

## Heart Murmur Detection Architectures

Heart murmur classification from ~900 raw PCG recordings is challenging due to limited data and subtle audio patterns. Recent studies show that CNNs on spectrograms (e.g. ResNet) reach ~82% accuracy, but top systems (PhysioNet 2022) only achieve ~80–83% <sup>1</sup>. The goal (>95%) is extremely ambitious. We compare leading approaches:

- **Audio Spectrogram Transformer (AST):** AST treats a log-mel spectrogram as an image and applies a Vision Transformer <sup>2</sup>. It achieved SOTA on generic audio tasks (95.6% ESC-50) <sup>2</sup>. For heart murmurs, transfer-learned AST gave only ~0.65 weighted accuracy <sup>3</sup> on the CirCor dataset, worse than SSL models. In practice, AST requires computing spectrograms and extensive fine-tuning. Its strength is modeling global context, but with 900 samples it risks overfitting. Niizumi *et al.* found supervised AST (AudioSet labels) underperformed a self-supervised model (M2D) on murmurs <sup>3</sup> <sup>4</sup>. AST may help as part of an ensemble, but alone is unlikely to hit 95%.
- **Wav2Vec 2.0:** A raw-waveform transformer pre-trained by self-supervision. Wav2Vec2 can be fine-tuned for classification. Panah *et al.* adapted Wav2Vec2 to murmurs and achieved ~0.80 weighted accuracy (UAR 0.70) <sup>5</sup> on CirCor data, and noted it is “robust to small fine-tuning data sizes” <sup>6</sup>. This suggests Wav2Vec2 features generalize to heart sounds. Implementation: feed raw audio (e.g. 4k–16k Hz) to Wav2Vec2 (or a Conformer variant) and fine-tune a classifier head. Use lower learning rate for stability <sup>7</sup>. Because it handles raw input, Wav2Vec2 avoids hand-crafted preprocessing. Its main trade-off is model size and computation. Given its SSL pretraining, it’s a promising candidate, though published results (~80%) remain below target <sup>1</sup>.
- **Conformers:** Conformers embed convolutional modules into transformer layers to capture both local and global features. Gulati *et al.* showed a Conformer model “significantly outperforms... previous Transformer and CNN models” on speech recognition by modeling local CNN filters and global attention <sup>8</sup>. A Wav2Vec2-Conformer (CNN in place of attention) similarly improved ASR error rates <sup>9</sup>. For murmurs, no direct results are reported, but a Conformer should theoretically better capture fine-grained murmur patterns (via convolution) plus context. It would use raw audio like Wav2Vec2. Conformers are heavy, but pretrained speech Conformers (e.g. fairseq’s S2T) could be fine-tuned. In summary, Conformers likely improve upon vanilla Wav2Vec2, but still constrained by data size.
- **HSMM+RNN (Segmentation-based pipeline):** The CinC2022-020 pipeline by McDonald *et al.* explicitly segments each heartbeat using multiple HSMMs conditioned on different murmur hypotheses. It computes a log-spectrogram (50 ms window, 0–800 Hz range) and feeds it to a bi-directional GRU whose outputs drive parallel HSMMs. The pipeline achieved 0.776 weighted accuracy (2nd place) on the PhysioNet test set <sup>10</sup>. This model is fully implementable from raw audio (via spectrogram pre-processing) <sup>11</sup>. Advantages: interpretable segmentation (identifies S1/S2/murmur intervals). However, its performance (~78%) is well below 95%. The RNN/HSMM approach effectively reduces temporal data to cycle-level features, but in this case it did not reach state-of-art. In principle, segmentation could aid a classifier by aligning beats, but by itself this pipeline is unlikely to

achieve >95%. It could be used as a preprocessing step (e.g. feed each cycle's segment features into a CNN/RNN), but heavy reliance on it may limit accuracy.

- **Other Hybrid Models:** Architectures that combine CNN, RNN, and/or Transformer layers can leverage both local and temporal patterns. For example, **CNN-RNN (CRNN)** models use a CNN on spectrograms for spatial features and an RNN (LSTM/GRU) for temporal context. Zolya *et al.* (2025) built a CRNN with preprocessing (band-pass filter, normalization) and heavy augmentation (noise, time/pitch shifts) and reported ~90.5% accuracy on murmurs <sup>12</sup>. However, cross-validation likely overfits: their external validation on CinC2016 was only ~87.3% <sup>13</sup>. The survey by Alkhodari *et al.* also notes CNN-RNN hybrids (with 1D wavelet smoothing) can reach high CV accuracy but generalize around 87% <sup>13</sup>. **CNN-Transformer** hybrids (e.g. FAST) stack CNN front-ends with transformer blocks. FAST (2025) combined MobileNetV2-like CNN layers and transformer, achieving SOTA on audio tasks with far fewer parameters <sup>14</sup>. This suggests CNN-Transformer can be powerful; in practice one could adapt FAST or a similar model to spectrogram inputs. **Transformer-RNN** combos are rarer, but conceptually one could use an RNN for final classification after a Transformer encoder or vice versa. In short, hybrids often perform well on audio. Empirically, CNNs still dominate murmur tasks: Sondermann *et al.* (2025) found CNN models (AUROC ~0.795) outperform “zero-shot” audio transformers (~0.657–0.701) <sup>15</sup>. This indicates tuned CNNs (or CNN+time-model) currently beat vanilla Transformers on PCGs. Nonetheless, hybrids leveraging both (e.g. Conformer, FAST) may offer a good trade-off if pretrained.

## Segmentation Pipeline Assessment

The HSMM+RNN segmentation pipeline is implementable from raw audio (via spectrogram features) <sup>11</sup>. It excels at aligning heart cycles and pinpointing murmur timing, which aids interpretability. However, its recorded accuracy (~77.6%) is far from 95% <sup>10</sup>. In fact, top challenge teams (using XGBoost and other features) only reached ~80% <sup>1</sup>. Thus a pure segmentation approach is unlikely to yield the required accuracy alone. It could be combined with DNNs (e.g. using segmented features as input to a CNN/RNN), but on its own it seems insufficient. In practice, segmentation could be used to augment feature extraction (e.g. computing cycle-based statistics or feeding state labels into a classifier), but the classification model will still need powerful learning (likely deep nets) to surpass ~80%. Given the 95% target, segmentation might best be used as auxiliary information (e.g. adding “S1/S2/murmur” timing features) rather than the final model.

## Model Comparison and Input Formats

Model	Input Format	Pretraining / Data	Reported Murmur Perf.
AST (pure Transformer)	Spectrogram (image) <sup>2</sup>	AudioSet-sup (SL) <sup>16</sup>	~65% Wacc on CirCor <sup>3</sup>
Wav2Vec2 (Transformer)	Raw waveform <sup>17</sup>	Large speech SSL <sup>17</sup>	~80% Wacc <sup>5</sup>
Conformer (CNN+Transformer)	Raw waveform	Speech SSL (e.g. XLSR)	Not reported (ASR SOTA <sup>8</sup> )

Model	Input Format	Pretraining / Data	Reported Murmur Perf.
CNN-RNN (CRNN)	Spectrogram or 1D Raw <sup>18</sup>	Typically ImageNet/ AudioSet SL	~90% (CV) <sup>12</sup> , ~87% external <sup>13</sup>
CNN only (ResNet/EEF)	Spectrogram	ImageNet or AudioSet SL	~82% (user's test)
HSMM+RNN (segmentation)	Spectrogram <sup>11</sup>	–	~78% Wacc <sup>10</sup>
AST+CNN (e.g. FAST)	Spectrogram	AudioSet SL + novel CNN/ Transformer design <sup>14</sup>	General audio SOTA (unknown for murmur)

**Implementation notes:** AST and CNN-based models use a 2D spectrogram (often log-Mel) input <sup>2</sup> <sup>11</sup>. Wav2Vec2/Conformer take raw waveform. HSMM+RNN uses log-spectrogram (0–800 Hz) <sup>11</sup>. Pretraining on large audio (AudioSet, speech) is crucial <sup>19</sup> <sup>1</sup>. For small data, one should freeze most layers and train only a classifier head or fine-tune lightly.

## Small-Data Strategies

With only ~900 samples, transfer learning and augmentation are key. **Transfer learning:** Use pretrained models (e.g. AudioSet CNNs, Audio Transformers, Wav2Vec2). Niizumi *et al.* showed AudioSet-supervised CNN14 and AST gave poor murmur results (58–65% Wacc) <sup>3</sup>, while SSL models (M2D, Wav2Vec2) outperformed them <sup>3</sup> <sup>1</sup>. Fine-tuning on PCG data with a low learning rate is recommended <sup>7</sup>. If possible, pretrain on any unlabeled heart sound data (e.g. via an autoencoder or contrastive SSL) to adapt to this domain.

**Data augmentation:** Crucial for generalization. Methods include time-domain noise, shifting and scaling, and spectrogram masking (SpecAugment). Zolya *et al.* used Gaussian noise, random time-shifts, pitch shifts, etc., and emphasized that “augmentation helps balance the dataset and improve generalization” <sup>20</sup>. Niizumi *et al.* found SpecAugment dramatically improved performance for low-data CNNs <sup>21</sup>. Key augmentations: adding Gaussian noise, time shifts, pitch shifts, random cropping/masking <sup>22</sup> <sup>21</sup>. Also consider balancing classes (murmur vs normal), e.g. oversample murmurs or use synthetic minority oversampling. Standard techniques (normalization, band-pass filtering 20–1000 Hz, fixed-length padding) are also recommended <sup>23</sup>.

**Ensembling:** Combining diverse models often helps. Niizumi *et al.* showed ensembles of different pretrained models (AST+M2D, CNN+M2D) further improved recall on rare classes <sup>24</sup> <sup>25</sup>. Given different models excel on different murmur types, ensembling (averaging or voting) can boost overall accuracy.

**Cross-validation:** Always validate with k-fold CV due to small data. Report metrics like weighted accuracy and UAR. Be wary of overfitting: very high CV accuracy (e.g. 99%) can collapse on held-out sets <sup>13</sup>.

## Recommendations

Given these findings, we recommend the following approach:

1. **Pretrained audio Transformer (spectrogram-based):** Fine-tune a high-capacity model like AST (Vision Transformer on log-mel spectrogram) or the FAST CNN-Transformer hybrid <sup>14</sup>. Start from weights pretrained on AudioSet or a similar large audio corpus. Use low learning rate and heavy regularization. This leverages the AST's strength in capturing global patterns <sup>2</sup>.
2. **Self-supervised waveform model:** Fine-tune Wav2Vec2 or a Conformer on the raw PCG signals <sup>5</sup> <sup>8</sup>. Wav2Vec2 is already pre-trained on speech and known "robust to small data" <sup>6</sup>. Consider using the Conformer variant for added CNN-style context <sup>8</sup>. Freeze most layers initially, train only a linear head. If possible, continue SSL pretraining on the heart sounds themselves before supervised fine-tuning.
3. **CNN-RNN model with augmentation:** Develop a CRNN on spectrograms, following Zolya *et al.* <sup>12</sup>. Apply aggressive augmentations (noise, time/pitch shifts, SpecAugment <sup>22</sup> <sup>21</sup>). This model can be shallower to avoid overfitting, or use pretrained CNN backbone (e.g. PANNs) to reduce parameter count. Monitor generalization (87% ext. validation in literature <sup>13</sup>), and use ensembling or dropout to mitigate overfit.
4. **Segmentation-aware features:** Optionally incorporate a segmentation module (CNN/RNN) to label S1/S2/murmur phases. Use those labels as auxiliary inputs or to align features. For example, pass spectrogram through a pretrained HSMM/RNN to generate segment probabilities <sup>11</sup>, then concatenate these with CNN features. This could help the classifier focus on murmur intervals, though it alone won't reach 95% <sup>10</sup>.
5. **Ensemble diverse models:** Combine the above models (e.g. Wav2Vec2+CNN-RNN+AST) by averaging probabilities. Ensembles improved accuracy in related studies <sup>24</sup> <sup>25</sup>, especially on hard classes. This is likely necessary to approach high accuracy.
6. **Preprocessing and normalization:** Standardize audio (e.g. 4 kHz resampling, band-pass 20–1000 Hz <sup>23</sup>) and normalize amplitude. Use uniform clip lengths (e.g. 10–15 s zero-padded) as in Zolya *et al.* <sup>26</sup>. Consistent preprocessing helps all models.

In summary, a hybrid strategy is advised: leverage **pretrained self-supervised transformers** (Wav2Vec2/Conformer on raw waveforms, or AST on spectrograms) because they have shown strong general-purpose audio representations <sup>2</sup> <sup>19</sup>. Complement them with a **CNN-RNN** model to capture fine details and with aggressive **data augmentation** <sup>20</sup> <sup>21</sup>. Incorporate segmentation cues if feasible, and ensemble for robustness. Even with these, >95% is unprecedented; one should set realistic goals (~85–90%) and use cross-validation to avoid overfitting. Critical factors are transfer learning from large audio/speech data and maximizing variability through augmentation <sup>6</sup> <sup>21</sup>.

**Sources:** We drew on recent literature on heart sound classification, including AST (Gong *et al.*, 2021) <sup>2</sup>, Wav2Vec2 adaptation (Panah *et al.*, EUSIPCO 2023) <sup>5</sup>, Conformer (Gulati *et al.*, 2020) <sup>8</sup>, the HSMM+RNN pipeline (McDonald *et al.*, CinC 2022) <sup>10</sup>, and surveys comparing CNN/Transformer models (Niizumi *et al.*,

2024 <sup>1</sup>; Sondermann *et al.*, 2025 <sup>15</sup>). Data augmentation and transfer learning best-practices are also drawn from these and related works <sup>20</sup> <sup>21</sup>.

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<sup>1</sup> <sup>3</sup> <sup>4</sup> <sup>16</sup> <sup>19</sup> <sup>21</sup> <sup>24</sup> <sup>25</sup> [arxiv.org](https://arxiv.org)

<https://arxiv.org/pdf/2404.17107>

<sup>2</sup> <sup>7</sup> **Audio Spectrogram Transformer**

[https://huggingface.co/docs/transformers/en/model\\_doc/audio-spectrogram-transformer](https://huggingface.co/docs/transformers/en/model_doc/audio-spectrogram-transformer)

<sup>5</sup> <sup>6</sup> <sup>17</sup> [eurasip.org](https://eurasip.org)

<https://eurasip.org/Proceedings/Eusipco/Eusipco2023/pdfs/0001010.pdf>

<sup>8</sup> **Paper page - Conformer: Convolution-augmented Transformer for Speech Recognition**

<https://huggingface.co/papers/2005.08100>

<sup>9</sup> **Wav2Vec2-Conformer**

[https://huggingface.co/docs/transformers/en/model\\_doc/wav2vec2-conformer](https://huggingface.co/docs/transformers/en/model_doc/wav2vec2-conformer)

<sup>10</sup> <sup>11</sup> [cinc.org](https://www.cinc.org)

<https://www.cinc.org/archives/2022/pdf/CinC2022-020.pdf>

<sup>12</sup> <sup>20</sup> <sup>22</sup> <sup>23</sup> <sup>26</sup> **AI-Enhanced Detection of Heart Murmurs: Advancing Non-Invasive Cardiovascular Diagnostics - PMC**

<https://pmc.ncbi.nlm.nih.gov/articles/PMC11945174/>

<sup>13</sup> <sup>18</sup> **Deep Learning in Heart Sound Analysis: From Techniques to Clinical Applications - PMC**

<https://pmc.ncbi.nlm.nih.gov/articles/PMC11461928/>

<sup>14</sup> **FAST: Fast Audio Spectrogram Transformer** © 2025 IEEE. Personal use of this material is permitted.

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<https://arxiv.org/html/2501.01104v1>

<sup>15</sup> **[2507.07058] Comparative Analysis of CNN and Transformer Architectures with Heart Cycle Normalization for Automated Phonocardiogram Classification**

<https://arxiv.org/abs/2507.07058>