



# FOUNDATION™ Digital Single or Dual Channel DSP Compression System

GA3216 PRELIMINARY DATA SHEET

GA3216

## FEATURES

- efficient, high fidelity 1 or 2-channel WDRC signal processing
- fully programmable via serial data interface
- SOUNDDESIGN™ high-fidelity audio quality
- four trimmer inputs plus volume control
- flexible trimmer/parameter assignments
- optional two-terminal or three-terminal trimmers
- choice of wideband or independent 2-channel level detection
- choice of two strategies for AGC-I parametric adjustment
- 6, 12 or 24 dB/octave band split filter or configurable as single-channel compressor.
- in-channel, low level squelch control (1:2 expansion)
- output compression limiting (AGC-O)
- flexible pre- and post-emphasis filters
- four independent memories
- pulse-density-modulated output stage drives zero-bias 2-terminal receivers

## thinSTAX™ PACKAGING

Hybrid typical dimensions:

0.190 x 0.123 x 0.060in  
(4.82 x 3.12 x 1.52mm)

## DESCRIPTION

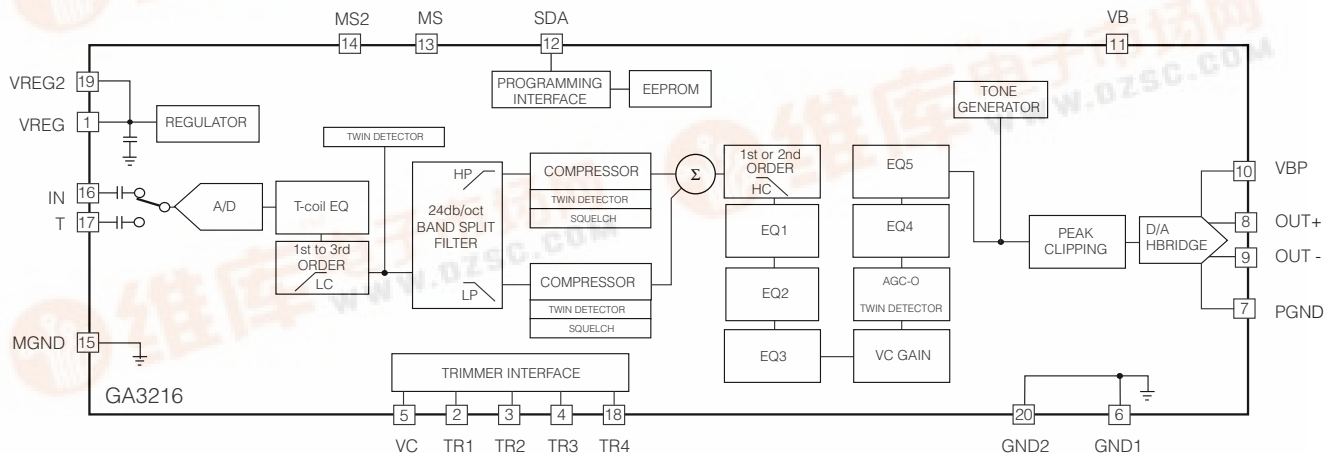
The GA3216 hybrid is a trimmer-configurable DSP system based on a two-channel compression circuit. It can efficiently replace traditional hearing-aid compression circuits without compromising fundamental performance requirements.

A trimmer interface supports manual circuit configuration. It continuously monitors trimmer positions and translates them into the hearing-aid parameters of choice. A serial data interface provides full programmability both at the factory and in the field.

The GA3216 includes in-channel squelch to attenuate microphone and circuit noise in quiet environments. It also includes low-distortion compression limiting and programmable high and low cut filters as well as five configurable equalization filters. Unused blocks can be powered down to save battery current, for example, when using the device in single-channel mode.

The GA3216 Hybrid code programmed into the GC5020 controller chip is '1'.

This datasheet is part of a set of documents available for this product. Please refer to [Getting Started with Foundation Digital](#), document #25786 for a list of other documents.



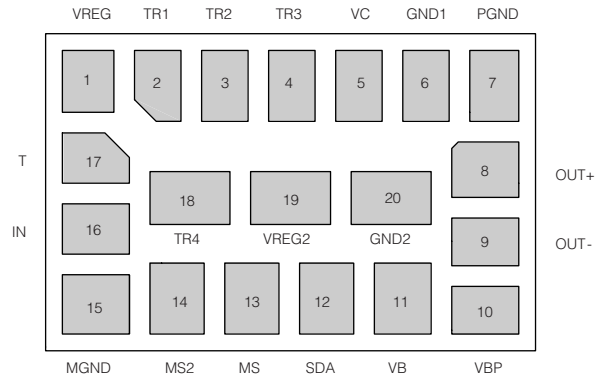
BLOCK DIAGRAM



## ABSOLUTE MAXIMUM RATING

Operating Temperature Range	-10°C to 40°C
Storage Temperature Range	-20°C to 70°C
Absolute Maximum Power Dissipation	25mW
Input ESD Voltage	2000V
Maximum Operating Supply Voltage	1.5VDC
Absolute Maximum Supply Voltage	2VDC

## PAD CONNECTION



**CAUTION**  
ELECTROSTATIC  
SENSITIVE DEVICES  
DO NOT OPEN PACKAGES OR HANDLE  
EXCEPT AT A STATIC-FREE WORKSTATION



**CAUTION**  
LEVEL 3 MOISTURE  
SENSITIVE DEVICES  
DO NOT OPEN PACKAGES EXCEPT UNDER  
CONTROLLED CONDITIONS



## ELECTRICAL CHARACTERISTICS

Conditions: Supply Voltage  $V_B = 1.3V$ ; Temperature = 25°C, 16 kHz bandwidth

PARAMETER	SYMBOL	CONDITIONS	MIN	TYP	MAX	UNITS
Hybrid Current	$I_{AMP}$	All functions, 24kHz sampling rate	-	540	-	$\mu A$
	$I_{AMP}$	All functions, 32kHz sampling rate	-	650	-	$\mu A$
Minimum Operating Supply Voltage	$V_{BOFF}$	Ramp down	0.95	-	0.980	V
Supply Voltage Turn on Threshold	$V_{BON}$	Ramp up	1.065	1.10	1.160	V
Supply Voltage Hysteresis		$V_{BON} - V_{BOFF}$	-	150	-	mV
Low Battery Warning Voltage		Ramp down	1.06	1.10	1.14	V
Supply Voltage During Communication	$V_{BC}$	During Communication	1.20	1.30	-	V
EEPROM Burn Cycles		Note 2	100k	-	-	cycles
Low Frequency System Bandwidth			-	130	-	Hz
High Frequency System Bandwidth		32 kHz sampling frequency	-	16	-	kHz
		24 kHz sampling frequency	-	12	-	kHz
Total Maximum System Gain	$A_V$	$V_{IN} = -95$ dBV @ 3kHz; squelch disabled See Note 1.	82	83	84	dB
Converter Gain	$A_{CONV}$	A/D + D/A gain.	28	29	30	dB
Total Harmonic Distortion	THD	$V_{IN} = -40$ dBV	-	-	1	%
THD at Maximum Input	THD <sub>M</sub>	$V_{IN} = -14$ dBV, HRX - ON	-	-	3	%
Clock Frequency	$f_{clk}$		1.963	2.048	2.115	MHz
<b>REGULATOR</b>						
Regulator Voltage	$V_{REG}$		0.87	0.90	0.93	V
Regulator Supply Rejection	PSRR <sub>REG</sub>		-	36	-	dB
<b>VOLUME CONTROL AND TRIMMERS</b>						
Volume Control or Trimmer Resistance	$R_{VC}$	Two-Terminal Trimmer. See Note 3.	-	200	-	k $\Omega$
Volume Control or Trimmer Resistance	$R_{VC}$	Three-Terminal Trimmer. See Note 3.	0.1	-	1	M $\Omega$
Volume Control Range	$\Delta A$		-	48	-	dB

**ELECTRICAL CHARACTERISTICS** (Continued)Conditions: Supply Voltage  $V_B = 1.3V$ ; Temperature = 25°C, 16 kHz bandwidth

PARAMETER	SYMBOL	CONDITIONS	MIN	TYP	MAX	UNITS
<b>MS AND MS2 INPUT</b>						
Pull Down/Up Resistance			-	1	-	MΩ
Logic 1 Voltage			-	$V_{REG}$	-	V
Logic 0 Voltage			-	0	-	V
<b>INPUT</b>						
Input Referred Noise	IRN	Bandwidth 100Hz - 8kHz	-	-	-108	dBV
Input Impedance	$Z_{IN}$		13.5	16	18.5	kΩ
Crosstalk		Between microphone and telecoil inputs	-	-60	-	dB
Maximum Input Level			-	-14	-	dBV
Input Dynamic Range		HRX - ON, Bandwidth 100Hz - 8kHz	-	95	-	dB
A/D Dynamic Range		Bandwidth 100Hz - 8kHz	-	86	-	dB
<b>OUTPUT</b>						
Maximum RMS Output Voltage		0dBFS $f = 1kHz$	-	-1	-	dBV
D/A Dynamic Range		Bandwidth 100Hz - 8kHz	-	83	-	dB
Output Impedance	$Z_{OUT}$		-	10	20	Ω
<b>SDA INPUT</b>						
Logic 0 Voltage		Note 2	0	-	0.3	V
Logic 1 Voltage		Note 2	1	-	1.3	V
<b>SDA OUTPUT</b>						
Synchronization Time (Synchronization Pulse Width)	$T_{SYNC}$	Baud = 0 Baud = 1 Baud = 2	237 118 59	250 125 62.5	263 132 66	μs μs μs

NOTE 1: Total System Gain consists of: *Wideband System Gain + High and Low Independent Channel Gains + Converter Gain*

Total System Gain is calibrated during Cal/Config process.

NOTE 2: Sample tested.

NOTE 3: Volume control is log taper, trimmers are linear taper.

**ELECTRICAL CHARACTERISTICS** (Continued)

PARAMETER	MIN	MAX	UNIT	ACCURACY
<b>TELECOIL</b>				
Telecoil Gain	-8	23	dB	type 3
Low Pass compensation Filter	0.5	1	kHz	type 1
<b>FREQUENCY SHAPING</b>				
Crossover Frequency	0.5	4.25	kHz	type 1, 2
High Cut Filter	1	16	kHz	type 1, 2
High Cut Filter Order	6 or 12		dB/Octave	N/A
Low Cut Filter	0.01	3	kHz	type 1, 2
Low Cut Filter Order	6, 12 or 18		dB/Octave	
Equalization Filter Center	0.125	16	kHz	type 1, 2
Equalization Filter Depth	-30	30	dB	
Equalization Filter Q	0.7079	70.7946		

**ELECTRICAL CHARACTERISTICS** (Continued)

INDEPENDENT CHANNEL PROCESSING				
Bandsplit Filter Slopes	6, 12 or 24		dB/Octave	type 1, 2
Low Level Gain	-18	42	dB	type 3
High Level Gain	-18	42	dB	type 3
Lower Threshold	30	110	dBSPPL	type 3
Upper Threshold	70	110	dBSPPL	type 3
Compression Ratio	1:1	$\infty$ :1	Ratio	type 3
AGCi Attack Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
AGCi Release Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
Squelch Expansion Ratio	1:2		ratio	N/A
Squelch Threshold	20	60	dBSPPL	type 3
Squelch Attack Time Constant	0.25	8192	ms	type 1, 3
Squelch Release Time Constant	0.25	8192	ms	type 1, 3
WIDEBAND SYSTEM GAIN				
Wideband System Gain	-36	12	dB	type 3
Wideband Attack Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
Wideband Release Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
External VC	-48	0	dB	type 3
Internal VC Attenuator	-48	0	dB	type 3
TOTAL SYSTEM GAIN				
Total System Gain	-19	83	dB	Note 1
AGCo				
AGCo Output Limiting	-30	-1	dBFS*	type 3
AGCo Compression Ratio	$\infty$ :1		Ratio	N/A
AGCo Attack Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
AGCo Release Time Constant (Fast & Slow)	0.25	8192	ms	type 1, 3
PEAK CLIPPER				
PC Output Limiting	-40	0	dBFS	type 3
TONE GENERATOR				
Pure Tone Frequency (memory and low battery indicator)	0.25	16	kHz	type 1, 2
Pure Tone Amplitude (memory and low battery indicator)	-50	0	dBFS	type 3

\* peak output is defined as largest sine wave possible at the resonant frequency of the receiver

NOTE 1: Total System Gain consists of *Wideband System Gain + High and Low Independent Channel Gains + Converter Gain* and accuracy of this parameter is dependent on accuracy of the components.

Accuracy definitions:

type 1: accuracy is determined by the clock frequency deviation

type 2: accuracy is determined by the quantization error of 16bit coefficient and 20bit or higher data word.

type 3: accuracy is determined by the quantization error of a parameter word (see table 2 for word length) and 20bit or higher data word.

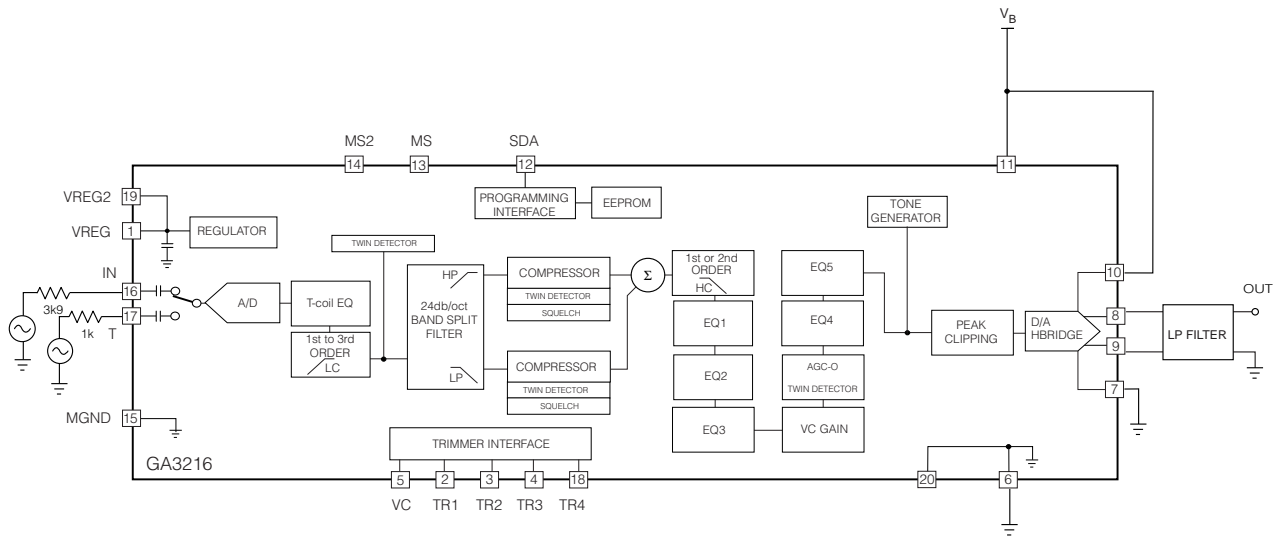


Fig. 1 Test Circuit

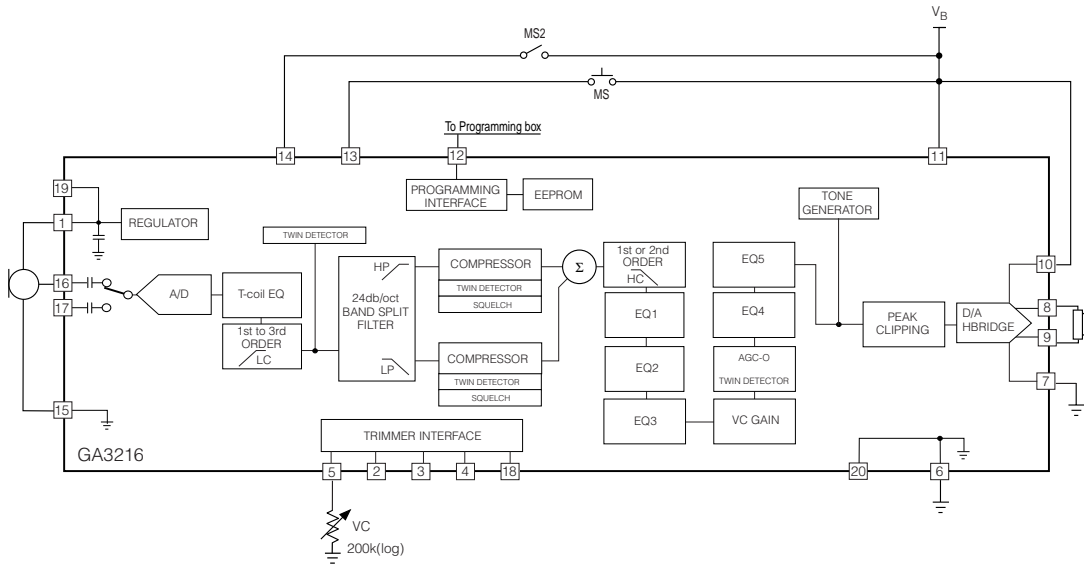


Fig. 2 Example of Programmable Application Circuit

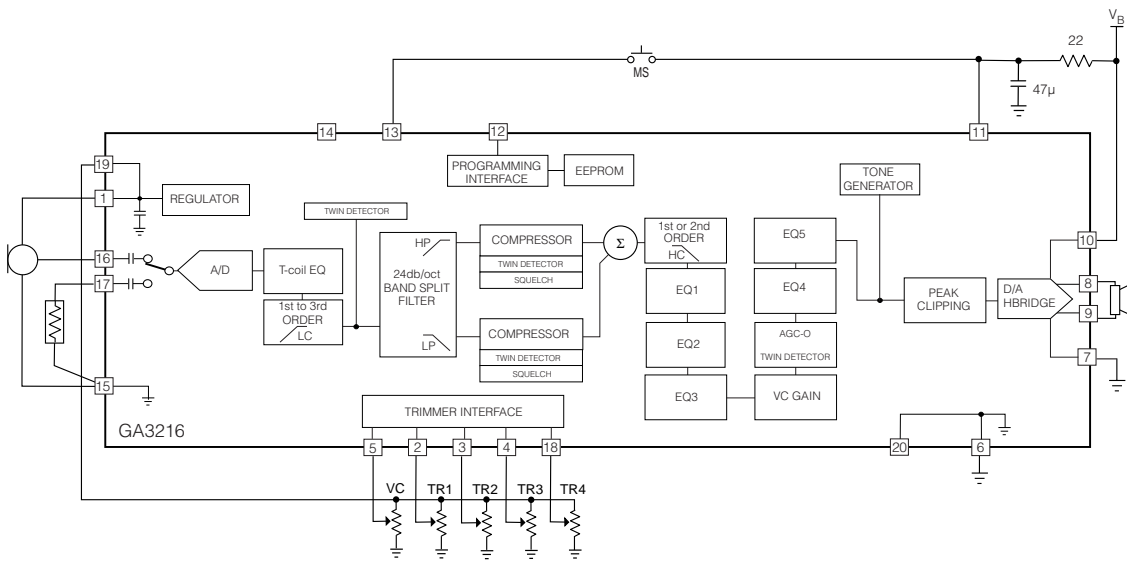


Fig. 3 Example of Trimmer Application Circuit

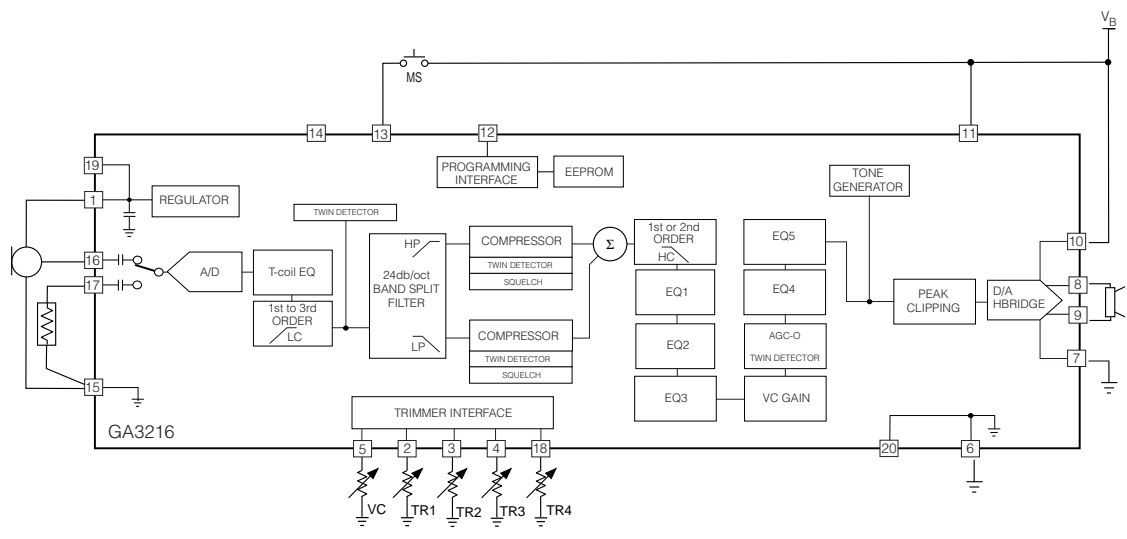


Fig. 4 Application Circuit for Figure 5

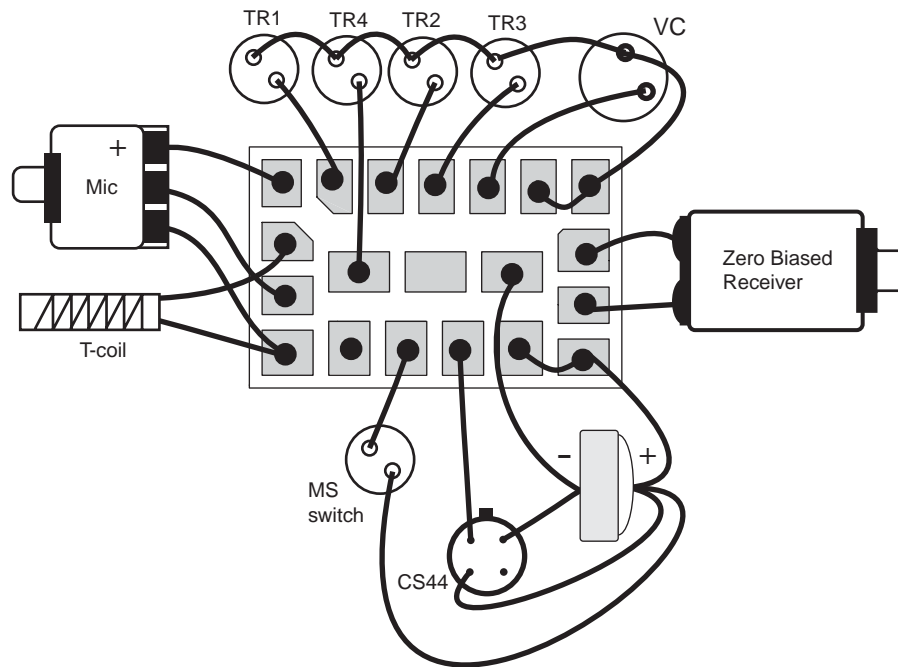


Fig. 5 Example of Assembly Diagram For Two-terminal Trimmer Circuit

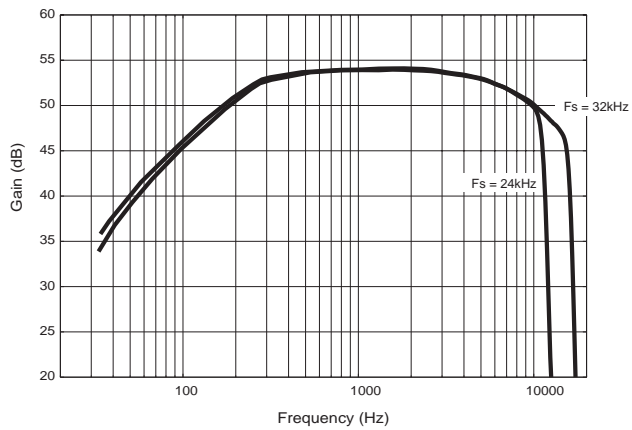


Fig. 6 Frequency Response at 32kHz and 24kHz Sampling Frequency

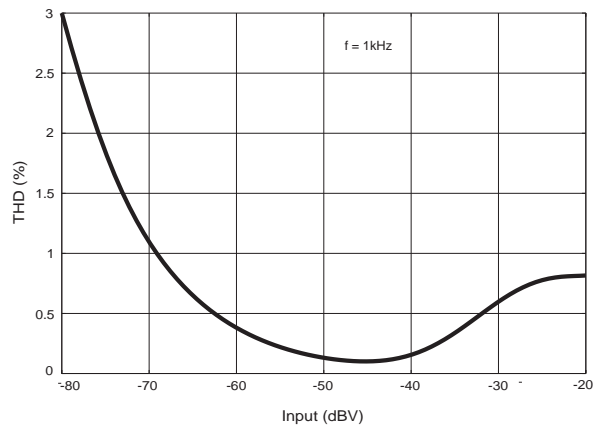


Fig. 9 THD vs Input Level

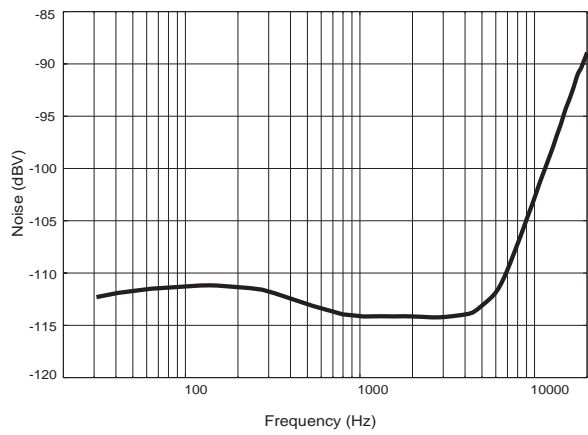


Fig. 7 Output Noise

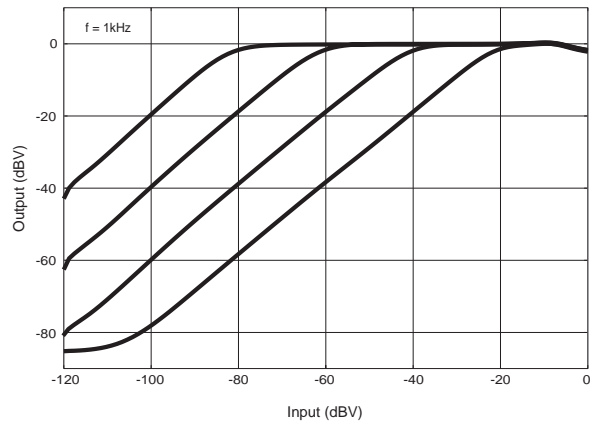


Fig. 10 Input/Output Curves at Various Gains

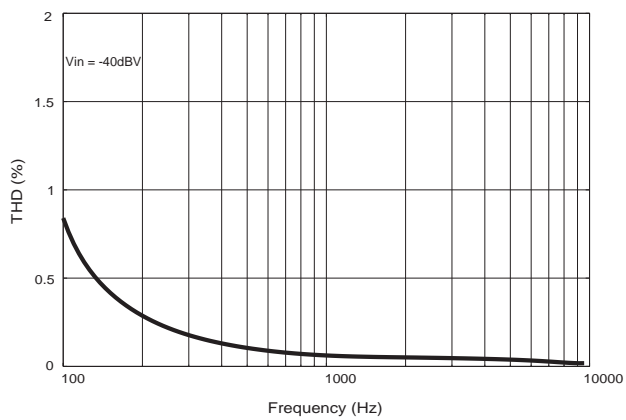


Fig. 8 THD vs Frequency



## INTRODUCTION

The GA3216 hybrid comprises a configurable two-channel compressor circuit with two methods of operation, programmable mode and trimmer mode. It may be configured as a one or two channel device with linear or WDRC processing. Configuration data stored in non-volatile memory defines hearing-aid parameters. This data needs to be uploaded to the hybrid before the circuit becomes functional. The GA3216 hybrid is programmed via the SDA pin using industry-standard programming boxes.

Configuration data is generated by an ARK product component library (DLL). Like the PARAGON™ Digital products, the GA3216 is fully supported by Gennum's software tools available from the Gennum ARK web site <http://ark.gennum.com>.

During normal trimmer mode operation, a low-speed A/D circuit monitors the positions of up to four manual trimmers and a VC potentiometer. Trimmer position changes are immediately interpreted and translated into new circuit parameter values, which are then used to update the signal path.

Gennum's Library Manager tool allows one to predefine trimmer assignments for all common functions such as low-cut, high-cut, notch and resonant-peak-shift filtering, AGC parameters, wideband gain, and maximum power output (MPO).

## SIGNAL PATH

There are two main audio input signal paths. The first path contains the Microphone and second path contains the Telecoil input as selected by a programmable MUX. The microphone input is intended as the main audio input for single-microphone applications. The two audio inputs are buffered, sampled and converted into digital form using an A/D converter. The digital output is then converted into a 32kHz 20-bit digital audio signal. It is possible to save current by reducing the sampling rate to 24kHz, which will also reduce the systems bandwidth from 16kHz to 12kHz.

In Telecoil mode, gains are trimmed during the Cal/Config process to compensate for microphone/Telecoil mismatches. The Telecoil input may also be used as a second microphone input. In this case the Telecoil compensation would be disabled. This can be done via the ARKonline wizard.

The wideband detector circuit output is routed to a band-split filter that divides the signal into two frequency bands. There is an option to configure the hybrid as a single channel device that simply shuts down and bypasses the band-split filter. The signal in each frequency band is processed by an independent AGC loop. The gain in any band is a function of the energy detected in that band or the overall detected wide-band energy. The two frequency bands are then summed back into a wide-band signal.

Further processing capabilities include a 1st to 3rd order low cut filter before the band-split filter. After the two channels are summed together there is a 1st or 2nd order high cut filter, three EQ filters followed by volume control, low-distortion AGCo, and two post-AGCo EQ filters.

All AGC loops use a feed-forward topology to ensure system stability. Detectors in the AGCi path and AGCo path are twin fast/slow detectors with independently adjustable attack and release time constants. The squelch detector (which resides in the AGCi path) is implemented as a single fast/slow detector.

## FUNCTIONAL BLOCK DESCRIPTIONS

### A/D AND D/A CONVERTER

The system's A/D converter is a 2nd-order sigma-delta modulator, which operates at a 2.048MHz sample rate. The system's input is pre-conditioned with anti-alias filtering and a programmable gain pre-amplifier. The analog output is oversampled and modulated to produce a 1-bit pulse density modulated (PDM) data stream. The digital PDM data is then decimated down to pulse-code modulated (PCM) digital words at the system's sampling rate of 32kHz.

The D/A is comprised of a digital 3rd-order sigma-delta modulator and an H-bridge. The modulator accepts PCM audio data from the DSP path and converts it into a 32-times oversampled, 1-bit PDM data stream, which is then supplied to the H-bridge. The H-bridge is a specialized CMOS output driver used to convert the 1-bit data stream into a low-impedance, differential output voltage waveform suitable for driving zero-biased hearing aid receivers.

### CHANNEL PROCESSING

Figure 1 represents the I/O characteristic of independent AGC channel processing. The I/O curve can be divided into four main regions:

- low input level expansion (squelch) region
- low input level linear region
- compression region
- high input level linear region (return to linear)

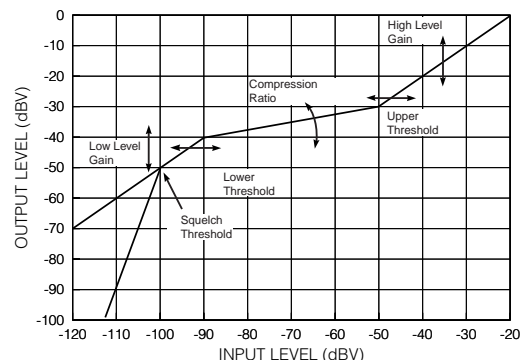


Fig. 11 Independent Channel I/O Curve Flexibility

The I/O characteristic of the channel processing can be adjusted in six ways:

- squelch threshold (SQUELCHTH)
- low level gain (LLGAIN)
- lower threshold (LTH)
- high level gain (HLGAIN)
- upper threshold (UTH)
- compression ratio (CR)

During Parameter Map creation, constraints are applied to the compression parameters to ensure that the I/O characteristics are continuous. In both Programmable mode and Trimmer mode predefined parameter adjustments support two popular styles of compression ratio adjustment:

- The compression region of the I/O curve pivots about the upper threshold. As the compression ratio trimmer is adjusted, high-level gain remains constant while the low-level gain changes.
- The compression region of the I/O curve pivots about the lower threshold. Low-level gain remains constant as the compression ratio trimmer is adjusted.

The two compression channels can be controlled in tandem using a common wideband level detector, or independently using dedicated in-channel level detectors. Parameters such as low level gain, lower threshold and compression ratio can be combined and controlled by a single trimmer.

The squelch region within each channel implements a low level noise reduction scheme (1:2 expansion) for listener comfort. This scheme operates in quiet listening environments (programmable threshold) to reduce the gain at very low levels.

Single-channel compressor operation is supported by disabling the band split filter and one of the channel compressors. The remaining compressor can be configured as a wide-dynamic-range compressor (WDRC), or as an input compression limiter (ICL).

#### **HRX HEAD ROOM EXPANDER**

The GA3216 has an enhanced Head Room Expander (HRX) circuit, which increases the input dynamic range of the FOUNDATION Digital without any unwanted audible artifacts. This is accomplished by dynamically adjusting the preamplifier's gain and the post-A/D attenuation depending on the input level.

#### **TELECOIL PATH**

The telecoil input is calibrated during the Cal/Config process. To compensate for the telecoil/microphone frequency response mismatch, a first order filter with 500Hz corner frequency is implemented. Through ARKonline it is possible to implement a telecoil compensation filter with an adjustable corner frequency. To accommodate for the gain mismatch, the telecoil gain is adjusted to match the microphone gain at 500Hz or 1kHz (default) and is selectable in ARKonline. There is also a telecoil gain adjustment parameter, which can be enabled in ARKonline and set in IDS that will allow for manual adjustment of the telecoil gain compensation.

#### **VOLUME CONTROL & TRIMMERS**

All parameters can be controlled via the SDA or assigned to the trimmers. The four trimmers have flexible parameter assignments so any of the four trimmers can be assigned to any available parameters (A complete list of parameters is available on ARKonline).

Both the external VC and trimmers can be configured to work with either a two-terminal 200k $\Omega$  variable resistor or a three-terminal 0.1M $\Omega$  – 1M $\Omega$  variable resistor. The volume control should have a log taper, while the trimmers should have a linear taper.

In two-terminal configuration the trimmers and VC are connected between GND and the trimmer Input and in three-terminal configuration between GND, Vreg and the trimmer Input. To enable the device to use two terminal trimmers, in IDS under Settings in the Cal/Config menu, click the trimmers tab and select two terminal trimmers. If using two terminal trimmers, they must be calibrated before use. Calibration is not necessary with three terminal trimmer pots. Hysteresis is built into the circuitry to prevent unintentional level toggling.

#### **EQUALIZATION FILTERS**

There are five equalization filters provided on the FOUNDATION Digital for additional frequency shaping. Each EQ filter has three adjustable parameters, centre, depth and Q. One of these parameters can be selected as trimmer adjustable for each filter. For added flexibility it is possible to combine EQ2 and EQ3 or EQ4 and EQ5 together and have them adjusted by one trimmer.

#### **AGCo**

The AGCo module is an output limiting circuit with a fixed compression ratio infinity:1. The limiting level is programmable as a level measured as dB from full scale. 0 dBFS is the maximum output of the device. The AGCo module has its own twin level detector, with programmable attack and release time constants.

**MS AND MS2 SWITCHES**

There are two, two-pole Memory Select switches available on the GA3216, which allows the user tremendous flexibility in switching between configurations. These switches may be either momentary or static as set up in ARKonline and are configurable to be either pull-up or pull-down through the setting tab in IDS. In programmable mode the device must be set to pull-down.

Up to four memories can be configured in programmable mode, or one memory in trimmer mode. Enabled (valid) memories must be sequential. For example, if three memories were required, memories A, B and C would be enabled. Memory A must always be valid. All memory select options are selectable via the ARKonline wizard.

In simple trimmer mode there is one method for memory select. In this mode, when the telecoil is enabled, MS (Pin 13) switches the GA3216's input between the microphone and the telecoil using either a momentary or static switch which is selectable using ARKonline.

There are four memory select modes available in programmable and multi-memory trimmer mode:

**Momentary Switch on MS**

This mode uses a single momentary switch on MS (Pin 13) to change memories. Using this mode will cause the part to start in Memory A and whenever the button is pressed the next valid memory will be loaded. When the user is in the last valid memory, a button press will cause memory A to be loaded.

Examples:

- If 4 valid memories ABCDABCD...A...
- If 3 valid memories ABCABCA...A...
- If 2 valid memories ABABA...A...
- If 1 valid memories AAA...A...

**Momentary Switch on MS, Static Switch on MS2 (jump to last memory)**

This mode uses a static switch on MS2 (Pin 14) and a momentary switch on MS (Pin 13) to change memories. If the static switch is OPEN, the part will start in memory A and it will behave like momentary with the exception that the last valid memory will not be used. If the static switch on MS2 is set to HIGH, the part will automatically jump to the last valid memory (this will happen on startup or during normal operation). In this setup, the momentary switch's state is ignored. This prevents memory select beeps from occurring. When MS2 is set to OPEN, the part will load in the last select memory.

Examples:

- If MS2 = OPEN and there are 4 valid memories: ABCABCA...
- If MS2 = OPEN and there are 3 valid memories: ABABA...
- If MS2 = HIGH and there are 4 valid memories: D...
- If MS2 = HIGH and there are 3 valid memories: C...

**Table 1: Dynamic example with 4 valid memories.**  
T = momentary switch is toggled, 0 = OPEN, 1 = HIGH

MS2	0	0	0	1	1	1	0	0	0	1	0	0	0	0	0	0
MS	0	T	T	0	T	T	0	T	T	0	0	T	T	T	T	T
Memory	A	B	C	D	D	D	C	A	B	D	B	C	A	B	C	A

**Static Switch on MS and MS2**

This mode uses two static switches to change memories.

The following table describes which memory is selected depending on the state of the switches.

In this mode it is possible to jump from any memory to any other memory simply by changing the state of both switches. If both switches are changed simultaneously then the transition will be smooth, otherwise, if one switch is changed and then the other, the part will transition to an intermediate memory before reaching the final memory.

The part will start in whatever memory the switches are selecting. If a memory is invalid the part will default to memory A.

MS	MS2	Memory
OPEN	OPEN	A
HIGH	OPEN	B (if valid otherwise A)
OPEN	HIGH	C (if valid otherwise A)
HIGH	HIGH	D (if valid otherwise A)

**Static Switch on MS, Static Switch on MS2 (jump to last memory)**

This mode uses two static switches to change memories.

Unlike in the previous example, this mode will switch to the last valid memory when the static switch on MS2 is HIGH. This means that this mode will only use a maximum of three memories (even if four valid memories are programmed). The following table describes which memory is selected depending on the state of the switches.

MS	MS2	Memory
OPEN	OPEN	A
HIGH	OPEN	B (if valid otherwise A)
OPEN	HIGH	Last Valid Memory
HIGH	HIGH	Last Valid Memory

In this mode it is possible to jump from any memory to any other memory simply by changing the state of both switches. If both switches are changed simultaneously then the transition will be smooth, otherwise, if one switch is changed and then the other, the part will transition to an intermediate memory before reaching the final memory.

When MS2 is set HIGH, the state of the switch on MS is ignored. This prevents memory select beeps from occurring when switching MS when MS2 is HIGH.

The part will start in whatever memory the switches are selecting. If the device starts up in a memory other than A, and the memory beep tones are enabled, the device will emit the corresponding tones for that memory. If a memory is invalid, the part will default to memory A.

#### AUDIBLE MEMORY CHANGE INDICATOR

The FOUNDATION Digital can be configured to produce tones to indicate a memory change.

Through ARKonline the GA3216 can be configured to either enable or disable the Memory Change Indicator. When the Memory Change Indicator is enabled, there is an option to have a single beep for each memory change or multiple-beeps.

The amplitude and frequency of the memory change tone can be selected independent of the Tone Generator settings and can be individually selected for each memory.

When the memory change multiple beep is enabled and the memory change tone is enabled, then during a memory change operation the selected tone will beep a code to indicate which memory has been selected. The beep sequence will be 150ms ON followed by a 150ms OFF time between the beeps. The memory change beeping code is deciphered in the table below.

SELECTED MEMORY	# OF BEEPS
A	1
B	2
C	3
D	4

#### STONE GENERATOR

The tone generator is capable of producing programmable tones. When the Tone Generator is enabled, the FOUNDATION Digital connects the output of the tone generator to the input of the D/A converter. The programmed tone is the output until the Tone Generator is disabled. When disabled, the normal audio signal path is again connected. The tone generator will produce a beep to indicate a memory select change, a microphone/telecoil selection change or to warn the user of low battery voltage.

NOTE: When the sampling frequency is set at 24kHz, the tones will scale. e.g. A 1kHz tone at 32kHz sampling would be approximately 667Hz at 24kHz sampling.

#### LOW BATTERY INDICATOR (POWER-ON/POWER-OFF)

During power-on the FOUNDATION Digital is held in a reset state until  $V_b$  reaches a turn-on threshold, and the hybrid's internal control logic determines that the supply is stable. When this occurs the hybrid begins its power-on sequence.

When a low battery condition (below turn on threshold) is detected, the FOUNDATION Digital sends out a series of 3 beeps (225ms ON, 225ms OFF, 225ms ON, 225ms OFF, 225ms ON) to indicate the battery is low. This will repeat every 5 minutes until the device reaches the turn-off threshold.

If  $V_b$  drops below the turn-off threshold then the FOUNDATION Digital is returned to its reset state and the audio output is muted.

There are four reset modes which are selectable through ARKonline.

The first is the shallow-reset mode which, during the power-ON sequence circuit starts when the supply voltage rises above the turn-ON threshold after shutdown. The device will function until the supply voltage drops below the turn-OFF threshold but will recover when the device rises above the turn-ON threshold again.

The second is deep-reset mode which begins when the supply voltage rises above the turn-ON threshold after shutdown. Once the supply voltage drops below the turn-OFF threshold the GA3216 will shut down and remain there until the supply voltage drops below approximately 0.3V and subsequently rises above the turn-ON threshold after shutdown.

The third is a combination of these two modes, where the device starts up in shallow-reset mode initially, then changes over to deep reset mode after 5 minutes.

The fourth option is to have the gain reduced as the battery voltage drops. When the supply falls below the low battery threshold the low battery tones will be emitted and the wideband gain will be reduced by 3dB. As the battery voltage continues to drop, the low battery tones will continue and the wideband gain will continue to be reduced until the turn-OFF threshold is reached when the device will shut down.

#### **CURRENT CONSUMPTION**

Typical current consumption of the FOUNDATION Digital, as stated in the Electrical Characteristics section, is measured at a specific configuration and settings. If lower current is desired, it can be achieved by selecting a 24kHz sampling rate from the settings tab in IDS. This will reduce the systems bandwidth from 16kHz to 12kHz.

#### **SOFTWARE**

Full software support is provided for every stage of development from design to manufacturing to fitting. Please refer to the "[Getting Started with ARK Guide](#)," document # 27217.

#### **SDA COMMUNICATION**

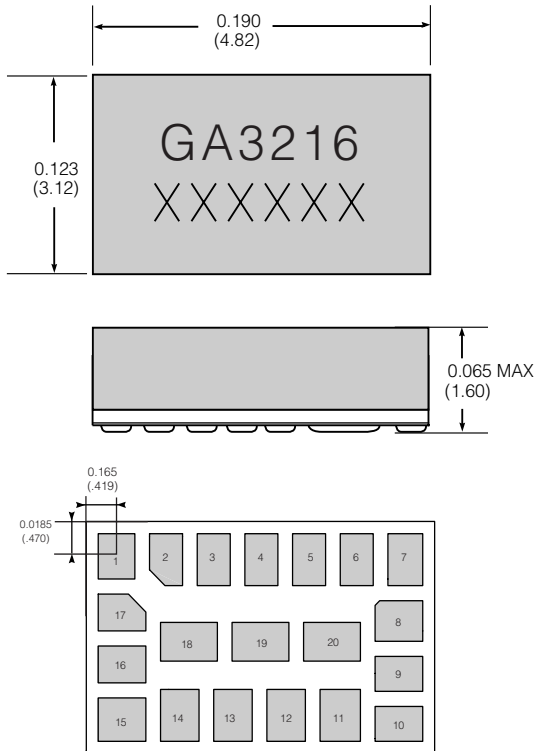
The FOUNDATION Digital is programmed via the SDA pin using industry standard programming boxes. During parameter changes the main audio signal path of the hybrid is temporarily disable and replaced with a low gain "bypass path" to avoid the generation of disturbing audio transients. Once the changes are complete, the main audio path is re-activated.

Any changes made during programming will be lost at power-off unless they are explicitly burned to EEPROM memory.

#### **POWER MANAGEMENT**

The FOUNDATION Digital was designed to accommodate high power applications. AC ripple on the supply can cause instantaneous reduction of the battery's voltage, potentially disruption the circuit's function. The GA3216 has a separate power supply and ground connection for the output stage. This allows hearing instrument designers to accommodate external RC filters in order to minimize any AC ripple from the supply line. Reducing this AC ripple greatly improves the stability of the circuit and prevents unwanted reset of the circuit caused by spikes on the supply line. For more information on properly designing a filter to reduce supply ripple, please refer to information note "[Using the GB3211 PARAGON Digital in High Power Application Initial Design Tips](#)" document #24561.

## PACKAGE DIMENSIONS



Dimension units are in inches.

Dimensions in parentheses are in millimetres, converted from inches and include minor rounding errors.

1.0000 inches = 25.400mm

Dimension tolerances:  $\pm 0.005$  ( $\pm 0.13$ ) unless otherwise stated.

Work order number: XXXXXX

This Hybrid is designed for either point-to-point manual soldering or for reflow according to Gennum's reflow process (Information Note 521-45).

## PAD LOCATIONS

PAD NO.	PAD POSITION		PAD DIMENSIONS		
	X	Y	Xdim	Ydim	
1	0	0	20	24	MIL
2	27	-1.75	18	27.5	
3	53	-1.75	18	27.5	
4	79	-1.75	18	27.5	
5	105	-1.75	18	27.5	
6	131	-1.75	18	27.5	
7	157.5	-1.75	19	27.5	
8	154	-34.25	26	21.5	
9	154	-62.25	26	18.5	
10	154	-88.75	26	18.5	
11	122	-84.25	22	27.5	
12	92.5	-84.25	21	27.5	
13	63.5	-84.25	21	27.5	
14	34.5	-84.25	21	27.5	
15	3	-86.5	26	23	
16	3	-57.25	26	19.5	
17	3	-29.75	26	19.5	
18	39.5	-45.25	31	20.5	
19	78.5	-45.25	31	20.5	
20	117.5	-45.25	31	20.5	
1	0	0	0.508	0.610	mm
2	0.686	-0.044	0.457	0.699	
3	1.346	-0.044	0.457	0.699	
4	2.007	-0.044	0.457	0.699	
5	2.667	-0.044	0.457	0.699	
6	3.327	-0.044	0.457	0.699	
7	4.001	-0.044	0.483	0.699	
8	3.912	-0.870	0.660	0.546	
9	3.912	-1.581	0.660	0.470	
10	3.912	-2.254	0.660	0.470	
11	3.099	-2.140	0.559	0.699	
12	2.350	-2.140	0.533	0.699	
13	1.613	-2.140	0.533	0.699	
14	0.876	-2.140	0.533	0.699	
15	0.076	-2.197	0.660	0.584	
16	0.076	-1.454	0.660	0.495	
17	0.076	-0.756	0.660	0.495	
18	1.003	-1.149	0.787	0.521	
19	1.994	-1.149	0.787	0.521	
20	2.985	-1.149	0.787	0.521	

GA3216

### DOCUMENT IDENTIFICATION

#### PRELIMINARY DATA SHEET

The product is in production. Gennum reserves the right to make changes at any time to improve reliability, function or design, in order to provide the best product possible.

### REVISION NOTES:

Updated for ARKbase 3.4.4 release.

GENNUM CORPORATION

MAILING ADDRESS:  
P.O. Box 489, Stn A, Burlington Ontario, Canada L7R 3Y3  
Tel. +1 (905) 632-2996 fax: +1 (905) 632-2814

SHIPPING ADDRESS:  
970 Fraser Drive, Burlington, Ontario, Canada L7L 5P5

GENNUM JAPAN CORPORATION

Shinjuku Green Tower Building 27F, 6-14-1, Nishi Shinjuku,  
Shinjuku-ku, Tokyo, 160-0023 Japan  
Tel. +81 (03) 3349-5501, Fax. +81 (03) 3349-5505

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