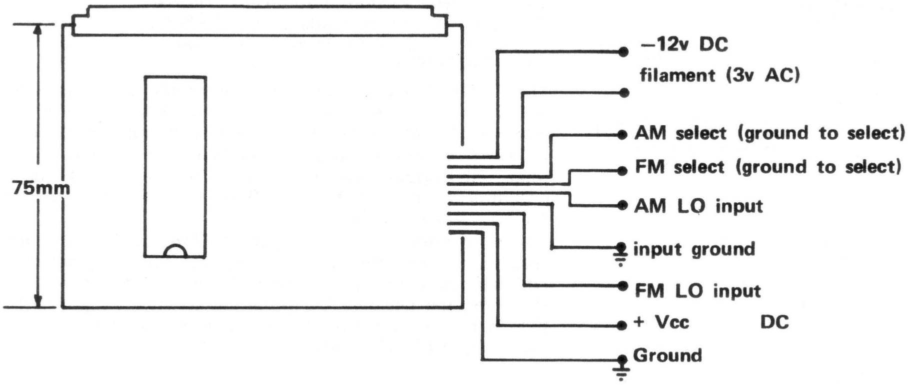
Fluorescent display, static drive LW/MW &FM frequency readout

Display format



The DFM4 uses the same basic circuit as the DFM2 - except that the MSM5525 and a fluorescent display are used instead of the LCD and MSM5526. The temperature range of the fluorescent display enables this type of unit to be used in far more demanding enviroments (cars etc) - where an LCD would be unsuitable.

Overhead view of connections



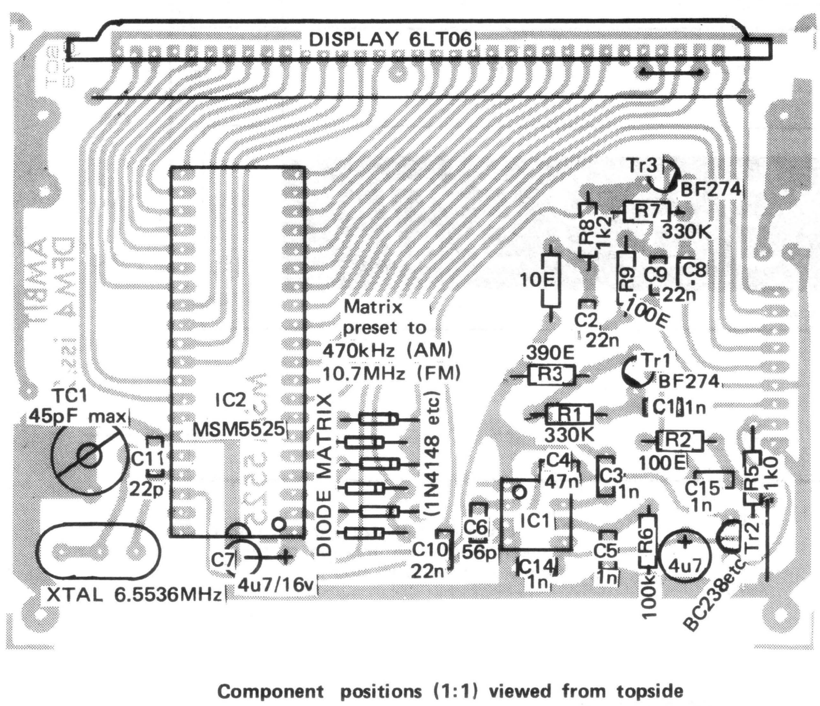
Notes on the DFM4

The same basic performance parameters apply to the this unit as for the DFM2.

The main additional requirement is for a 3v AC supply at approx. 50mA for the filament of the 6LT06 readout tube.

The light from the tube is basically green, although it may be filtered with either green or blue to provide a wide range of possible tints.

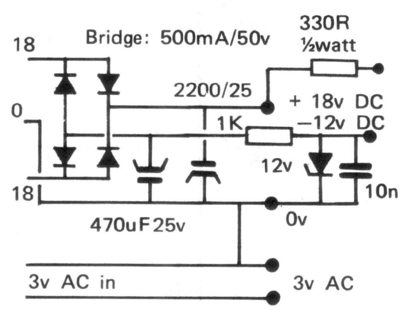
A negative rail of approx. 12v must be added between the filament and the circuit ground to provide drive for the tube itself. It may be connected to either side of the filament supply.



Component positions (1:1) viewed from topside

A suitable transformer, type DFM4 TRANSFORMER (Simple, eh?) is available with all necessary windings to drive the DFM4, and the tuner associated:

18-0-18v
3v AC
120-0-120v AC input



NB The transformer supplied is marked with the DC voltage after rectification. Thus 18v DC = 12.7v AC
The bridge rectifier may require 100nF ceramic (or sim) placed across each diode, if used in proximity to an antenna input stage.

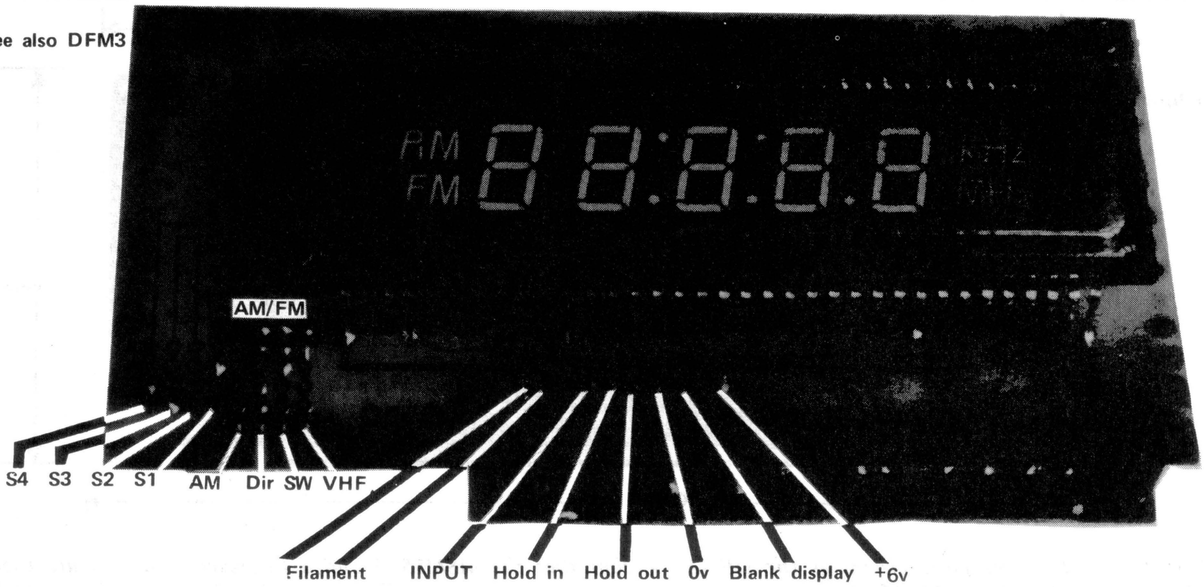
Power Supply based on the DFM4 transformer

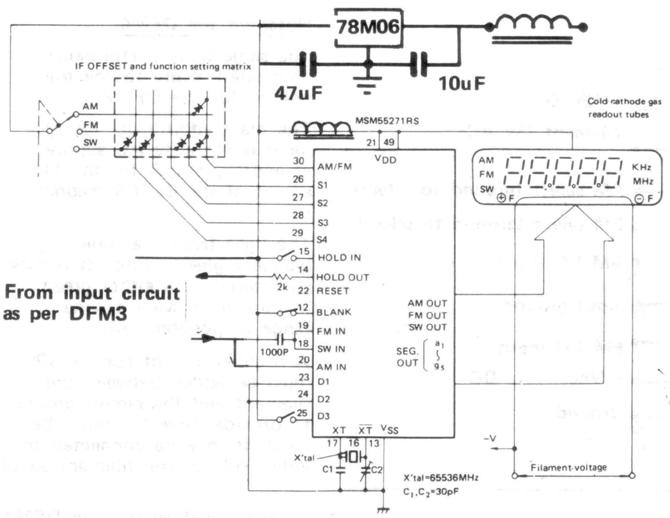
ambit
INTERNATIONAL

DFM 6

with large, easy to read green fluorescent display
AM (100Hz res. to 3999.9MHz), SW (1kHz). VHF (10kHz)

See also DFM3

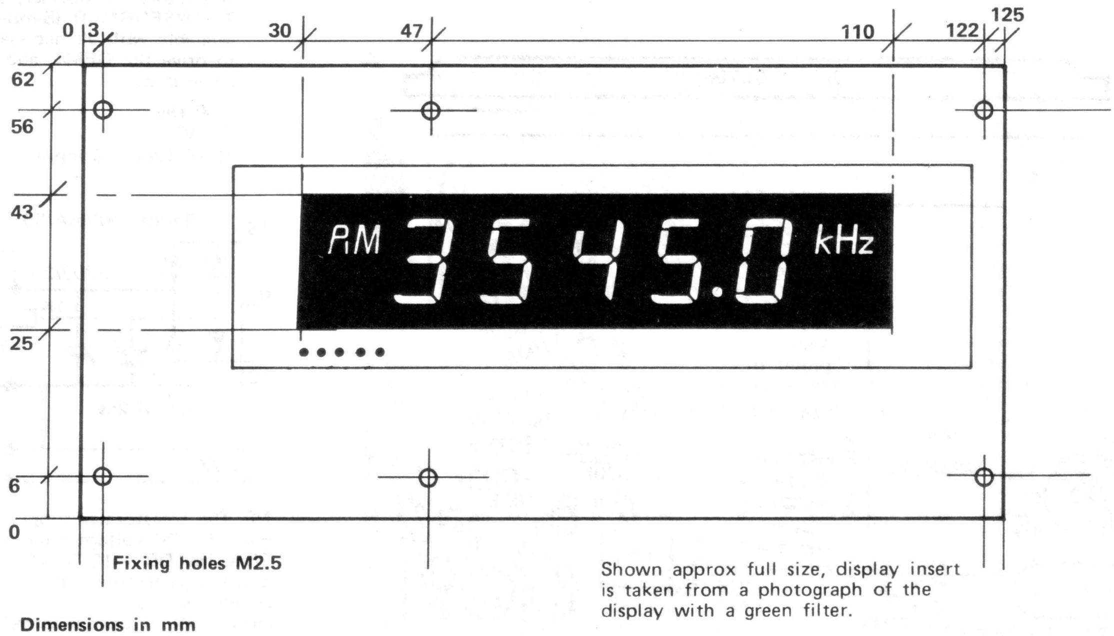




THE DFM6

The input circuitry and general specification of the AMBIT DFM6 is the same as the DFM3 - except that the DFM6 uses a Futaba vacuum fluorescent display.
Reference should be made to the DFM3 data sheet for details of the IF offset programming and input details - and to the OKI MSM5527/55271 data sheets.

The DFM6 operates over a broader temperature range (-20 to +70^{o}), and may thus be better suited to harsher enviroments that may be encountered in communications applications.
A 3v AC supply is required, together with a negative supply (max -20v) applied to the filament for the illumination potential at between 12 and 20mA max. for the grid, and 1.2 to 2.4mA per full digit.
The transformer supplied for the DFM4 provides the necessary voltages to drive this display - and the PSU diagram on the DFM4 data sheet may be used in conjunction with this unit. Reducing the negative DC potential to -12v via a regulator (which may be a simple zener diode/pass transistor) will ensure long life (100,000 hours +)

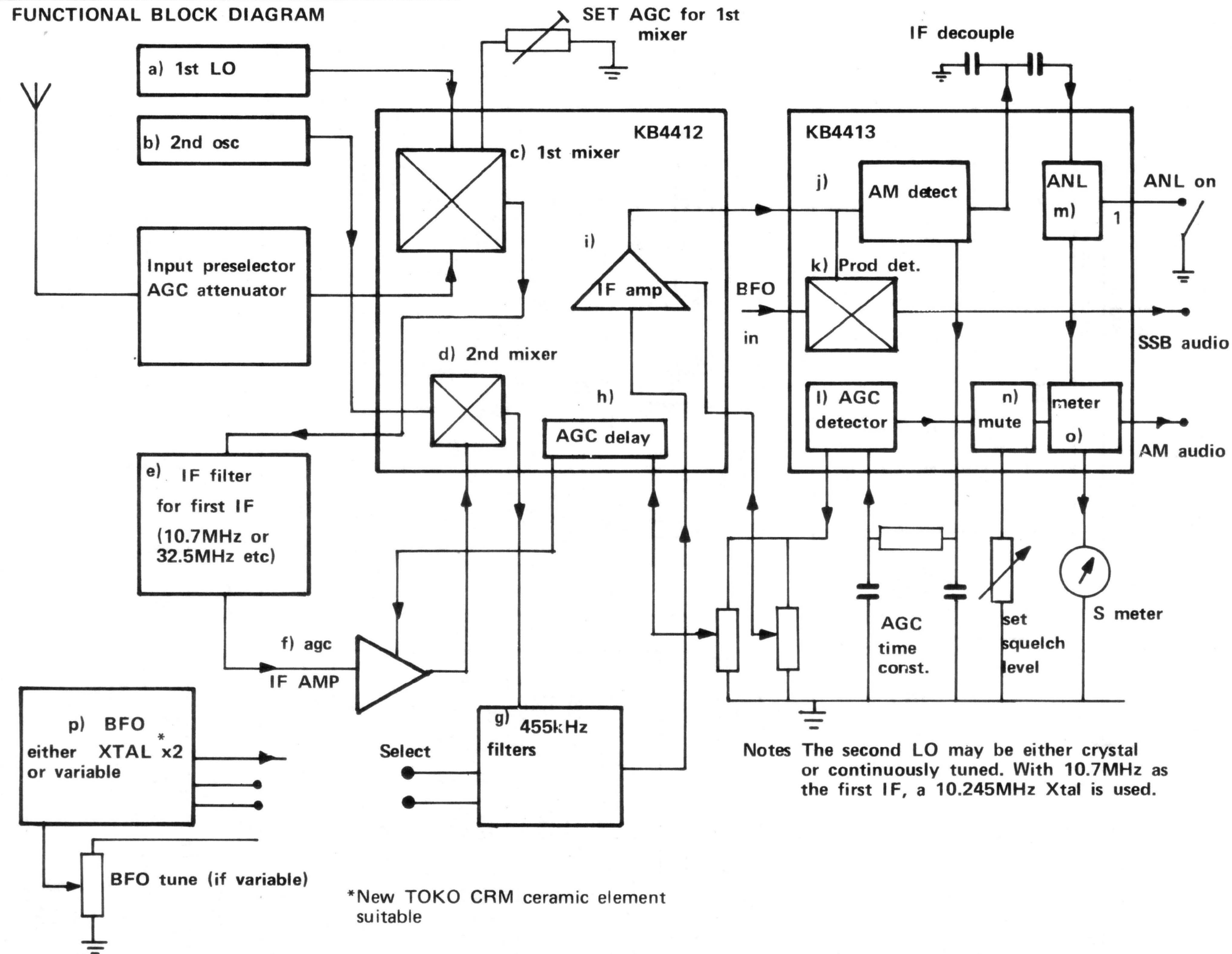


Shown approx full size, display insert is taken from a photograph of the display with a green filter.

AMBIT 4412/3 SYSTEM - 7.5kHz to 30MHz

AM, SSB, CW radio tuning block

FUNCTIONAL BLOCK DIAGRAM



General

Ever since the first of this catalogue series appeared, with details of the block approach to making a continuously tuned HF receiver system, requests have been sent in for "more details".
We had hoped that the original idea would have provided the basis for some of the more experimentally minded to have a go, but to date - we have recieved no requests for component sponsership along the lines of our oft repeated offer, for this type of project.
Continuously tuned HF recievers are nothing very new, but the approach in terms of the components available is evolving very rapidly. One of the major problems of such a receiver is the oscillator stability requirement. For unlike a narrow band system involving much switching and tracking, the oscillator is designed to tune (1st IF + 0 to 30MHz) to give 0-30MHz output at the first mixer. The scope for oscillator noise/hum/jitters is thus very substantial - and can only be met with a fully synthesised design. Words on synthesis appear elsewhere in this issue, but at the time of publication, we do not yet have full details of our HF synthesiser (based on the Mullard L123/4 system).
However, the First LO input is fully accessible on the board, and so there is nothing to stop this system being used as the "back end" of a more conventional approach, using a switched coilpack.
The on-board 1st oscillator is a balanced FET arrangement, which is sufficiently stable to provide coverage from 5kHz to 10MHz, using a 10.7MHz first IF. (Osc tunes 10.7 to 20.7MHz).
Using the Ambit DFM3 or DFM6, a 1kHz resolution readout is possible, as these counters have a 10.7MHz offset on SW.
One of the many pseudo synthesised stablizer circuits may be used to provide frequency correction of the oscillator, since long term stability is not sufficient for communications SSB applications, although sufficient for the interested enthusiast to use as the basis for experimental work.

Various options are left available, such as the use of either a crystal controlled, or VFO 2nd oscillator. If the first oscillator is locked onto 1kHz steps, interpolation may be desireable at the second LO.
The board may also be used a 2-3MHz, or 3-4MHz tuneable IF, in conjunction with a suitable coilpack front end. The DFM3/6 then provides 100Hz resolution in the "AM" mode. (Max. 3.9999MHz). In such a case, the first IF filter will be omitted, with filtering at 455kHz instead.

The 455kHz filtering uses two switched ceramic ladder units, (AM/SSB).

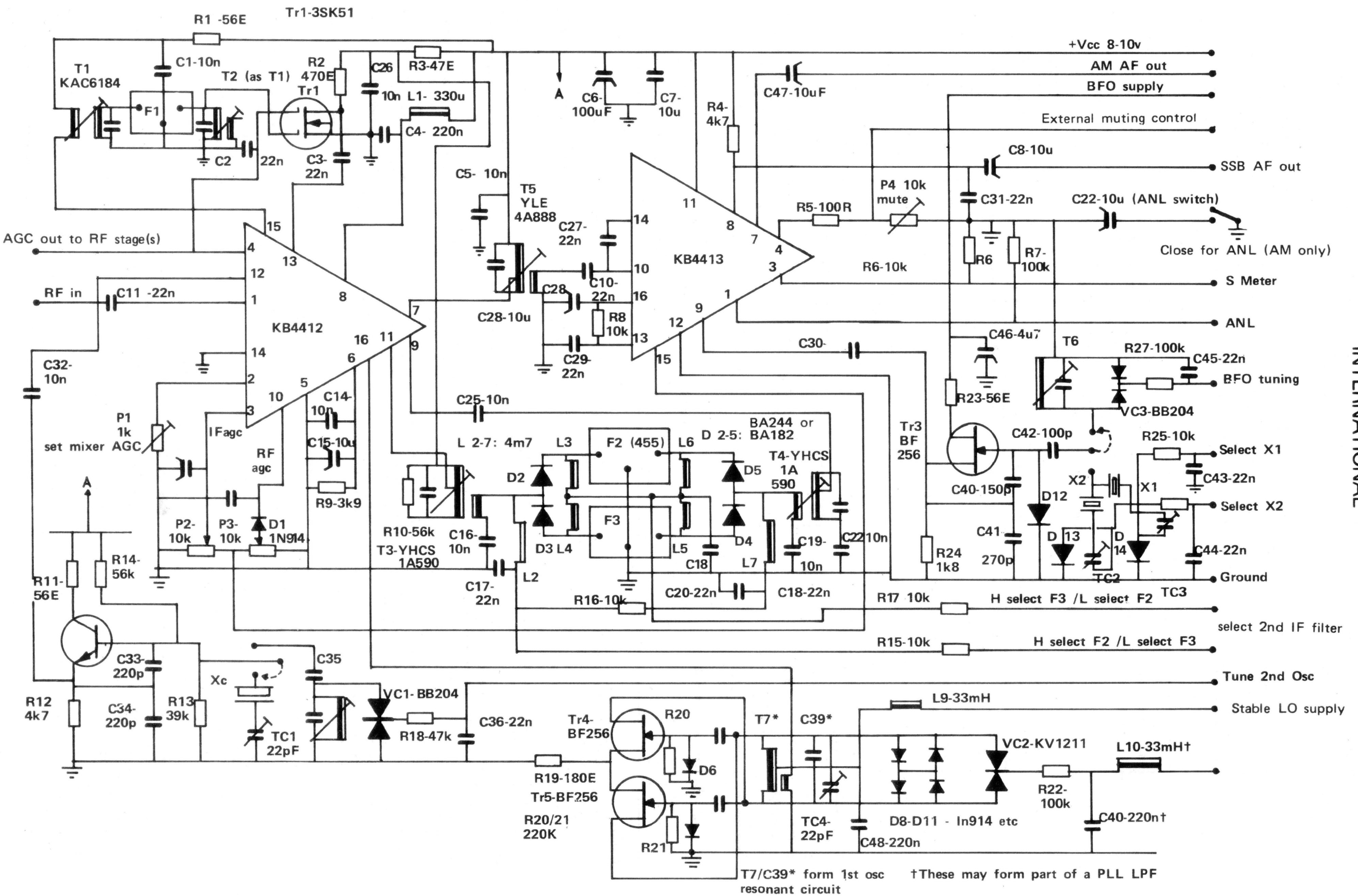
If coverage were restricted to 5kHz to 4MHz - by adding in a padding capacitor at C39 - the longterm stability of the on-board 1st oscillator is excellent, and the unit could be used as the basis for a very high performance LW/MW RX.

So, in other words, the board is offered as a flexible building block providing RF/IF signal processing functions, which are very largely programmable by the user.
The lowest frequency is determined by the bandwidth of the first filter, so the 8pole TOYO series filters will enable reception of signals of down to about ½ the nominal filter bandwidth. Not that much goes on around there! A narrow 1st filter also provides immunity against intermodulation and overloading in the subsequent stages.
However, the on-board first mixer is designed for use with some preceeding selectivity - as the effects of simply hanging on an HF antenna will be overpowering. Alternatively, the first mixer may be placed outside, using a passive mixer such as the SBL1 (or higher level) so that the 3rd order IM intercept is extended.
If used in the LW/MW bands, approx. 10 feet of antenna, and a well designed low pass filter with Fc of 2MHz enables the unit to be used with no additional tuning required. Locations and the presence of strong local transmissions will be a major consideration for some users. One of the neatest ways of dealing with a case of an MW transmitter in the locality is simply to trap it out in the antenna filter line.

Specifications

In its present form, the unit is offered as the basis for a wide variety of applications. However, when configured as a basic 0-10MHz receiver, with 8 pole 10.7MHz first filter, the input sensitivity for AM is 1uV, and 0.6uV for SSB/CW.
Some 100dB of useful AGC is available, but the first mixer overload reduces the practical range to about 70-80dB.

The unit is being developed further, and more information will be made available in due course, meantime, the basic PCB and all components are available. We welcome reports on applications and systems developed using this unit as a basis.


Tr1-3SK51
R1 -56E
T1
KAC6184
C1-10n
T2 (as T1)
R2
470E
Tr1
C26
10n
R3-47E
L1- 330u
C4- 220n
F1
C2
22n
C3-
22n
+Vcc 8-10v
AM AF out
BFO supply
External muting control
SSB AF out
A
C6-
100uF
C7-
10u
C47-10uF
R4-
4k7
C8-10u
C31-22n
C22-10u (ANL switch)
Close for ANL (AM only)
S Meter
ANL
AGC out to RF stage(s)
RF in
C11 -22n
KB4412
C5- 10n
T5
YLE
4A888
C27-
22n
C28-10u
C28
C10-
22n
R8
10k
C29-
22n
KB4413
R5-100R
P4 10k
mute
R6-10k
R6
R7-
100k
C32-
10n
P1
1k
set mixer AGC
IFagc
RF
agc
C14-
10n
C15-10u
R9-3k9
P2-
10k
P3-
10k
D1
1N914
C25-10n
L 2-7: 4m7
L3
F2 (455)
L6
D 2-5:
BA244 or
BA182
T4-YHCS
1A
590
D2
D5
F3
D3 L4
L5
D4
C18
L7
C19-
10n
C22 10n
R10-56k
T3-YHCS
1A590
C16-
10n
C17-
22n
L2
R16-10k
C20-22n
C18-22n
C30-
C46-4u7
T6
R27-100k
C45-22n
BFO tuning
VC3-BB204
R23-56E
Tr3
BF
256
C42-100p
R25-10k
Select X1
C43-22n
X2
X1
Select X2
C40-150p
D12
C41
270p
R24
1k8
D 13
D
14
TC2
C44-22n
Ground
TC3
R17 10k
H select F3 /L select F2
select 2nd IF filter
R15-10k
H select F2 /L select F3
Tune 2nd Osc
Stable LO supply
A
R14-
56k
R11-
56E
C33-
220p
R12
4k7
C34-
220p
R13
39k
Xc
TC1
22pF
C35
VC1- BB204
R18-47k
C36-22n
Tr4-
BF256
R20
D6
R19-180E
Tr5-BF256
R20/21
220K
R21
T7*
C39*
L9-33mH
VC2-KV1211
L10-33mH†
R22-
100k
C40-220n†
TC4-
22pF
C48-220n
D8-D11 - In914 etc
T7/C39* form 1st osc
resonant circuit
†These may form part of a PLL LPF

Ambit EF5804 series: Band two and general VHF coverage voltage tuned front ends

Background

The original EF5800 series is getting on for its third year of production - during which time it has been developed considerably to absorb new component developments, in the shape of better MOSFETs, and now most recently, very accurately matched sets of tuning varicaps.
The basic approach has been based on concerntrating on the design of a tuner to suit the majority of English conditions - as opposed to most commercial efforts which are obliged to take into account the very high signal levels that are frequently found in continental and North American applications. The aim is thus for high gain and very low noise - and whilst this may be also achieved in conjunction with huge dynamic range, the techniques are very expensive in domestic applications. In the UK, most of listeners are quite a distance from an FM transmitter, so gain is the prime consideration, together with low noise. But gain without good RF selectivity is a problem even in the UK, since there are still too many public service transmissions in Band 2 around the 100MHz region. These are frequently far more local than the genuine broadcast signals, and thus place an unfair strain on the recieving system.

Definitions of tunerhead design

1 Gain

The gain of the front end is defined as the IF output voltage, Vo, across a specified load resistance, divided by half the antenna input voltage, Vs. In dB:

Gv = 20 log(2Vo/Vs) dB

The more common power gain term is related to the antenna resistance, Ra, and load resistance, RL:

Gp = $10\log(4Vo^2.Ra/Vs^2.RL)$ dB

so Gv = Gp + 10log(Ra/RL) dB

2 Noise

Factors relating to noise behaviour are covered elsewhere in this catalogue edition.(See the SBL1 DB mixer and application notes.)
The noise figure of the front end is defined as being the ratio of the input power S/N, compared with the output power S/N, and is usually expressed in dB:

$$NF = \frac{\text{input power S/N power ratio}}{\text{output power S/N power ratio}}$$

The noise figure is the prime function of the amplification devices employed, and the extent to which they are correctly matched to achieve optimum NF.

3 RF bandwidth

The RF bandwidth is simply the difference between two input frequencies at which the input to the mixer stage has dropped by 3dB. The effects of AGC should be disabled for this test. Since the effects of coupling between bandpass tuned circuits are frequency dependent, the RF bandwidth will tend to vary slightly from one end of the band to the other.

4 IF bandwidth

The IF bandwidth of a front end is the difference between two frequencies, either side of the IF centre frequency, at which the IF output has dropped by 3dB. The IF bandwidth is usually 300kHz or so, being determined by either a single or double tuned IF filter stage.
The RF and IF bandwidths will tend to be level dependent, since saturation effects on the cores will vary the Q and the tuned frequency under severe overload conditions. The effect will get progressively more noticable as the signal proceeds through the amplification stages, although the AGC levels available in the EF5800 series of front ends will cope with inputs of up to 200mV. The EF5804 has been provided with a closed loop PIN attenuator system, and can take up to 1 or 2 volts before detuning becomes serious.

5 Spurii

Much the same considerations apply here as are discussed in the section on the SBL1 and its applications. The major problems for the VHF tunerhead are created by intermodulation products of strong input signals, and the local oscillator.
The prime problem spot is the product between twice the input and twice the LO frequencies:

example Fantenna = 90MHz, Fosc = 100.7MHz, so twice the LO is 201.4MHz.
And 201.4 - 10.7 = 190.7MHz, implying the frequency of the strong interfering signal is (190.7/2) = 95.35MHz.
Which you will see is half the IF away from Fantenna.
The answer is design with a narrow RF bandwidth, and a pure LO signal. Since this spurii is one of the major rpoblems in good front end designs, the output from the LO may be fed via a simple LPF aimed at trapping out the 2nd harmonic.
However, most of the harmonic generation will take place in the mixer device itself, although every little helps ! A simple ferrite bead placed on the appropriate leads can make a very surprising contribution to the suppression of this type of signal.

As with all FM measurements, the figures derived will vary according to the power levels injected. The term used in nearly all the measuremetns is the dimensionless dB - and it may be very misleading to quote a spurious level at a relatively low input level.

6 Double beat intermodulation products

When two strong antenna signals interact in the amplification stages of the front end, various products occur:

Frequency 1 = F1
Frequency 2 = F2
LO = Fo

Double beat (DB) interference is worst when the results are:

Fif = nF1 + mF2 + pFc
(Where n,m & p are positive or negative integers)
The products that will fall in the band (that is, 88-108) are:
Fif =Fo - (2F1-F2)
or =Fo - (2F2-F1)
or =2Fo - (F1+F2)

7 Continuous beat intermodulation products

Where two strong input signals occur such that their fundamental frequencies, or harmonic frequencies, are spaced by the IF, the intermod products are constant, and independent of the Lo. The effects will vary with tuning however, since the RF bandwidth varies as well as the LO.
The intermod usually occurs in overloaded amplification stages, and so the problem can be assisted by the effects of AGC - if the desired signal is of sufficient amplitude to cause AGC onset.

8 Image frequency

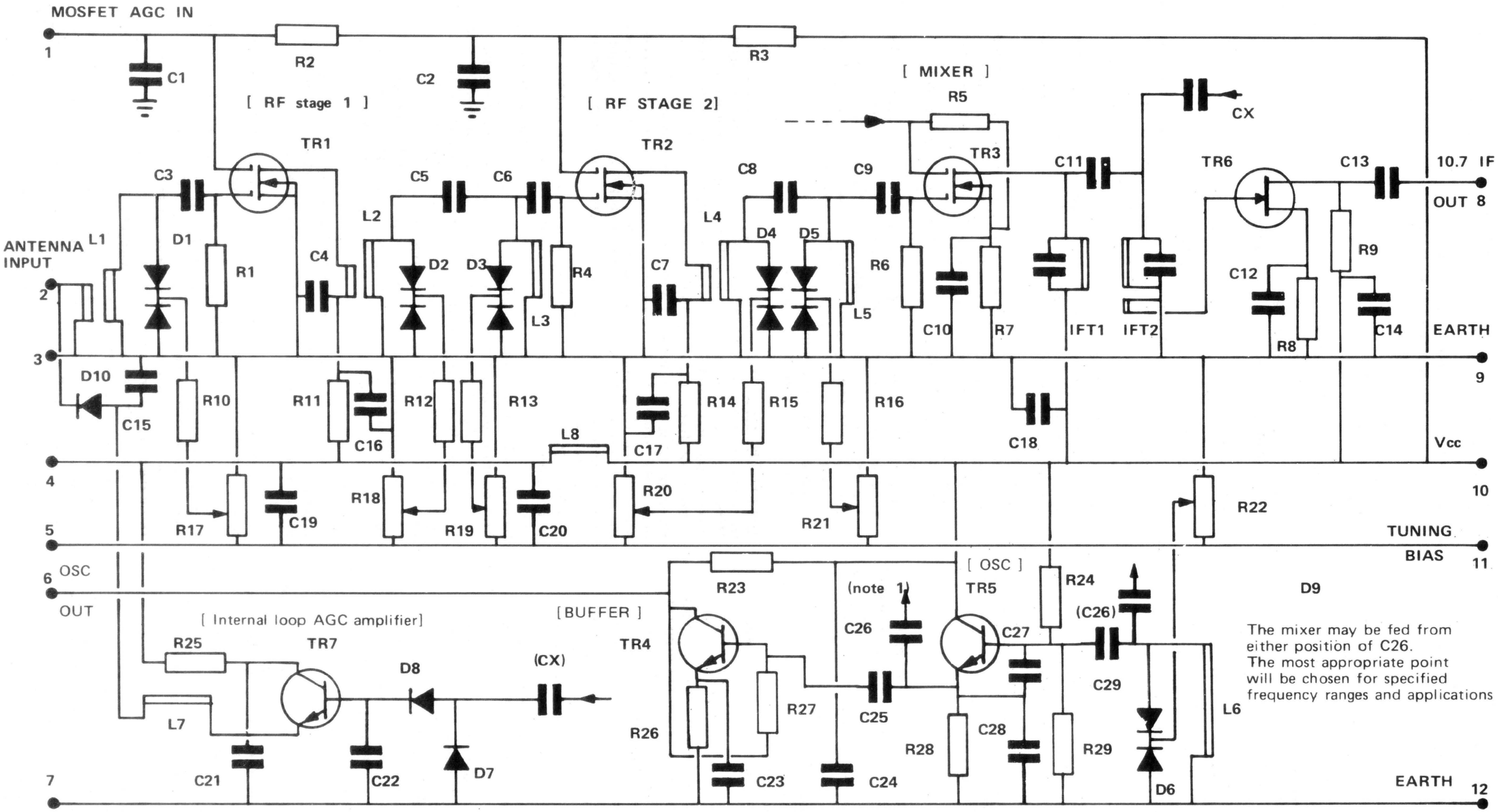
Probably the most widely used term in determining front end quality, image frequency relates to the response of the unit to signals that are exactly spaced by the IF on the other side of the LO. ie
Fif = Fantenna - Fo (in the case of the oscillator usually being placed on the HF side of Fin.)
Image signals occur in the upper reaches of band 2, but since they are by definition 21.4MHz away from the desired signal, the UK problems are not caused by other broadcast stations - since UK FM ends at about 98MHz. However, the image band is right in the midst of public service intrusions, and so is an important consideration.
Again, good RFselectivity is the answer.

The EF5804 solutions

Gain and low noise are ensured by the use of the latest dual gate MOSFETs. Overall NFs as low as 2dB have been achieved with special care, and the NF usually runs at about 3.5dB, with a gain in excess of 40dB. Typically 46dB.
Rejection of spurii is achieved with multiple bandpass tuned high Q RF tuned circuits. However, to ensure the excellent image and spurious rejections are maintained as the signal level increases, the unit is symetrically screened about each RF circuit. This not only prevents unwanted signal coupling, but ensures good tracking is maintained across the band.
The EF5804 also uses the trimming technique of the EF5402 - where a preset resistor is used in place of a trimmer capacitor. The technique ensures low residual capacity, and enables the whole band to be covered with 8v maximum.
This places extra emphasis on the stability of the tuning voltage and associated parts, but it also means the EF5804 is directly able to be driven from 12v systems incorporating frequency synthesis. In systems where greater voltages are available, the EF5804 will tune 88-130MHz with approx. 15v bias max.
The internal PIN agc system takes over when the IF derived agc runs out, providing attenuation of exceptionally strong signals that might otherwise lead to any of the various intermod products described.
Additional amplification is provided at IF, with an FET that presents a purely resistive load to subsequent stages. This technique assists more reliable matching into various forms of IF filter or amplifier.
A buffer stage on the oscillator supplies sufficient drive to almost any digital frequency counter/synthesiser system.

The net result is a tuner that will cope with almost all FM reception problems encountered in the UK - and most overseas. The very high gain of the system, coupled with the narrow RF bandwidth and low noise figure make the unit suitable for applications beyond those described as fringe. In areas where a local very strong broadcast or public service transmitter cause probelms, the best solution is to employ a trap in the antenna line. A single L/C circuit will provide about 30dB loss over a 1MHz span, and this is usually more than enough to ensure spurious free operation.
Using the Hyperfi IF amplifier (type 7230), typical 30dB S/N (mono) sensitivities of 0.5uV have been achieved, with a dynamic range of over 100dB in practical applications, where AGC is employed.

A useful and informative explanation of low noise basics is provided on page 149 of the Plessey Radio Communications Handbook. (Now available from Ambit)



RESISTORS

R1	100-180K	8	150-180E	15	as R11	27	220-330K		
2	47-68K	9	330-390E	16	as R11	28	1k5-1k8		
3	220-330K	10	56-100K	17-22	100K preset	29	8k2-12K		
4	as R1	11	82-120E	23	390-470E				
5	100-220K	12	as R10	24	4k7-5k6				
6	as R1	13	as R10	25	390-470E				
7	120-180E	14	as R11	26	82-120E				

CAPACITORS

C1	10-47nF	8	as C5	15	as C1	22	as C1	28	18-22pF
2	as C1	9	22-33pF	16	as C1	23	as C1		
3	15-22pF	10	as C1	17	as C1	24	as C1		
4	as C1	11	2p2-3p3	18	as C1	25	10-18pF		
5	0.75-1pF	12	as C1	19	as C1	26	1n0-2n2		
6	39-56pF	13	220-1000pF	20	as C1	26b	10-22pF		
7	as C1	14	as C1	21	as C1	27	15-18pF		

TRANSISTORS

Tr1-3	BF960/1, 3SK51
Tr4,5	BF273/224
Tr6	BF256
Tr7	BC238/9

D1-6	BB204
D7,8	1N914
D9	1N914
D10	BA479

COILS
Selected to suit desired frequency range/coupling

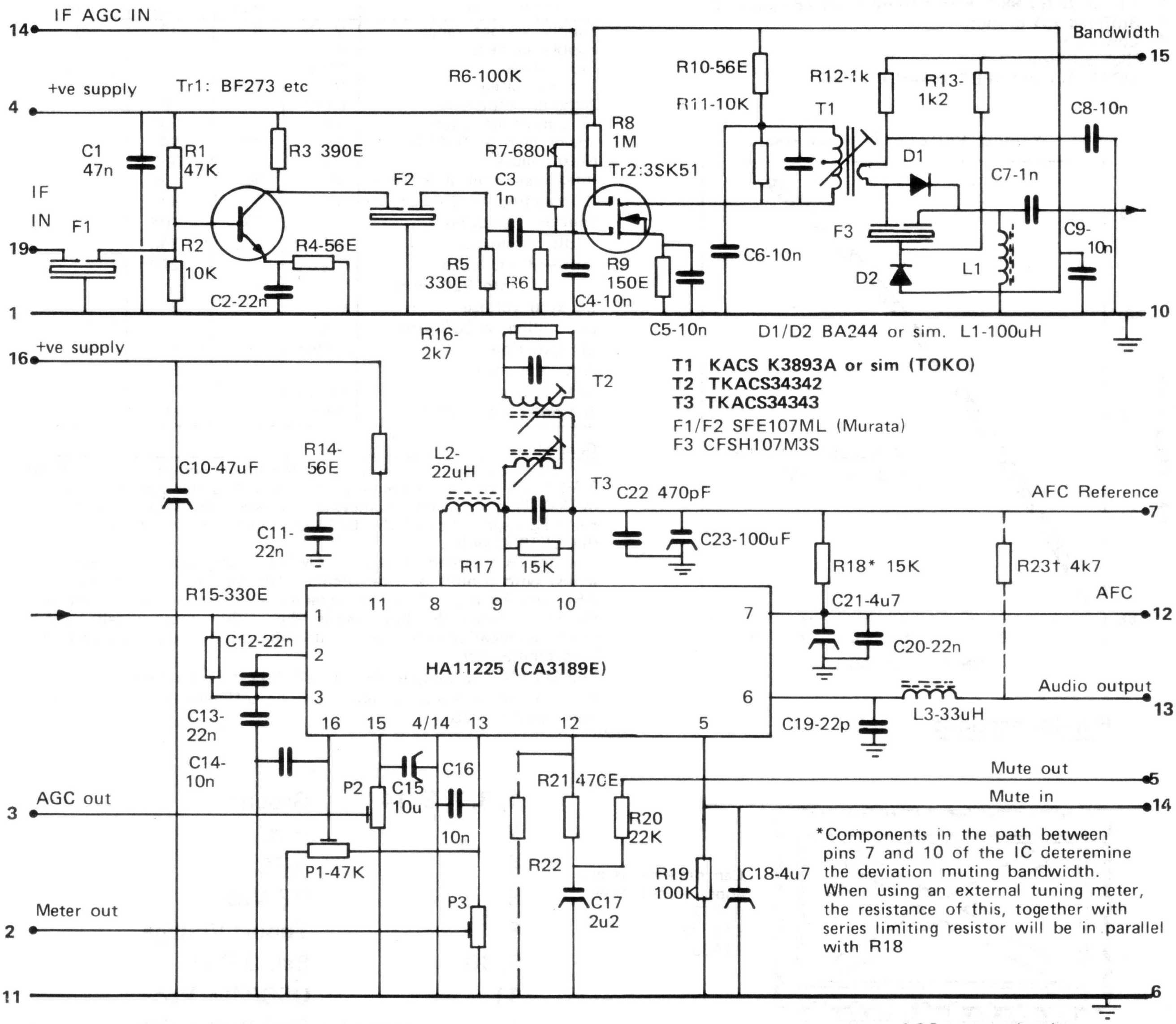
AMBIT EF5804 series : Hyperfi range VHF tunerhead

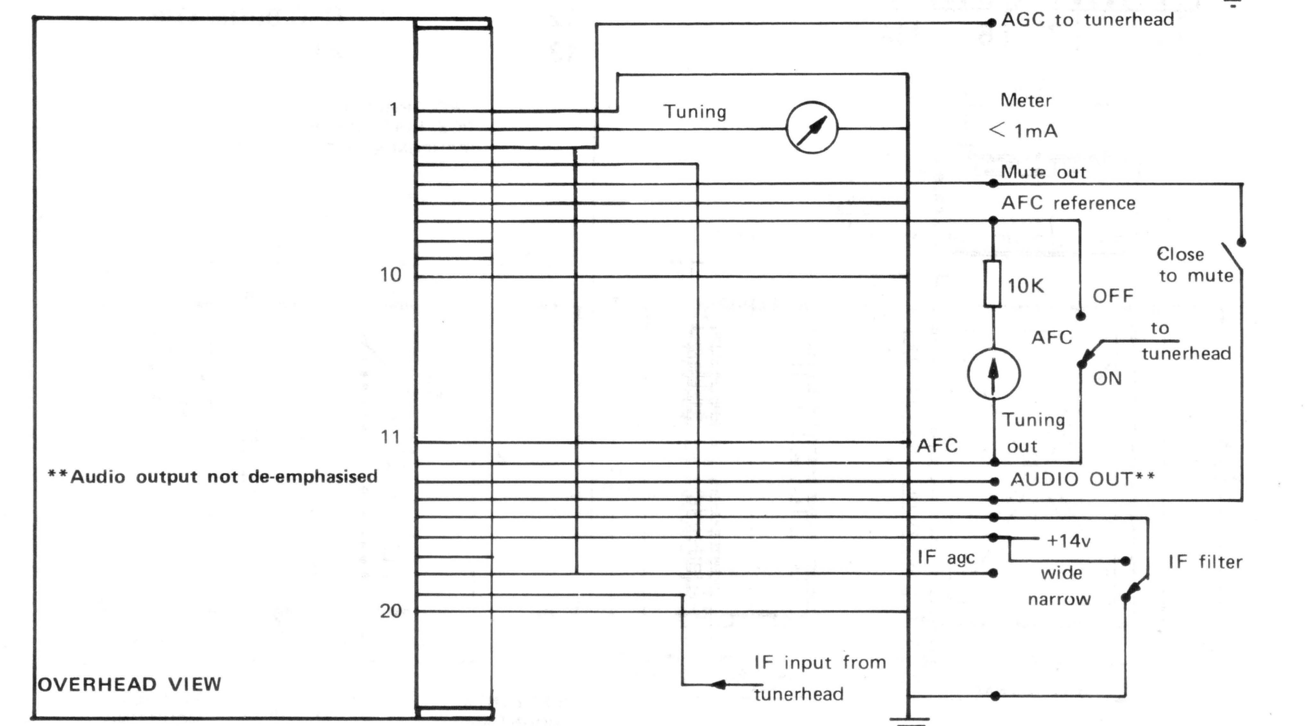
For connections to the EF5804, see EF5800/1/3 in previous issues

ambit INTERNATIONAL

91225: Fully screened FM IF amplifier

***Muting (signal level and deviation) *AFC *AGC *IF preamp *Electronically switched IF bandwidth *80dB S/N+**

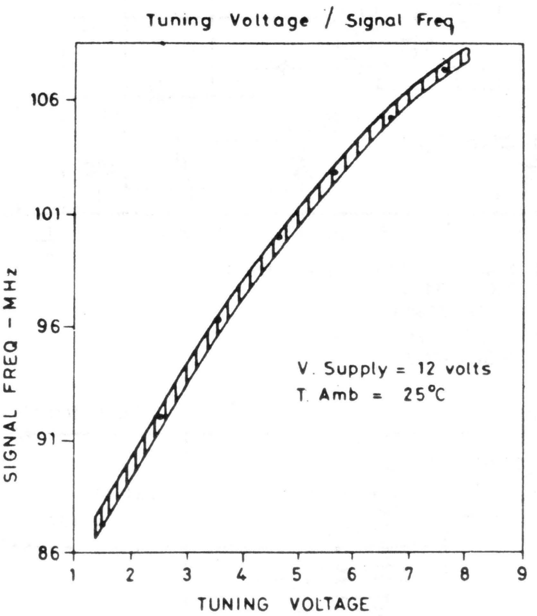




UM1181
FM varicap tunerhead

Features

* 1.5 to 8v for 88-108MHz (synthesiser compatible)
* Buffered LO output
* Low cost
* High performance
* MOSFET input/JFET mixer/IF preamp



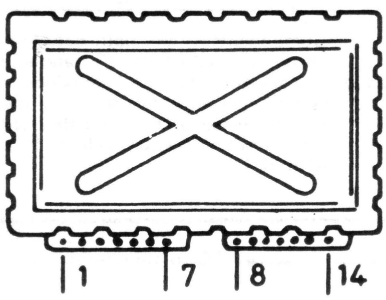
Specifications

(V supply 12v)		min	typ	max
Supply voltage range	V DC	8	12	20
Supply current	mA			30
Freq range	MHz	87		108
Tuning voltage	V	1.5		8
Antenna impedance	ohms		75	
IF output impedance	ohms		330	
Power gain at 98MHz	dB	30	34	
Noise figure	dB		3.6	4
Image rejection at 98MHz	dB	90	100	
IF rejection at 98MHz	dB	90	97	
Spurious rejection	dB	93	104	
−3dB bandwidth	kHz	300	340	
Buffered LO output into 220 ohms	mVp-p		220	
LO stop voltage	v		3	
LO pulling at OdBm in	kHz			20
LO radiation	−dBm		60	55
AFC reference	v		6	
AFC shift	kHz/v		75	
3rd order intercept	dBm		-9	

General

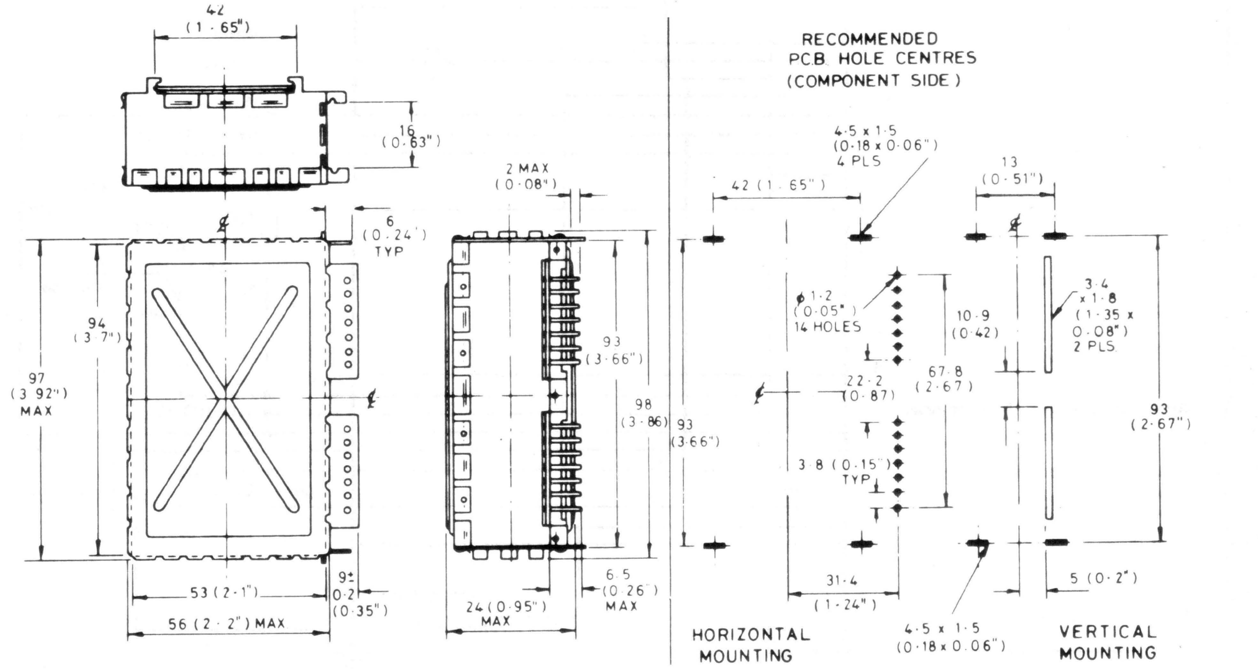
The UM1181 tunerhead is a 5 varicap tuned circuit unit, for band 2 VHF FM broadcast receivers. It neatly fills the slot in the range of AMBIT tunerheads - somewhere between the strong signal handling capabilities of the EF5400, and the very high gain of the EF5800 series.
The unit is supplied (as are all AMBIT tunerheads 5400/5800 series) with a built in local oscillator buffer. A limited range of AFC may be applied via the separate AFC diode arrangement on the LO, although for best results, the method more widely used in our applications involving controlling the entire tuning voltage is recommended.
The UM1181 is compatible with all Ambit IF modules.
It is not easily made available covering different specifications to those outlined above.

Pin Connections



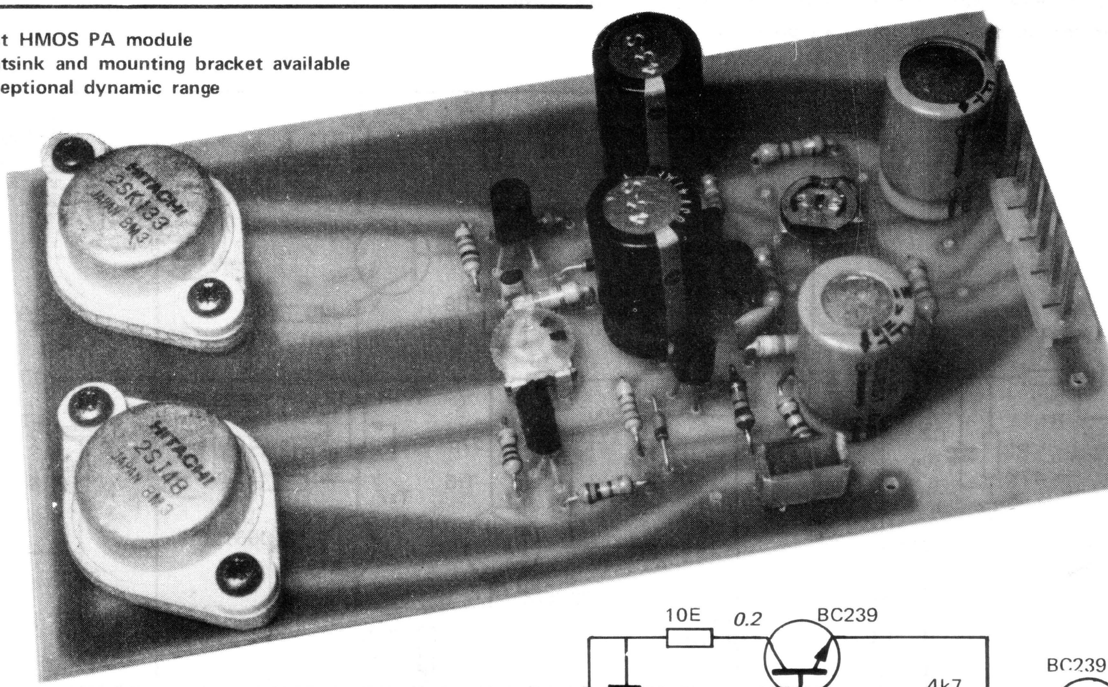
Can dimensions also apply to the new
91072
911225
94438

1, 3, 7, 8, 14	Ground
2	Ant.
4	AGC
5	RF Vcc+
6	Tuning Voltage
9, 10	Bal. O/P IF
11	OSC/MIX Vcc+
12	OSC Buffer O/P
13	AFC



PA100

* First HMOS PA module
* Heatsink and mounting bracket available
* Exceptional dynamic range



General

There are, as you may have noticed, a lot of audio amplifier modules on the market. Moreover, there are approx. 'n' times as many different opinions on the subject of audio amplifier design. In the desperate search for a gimmick or two, some of the commercial companies engaged in the design and supply of audio equipment have been found searching the walls of certain palaeolithic caves in an attempt to revert to some long lost technology that might provide the answer for next year's sales feature.

Valves have been cited as the marvel of the age by some, but then again, if you read widely enough in the audio press, you are likely to come across some character advocating the application of HT across a watermelon to reduce TMKLD.

A well known manufacturer who doesn't in fact make what the name implies, produces a very excellent 100 watt amplifier that has been variously raved over, and raged over. It wouldn't be very surprising to learn that audio reviewers are really as bemused by the difference between A and B amplifiers as most mortals confess to being. But on the assumption that with 1% distortion from the disc/tape/microphone etc, and 3-5% from the loud-speakers - that the difference between 0.001% and 0.01% from the amplifier is really all that important, we offer you the new PA100, using some of the fastest and least obtrusive output devices known to man.

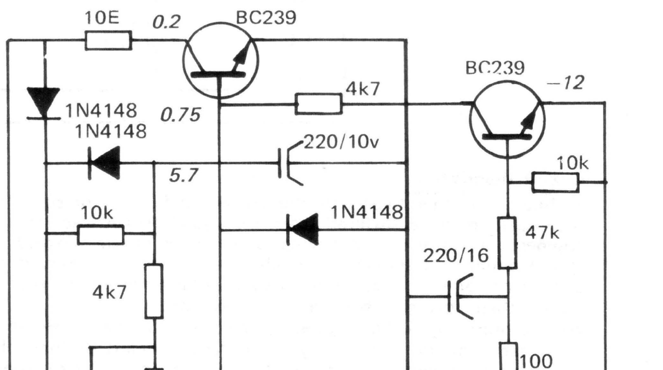
When you compare the frequency response of a good loud-speaker with that of a good amplifier, you are forgiven for wondering what all the fuss is about. Still, to make the best of the situation, the obvious answer is for the loudspeaker to be designed totally around the characteristics of a specific amplifier. Philips motional feedback seems like such an obvious concept, that it is a mystery why the practise is not far more widespread.

Notwithstanding all this cynicism, as long as you are determined to have a go at HiFi, then you might as well start with an amplifier that is as inobtrusive as possible. Ambit's PA100 is designed with the output transistors on the board, thus giving known and repeatable results – the heatsink bracket is predrilled to fit between the output devices and the PCB.

The dynamic range of the unit is in fact over 100dB, using Hitachi low noise input stages. The low drive requirement of the HMOS output stage reduces the number of active devices to a bare minimum - making the amplifier both simple to build and simple to set up.

Since we advocate the use of DC amplification (direct coupled) - alongside here is a design for a relay operated offset sensor. It is not guaranteed to switch off in time to save every amp/speaker combination from nanosecond destruction, but it will preserve the vast majority of voice coils from accidental damage. The relay contacts are paralleled to reduce contact resistance to a minimum.

The unit provides a slow switch on build up time delay, thus providing a welcome respite from the usual 'thump' that rattles the flight of plaster ducks on the living room wall.



DC offset protection

Correct operation causes the relay to close

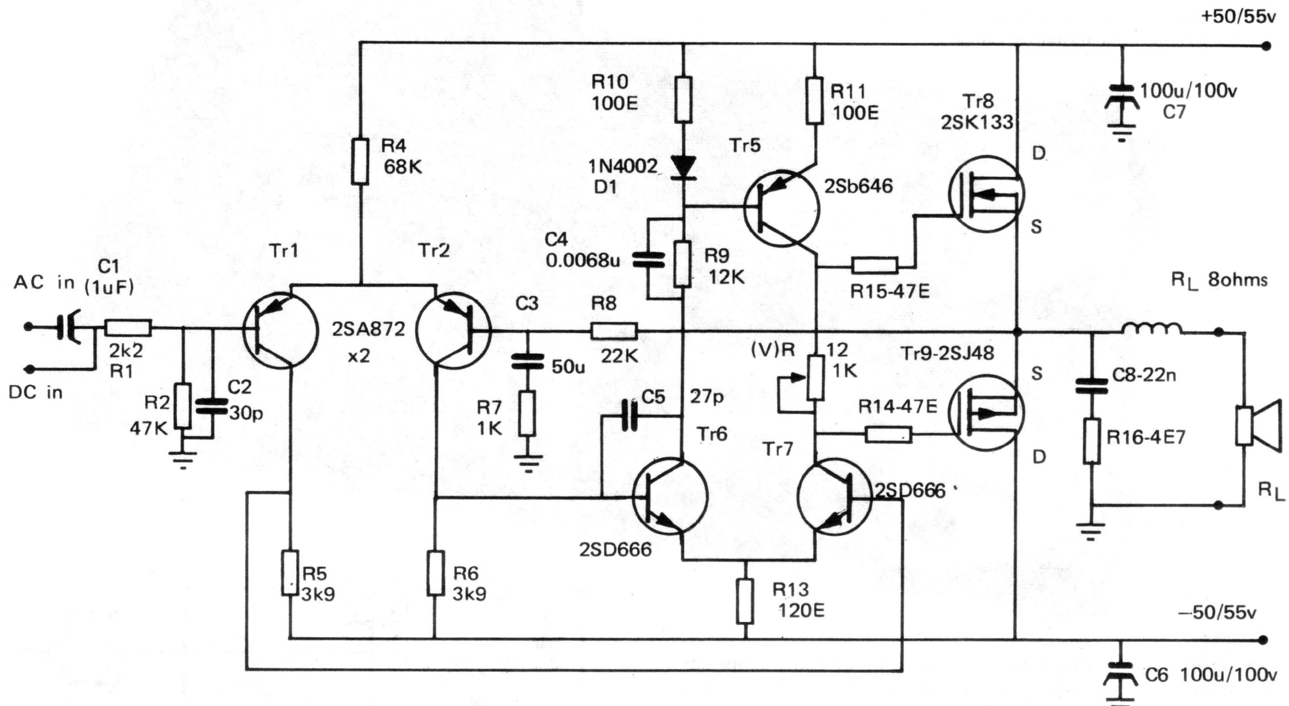
The circuit may be driven from any number channels (via the 100K resistors on the base of the BC309) - since if an offset condition occurs in any output, the circuit will cause the relay to drop out. The relay contacts are not shown paralleled in the diagram, but for best results, as many gold plated contacts as possible should be connected in parallel for minimum resistance.

Figures in italics indicate test voltages

100W MOSFET PA unit - PA101

The circuit below is a development from the original PA100 - and it is relatively easy to adapt existing boards to the new suggested configuration. The major advantages being the improved stability of the bias arrangements - enabling the offset adjustments to be disposed of. The net result produces virtually no switch on/off transient - although we still recommend the use of an offset sensing circuit with relay control for the remote eventuality of a breakdown.

Facilities for AC and DC coupling are provided.



Circuit description

Although we have been offering power MOSFETs for some time, they seem to have suffered from the usual British electronic engineers' malaise of not having appeared in Wireless World. If a product is described in Wireless world, be it a heap of old bean cans, or the most sophisticated audio amplifier in the world, then engineers believe in its existance. At the moment, audio engineers seem to treat power MOSFETs on a par with the Yeti - they've just about heard of them, but until one is caught and displayed in public (in this case, some noble tome like WW), then they might as well not exist.

Those intrepid explorers who have dared to cross this uncharted territory, and found their first experience to be anything less than shattering have been overheard to take an uncharitable view of the devices. Thus have complementary HMOSFETs been dismissed as transient figments of some crazed designer's imagination in some quarters.
The first time an engineer tried to get to grips with a bipolar output system, no doubt he went scurrying back to the 807s.

Enough of this - on the positive side, we have been developing with these devices for over a year, the results of which are shown above. The circuit isn't much changed, but various PC layouts have been tried to achieve optimum results.

C1/R1 are used in AC coupled applications, whereas C1 is bypassed for DC operation, when the whole amplifier may be treated as a large DC opamp, with Tr2 being the inverting 'input'. Thus the gain is set by the ratio of the two resistors in the negative feedback path - R7 and R8.

The input and driver stage transistors are all characterized for high voltage, low noise audio amplification. The ratings of the devices are generous, enabling the whole circuit to tolerate a high degree of mal treatment before irrepairable damage is done.

The bulk of the driver circuitry is devoted to classical voltage gain configurations, since the output FETs require little power for operation - merely enough to cope with the charging of the input capacities.
VR12 sets the output quiescent current of the FETs in a class AB configuration - without the need for considerations of secondary breakdown and thermal runaway, encountered in bipolar design techniques. 80mA is a typical value, selected by checking for minimum distortion.
The speed of operation of the MOSFET, coupled with the very low drive requirements, makes the calculation of phase correction components simple. For the most, part, they are not needed. Some very fine tuning can be achieved in some circuits by placing a small capacitor (2p2 -22p) across R8.

C5 is not intended for phase correction, but for stability in the drive circuits. This component is layout sensitive, and thus also depends on the types of components used, together with their lead lengths.

Major considerations of power audio amps

The first concern is usually good signal to noise+distortion. 100dB S/N should be easily achieved with most modern semiconductor technology - and in the case of the PA101, this can be extended to 120dB + (input shorted).
Most of this is dependant on good earthing practise, and avoiding taking the input earths across the output and power earths. The currents flowing in high power amplifiers are substantial - and even an apparently low impedance piece of copper track (or connection wire) can possess the few fractions of an ohm necessary for a substantial potential to be set up along a single earth track.

If a wire of resistance 0.05 ohm carries 4 amps, then the PD is (4x.05) V = 0.2V. If this PD happens to lie across the path of a feedback loop - then the result is distorion, since the feedback loop is no longer referred to the earth. It is thus necessary to run all earths to a centra earthing system via low impedance connections.
If the earth loop current happens to be in phase with the amplified signal, then positive feedback results, and the circuit oscillates at the 'resonant' frequency.

Capacitors C7 and C6 may seem superfluous if you are using 10,000uF smoothing at the power supply, but if they are left out, the effects are the same as with earth loops, since for example, the input stage emitter resistor is now R4, plus the impedance of the +ve rail supply to the reservoir capacitor. R4 is thus on the end of a 'modulated' HT rail, and must be decoupled at the board end of the supply.

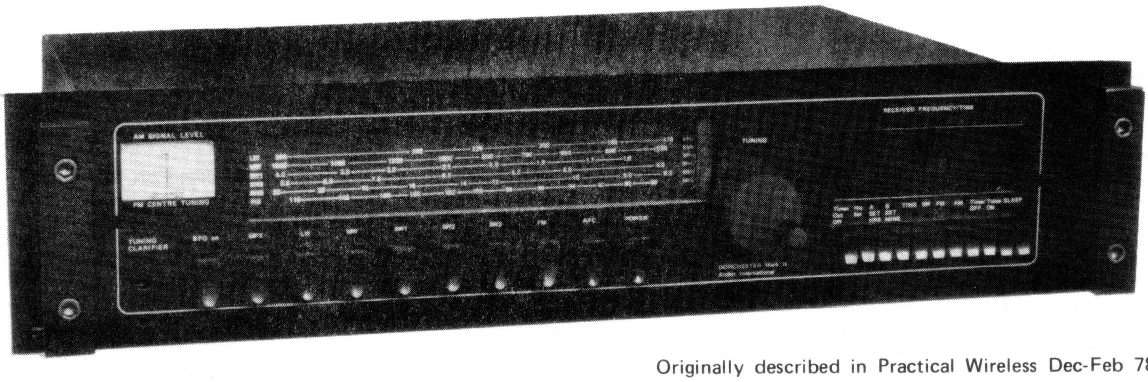
If two or more amplifiers are used with a single supply, then the supply leads - as well as the earth leads, must be taken to a single point source. Otherwise, simply providing a ring main enables one amplifier to modulate the other(s), resulting in cross talk.

Stability and varying load conditions is one point that frequently gets overlooked. The fact that an amplifier works immaculately into an 8 ohm resistive load may be very misleading. Most people have heard apparently similarly specified amplifiers making entirely different sounds on the same speakers - and an amplifier that completely changes its characteristics when fed into different speaker loads.

Capacitive loading changes the voltage/current phase at the output by drastic amounts. Sometimes, sufficiently for the amplifier feedback to be driven into a positive condition resulting in instability. Loudspeaker crossovers are usually full of capacitors - so a check on the stability is essential.
Although there are many detailed differences with a power MOSFET v Bipolar, most of the basic criteria for sound amplifier design apply to both types. We invite the daring to advise us of their exploits, and we will provide a £100 credit note prize for the first design we examine that exceeds -70dB for both THD/TID from 30Hz to 20kHz at 75W RMS into 8 ohms.. (Using our MOSFETs, of course!)

'DORCHESTER' series

- *All broadcast band AM/FM tuner*
- *Ceramic FM filters, mechanical AM filter*
- *Available with DFM options*



Originally described in Practical Wireless Dec-Feb 78/9

DORCHESTER SERIES : MULTIBAND AM/FM TUNERS

General

The "Dorchester" series of multiband tuners are derived from an original prototype described in a hobbyist magazine. The general concept of superhet multiband receivers does not lend itself to 'DIY', since the general bandswitching complexity of a 6 band tuner such as the Dorchester A series is very critical.

However, recent developments in single IC AM/FM systems has offered a short cut solution which has been exploited in the Dorchester concept. In fact, take away the tuned circuits which are indespensable to any radio design, and there isn't really a lot left on the circuit diagram.

Nevertheless, the Dorchester is a full superhet system, with ceramic FM IF filters, and a mechanical AM IF filter.

The tuning system of A series is a simple mechanical arrangement, using an air gang capacitor for AM and FM. The FM input tunerhead uses an FET stage.

The AM system uses the double balanced input mixer system of the ULN2242 IC, and provides an excellent all round broadcast performance - though please do not confuse broadcast with communications performance, where many factors are far more critical than can be dealt with in a unit of this price range.

The Dorchester range of equipment has been developed as much as a generally stylish housing for genuinely 'Home Brewed' radio equipment, as an end in its own right. All parts of the hardware kit are available separately, and we invite the adventurous to submit details of the type of equipment they have installed themselves.

Construction of the Dorchester should not be undertaken by inexperienced radio constructors-unless you have access to assistance from a competent engineer/enthusiast with radio knowledge. The service facilities offered by Ambit are not supplied free - and accordingly, a poorly made kit may be very costly to repair.

Options in the Dorchester Series

Digital frequency indicator unit, Ambit type DFM1. Providing frequency readout to the nearest 1kHz in AM ranges, and 100kHz in FM range. Also with various clock/timer functions as described in DFM1 details. Parts required

BF1	AM/FM oscillator buffer unit with FETs
Switch	11 way function setting switch
DFM1	Screened DFM/Clock timer unit

Specifications

A series

Coverage in six ranges:	
Longwave	150-275kHz (Ferrite rod)
Mediumwave	525-1600kHz (..)
SW1	1.6-4MHz (ext.antenna)
SW2	4.0-8.5MHz (..)
SW3	8.5-22MHz (..)
FM	88-108MHz (..)
Antenna impedances	75Ω (optional 300Ω FM)
AM IF FILTER	4 or 6kHz (specify)
FM IF FILTERS	Standard CFSE10.7
Meter outputs	Sig level on AM Centre tune on FM
BFO	Variable about 455kHz
SSB/CW	A MOSFET product detector is included for AM/SSB within the range SW1-SW3, but please note this is primarily intended for use without large antennas which will lead to overload if used under 'communications' conditions.
Stereo	PLL stereo decoder included on FM, together with input Birdy filter and output pilot tone filters.
Audio output	200mV each channel
Power requirement	240/220v AC mains (12vDC optional)

MA1023 LED digital clock/timer module. A low cost alternative to the DFM1, providing 12 or 24 hour display, mains driven only (ie no 12v DC option available on this unit)
Parts required:

MA1023	LED clock module
Switch	11 way function setting switch as per DFM1 - since the metalwork is designed to accept this type of unit. Obviously there will be spare functions according to the panel ledgend, which is designed for the DFM system as well.

VHF NBFM Monitor receiver unit : Ambit 96640

General

The 96640 is a designed based on a PW feature, which itself is based on a Plessey design by James Bryant (G4CLF).
Having said that, the circuit largely speaks for itself, being a combination of high gain, low noise MOSFETs in the RF and mixer stages, followed by an 8 pole crystal filter (we now use TOYO 10M series, with matching IFTs to achieve better overall performance in this configuration.
Both 25kHz, and 12.5kHz channel spacing filters are available, and the overall performance of the unit is designed to comply with standards required of professional equipment for PMR and marine applications. However, in the hands of the home constructor, it is not possible to guarantee performance without carrying out tests and alignment - and so we offer an alignment (not a complete debugging and fault finding) service for units which have been correctly constructed, and are showing some signs of life.
The unit uses TOKO S18 coils, providing coverage of various 2MHz bands from 30 to 200MHz maximum. Bands cannot be switched in this design, although up to nine crystal controlled channels may be switched using DC only switching.
The oscillator uses cheap fundamental mode (RX frequency, minus 10.7MHz, divided by 9) crystals, which can be pulled onto frequency with the trimmers supplied. In fact, by using this type of crystal, it is usually possible to pull a crystal to one channel either side of the desired centre frequency.

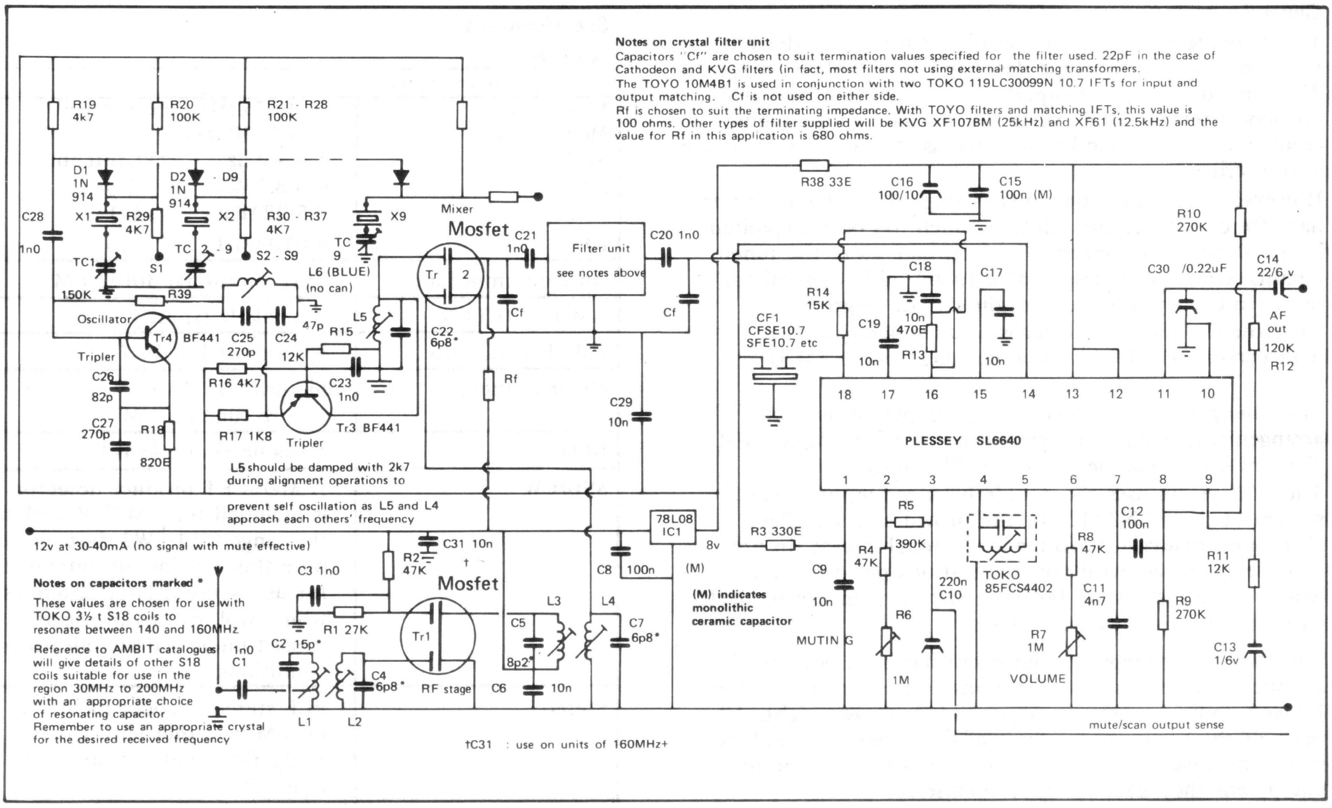
Specifications

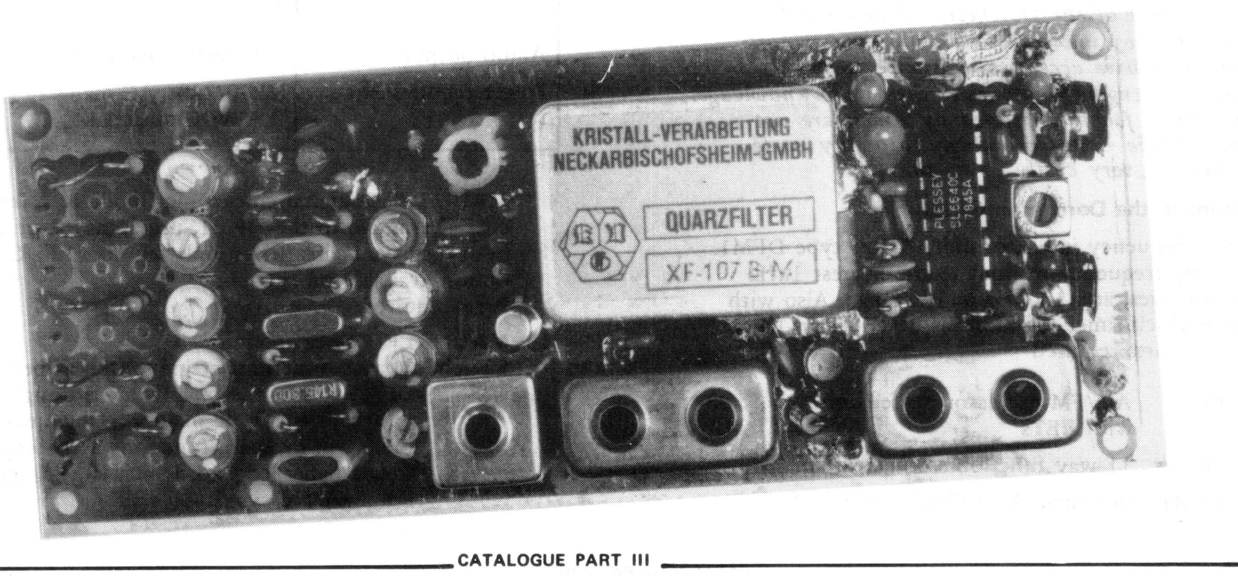
Parameter	min	typ	max	units
Supply voltage	10	12	18	volts
Supply current at 12 volts		32		mA
Input sensitivity to operate mute circuit (50 ohms PD)		0.6	1.0	uV
AF output with 10uV in		100		mW/8ohms
Adjacent channel				
a 25kHz filter	90			dB
b 12.5kHz filter	90			dB
Filter shape	2:1			6/60dB

For additional information, refer to SL6640 data (in part 2 cat.)

Although MEM680 MOSFETs are indicated on the circuit below, we reserve the right to supply either
MEM680
3SK51 3SK60
BF960/1/3
According to availability and our assessment of batch samples.

The switching speed of the oscillator should be kept to 100mSec sampling in sampling applications, although particular examples may be capable of faster strobing. The output at pin 3 of the SL6640 may be used to sense signal presence - but C10 may require adjustment to suit reaction time required.





PW Sandbanks metal locator etc

General

Following an Ambit tradition, this catalogue also includes some details of a metal locator. The Sandbanks kit has been developed to produce a very rugged overall design - using injection molded ABS housings - and precut and prepared brackets etc. In fact, the housing are available separately, since they provide the basis of many DIY sets of electronics for the more adventurous constructors.

Home Office Approved Design

- Pulse induction power
- Immunity to ground effects/silver paper

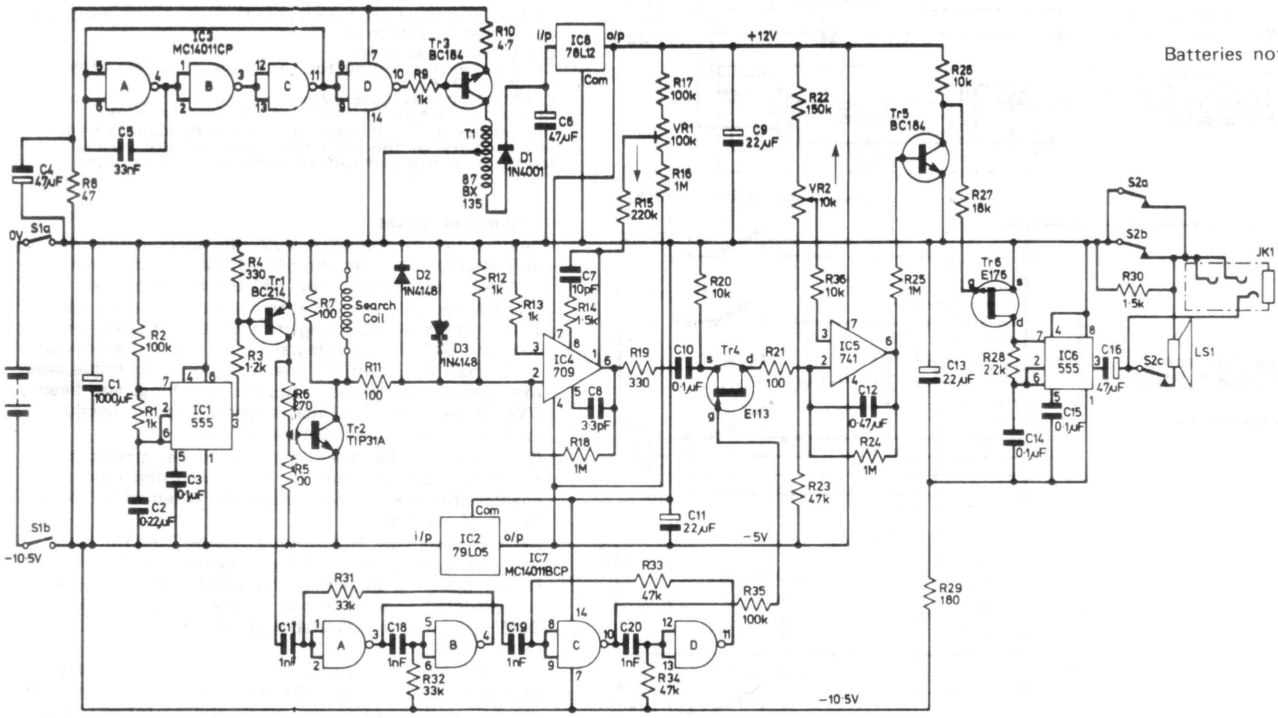
The pulse induction method of metal location is based on a principle of operation that may be likened to radar or sonar. A signal is first transmitted, and subsequently received back and detected. However, the signal is not strictly speaking a true reflection as such, since the metallic object involved has basically been electromagnetically charged with the output pulse - and the detector senses the decaying magnetic field when the pulse is removed.
As long as there is a change in magnetic flux in the system, the effect will produce a signal in the pickup coil.
Extensive quantitative analysis of this type of detection technique is not very reliable, but it is fair to say that the range of detection will be function of the power of the output pulse. The various laws relating to the diminution of power with respect to distance are very much in evidence, and whilst prodigious distances of several feet for coin are achieved in some systems using PI techniques - power levels in excess of kW have been known.
The choice of pulse timing and pulse width is a result of various compromises to make the best of the coil characteristics. Whilst a bigger coil will result in better range (bigger inductance that is) the mass of the metal in the coil itself will cause the collapse of the coil field after the output pulse, to interact with the small signals the unit is trying to detect.
Metal detection is still a field where this is plenty of scope for the experimentally minded - so it is a matter of some regret to us that so few original approaches ever seem to be published in the hobby press.

The construction includes a set of U2 (D cell) batteries, as an integral feature of the tubular design. Replacement is achieved by a simple half-turn cover at the rear of the box section.

In its form as a PULSE INDUCTION detector, the unit consumes approx 80-100mA, but provides simple single knob tuning operation - and a 2p detection range of 8 to 10 inches.
The PI techniques gives almost total immunity to ground effects of damp grass, water etc - and a substantial degree of immunity to silver paper interference.
The PI technique is particularly sensitive to highly conductive metals - and fortunately, most precious metals come into this category. However, it is also sensitive to iron - and whilst this is of great benefit to those of you looking for manhole covers and iron pipes - it can interfere with the detection of other objects.

- Makes your donger grow longer

Batteries not included

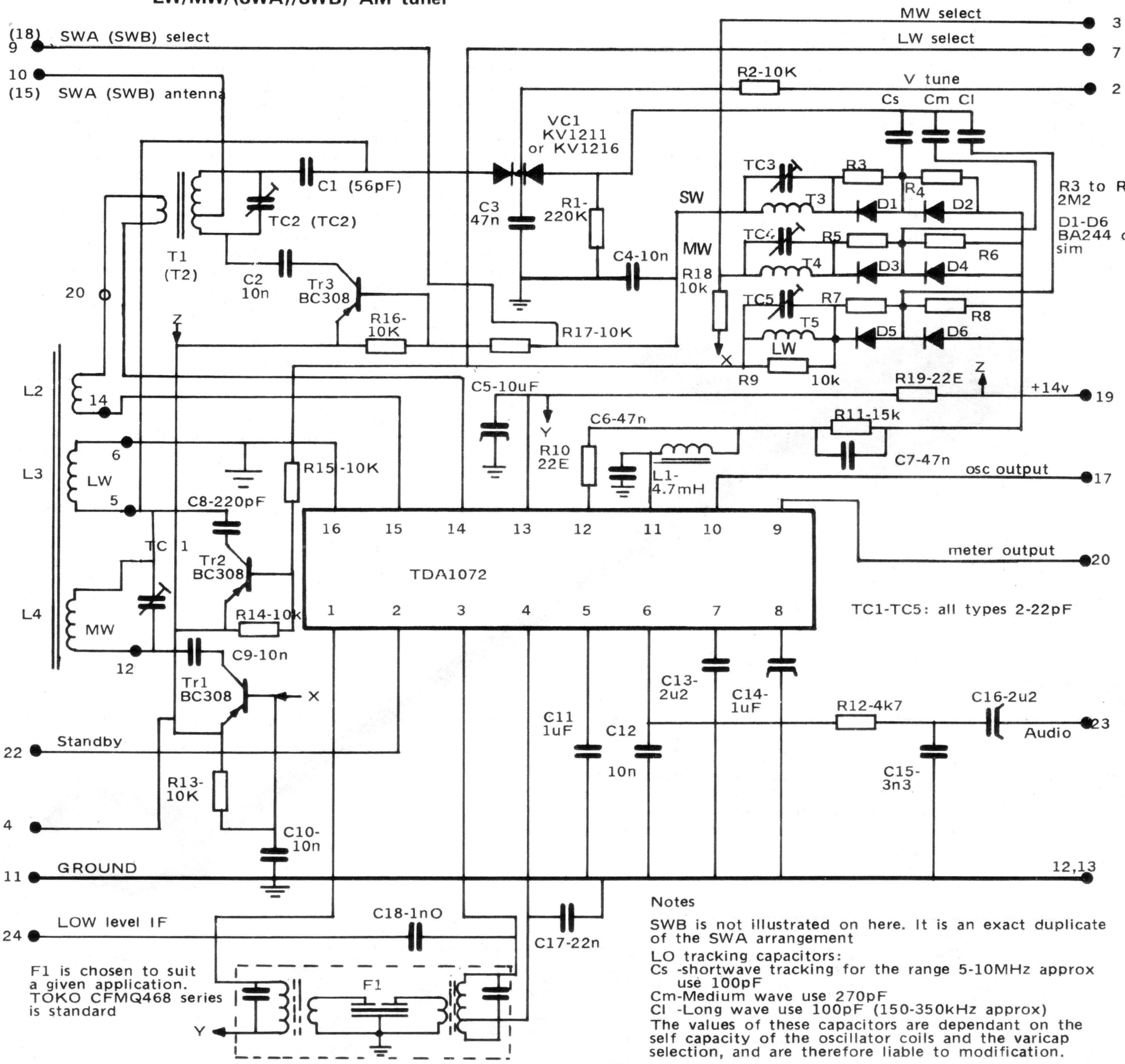


Many of the modern radio ICs lend themselves to being used in metal locator designs. Anything from BFO, to IB and VLF can be accomodated in something like the TDA1083. And with the very wide range of operating voltages, low current and built in audio, it would seem to offer all you need.

If anyone is sufficiently interested, then remember our various incentive offers to refund all component costs for anything that ends up getting published (we like to hear about it in the first place, though.)

ambit INTERNATIONAL

91072 series - DC switched and DC tuned LW/MW/(SWA)/SWB) AM tuner



Notes

SWB is not illustrated on here. It is an exact duplicate of the SWA arrangement

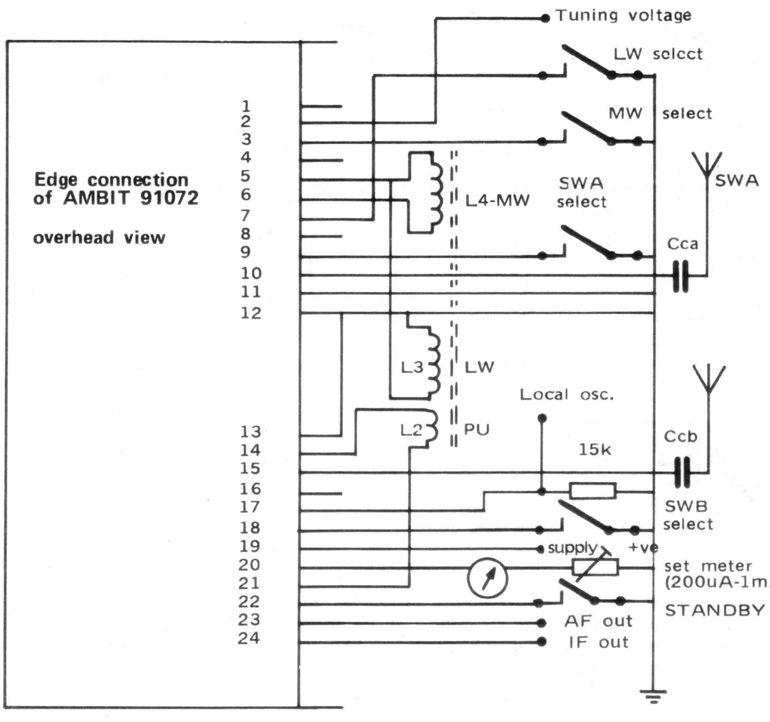
LO tracking capacitors:

Cs -shortwave tracking for the range 5-10MHz approx use 100pF

Cm-Medium wave use 270pF

Cl -Long wave use 100pF (150-350kHz approx)

The values of these capacitors are dependant on the self capacity of the oscillator coils and the varicap selection, and are therefore liable to modification.



Application notes

Being based on the TDA1072, the specification of this unit is very much as per the IC data sheet - enabling optimum reception of the AM broadcast bands.

MW/LW reception is via a ferrite rod antenna, using a slightly unusual coupling technique - whereby the coupling coil that is integral in the LW tuned circuit is retained in circuit for the MW. This technique saves the need for additional complex switching, and keeps MW 'Q' exceptionally high since the coil is barely loaded by the coupling arrangements.

Band selection has been designed to be 'ground to select' (pullups included for open selection pins). This enables the most flexible approach when using remote forms of switching.

SWA is nominally 5-10MHz (covering 39-41m bands), and provision is included for a further band SWB. The circuit design is such that this band need not however be restricted to any section of the region 150kHz to 30MHz, and is left available as an option for custom requirements.

The unit is arranged such that LW is a completely separated section from MW (not simply using the MW osc coil with padding capacity), and so tunes through the marine DF band continuously.

Cca and Ccb are left external to the unit, since depending on the facilities available for the SW external antennas - these may need to be in the region 22pF to 100pF, and must be determined according to the characteristics of the external antenna.

The 91072 is ideally suited for operation with digital tuning synthesiser systems (Ambit HD944752 etc)

ambit
INTERNATIONAL

CATALOGUE PART 3 - STOP PRESS

(NOVEMBER 1979)

Items listed here have arrived too late to catch the main section of this catalogue

Cage Jacks These are individual sockets for component legs, that enable plug in facilities to be devised for virtually any component. Including hitherto impossible things, like coils, DBMs etc. We stock 3 standard sizes, covering components from MOSFETs to crystals. Being gold plated, and precision made, they are not a cheap alternative to an IC socket. But for engineering and development samples, they are a real boon.

- CG1 for 1mm pins
- CG2 for 0.56mm pins
- CG3 for 0.76 to 0.81mm pins

IC sockets Not perhaps the earth shattering excitement you have come to expect from Ambit, but after a few of the TI sockets with their brittle legs and micron sized entry holes, you will appreciate our new Zetronic DIL sockets. They are same overall size as TI low profile types, but the contact is side wipe, with a sloped entry cap, enabling easy insertion. The legs may be bent 90° and returned to vertical, without dropping off. We unhesitatingly recommend them as being the best value on the market.

Books **Plessey Radio communications handbook**
Signetics phase locked loop handbook
In stock (usually) see PL for price

Changes **The following are discontinued**
MFL series mechanical filters - replaced by a 455kHz crystal filter with 1.3:1 shape factor to be announced in PL
MFH series replaced by the CFM2 and the matching IFTs (see this issue)
CFX series replaced by the CFM2 as well
E176 JFET replaced with J176

Addenda **Coils**
199CCA127EK a 27MHz 7KC (built in resonanting capacitor) for the RCME RX and the PW RC series
113KN2K241DC 7KN 27MHz coil
113CN2K509ADZ 7KN 27MHz coil

YLE4A888EK lower Z 455/470kHz IFT with 2:1 secondary

YWOS6A356EK low self capacity version of the YMRS80046N MW oscillator coil

7P style input/output matching IFts for 2K impedance ceramic filters

Semiconductors
3SK60 dual gate MOSFET replaces the MEM680 with similar and in some parameters, superior specifications.
2SB720/2SD760 Hitachi 200v complementary driver devices for very high power audio outputs.

Filters
Monolithic 10.7MHz SSB crystal filters, 8 pole design, type 10M22D
Alternative 34.5MHz HF receiver filter, type B34 F8A (slightly elongated design)
455kHz crystal filter for SSB, type AOOF24A.

Modules
7230 Hyperfi IF, based on our Mk 3 tuner connector system, this is basically the 911225 described herein, with slope controlled AFC added into the main tuning voltage. ie The gain of the AFC is controlled by signal level, providing very little pulling effect when tuning, but 'locking' on to a signal which remains in the passband. The next best thing to a synthesised tuner.

94438/7 Hyperfi stereo decoder. Our 944378 decoder squeezed down into the new shield can system of the UM1181, 911225 and 91072.

92242 A TDA1090 based AM section with MW/LW, and FM section with IF preamp and twin ceramic filters. Again, squeezed into one of the new style screening boxes.

Double Balanced Mixers
Since introducing the SBL1 in our price lists, it has been evident that interest in this type of mixer is growing. We have now included the SBL1-8, which is suitable down to 100kHz (100kHz to 200MHz), all other major paramters are the same, although the case is slightly taller to accomodate larger transmission line transformer cores. Our CG3 cage jacks may be used to provide plug-in capability.
We are proposing to extend the range with other DBMs, including some high level (17dBm LO) and 1000MHz types. We welcome any comments from potential users.

LARSHOLT TUNERSET TYPE 7254
Write in for an illustrated leaflet on the Larsholt tunerset range, together with the details of their tuner and audio amplifier kits.
We unhesitatingly recommend Larsholt tuner and amplifier kits to either the newcomer (who is able to at least make a workable solder joint) - or the more experienced constructor who simply doesn't have the time to assemble the more intricate systems with multiple options.
A new tunerset, the 7254 is now available, and this uses the TDA1062 in the front end, with a HA1137/KB4420 type IF, and KB4400 stereo decoder. A buffered LO output is also available, together with a companion LCD DFM module (type 9005).

FRG7 OWNERS:
Since the cessation of the MFL SSB filter, we have been looking for a suitable alternative unit for an FRG7 conversion. As you know, the basic filter is pretty dreadful for SSB, being neither really wide enough for decent AM, or narrow enough for SSB.
We have thus designed a unit using the exceptional new A00F24A, together with a CFM series ladder filter of good AM passband (nom. 6-7 kHz). Depending on the filter deliveries (expected for Late Jan 80) the Ambit **455F** kit (as it is known) will be available to transform the otherwise indifferent bits of the FRG7 into a nearly professional class communications receiver.
Regarding fitting DFMs to this set, the problem is that the last tuneable IF runs from 3 to 2MHz, tuning in reverse, due to the choice of the first IF frequency. By retuning the first IF to invert the tuneable IF. The resultant 2 to 3MHz can be displayed to the nearest kHz on a DFM2A, or to 100Hz on a DFM3A or DFM6A. But we do not suggest you perform this trick unless you are very certain of your capabilities. We accept no liability if you get stuck !! Retuning this IF puts a big birdy into to the 2-3MHz range, but this seems to be the only major spurii "extra", so for most users, the modification is worthwhile.
The 455F kit is fully DC switched (using the existing mode switch facility.)

PW RADIO CONTROL SYSTEM
We had hoped to have been moved a month or so before this series started to appear in print. As it happened we weren't, so whilst we are able to supply all parts, we are not yet rash enough to claim to have a complete 'kit' ready. (Nov.79)
A separate leaflet will be printed with fuller details - but meantime, all parts appear in the latest price lists. If you want details, please deposit a large SAE with us, and we will send them when ready.

The NEXT CATALOGUE !!
When part 4 is ready, we anticipate having also edited parts 1 to 3 to bring them up to date in one cover. There will not be much reduction since all catalogues are supplemental - but if your copies are getting dogged you can place an order now for both the new part four, and the Amnibus edition of parts 1 to 3. About Sept 80 (or sooner, we hope). Amnibus £1.75, Pt.4 75p.

CHECK CURRENT PL FOR PRICE AND AVAILABILITY INFORMATION

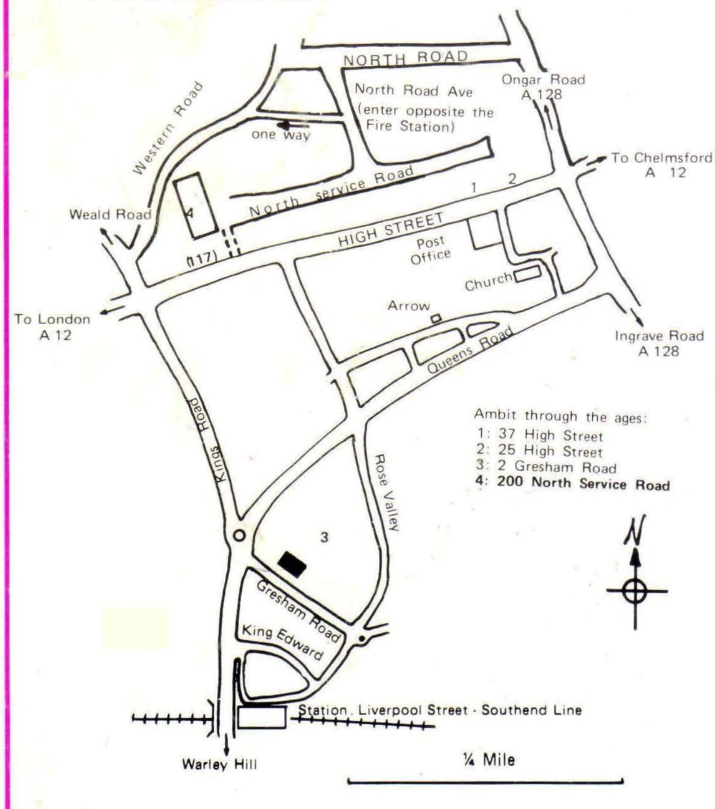
........*has now moved to larger premises at:*

200 NORTH SERVICE ROAD, BRENTWOOD, ESSEX

Telephone number changed to (0277) 230909, Telex 995194

* *Still only 5-10 minutes from Brentwood station (Eastern Region - Liverpool Street Terminal)*
* *Trains every 20minutes weekdays*
* *Shortly to be reached by the new M25*
* *Parking when you reach us, or pedestrian access from the High Street*
* *Brentwood has more pubs per capita than any other town in the country (over 15 within about 10 minutes of us !) - Including one with Greene King*
* *Four "Fast Food" shops in the High Street*
* *Greatly expanded trade and "caller" services, with permanent displays of our products (at last!)*

How to find us: —



In our next issue........

* *Cannelloni Hunting in Sicily (a follow up to our immensely popular feature on the Spaghetti farms of Tuscany)*
* *How to tell the difference between a microcomputer and Basil the rat.*
* *How to cope with being over eighty, and yet still retain a healthy interest in electronics.*

WARNING by H.M. GOVERNMENT

Reading this catalogue may potentially be injurious to your bank balance, since it contains details of far too many tempting innovations, new ideas and other exciting stuff. Beware !!