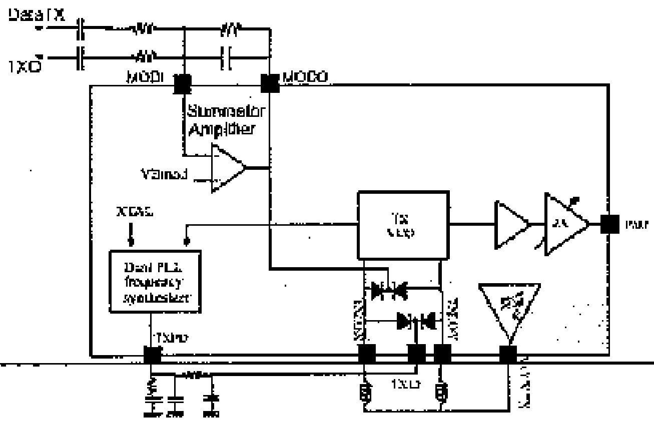
3.3 Transmit Section

The block diagram of transmit section is shown below:



3.3.1 Tx Amplifiers

A programmable power amplifier is used to provide the necessary gain for the transmit section. It amplifies the signal from the Tx VCO to apply the correct output power into the antenna (-3 dBm for both handset and base).

There is a Chebyshev low-pass filter between duplexer Tx_in port and PAO pin. The cutoff frequencyof the filter =915MHz, 3 poles and ripple <.1dB. The source and load impedence = 50 ohms. It is used to reduce the 900 MHz harmonics power level.

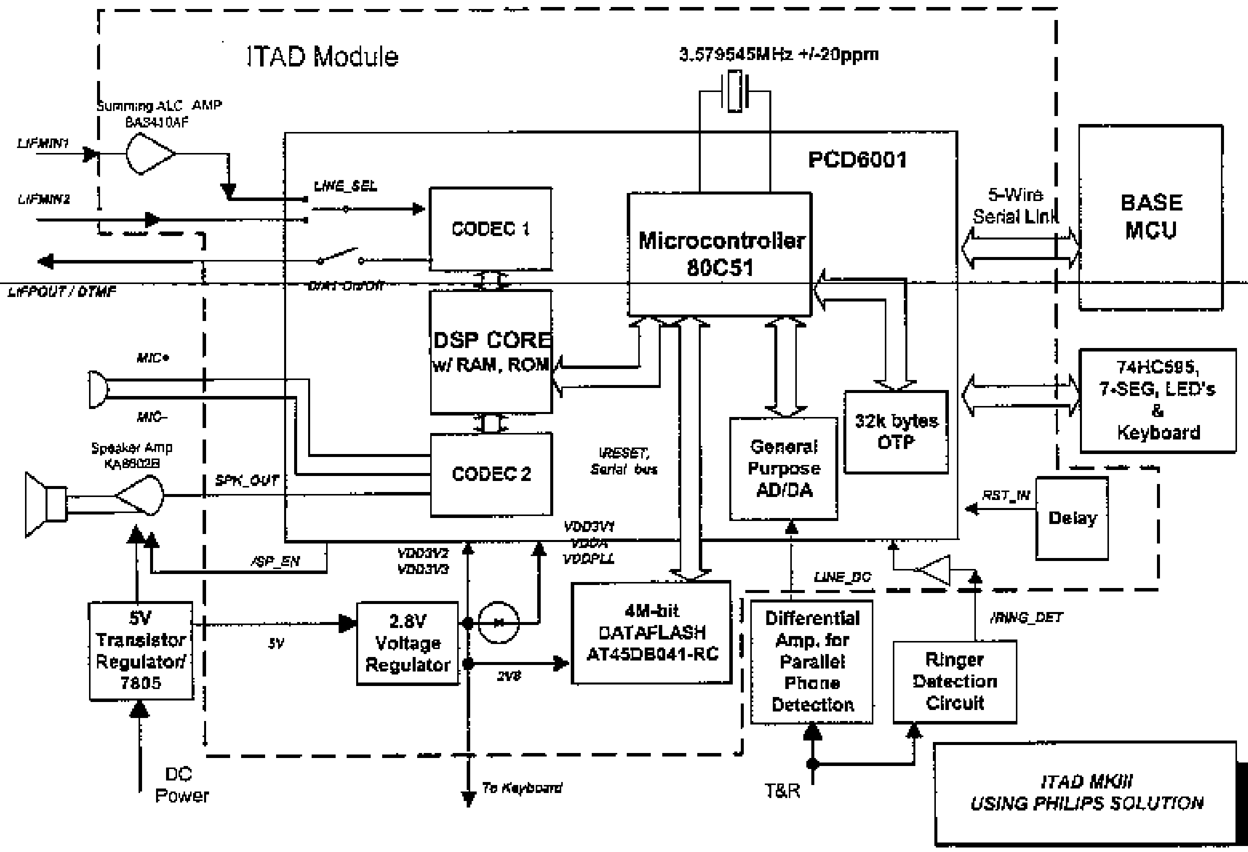
3.3.2 Tx VCO and PLL

The basic operation of the Tx VCO is the same as the Rx VCO, except that the Tx VCO is modulated by the transmit voice and data. The audio deviation is adjusted to a nominal value of 25 kHz by adjusting the summator amplifier gain.

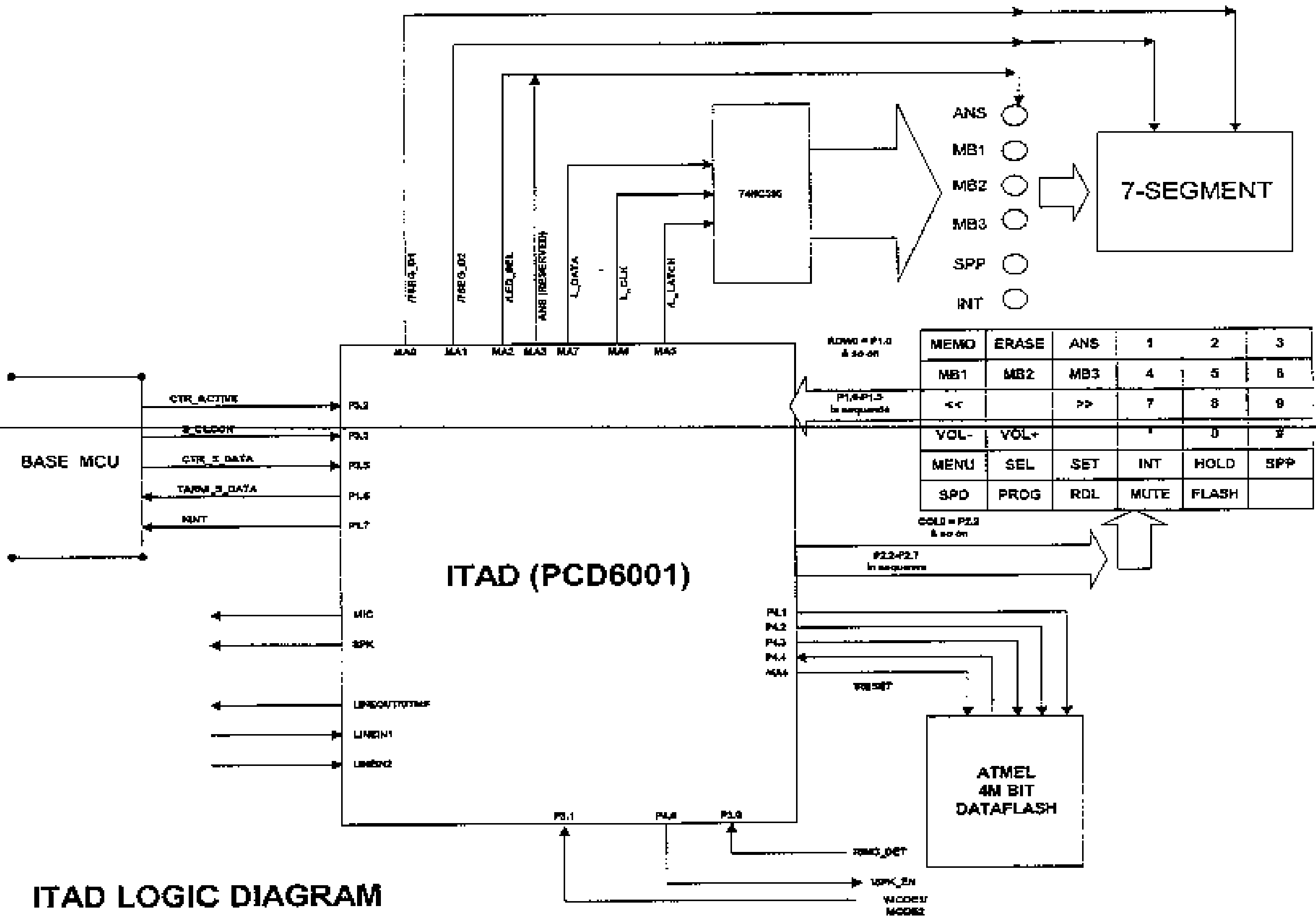
The Tx PLL is combined together with Rx PLL in the block "Dual PLL frequency synthesizer". The loop filter cut-off frequency is about 60-70 Hz. This allows the data and audio modulation to include frequencies down to about 100 Hz. The power-up time of the Tx PLL is not critical.

4 ITAD Mkiii Module :

4.1 General Block Diagram :



4.2 Logic Block Diagram :



ITAD LOGIC DIAGRAM

4.3 Matching with the New ITAD Module in Base Unit :

4.3.1 Additional Circuitry on Base Main :

Additional Circuits	Description
1. Power AMP	Amplifies SPK_OUT signal from ITAD.
2. +5V regulator	Generate +5V from adapter input to provide power to the power AMP and ITAD module.
3. MIC+ & MIC-	Traces of MIC+/- will be laid.
4. SPK+/-	Speaker output from the power AMP.
5. T&R DC monitor	Using LM358 to give the DC level at T&R for parallel phone detection while filtering the ac components. It inputs to the A/D converter of ITAD Module.

4.3.2 Communication with Base MCU:

In this ITAD Module, two communication modes are supported:

5-Wire *Serial Communication Mode*

Stand-alone mode.

The mode can be selected by P3.1. Connecting a 0-Ohm to Vss will set to Mode1. Otherwise the unit will work in Mode2.

4.3.2.1 Mode 1: 5-Wire Serial Communication Mode

In mode 1, ITAD module can be configured as Stand-alone with Digital Speakerphone & Handset Remote option. A 5-wire serial link is built between ITAD module and Base MCU so that they exchange data/command.

4.3.2.2 Mode 2: Stand-alone Mode

In this mode, no direct data communication is made between Base MCU & ITAD Module. This interface is exactly the same as ITAD MKII-A. Simple monitoring of each party is done by several logic:

BU_HOOK - Hook switch input from the base MCU, which indicates base cordless, takes the line. A high level indicates that the base is off-hook.

- *RING_DET* - Logic input from the ringer detection circuit.
- *HOOKSW - Logic* output from ITAD as hook switch.
- *PARA_DET and LINE_DC* - *PARA_DET* Logic input from parallel phone detection circuit. A zero pulse of minimum length 80ms will be recognized as parallel phone pick-up. Note that in Mode2, Line_DC will also be monitored. If one of these inputs meets the requirement of parallel phone detection, ITAD will go on-hook.

Thus there is no need to reserve extra I/O's on the base MCU when adding the ITAD module into the basic platform.

Important notes :
When interfacing with the base circuit, one must be very careful on the high logic level feeding into ITAD input pins (including LINE_DC). Except for the \RING_DET pin, other input pins cannot withstand a voltage of 3.2Vdc. Any higher voltage will burn the DSP chip! For the LINE_DC, the maximum convertible input voltage is only up to 2V. Any voltage higher than 2V will only give the maximum digital value.

4.3.3 Job Sharing Between Base MCU & ITAD Module :

The ITAD module in SPP-A941 works in Mode 2. ITAD will perform the following functions independently:

It will detect the exact ringer pattern from the *\Ring_Det* logic of the ringer detection circuit in the base main.

It receives *\Para_Det* input from the parallel phone detection circuit located on the base main. It also monitors Line_DC pin. If one of these inputs meets parallel phone detection condition, ITAD will go on-hook.

It also monitors the base hook switch status by *Bu_Hook*.

ITAD will get the line through *\HookSW by* itself.

4.4 *Digital Logic Interface:*

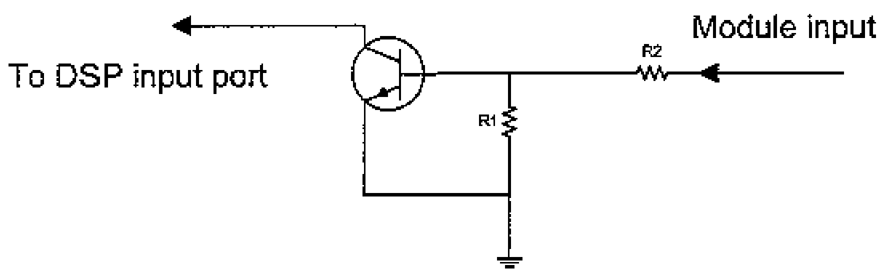
Internally, the DSP chip I/O operates at 2.8V. To interface with external logic, level shifting is necessary. Basically, ITAD MK3 is designed to interface directly with 3V systems.

4.4.1 Digital Input:

There are two kinds of digital inputs: Internal pull-up (ROWs & 5-wire input logics) and Internal pull-down (\Ring_Det). All inputs can withstand 3.2V maximum. For \Ring_Det pin, it can even up to 5V due to internal buffer built.

Internal pull-up inputs are directly fed to the input ports P1 & P3 of the DSP, which are internally pull-high with resistance about 100k. Pull-up resistors thus can be eliminated for the keyboard scanning input ports.

\Ring_Det is an internal pull-down input are connected with resistor network before feeding into the DSP input ports :



For \Ring_det input logic, R1=47k; R2=47k and one transistor is inserted before going into the DSP input port. Such a high value of R1 & R2 is to prevent loading the ringer detection circuit in the base main.

4.4.2 Digital Outputs :

There are two kinds of digital outputs that are needed to connect with the external world: Open-Drain (5-Wire outputs) & Push-pull (other outputs).

Open-Drain outputs can be pulled up to maximum 3.2V. In order to obtain high speed (especially for the 5-wire outputs), the pull-up resistor is recommended to be 1k Ohm.

\SP_EN is internally pulled up to ITAD Vcc (2.8V) through a 4.7k resistor.

Push-pull output will only connected to keyboard to control serial latch, LED and keyboard scanning. They are directly fanned out from the DSP output ports and thus can only provide high level at 2.8V. To eliminate level-shifting circuits, the components like serial latch, LED powers is necessary to connected to ITAD Vcc (2.8V) which is the same as that of DSP I/O core supply.

4.5 LED & 7-SEG Driving Method:

To increase the flexibility & simplicity, the LEDs like Mailbox, ANS, will be put together as another "7-segment digit" and will be driven as if the third 7-SEG digit using the eight bits of 74HC595 & \LED_SEL. This can allow the user to expand the LED outputs as large as eight without adding extra pin-outs from ITAD. The user can add the extra two LEDs defined by Base MCU that can be controlled through the serial link from the Base MCU. For example, one can add "Privacy" LED by just connecting the LED to the latch output and of course, the Base MCU must take care of the LED definition by itself. Finally, this method can also help to reduce the static current drawn.

The user must be careful in choosing the 7-SEG LED. The reason is now the driving voltage is only at 2.8V but in the market, 7-SEG LED has high forward voltage drop (~ 2V). The series resistors to the LEDs should be kept at a reasonable higher value so that the variation in the diode forward drop will not make a significant change in the LED current. Basically, the rule is: either chooses a 7-SEG with a lower forward drop or use a brighter one so that the driving current can be reduced.

4.6 Tunable Parameters:

To increase the flexibility of the ITAD module, some parameters can be set by the user so as to match the ITAD module with his own base design, for example, changing the line input gain, using a low-sensitivity microphone, etc. These parameters are preprogrammed into a specific area (4k byte-size) of the Flash in the voice prompt injection process during production. The DSP will look for a *parameter download byte* to get the parameter from the Flash, otherwise default settings are used.

Parameter	*Description*	*Default Setting (hex)*
VOX_rec	VOX detection threshold during line record (7FFF-0000); see table 1*	0B00
DFD_rec	DTMF detection threshold during line record (7FFF-0000); see table 2*	0333
DFD_play	DTMF detection threshold during line play (7FFF-0000); see table 3*	0033
BUSY_rec	Busy tone detection threshold during line record (7FFF-0000); see table 4*	00C0
MIC_rec	Microphone gain in local recording (0-F; from +22dB to +37dB)	4 (=+26dB)
Line_Drop	Line DC drop to be recognized as parallel phone (01-FF; 7.8mV – 2000mV)	0A (=0.078V)
Drop_delay	Delay time for parallel phone detection to operate after ITAD being off-hook (01-FF; 0.01sec – 2.55sec)	64 (=1s)
Drop_dur	Parallel phone continuous off-hook time to be recognized (0-F; 0ms – 1500ms)	5 (=500ms)
Tbd	Speakerphone parameters	tbd

4.7 *Audio Quality:*

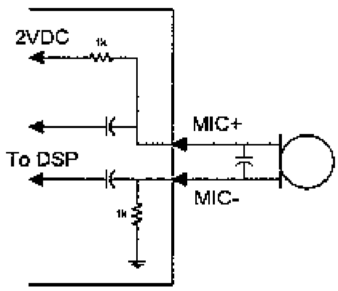
The compression algorithm in Philips chip is called *"Harmony"*. Generally, it can reproduce the compression voice with good quality and lower bit rate compared with the traditional. However, intensive tests show that there are actually some precautions in applying this *"Harmony"* encoder, especially in the MIC recording path!

4.71 "Scratching noises"

This noise will appear when the recording male voice signals which are enriched with high frequency components and in a quiet environment. The reason for this terrible noise is that the *"Harmony"* algorithm is very sensitive to high frequency. To make sure a good voice reproduction, a flat response at high frequency is necessary.
On the other hand, in the line-recording path, experimental results show that it can hardly reproduce this phenomenon. It is due to the fact that the line signal normally is band-limited. For safety, as the sender telephone may have a high pre-emphasised sending response, it is recommended to have a 3dB roll-off at about 3kHz when feeding to ITAD line recording input (LIFMIN1).

4.72 How to avoid the "Scratching noises" in MIC recording

Simply by adding a capacitor across the MIC as shown, you can damp the high frequency components from the MIC.



To find out the final acoustic performance, you can enter the test mode and sweep the frequency response for the path "from MIC to line". By setting the line interface without any roll-off, the measured frequency response actually shows the response from MIC to the DSP encoder.

Important notes:

The fine-tuning must be done with the whole cabinet and MIC holder in a quiet environment. All these things have a large effect on the acoustic response from MIC to the DSP encoder.

4.8 Audio Level Settings:

4.8.1 Line Recording to DSP :

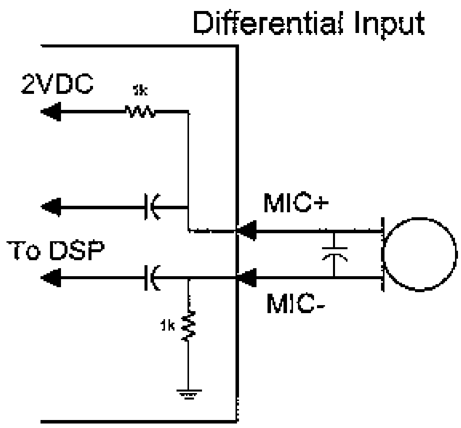
For line recording, there is a built-in analog ALC circuit inside the ITAD module. Signal will pass through this circuit in ICM recording from LIFMIN1 pin and the DSP will select LIFMIN1 pin signal input during line recording. With reference to Sony's product, there is a requirement for the ALC output curve (shown in Section 7.5).

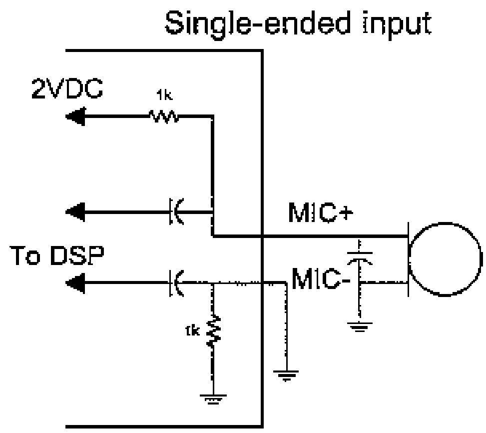
In the application circuit, LIFMIN1 level will have an attenuation of 8dB from the signal at T&R. The ALC can be controlled curve by adjusting the gain from T&R to LIFMIN1. However, it must take care of all the thresholds (e.g. DFD_rec, VOX_rec & CPD_rec) and must set to other values when adjustment of this gain is made.

The designer must also note that LIFMIN1 maximum allowable level is 3Vpp (+3dBm).

4.8.2 MIC to DSP :

For this path, the signal from MIC is differentially fed to the DSP. The bias is provided by ITAD's MIC+ and MIC- pins automatically. With this configuration, common-mode noise rejection can be optimised. However, single-ended input is also feasible by connecting MIC- to GND as shown. The internal DSP gain will be the same in both cases but the overall gain may be different due to change in the input impedance. The parameter *MIC_rec* can be changed according to the type of microphone used. Also the user must be aware of the maximum driving current for the MIC is limited to 0.4mA and this results in minimum MIC biasing voltage for differential and single-ended inputs are 1.2V & 1.6V respectively.





4.8.3 Local Playback & Call-screening :

For both local playback & call screening, there are 8 levels of volume controls (Level 1 of call screening is speaker-mute).