Layman’s Handbook of type F22A Data Sheet
[English]
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Foreword

This description of type F22A data sheet is meant to give basic understanding of the graphs shown in the technical data sheet without using deep engineering language. It is not intended to explain all of the technical detail in depth. To select or judge a loudspeaker’s performance, we agree that one should listen to the product directly. Generally speaking, audio is a field that is split. 50% of it is art (subjective) and 50% is science (objective). With this handbook, we would like to bring the reader closer to the 50% that is the scientific portion.

Frequency Response and General Description

Frequency is the number of cycles per unit interval time. For a sound wave, this denotes the number of waves passing a point in a certain time. It determines the pitch of a sound, such as how high or low a musical note is. The unit is cycles per second or widely known as Hz (Hertz). One Hz is equal to one wave per second.

Frequency response shows the relationship between the device’s input and output with regard to frequency and amplitude. Ideally, when a loudspeaker is given equal voltage at different frequencies, the output sound pressure level (SPL) should be the same at different frequencies. But of course it is not the case. When given equal voltage at different frequencies, a loudspeaker’s output will result in a frequency response graph that rises up at the very low frequencies, rolls down at the very high frequencies and results in a curve that has ups and downs.

Human hearing ranges from about 20 Hz to 20000 Hz. Well, that is for a newborn infant with a good normal hearing. By the time a person reaches 30 years old, the hearing range typically becomes narrower to about 30 Hz to 14000 Hz. In loudspeaker design, no single component can perfectly reproduce 30 Hz to 14000 Hz with near flat frequency response. To aim for a very wide output frequency range that is close to the human hearing range, a loudspeaker output is typically handled by different components, such as a woofer to handle 1000 Hz and below, and a tweeter for 1000 Hz and above. If a loudspeaker uses two components to handle two different frequency ranges, it is called a two way system. The point where each component overlaps is typically called the crossover frequency point.

Type F22A frequency response can be seen in figure 1 (black curve).

![On-Axis Frequency Response Preset #1](image)

**Figure 1**

As the reader can see, the lower frequency curve rises up to about 45-50 Hz before it becomes steadily even and straight. This denotes that the limitation of the low frequency reproduction of the loudspeaker is about 45-50 Hz. Usually, the nominal frequency response range is specified at the -3dB limit, which is 45 Hz to 21500 Hz for type F22A. Why -3 dB? A minus 3 dB is equal to half power.

A frequency response of a good loudspeaker is expected to be a flat line. Flat denotes a visual cue. In audio, it means a low amplitude variation at different frequencies within its nominal range. For accurate signal reproduction, a loudspeaker frequency response should have a frequency response curve variation of equal to or less than ±3 dB (or also called 6dB variation). For human hearing system, a 3 dB SPL change denotes a perceived just noticeable difference.
This variation also varies with the curve smoothing in a frequency response graph. Smoothing is usually indicated by 1 octave smoothing or 1/3 octave smoothing or 1/6 octave smoothing and so on. Figure 2 shows a zoomed in frequency response of F22A at 500 Hz – 12000 Hz. Please note that figure 2’s Y-axis is only indicating a 10 dB range, whereas figure 1’s Y-axis is indicating a 48 dB range.

![Figure 2](image)

Using a smaller smoothing value (1/6 octave is smaller than 1/1 octave, for example), one is able to see more detail up and down on the frequency response curve. Why do you need to smooth the curve? The measurement system has very fine frequency resolution: closer to a 1 Hz measurement resolution. However, our hearing does not work in that way. A 10 Hz change in the low frequency range is perceived as a much bigger change than in the high frequency range. One will be able to tell the difference between 50 Hz and 60 Hz very easily, but one will not be able to tell any differences between 5000 Hz and 5010 Hz. Also, in western music, one octave is divided into twelve tones which are closer to 1/12 octave smoothing. With the help of smoothing, a loudspeaker designer is able to narrow down the objective requirement for the loudspeaker’s processing requirement to achieve a desired goal.

**Phase Response**

In figure 1, the reader will notice the blue curve. This is the phase response of the loudspeaker. This curve can be thought as the relative time arrival between different frequencies. A less variation of the phase response curve denotes that the loudspeaker will preserve the shape of the input signal’s waveform as much as possible. The word ‘shape’ denotes another visual cue. Sound is not visual and it is known that the human hearing system is not sensitive to phase changes, especially for a single mono unit comparison.

In the development process, we focus on modern processing to find ways to ‘shape’ and manipulate sound reproduction. We completed many blind comparison tests of digital audio signal processing. For example we tested flat phase response down to 10 Hz versus no phase processing, the effects on stereo image (sound stage) changes and subjective phantom image creation, based on what is heard. The various blind tests (including reviewing many different components) were done over five years of intense research. We concluded that phase response can alter the sound stage, the stereo image and depth perception in stereo sound reproduction.
Side Techy Note: Why We Use 48 kHz Sample Rate

In music (equal temperament tuning), the middle C frequency is 261.6 Hz and an 88-key piano has a musical pitch ranging from 27.5 Hz (lowest A) to 4186 Hz (highest C). Have you ever noticed how middle C sounds much fuller than the highest C in a piano? That’s because at the highest C, the human’s ear can only hear the next 3 harmonics of 4186 Hz (which are 8372 Hz, 12558 Hz, and 16744 Hz). These frequencies are simply integer multiples of 4186 Hz, with the 4th harmonic of 4186 Hz already exceeding 14000 Hz, nearing high frequency hearing limit for a typical adult. However, when we listen to the middle C, our ears can hear many more of the harmonics of 262 Hz (which are 524 Hz, 786 Hz, 1048 Hz, 1310 Hz, 1572 Hz, 1834 Hz, and so on). The reader will note that even the 10th harmonics of 262 Hz (2620 Hz) are still far from the adult high frequency hearing limit. To successfully design a very good sounding loudspeaker, HX Audio Lab gives extra attention to the frequencies in the 100 Hz to 1000 Hz range, especially in regard to frequency and phase response.

While a lot of audio equipment uses 96 kHz sample rate to be able to reproduce frequencies up to 48 kHz, we stay with 48 kHz sample rate to focus on the best we can provide in the music and human frequency range. As many know, there is no free lunch in engineering. With current technology, it is not possible to flatten the phase down to 30 Hz without adding what we feel is too much processing delay (often called latency). We decided to limit the total processing delay to 5 ms and flatten the phase response to slightly lower than the middle C frequency.

A linear phase filter has a frequency resolution that follows for a simple equation: sample rate divided by the number of the filter’s taps. For example, a 96 kHz sample rate filter with 384 taps will result in a frequency resolution of 250 Hz. The effectiveness of the filter with 250 Hz resolution roughly starts at 4 times the resolution frequency, which is 1000 Hz and up. If we use a same number of taps with 48 kHz sample rate, then the filter’s frequency resolution becomes 125 Hz. The effectiveness of the filter starts half octave lower. Since the most important frequency range for music and speech is below 5000 Hz, we stay with 48 kHz sample rate digital signal processing to create an optimized linear phase filter with low processing delay and are able to flatten the phase response down to 200 Hz, even with an 8th order (48 dB/octave) crossover slope.

“So, is it true that a linear frequency response will result in pure sound reproduction?”

“Yes! Combined with linear phase response, the result will be the truest stereo sound stage reproduction”

How Transparent is the Loudspeaker?

Transparent is another visual cue, which is allowing light to pass through so that objects behind can be distinctly viewed. In audio, this subjectively means that the waveform’s shape is preserved in its natural state as much as it can. As explained above, linear frequency and phase response will provide a high level of transparency.

Transparent also means nothing is added. Nothing distorting the image. A loudspeaker cannot reproduce a single frequency absolutely purely. Due to the mechanical and physical properties of the magnet, voice coil, diaphragm materials and so on, there will be extra artifacts in the output. Harmonic distortion is a parameter to understand how much extra artifacts are added. When a loudspeaker is fed a single 1000 Hz sine wave input signal, the output will be a 1000 Hz (fundamental), a 2000 Hz (second harmonic) and also a 3000 Hz (third harmonic). The second and third harmonics are typically the most dominant extra artifacts relative to other distortion values.
Figure 3 shows the harmonic distortion of the type F22A when the fundamental level is at about 92 dB. The second and third harmonic components are about 40 dB or less at 100Hz and up. In human hearing, a sound will totally mask another sound when the level is more than 10 dB relative to the others. With such low harmonic distortions, the type F22A is capable of delivering a very transparent sound that you can trust immediately.

Why Can It be Used in Vertical Position Only?

Due to the driver arrangement (tweeter on the top and woofer on the bottom), the sound dispersion in the crossover range overlap (around 1200 Hz) will exhibit some unevenness. This is shown in the vertical off-axis frequency responses in figure 4 below.

When the type F22A loudspeaker is placed horizontally (sleeping position on its side), then the vertical side off-axis responses become the horizontal side off-axis responses. As noted in the graph, when you move beyond 30 degrees to the side, you will likely get an undesirable cancellation/null at around 1200 Hz.
When the type F22A is used vertically, figure 5 shows the horizontal side off-axis frequency responses. As noted, one can move up to 50 degrees to its side and not experience a more than 6 dB drop in the response. This ensures a very smooth frequency response change in the horizontal projection plane.

For musicians, composers and other users that do not have a fixed sweetspot listening location, the type F22A provides a wide sweetspot preset. This preset is slightly brighter when you listen on-axis, but will maintain a consistent spectral balance off-axis.

"Can I use it vertically with the tweeter on the bottom?"

"Yes! It will still result in a very smooth horizontal coverage. Do make sure the tweeter location is far from nearby boundaries or other furniture"

On-axis means being straight in front of the loudspeaker’s on-axis reference point. For type F22A, this is right in the front of the tweeter. Off-axis means being off from the loudspeaker’s on-axis reference point, but still inside the minus 6 dB coverage of the mid and high frequencies reproduction (also called beamwidth).
Figure 6 shows the minus 6 dB location of the type F22A’s sound dispersion. One can expect smooth frequency response changes over 120 degrees (60 degrees left and 60 degrees right) horizontal coverage and 60 degrees (30 degrees up and 30 degrees bottom) vertical coverage. Figure 7 illustrates the horizontal degree coverage with distance.

Assuming the straight on-axis distance of 1 m (3.3 ft.), the wide sweetspot preset will preserve the mid and high frequency spectral balance up to 1.7 m (5.5 ft.) to its side!

"Perfect! I am a composer with six keyboards. I can switch to different keyboards without losing the spectral balance even when jumping during a jam session!"

"Eerrr ... Yes, but calm down with the jumping part please"
Can You Explain the Time Response Graphs?

In the measurement world, loudspeaker measurement data can be observed in the frequency and time domain. One loudspeaker can have good frequency domain responses (on-axis response, off-axis response, distortion, and phase response), but can suffer resonances, not-so-tight low frequency reproduction or other time domain problems.

The waterfall and spectrogram of the type F22A are shown in figure 8 below. Both graphs show the frequency response over time, to see if there's energy stored in the mechanical structures that would 'ring' after the stimulus. Ideally nothing should persist after the stimulus signal stops.

![Figure 8](image)

From figure 8, it shows that 40 Hz area rings a little longer than others. This is a typical behavior of a vented/ported loudspeaker where the air oscillation inside the port moves slightly longer. The port is optimized so it does not create resonances at higher frequencies (above 100 Hz). The graph indicates very clean reproduction above 100 Hz where the driver movement halts when the input signal stops.

![Figure 9](image)

The step response graph is a look at the time domain of how a system responds to a transitory stimulus. When the driver’s diaphragm moves forward and creates a pressure wave front, the air molecules would respond to that pressure front with an inertial reduction in pressure and slow oscillation. The pressure then stabilizes to equilibrium. So an ideal step response will be a peak that decays. The step response graph shows signal coherency between the drivers. For a typical two-way loudspeaker, the step response graph can show two peaks, a sharper peak (typically from the tweeter) and a broader peak (typically from the woofer). However, with our precision signal and phase alignment method, type F22A step response only shows one peak shown in figure 9, indicating the drivers are very signal coherent.
May I Listen Closer Than 0.8 cm [32 in.] or Farther than 3 m [118 in.]?

The type F22A loudspeaker is optimized to give the best performance at a distance of 1 m [39 in.] or more. We do not recommend listening at a distance of less than 0.8 m [32 in.]. However, it is okay to listen beyond 3 m [118 in.]. Please understand that the direct sound pressure level of the loudspeaker decreases over distance. The type F22A is measured to give a maximum output of 114 dB at 1 m. At 3 m distance, the maximum output is expected to be 104 dB. As long you have a sufficient sound pressure level for your high level playback (subjective to each person), it does not matter how far you listen from the loudspeaker. Please be aware that hearing loss can occur when listening above 85 dB average for a long period.