# U3760MB
# Application and Adjustment Hints

**Authors:**

Wilhelm Rauter
Joachim Kühnle

**Publisher:**

TEMIC Semiconductor GmbH
Theresienstraße 2
P.O.B. 3535
D – 74025 Heilbronn
Phone : +49 7131 67-2594
Fax : +49 7131 67-2423
http://www.temic-semi.de

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# General

The U3760MB is suitable for low-voltage, low-end telephone applications. It consists of a dialer, a ringer and a speech circuit in a single chip.

The following sub-chapters describe the available functions of the dialer, the ringer and the speech circuit.

## Speech Circuit

The speech circuit contains a transmit amplifier suitable for dynamic, electret and piezo microphones, and a receive amplifier for driving dynamic earpieces. The latter is either connected symmetrically to RECO1/ RECO2 via CREC2 or asymmetrically connected to RECO1 against ground via CREC2 (see figure 7). The output amplifier can drive the piezo earpiece. The speech circuit also offers mute possibilities – internally by means of the dialer or externally by using Pin MUTE (internal pull-up resistor).

## Dialer

To control the dialer functions, 16 keys (Pin = C1/C2/C3/R1/R2/R3/R4) are available.

For indication that a key is pressed during pulse mode, the dialer has a key-tone output signal (Pin KT).

The break-make ratio can be selected either by connecting pin B/M to ground (resulting in a ratio of 33/67) or by leaving the pin open which leads to a ratio of 40/60.

DTMF mode can be selected by connecting MODE to High = $V_{DD2}$. PULSE mode can be selected by connecting MODE either to ground (then 20 pps is selected) or to leave MODE open, then 10 pps is selected (Pin MODE).

Flash times can be realized with the keys:

- F1 =  94 ms
- F2 = 250 ms
- F3 = 600 ms

Temporary DTMF dialing is available by using the key */T

- Changes from pulse dialing mode (MODE = GND or floating) via key */T to DTMF mode, whereas pressing the key */T does not cause a DTMF signal but a key-tone control signal to be sent out (see also timing diagram in the data sheet).
- If MODE = "High", key */T delivers a DTMF output signal

The re-dial function is available when key R/P is pressed first after going off-hook. If the number of digits is >32, re-dialing is impossible. When key R/P is not the key pressed first after the off-hook condition, the next digit is sent out after a pause of 3.6 seconds.

The dialer logic is driven by a 3.579545-MHz oscillator, stabilized by a crystal or a ceramic resonator of 3.579545 MHz. Due to internally integrated capacitors, no additional external components are needed when using a ceramic resonator. The latter is possible as the frequency deviation of the chip is low (see table 3).

## Ringer

The U3760MB has a two-tone ringer. Adjusting the oscillator frequency affects the repetition rate. Furthermore, the IC has an adjustable ringing impedance, an adjustable threshold for the ringing voltage detection and an adjustable ringer volume.

Table 1. Mode selection

| | MODE = | | |
|---|---|---|---|
| | Low = GND = 20 pps | Open = floating =10 pps | High = $V_{DD2}$ = DTMF |
| B/M = Low = GND = 33/67 | 16.5 ms/ 33.5 ms | 33 ms/ 67 ms | DTMF mode |
| B/M = Open = float. = 40/60 | 20 ms/ 30 ms | 40 ms/ 60 ms | DTMF mode |

pps = pulse per second

# Adjustment Procedure for the Speech Circuit

**Main steps in telephone adjustments:**

**Step 1**

Adjust the DC characteristic (due to the influence of RIMP1 and RIMP2, this step has to be carried out first).

**Step 2**

Adjust the AC impedance (also called return loss), for example, 600 R or complex impedance (e.g., in Germany).

**Note: Never change the impedance after the second step, otherwise all further steps must be carried out once again (especially the side-tone adjustment).**

**Step 3**

Adjust the transmit gain and the transmit frequency response.

**Step 4**

Adjust the AGC (Automatic Gain Control = gain correction dependent on the line current).

**Step 5**

Adjust the side tone (make sure that the impedance is already adjusted, see step 2).

**6. Step.** Adjust the receive gain and the receive frequency response.

**The following adjustment steps are independent from step 1 to 6 (i.e., they can be carried out before or after step 1 to 6 ).**

**Step 7**

Adjust the ringer impedance.

**8. Step.**

Adjust the ringer oscillator.

**9. Step.**

Adjust the ringer threshold.

**10.Step.**

Adjust the ringer volume.

# Adjustment Hints

## DC Characteristic

The resistor RDC1, connected from Pin RDC to ground, sets the slope of the DC characteristic (see figure 3).

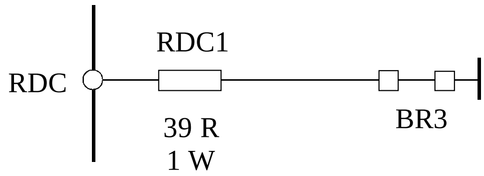


Figure 1.

Instead of using the bridge BR3, a diode can be added to shift the complete DC characteristic in parallel.

At the maximum possible current in an application, adjust the voltage to less than 10 V.

Calculation of the circuit's power dissipation in speech-/DTMF mode:

$$P_{U3760MB} = V_L \times I_L - [RDC1 \times IRDC1] - [(RIMP1 + RIMP2) \times 2\ mA]$$

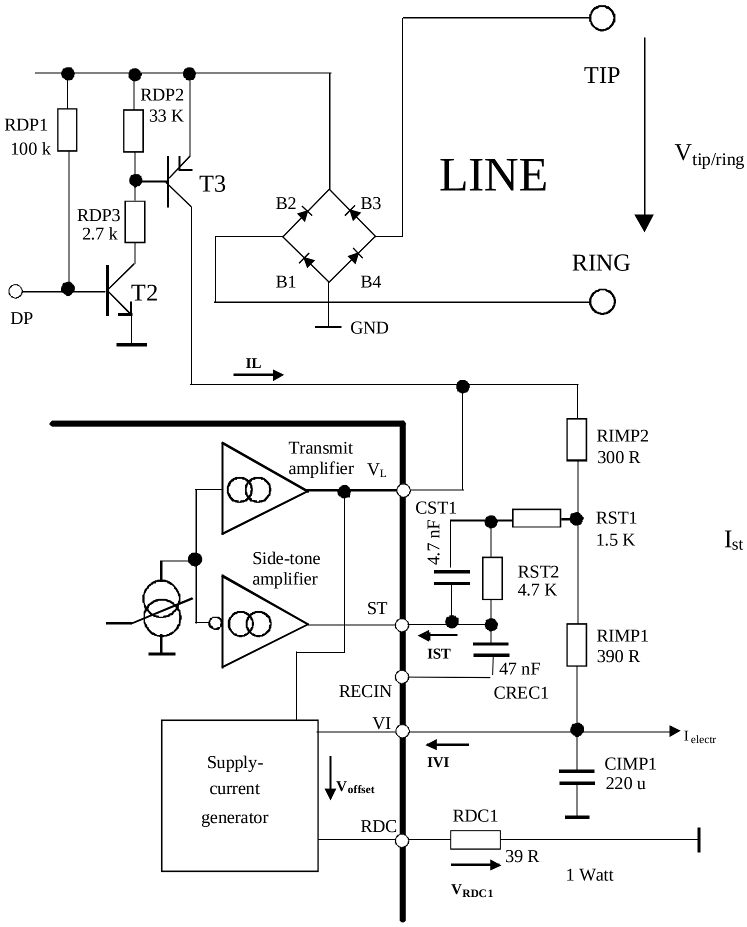


Figure 2.

The DC operating point in speech mode is given by:

$$\begin{aligned} V_{tip/ring} = \; & V_{RDC1} + V_{offset} \\ & + I_{VI} \times (RIMP1 + RIMP2) \\ & + I_{st} \times RIMP2 \\ & + (RIMP1 + RIMP2) \times I_{electr} \\ & + V_{cesat}\,(T3) \\ & + 2 \times V_{diode} \end{aligned}$$

$I_{st} \approx 270\ \mu A$, $I_{VI} \approx 1.2\ mA$, $V_{offset} \approx 2.84\ V$

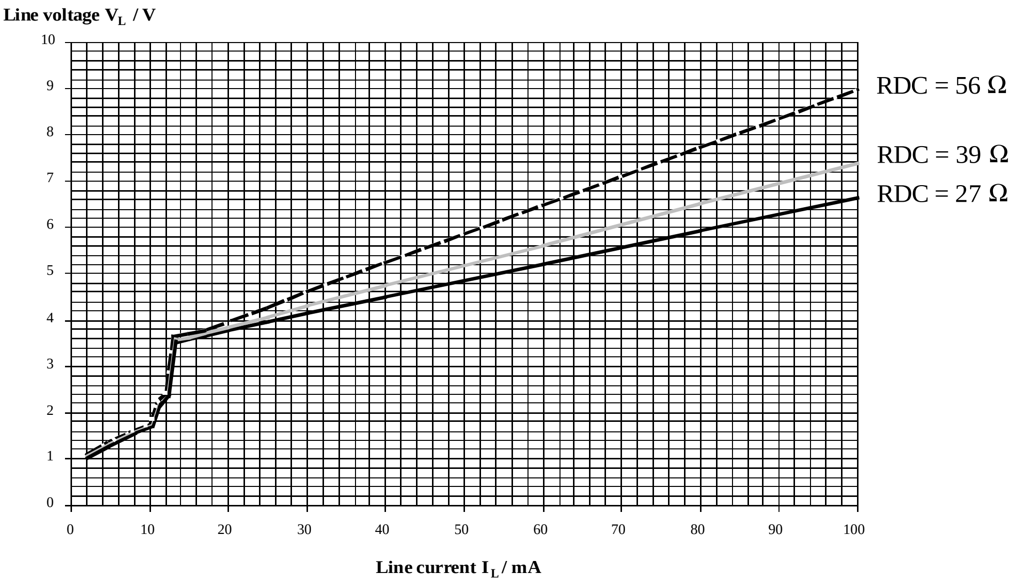


Figure 3. DC characteristic (typ.)

## AC Impedance

To adjust the AC impedance, see figure 4. For 600-R measurement results, see figure 5.

In general, the AC impedance consists of: (RIMP1 + RIMP2), CIMP1 and CIMP2.

In a telephone application, there might only be a resistive impedance such as, for example, 600-R; or a complex impedance such as ZR, e.g., in Germany.

Resistive impedance:

(RIMP1 + RIMP2), CIMP1;
where CIMP1 is for AC coupling only.

Complex impedance:

(RIMP1 + RIMP2), CIMP1 and CIMP2;
where CIMP1 is for AC coupling only and CIMP2 is used to realize the complex impedance together with RIMP1 + RIMP2.

The ratio RIMP1 to RIMP2 depends on the DC characteristic, the maximum transmit and the maximum receive level on $V_L$ (this depends on which country is concerned). For applications using high DC characteristics, RIMP1 may have the value 0. In this case, the AC impedance consists of RIMP2 and CIMP1 (resistive impedance) or RIMP2, CIMP1 and CIMP2 (complex impedance).

Please note that each low-ohmic resistive load from the positive supply of bridge-to-ground influences the AC-impedance adjustment. This is valid especially for the resistor RDP3 (necessary for biasing transistor T3). The characteristic of RPD3 is as if it were connected in parallel to RIMP1 + RIMP2.

**Example** (neglecting RDP3):

Line = 600 Ω ⇒ (RIMP1+ RIMP2) = 620 Ω

Line = 900 Ω ⇒ (RIMP1 + RIMP2) = 910 Ω

Table 2 shows the AC adjustment for several countries (application circuit as shown in figure 22, RDP3 = 2.7 k).

Table 2. AC adjustment

| Country | RIMP1 | RIMP2 | CIMP1 | CIMP2 |
|---|---|---|---|---|
| Japan | 390 | 300 | 220 u | 2.2 n |
| Great Britain | 330 | 560 | 220 u | 220 n |
| China | 330 | 390 | 220 u | 68 n |
| Germany | 560 | 330 | 220 u | 68 n |

The test circuit for line-impedance adjustment is shown in figure 4.

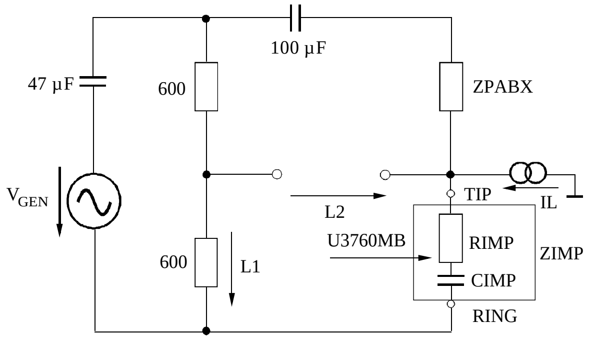


Figure 4. Test circuit for line-impedance adjustment

Best matching for the line impedance can be achieved when the bridge is balanced (i.e., ZIMP = ZPABX), where ZPABX is the impeddance which should be matched.

Return losses are acceptable if RL is minimized [RL (dB) = L2 (dBm) - L1 (dBm)].

Figure 5 shows an example of AC-impedance adjustment.

## Transmit Gain and Frequency Response

The entire gain consists of a fixed microphone-amplifier gain and a constant transmit amplifier gain.

The output level on the line is dependent on the AC impedance and the microphone-ampli-fier input level. The only method of adjusting the transmit level is to adapt the input circuit to the microphone sensitivity.

- In the case of an electret microphone, the level as well as the frequency response can be adjusted via RMI1/ RMI2/ RMI3/ RMI4/ RMI6 and CMI1/ CMI2/ CMI4.
  Both resistors – either RMI1/RMI2 or RMI6 – are recommended for gain adjustment.
- In the case of a dynamic microphone, the microphone can be connected directly to MIC1/MIC2. If a level reduction is necessary, this can be done via a parallel resistor. The frequency response can be affected with a parallel capacitor or by changing the housing conditions (damping material, additional holes or the hole diameters of the microphone).

Another possibility of adjusting the level as well as the frequency response is connecting a network between pins MICO and TIN. This is, however, not recommended, as a net between these two pins will not only influence the transmit signal but also the DTMF signal.

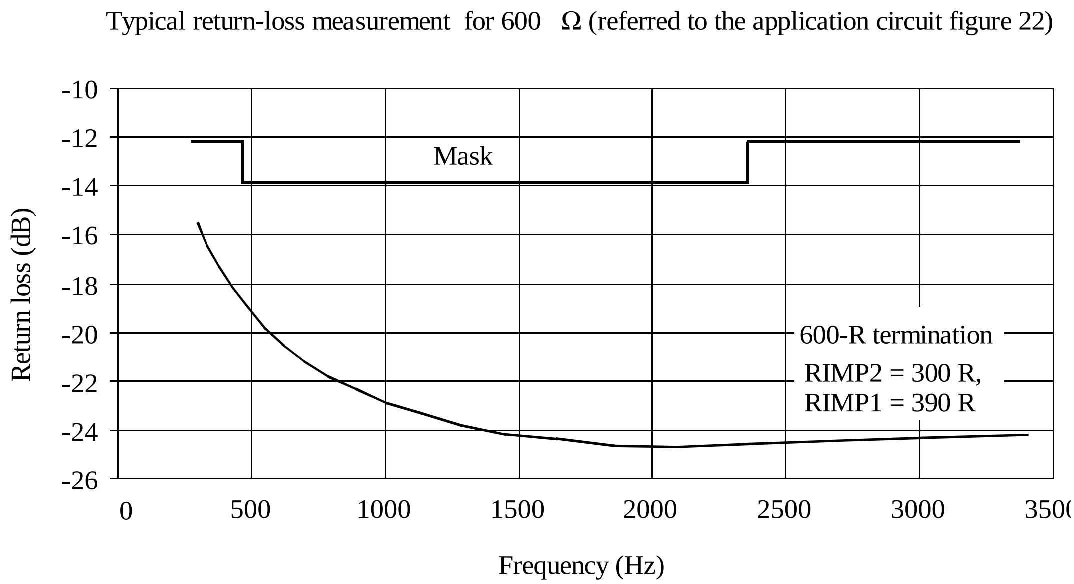


Figure 5. Example of ac-impedance adjustment

**Transmit frequency response**

- CMI4 cuts high frequencies and reduces electromagnetic influence (EMI).
- CMI1 and CMI2, along with RMI6, cut low frequencies.
- CTIN cuts low frequencies, and also affects the DTMF frequency response.

The **input impedance** of the microphone amplifier is 75 kΩ typically.

The **output impedance** of the microphone amplifier is

- 400 Ω typ. in speech mode
- 16 kΩ typ. (bias on) in dial mode
  the **input impedance of TIN** is 5.8 kΩ typ.

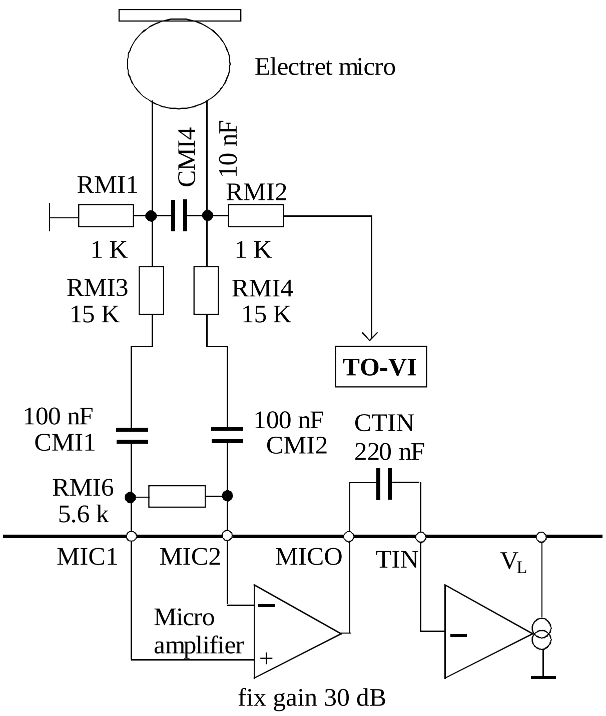


Figure 6.

## Receive Gain and Frequency Response

**Receive gain**

The receive gain can be adjusted by modifying the ratio of RIMP2 to RIMP1. RIMP2/RIMP1 is a voltage divider for the receive signal (see figure 2).

Maximum gain is achieved if RIMP2 = 0 R.

Make sure that the sum of RIMP2 and RIMP1 is always kept constant so as not to affect the AC impedance.

A further adjustment possibility is to use a resistor in series with the earpiece (RREC1).

**Receive frequency response**

The frequency response can be adjusted high ohmic via an RC net at RECIN (CREC1, 60 k typ.) or low ohmic at the ear-piece output RECO1/ RECO2 or directly at the ear piece.

- CREC1/ CREC2 cut low frequencies
- CREC3 cuts high frequencies
- An increase of RREC1 also increases the receiver’s dynamic range
- An increase of RIMP2 decreases the RECIN input (as a result, the total RX gain is smaller)

An adjustment at RECIN is not recommended as this could influence the side-tone characteristics.

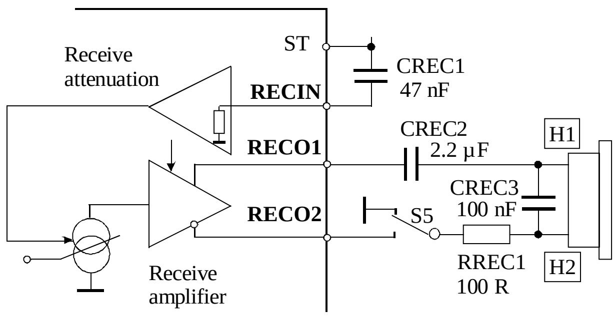


Figure 7.

## AGC Characteristic

The resistor RAGC1 defines the amount of gain reduction for transmit and receive mode. An open input de-activates the AGC.

Please note that AGC characteristic is influenced by the DC characteristic (RDC1). Therefore, the adjustment of the DC characteristic has to be carried out first, as it is stated in the chapter "Main Adjustment Steps", and must not be changed afterwards, otherwise the AGC must be adjusted once again.

Please note also that in dial mode, the AGC function is switched off.

A typical AGC characteristic is shown in figures 9 and 10.

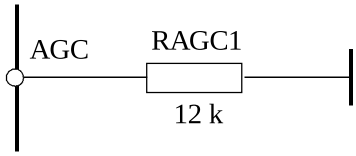


Figure 8.

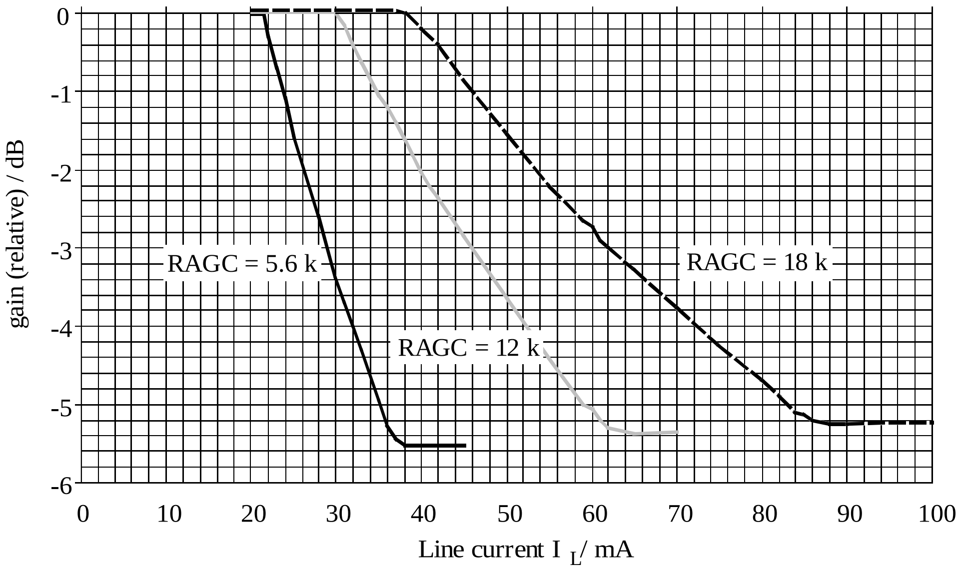


Figure 9. AGC characteristic (typ.), receive mode

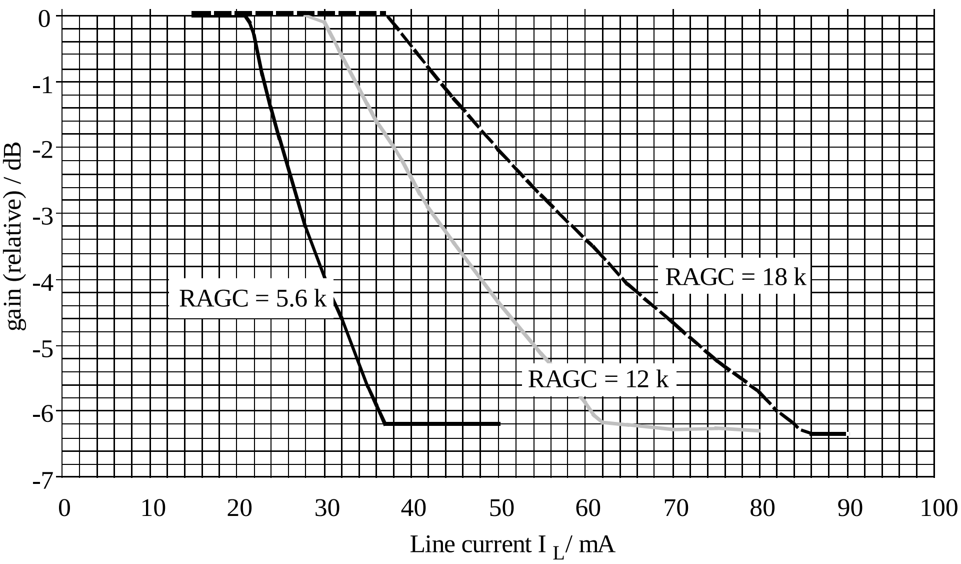


Figure 10. AGC characteristic (typ.), transmit mode

## Side Tone

The amplified microphone signal controls two current sources, $I_{mod}$ and $I_{st}$. The ratio of $I_{mod}/I_{st}$ is 20. An increase of $I_{mod}$ will also cause $I_{st}$ to increase. The line voltage $V_L$ will then go down, while the voltage across $Z_{st}$ ($V_{zst}$) will go up. If the side-tone network $Z_{st}$ is perfectly matched to the lumped line impedance $Z_L$ (impedance of the telephone RIMP1 + RIMP2 in parallel to the impedance of the line itself), the voltage at ST will remain constant, thus resulting in a perfect side-tone cancellation.

This description assumes, that RIMP2 is 0 Ω ($Z_{st}$ is directly connected to $V_L$). In all other cases, the voltage divider RIMP1/ RIMP2 has to be taken into account as shown in the equation for the constant a (see below).

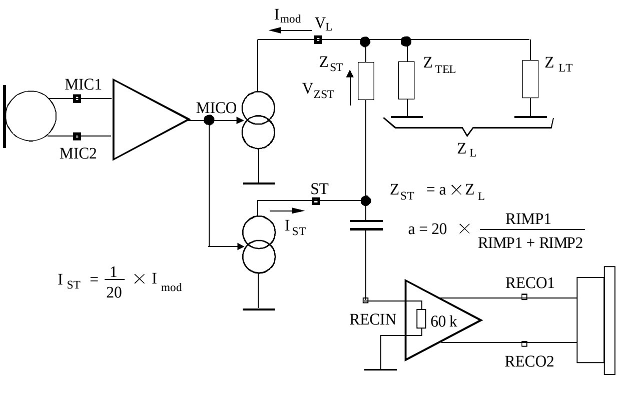


Figure 11.

### How to adjust the side-tone network

- Monitor RECO1/RECO2 while tuning RST1, RST2, CST1 until a flat frequency response characteristic is gained.
- For first try, use the following estimation:

  RST1 + RST2 is about 20 times the parallel impedance of RIMP1 + RIMP2 and the real part of ZLT, the line network terminal (do not forget the parallel impedance of RDP3).
- As a first approach, try to use the ratio of RST1/RST2 while keeping CST1 (about 2.2 nF) constant.
- Then change CST1 while keeping the ratio of RST1/RST2 constant.
- For fine adjustment, repeat the second and the third step to achieve a better result.

In certain cases, especially if the DC characteristic is flat, the receive level high and the line current low, the negative half cycle of the receive signal might cause dynamic range problems, thus making it impossible to adjust the side tone. In such cases, RIMP1 and RIMP2 act as a divider for the receive signal and can prevent the dynamic range problem. However, if the receive signal is divided, the receive output level is reduced.

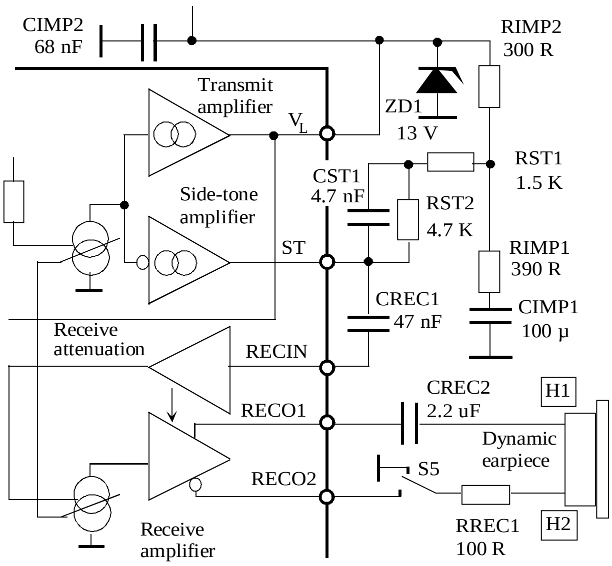


Figure 12.

## Dialer

### Pulse dialing (see figure 14)

For pulse dialing, there is no adjustment necessary. The dial pulse comes from pin DP and goes to T2/T3 to realize the break/ make.

For T2 and T3, it is recommended to use bipolar transistors instead of MOS transistors. This is because during low supply-voltage operation, a MOS transistor might not work due to his high threshold voltage.

## DTMF filter (see figure 13)

In most countries, a DTMF filter is not needed. If a DTFM filter is necessary, the following procedure has to be carried out:

Use the structure shown in figure 13 and adjust the components to fulfill the desired specification. The DTMF signal is finally fed into the transmit path on pin MICO.

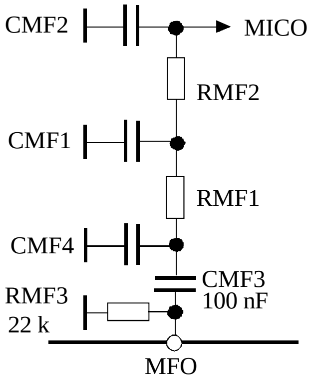


Figure 13.

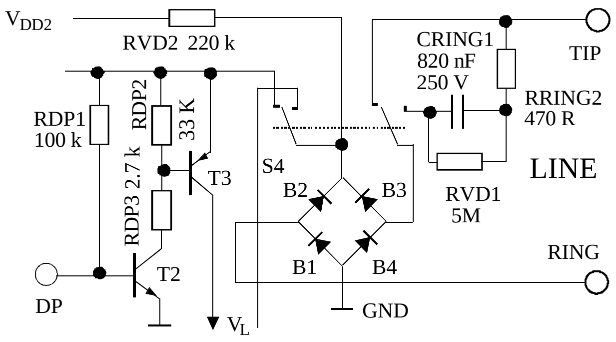


Figure 14.

## Mode switching

- Pin MODE selects DTMF/pulse (10/20 pps)
- B/M ratio in pulse mode is selected by BM
- The DTMF signal remains as long as the key is pressed

## DTMF output frequencies

Ceramic resonators from MURATA can be connected directly to XT and XTB; external components are not needed.

The maximum frequency error can be calculated by the formula:

$Error_{max}$ = (dialer error) + (xtal deviation from 3.579545 MHz) ± (xtal tolerance range)

Example: CSA3.58MG300 – FGA
given xtal frequency = 3.566 MHz
Deviation from 3.579645 MHz is -0.38%

The maximum frequency deviation gained from the specified value is:

- R2: -0.52% – 0.38% – 0.4% = -1.3%
- R1: +0.74% – 0.38% + 0.4% = +0.76%

Table 3. DTMF output frequencies
(oscillator: 3.579545 MHz)

| | Specified/Hz | Actual /Hz | Error % |
|---|---|---|---|
| R1 | 697 | 699 | +0.28 |
| R2 | 770 | 766 | -0.52 |
| R3 | 852 | 848 | -0.47 |
| R4 | 941 | 948 | +0.74 |
| C1 | 1209 | 1216 | +0.57 |
| C2 | 1336 | 1332 | -0.30 |
| C3 | 1477 | 1472 | -0.34 |

## MFO, MICO

MFO is a source-follower output. A 10-k to 22-k resistor at MFO to ground is necessary for current flow. Direct AC coupling overdrives the TIN input. Therefore, a series resistor has to be used to achieve an appropriate TIN level. Biasing of the MFO output is about ½ $V_{DD1}$ during dial mode only. MFO is grounded by an internal N-channel FET, except when bursting out the DTMF signal.

The MICO output bias is always 1.0 V.

## $V_{DD1}$, $V_{DD2}$

The line current is controlled by make/ break action during pulse dialing. During the break period, there is no $V_L$ from the line. Due to this behavior, $V_{DD1}$ is no longer fed from the line current and $V_{DD2}$ becomes the supply voltage of CVD1. For this reason, CVD1 needs a relatively large capacitor (≈ 220 µF).

Max. current consumption at $V_{DD2}$ = 2.5 V:

| | |
|---|---|
| Pulse dial mode | $I_{DD1} + I_{DD2} < 600\ \mu A$ |
| DTMF mode | $I_{DD1} + I_{DD2} < 1$ mA |

Flash mode $I_{DD1} + I_{DD2} < 500\ \mu A$

$V_{DD2}$ operation range: 2.0 V up to 5.5 V.

When the line current is high, a 5-V Zener diode is necessary at $V_{DD2}$. It also ensures ESD protection.

Please note that in pulse-/DTMF dial mode, microphone and receiver are both muted by the dialer. Muting can also be activated by an external switch to GND at pin MUTE.

### GND1, GND2

GND1 is internally connected to the speech- and ringer-block ground. GND2 is internally connected to the dialer-block ground. These two grounds must be connected together as close as possible to the pins GND1, GND2.

### DTMF monitor volume

The tone output can be monitored at the receiver during DTMF dialing. The U3760MB turns off the side-tone amplifier during DTMF output and mutes the receiver amplifier by 29 dB approx. This mute level is fixed. The $V_L$ output tone signal goes directly to RIMP2, to the side-tone network and to RECIN. Meanwhile, the side-tone amplifier is off. Therefore, the receiver sound is only determined by the line signal.

The tone output from $V_L$ is divided by RIMP1 and RIMP2, and fed to RECIN. Between RECO1/2 and RECIN, it is muted by 29 dB.

**Example:**

RIMP2 and RIMP1 are at a ratio of 1:1 between $V_L$ and $V_I$. The signal is fed from the connection of RIMP2 and RIMP1 to RECIN, the following tone-signal level is expected at RECO1/RECO2 regardless of the line conditions.

(monitor output at RECO1/RECO2) =
(DTMF output level at $V_L$ ) –
(loss of divider ≈ 6 dB) – (mute = -29 dB).

### Memory retention for re-dial

The re-dial function is possible when $V_{DD2} \geq$ 1.0 V and the last number is ≤ 32 digits. During on-hook state, there is no $V_{DD1}$ supply.

When CVD1 discharges and $V_{DD2} < 1.0$ V, the re-dial memory is not retained. To avoid this, the current must be supplied directly from the line through a high-impedance resistor. In the standard application circuit, this can be achieved by RVD1 = 5 MΩ/ RVD2 = 220 kΩ (see figure 14).

### Key tone

In pulse dialing mode, each key strike generates a 50-ms burst of 1240 Hz to indicate keyboard activities. This signal can be fed to the earpiece as shown on the demo board schematic. The loudness can be adjusted by means of RMEL1 and CMEL1. If RMEL1 > 10 k, CMEL1 can be omitted in some applications.

### Privacy

The "privacy" function is a convenient feature when muting during conversation is desired. The PRIVACY pin has an integrated toggle switch and can be controlled between normal mode and mute as its input switches from HIGH to LOW.

If this function is not used, PRIV pin can be left open. When the peripheral noise is high, this noise may switch the PRIV function on and off, resulting in unstable speech function. Connecting the capacitor to ground or a 100-kΩ resistor of from PRIV pin to $V_I$ increases the noise immunity. In order to debounce the key, a capacitor must be connected to ground.

Please note that the privacy function is reset automatically by the MUTE signal or by going on-hook.

## Ringer

### Ringer impedance

In on-hook condition, a telephone must have a well-defined input impedance for incoming ringing signals at Tip and Ring.

The U3760MB uses a standard input configuration (see figure 15) which consists of the series resistors RRING2 and RRING1 plus the decoupling capacitor CRING2 and the input impedance of the chip. The real part of the input impedance is mainly formed by the two resistors. A total value of about 2 kΩ will guarantee good results for most applications. Especially at low frequencies (< 20 Hz), the major part of the input impedance is given by the capacitor CRING2. A range of 0.8 µF to 1.5 µF will suit almost every specification.

The input impedance of the chip is in the range of 5 kΩ to 8 kΩ (see figure 21). This is valid as long as the chip is below the turn-on threshold and thus the output stage is off.

## Ringer threshold

The start-up threshold for the ringer output signal is adjustable in a very wide range by simply modifying the resistor RTH1(see figure 19). DC values between 10 V and 24 V at $V_{RIAC}$ can be chosen. Figure 20 shows the resulting adjustment range in the standard application.

The comparator in the chip uses a hysteresis with a fixed lower threshold of about 5.6 V at $V_{RIAC}$. Once the ringer has reached the upper threshold, it will continue to drive the buzzer as long as the $V_{RIAC}$ voltage remains above 5.6 V.

In some countries, the ringer must stop at a certain time after the input signal at TIP and RING has disappeared. In those applications, a diode, DRING1, must be added. The full-wave rectified signal at $V_{RIAC}$ will then be used to trigger an internal mono-flop. As long as the mono-flop is periodically re-triggered, the circuit will continue with the melody. As soon as the input pulses disappear, the chip will wait until the time of the mono-flop ($t_{off} \approx 65$ ms) has elapsed and automatically stop the melody generator.

If this feature is not needed, the diode DRING1 must be shorted.

## Ringer melody

The ringer circuit of the U3760MB generates a two-tone melody. The tone frequencies and the audio-sequence frequency are controlled by a common oscillator circuit.

Thus, changing tone frequencies will also change the audio-sequence frequency. The ratio of the higher output frequency to the repetition rate is fixed to $f_{1H}/f_2 = 80$.

The audio-sequence frequency can also be calculated by $f_2 = f_{osc}/320$.

The oscillator frequency is defined by R and C connected to pin RCK. It is recommended to use 75 kΩ ≤ R ≤ 330 kΩ
and 470 pF ≤ C ≤ 2.2 nF

The oscillator frequency is given by the following calculation (see also figure 17):

$$t_1 = (UB - UA) \times \frac{1}{I} \times C$$

$$t_2 = R \times C \times \ln \frac{UB}{UA}$$

| | | |
|---|---|---|
| where | UB = | 3.2 V typ. |
| | UA = | 0.65 V typ. |
| | I = | 420 µA typ. |

$$\text{thus, } f_{Osc} = \frac{1}{t_1 + t_2} = \frac{1}{T}$$

The audio-sequence frequency, $f_2$, is a function of the oscillator frequency, derived from internal dividers:

$$f_2 = \frac{f_{Osc}}{320}$$

Example:

Derived from the oscillator frequency and the ratio of low frequency to high frequency, the ringing frequency is $f_{1H}/f_{1L} = 5/4$ which means:

$$f_{1H} = \frac{f_{Osc}}{4} \text{ and } f_{1L} = \frac{f_{Osc}}{5}$$

Example: R = 150 k and C = 1 nF
(see figure 18)

$$f_{Osc} = \frac{1}{t_1 + t_2} =$$

$$\frac{1}{\frac{2.55\ V}{420\ \mu A} \times 1\ nF + 150\ k \times 1\ nF \times \ln 4.923} = 4079\ Hz$$

thus,

$$f_{1H} = \frac{1}{4} f_{Osc} = 1020\ Hz,$$

$$f_{1L} = \frac{1}{5} f_{Osc} = 816\ Hz,$$

$$f_2 = \frac{f_{Osc}}{320} = 12.7\ Hz$$

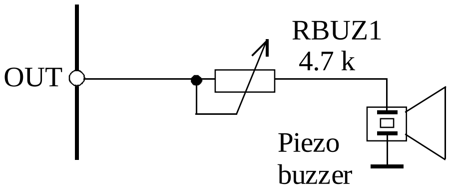


Figure 16.

### Ringing loudness

To adjust the ringing volume, a resistor in series to a piezo transducer is used (see figure 16).

### Impedances of some ports

- MFO
  Active: low impedance (source follower)
  others: ground (NMOS switch on)
- MICO
  Speech mode: ≈ 400 Ω
  dial mode: ≈ 16 kΩ (bias on)
- $T_{IN}$: ≈ 5.8 kΩ
- $V_L$: AC impedance > 50 kΩ
- ST: High input impedance
  (≈ 290 µA current source)
- RECIN: ≈ 60 kΩ (input)
- RECO1/RECO2: 10 Ω (differential, output)
- $V_{RIAC}$: diode × 2 (series) + 80 kΩ
- $T_{HA}$: ≈ 30 kΩ

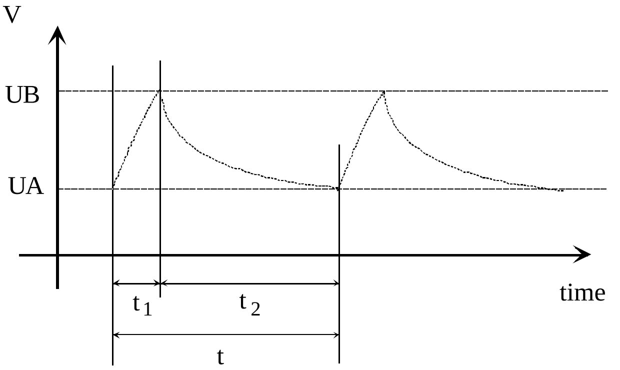


Figure 17.

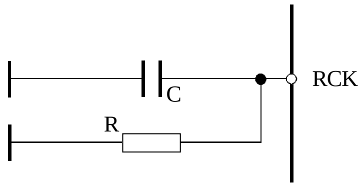


Figure 18.

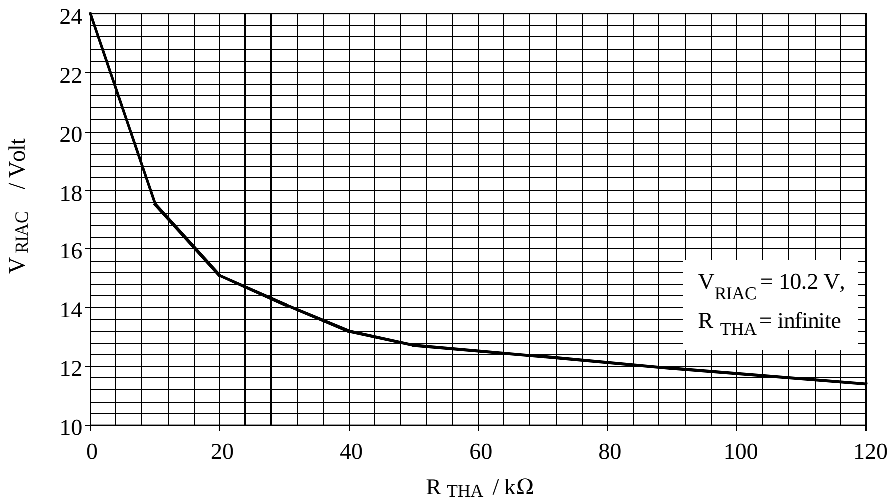


Figure 19. Threshold switch on ringer voltage ($V_{RIAC} = f_{RTHA}$ measured values)

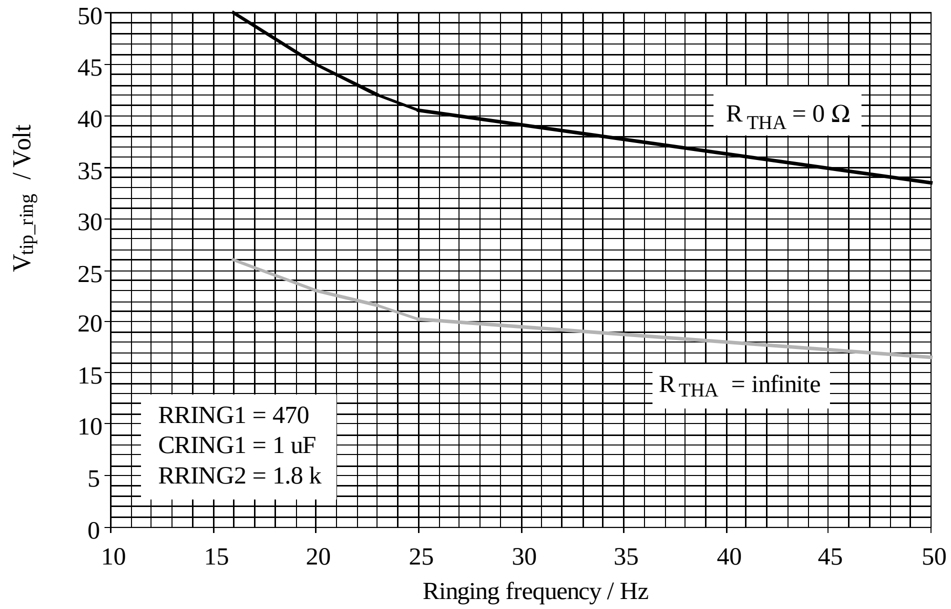


Figure 20. Ringing-voltage threshold (typ.) at tip/ring as a function of the ringing frequency

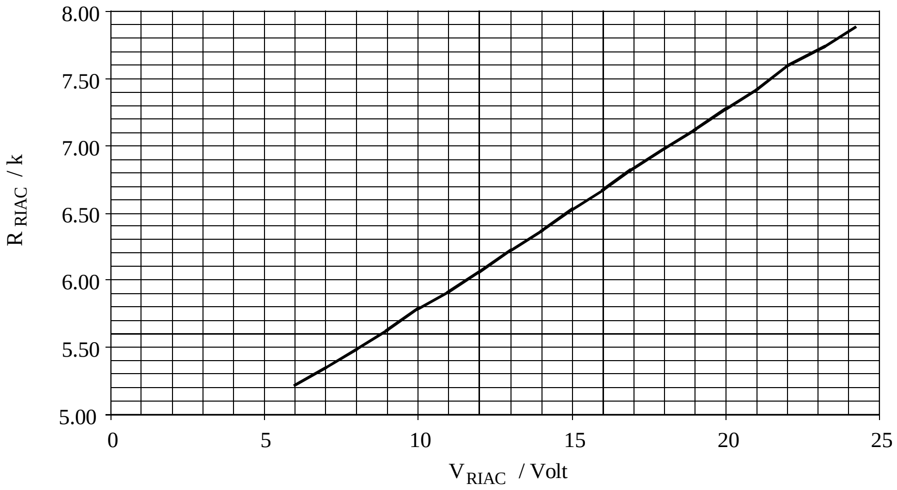


Figure 21. Ringer impedance in off-state (typ.)

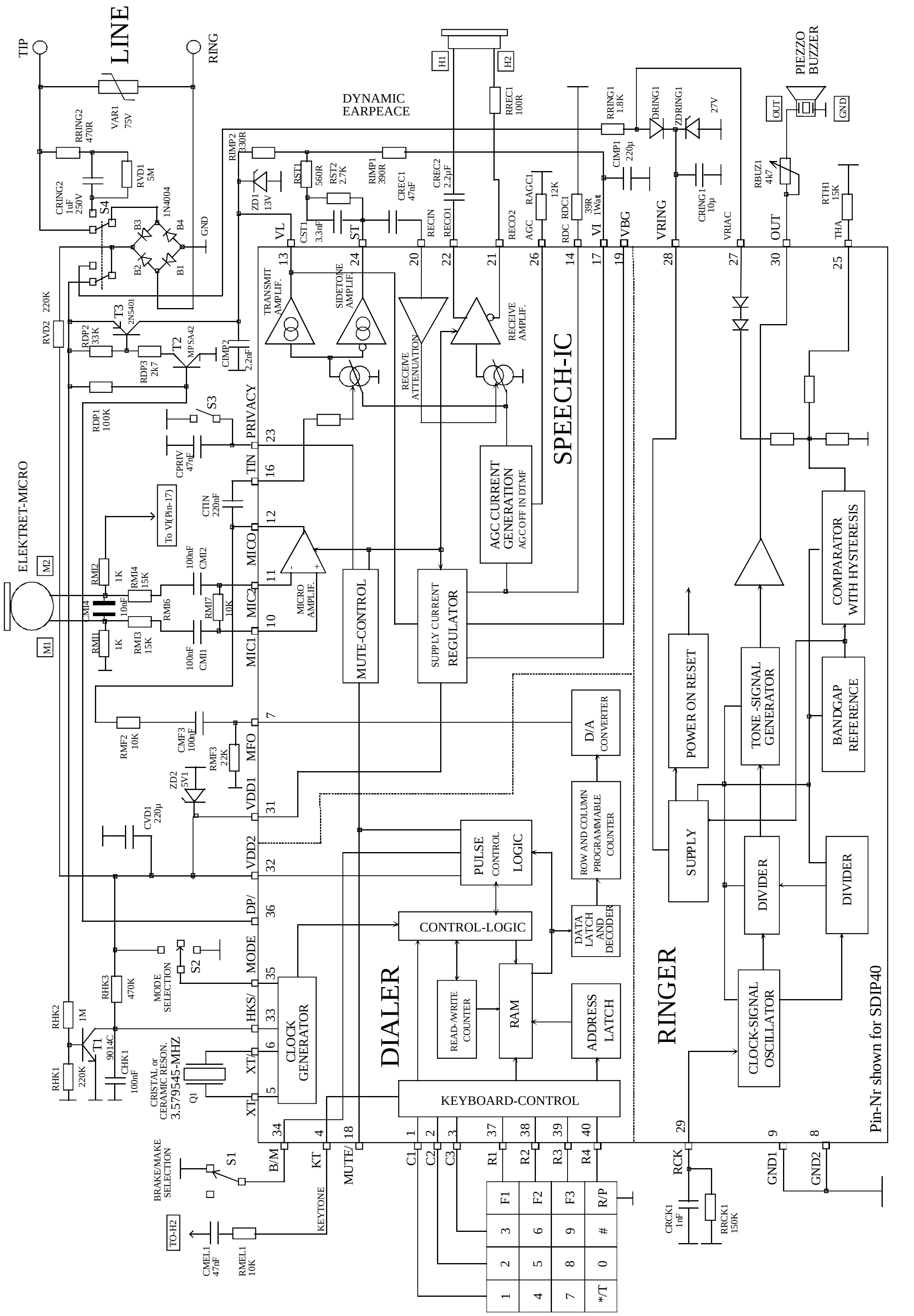


Figure 22. Demo board

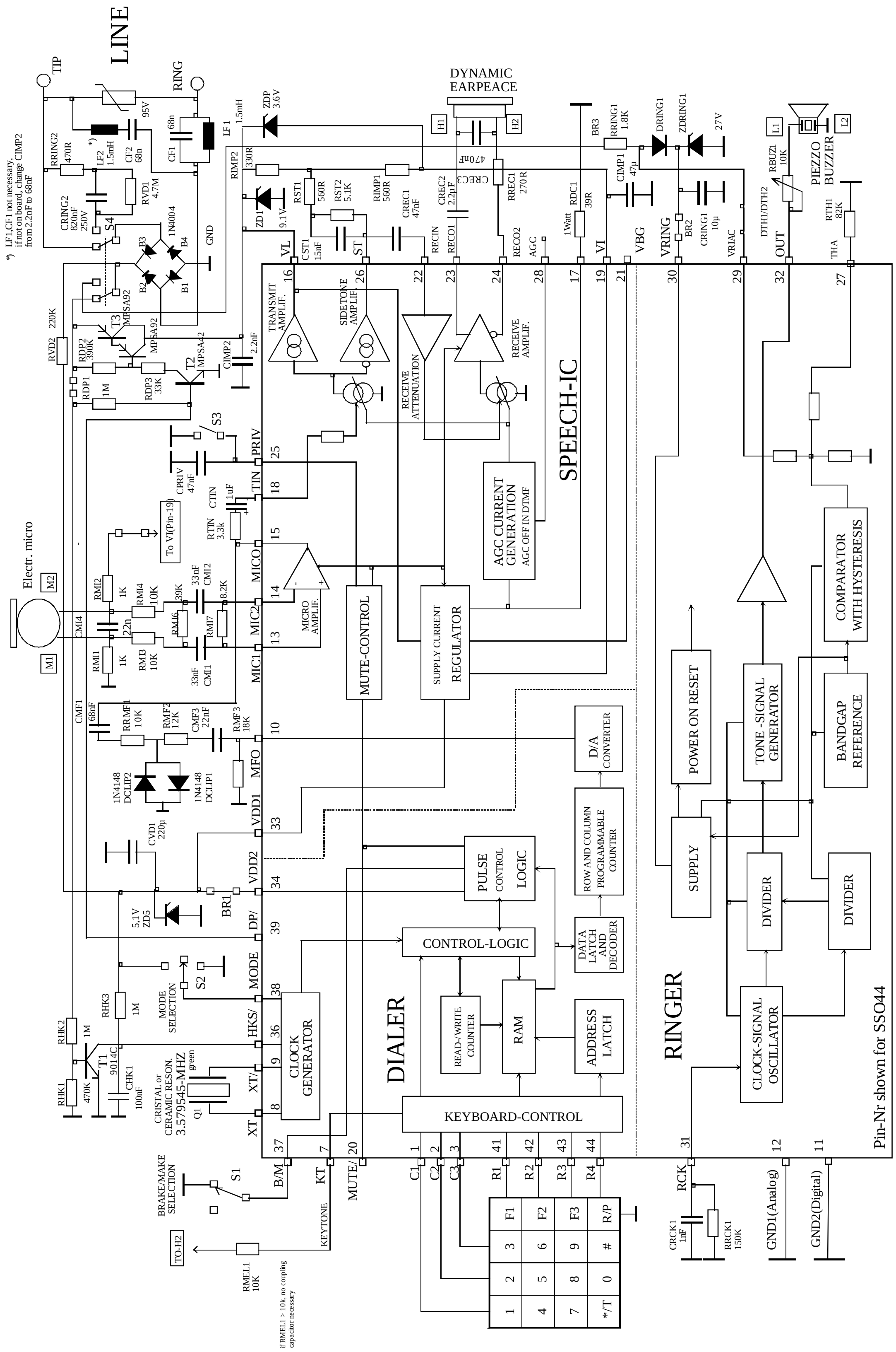


Figure 23. Typical application for Germany